



## Avaya Solution & Interoperability Test Lab

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# Application Notes for Configuring Avaya Aura® Communication Manager 6.0.1, Avaya Aura® Session Manager 6.1, and Avaya Session Border Controller for Enterprise 4.0.5 with AT&T IP Transfer Connect Service – Issue 1.0

## Abstract

These Application Notes describe the steps for configuring Avaya Aura® Communication Manager with SIP Network Call Redirection (NCR), Avaya Aura® Session Manager, and the Avaya Session Border Controller for Enterprise 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 Aura® Communication Manager. In addition, the Avaya Aura® Communication Manager NCR and SIP User-to-User Information (UII) features can be utilized together, in conjunction with the Data Forwarding option of the AT&T IP Transfer Connect service, to transmit UII within SIP signaling messages to the alternate destinations. Avaya Aura® Session Manager is a core SIP routing and integration engine that connects disparate SIP devices and applications within an enterprise. Note that these Application Notes are intended to supplement the separate document: *Applications Notes for Avaya Aura® Communication Manager 6.0.1, Avaya Aura® Session Manager 6.1 and Avaya Session Border Controller for Enterprise with AT&T IP Toll Free SIP Trunk Service*.

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

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# 1. Introduction

These Application Notes describe the steps for configuring Avaya Aura® Communication Manager SIP Network Call Redirection (NCR), Avaya Aura® Session Manager, and the Avaya Session Border Controller for Enterprise (referred to in the remainder of this document as *Avaya SBCE*), with the AT&T IP Transfer Connect service using **AVPN** or **MIS/PNT** transport connections<sup>1</sup>. 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<sup>2</sup> destinations based upon SIP redirection messages from Avaya Aura® Communication Manager. The AT&T IP Transfer Connect service is typically used by enterprises that have multiple call centers that are separated geographically or otherwise not interconnected. Using SIP NCR, trunk-to-trunk routing of certain inbound calls on Avaya Aura® Communication Manager can be avoided by requesting that the AT&T network transfer the inbound caller to an alternate destination.

In addition, the Avaya Aura® Communication Manager SIP User-to-User Information (UUI) feature can be utilized with the SIP NCR feature to transmit UUI within SIP signaling messages to the alternate destinations. This capability is used in conjunction with the Data Forwarding option of the AT&T IP Transfer Connect service to transmit a limited amount of call-related data between call centers to support enhanced, customer-friendly applications and/or support efficient use of call center resources. Examples of UUI data might include a customer account number obtained during a database query and the best service routing data exchanged between Avaya Aura® Communication Manager systems.

**Note:** These Application Notes are intended to supplement the separate document: *Applications Notes for Avaya Aura® Communication Manager 6.0.1, Avaya Aura® Session Manager 6.1 and Avaya Session Border Controller for Enterprise AT&T IP Toll Free SIP Trunk Service*.

## 2. General Test Approach and Test Results

The test environment consisted of:

- A simulated enterprise with Avaya Aura® System Manager, Avaya Aura® Session Manager, Avaya Aura® Communication Manager, Avaya phones, and an *Avaya SBCE*.
- A laboratory version of the AT&T IP Transfer Connect service, to which the simulated enterprise was connected.

### 2.1. Interoperability Compliance Testing

The interoperability compliance testing focused on verifying inbound call flows (see **Section 4.1** for descriptions) to Session Manager and subsequent routing to Communication Manager, and subsequent redirection messages to AT&T for rerouting to alternate destinations.

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<sup>1</sup> MIS/PNT transport does not support compressed RTP (cRTP), however AVPN transport does support cRTP.

<sup>2</sup> 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 compliance testing was based on a test plan provided by AT&T, for the functionality required for certification as a solution supported on the AT&T network. Calls were made from the PSTN across the AT&T network. The following features were tested as part of this effort:

- SIP trunking.
- Call redirection functionality utilizing 302 and Refer SIP call processing.
- Communication Manager features such as hold, resume, and transfer.

## 2.2. Test Results

The main test objectives were to verify the following features and functionality:

- Inbound AT&T IP Transfer Connect service calls to Communication Manager VDNs, agents, and phones.
- Inbound AT&T IP Transfer Connect service calls that are immediately redirected by a Communication Manager vector (pre-answer redirection) back to the AT&T IP Transfer Connect service for redirection to an alternate destination.
- Inbound AT&T IP Transfer Connect service calls that are answered by a Communication Manager vector and then redirected (post-answer redirection) back to the AT&T IP Transfer Connect service for redirection to an alternate destination.
- Redirected AT&T IP Transfer Connect service calls per above arriving on Communication Manager VDNs, agents, and phones (i.e., Communication Manager as the target party for the redirected calls).
- Recovery from unsuccessful post-answer redirection attempts per above due to busy or error conditions on the alternate destination.
- Call and two-way talk path establishment between callers and Communication Manager agents/phones.

The above test objectives with limitations as noted in **Section 2.2.1** were verified.

### 2.2.1. Known Limitations

1. The IP Transfer Connect service specifies that 18x responses should not be used in conjunction with 302 redirection calls. Therefore ring back should not be specified in Communication Manager 302 redirection vectors (see **Section 7.3**).
2. The Communication Manager 6.0.1 SIP trunk form may be configured to send either a 180 (default) or 183 session progress message (see **Section 7.2**). The session progress message selected alters the Communication Manager behavior upon receipt of a Notify form AT&T during Refer calls. If 180 is selected, then Communication Manager will issue a BYE upon receipt of the Notify/Ringing message from AT&T. If 183 is selected, then Communication Manager will issue a BYE upon receipt of the Notify/200OK from AT&T. In both cases the expected behavior was for the Avaya CPE to wait for AT&T IP Transfer Connect service to issue the BYE. However no issues were encountered during testing due to either behavior.

3. The Communication Manager Network Call Redirection (NCR) feature is required to enable Refer and 302 call redirection with the AT&T IP Transfer Connect Service (see **Section 7.2**). With this feature enabled, Communication Manager will also use the SIP parameter *SendOnly* to signal any hold call conditions. The *SendOnly* SIP parameter is not supported by the AT&T Flexible Reach service. Any customers that access both AT&T IP Transfer Connect and AT&T IP Flexible Reach services via the same Communication Manager environment, must use the procedures described in **Addendum 1** of this document that describes having the Avaya SBCE replace the *SendOnly* parameter with the *SendRecv* parameter that the AT&T Flexible Reach service does support.
4. An issue was found with Communication Manager Service Pack 7 (19528) and conference calls initiated by a Communication Manager station when NCR is enabled. During the conference process Communication Manager places the first call leg (e.g. PSTN) on hold (*SendOnly*), to place the second call leg. Sub sequentially; Communication Manager does not remove the first call leg from hold (*SendRecv*). This results in no inbound audio to the conference from the first call leg.
  - An MR was opened with Communication Manager support.
  - **Note** – A workaround for this issue may be implemented by following the procedure described in item 3 above and shown in **Addendum 1**.
5. An issue was found with the Avaya SBCE in regards to Refer call processing. When the initial call leg is disconnected after the Referred-To call is established, the IP Transfer Connect service issues a BYE. This BYE should be routed by the Avaya SBCE to Session Manager, for delivery to Communication Manager. Instead, the Avaya SBCE sends the BYE directly to Communication Manager. While the Referred-To call completes successfully, the resulting SIP dialog between the Avaya SBCE and the IP Transfer Connect service for the termination of the initial call leg, does not match expected behavior.
  - An MR was opened with Avaya SBCE support.
  - **UPDATE** – Avaya SBCE load 4.0.5.Q09 fixed this issue.
6. An issue was found with the Avaya SBCE in regards to Refer and 401 Unauthorized call processing. A test case specifies that the CPE generate a Refer using the wrong DNIS access number. The AT&T IP Transfer Connect service then responds with 401 Unauthorized. This 401 has a Contact header containing an AT&T internal IP address (not routable from the CPE). When the CPE generated a subsequent ReInvite, the Avaya SBCE used the IP address defined in the 401 Contact header as the Internet Protocol Destination header IP address, and not the AT&T Border Element IP address.
  - An MR was opened with Avaya SBCE support.
  - **UPDATE** – A script was written for the Avaya SBCE as a workaround for this issue, pending a fix from Avaya SBCE support (see **Addendum 2**).

## 2.3. Support

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

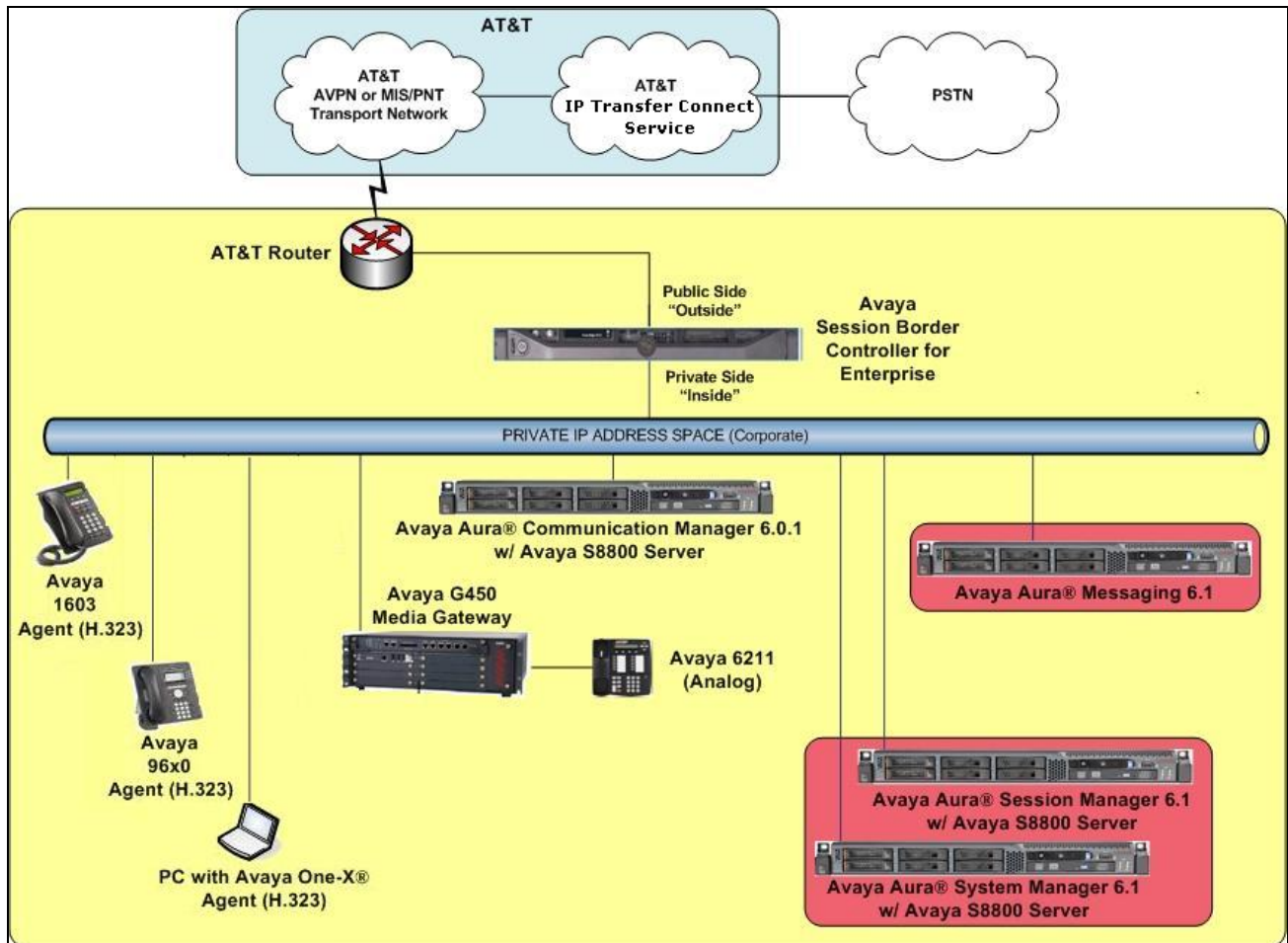
Avaya customers may obtain documentation and support for Avaya products by visiting <http://support.avaya.com>. In the United States, (866) GO-AVAYA (866-462-8292) provides access to overall sales and service support menus. Customers may also use specific numbers (provided on <http://support.avaya.com>) 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** and consists of several components:

- Avaya Aura® Session Manager provides core SIP routing and integration services that enables communications between disparate SIP-enabled entities, e.g., PBXs, SIP proxies, gateways, adjuncts, trunks, applications, etc. across the enterprise. Session Manager allows enterprises to implement centralized and policy-based routing, centralized yet flexible dial plans, consolidated trunking, and centralized access to adjuncts and applications.
- Avaya Aura® System Manager provides a common administration interface for centralized management of all Session Manager instances in an enterprise.
- Avaya Aura® Communication Manager provides the voice communications services for a particular enterprise site. In the reference configuration, Communication Manager runs on an Avaya S8800 Server in a Processor Ethernet (Procr) configuration. This solution is extensible to other Avaya S8xxx Servers.
- The Avaya Media Gateway provides the physical interfaces and resources for Communication Manager. In the reference configuration, an Avaya G450 Media Gateway is used. This solution is extensible to other Avaya Media Gateways.
- Avaya desk telephones are represented with Avaya 1603(H.323), 960x Series IP Telephones (running H.323), and 96x1 Series IP Telephones (running H.323), Avaya 6211 Series Analog Telephones, as well as Avaya one-X® Agent. Note that all agent telephones (hardware and software based) are H.323.
- The Avaya SBCE provides Back to Back User Agent (B2BUA) functionality, including address translation and SIP header manipulation between the AT&T IP Transfer Connect service and the enterprise internal network. UDP transport protocol is used between the Avaya SBCE and the AT&T IP Transfer Connect service.
- An existing Avaya Aura® Messaging system provides the corporate voice messaging capabilities in the reference configuration. The provisioning of Avaya Aura® Messaging is beyond the scope of this document.

- Inbound calls were placed from PSTN via the AT&T IP Transfer Connect service, through the Avaya SBCE to the Session Manager which routed the call to Communication Manager. Communication Manager terminated the call to the appropriate agent/phone. The H.323 phones on the enterprise side registered to the Communication Manager Processor Ethernet interface (Procr, see [1]).



**Figure 1: Reference Configuration**

## 4. 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 specific values for their own specific configurations.

**Note** - The AT&T IP Transfer Connect service Border Element IP address and DNIS digits, (destination digits specified in the SIP Request URIs sent by the AT&T Transfer Connect service), shown in this document are examples. AT&T Customer Care will provide the actual IP addresses and DNIS digits as part of the IP Transfer Connect provisioning process.

Component	Illustrative Value in these Application Notes
<b>Avaya Aura® System Manager</b>	
Management IP Address	192.168.67.207
<b>Avaya Aura® Session Manager</b>	
Management IP Address	192.168.67.209
Network IP Address	192.168.67.210
<b>Avaya Aura® Communication Manager</b>	
Procr IP Address	192.168.67.202
Avaya Aura® Communication Manager extensions	40xxx = H323 and Analog 41xxx = SIP
Voice Messaging Pilot Extension	36000
<b>Avaya Session Border Controller for Enterprise</b>	
IP Address of Outside (Public) Interface (connected to AT&T Access Router/IP Transfer Connect Service)	192.168.64.130
IP Address of Inside (Private) Interface (connected to Avaya Aura® Session Manager)	192.168.67.125
<b>Avaya Aura Messaging</b>	
Messaging Server IP Address	192.168.67.147
Messaging Mailboxes	4xxxx
<b>AT&amp;T IP Transfer Connect Service</b>	
Border Element IP Address	135.25.29.74
AT&T Access router interface (to Avaya Aura® outside)	192.168.64.254
AT&T Access Router NAT address (Avaya Aura® outside address)	135.16.170.55

**Table 1: Illustrative Values Used in the Reference Configuration**

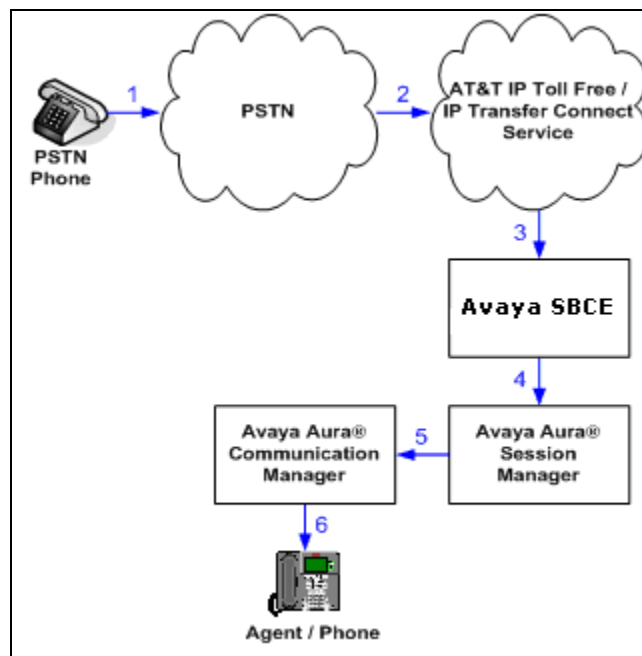


## 4.1. Call Flows

To understand how inbound AT&T IP Transfer Connect service calls are handled by Session Manager and Communication Manager, four general call flows are described in this section.

The first call scenario illustrated in **Figure 2** is an inbound AT&T IP Transfer Connect service call that arrives on Session Manager and is subsequently routed to Communication Manager, which in turn routes the call to a vector, agent, or phone. Note that no redirection is performed in this scenario, and thus the call flow is the same as that of an inbound AT&T IP Toll Free service call.

1. A PSTN phone originates a call to an AT&T IP Transfer Connect service number (an AT&T IP Toll Free service number that has been enabled with the AT&T IP Transfer Connect service option).
2. The PSTN routes the call to the AT&T IP Transfer Connect service network.
3. The AT&T IP Transfer Connect 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 Session Manager.
5. Session Manager applies any necessary SIP header adaptations and digit conversions, and based on configured Network Routing Policies, determines where the call should be routed next. In this case, Session Manager routes the call to Communication Manager.
6. Depending on the called number, Communication Manager routes the call to a) a vector, which in turn, routes the call to an agent or phone, or b) directly to an agent or phone.

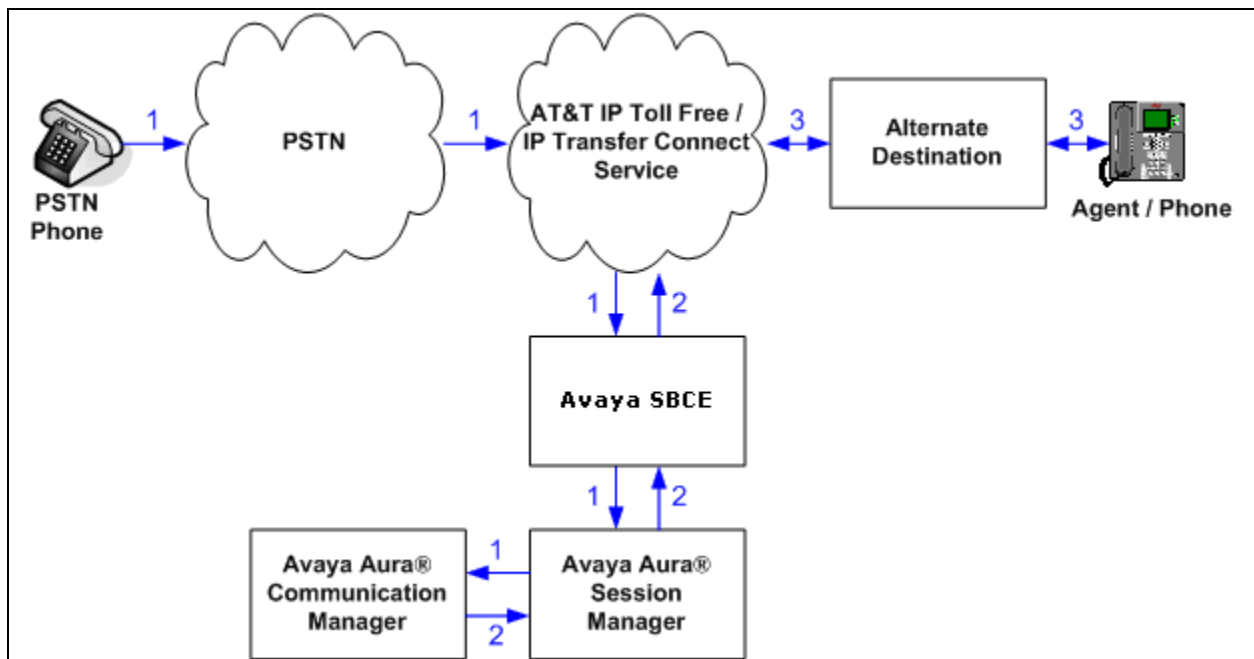


**Figure 2: Inbound AT&T IP Transfer Connect Call – No Redirection**

**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 AT&T IP Transfer Connect service are unrelated.

The second call scenario illustrated in **Figure 3** is an inbound AT&T IP Transfer Connect service call that arrives on Session Manager and is subsequently routed to Communication Manager, which in turn routes the call to a vector. The vector, without answering the call, immediately redirects the call back to the AT&T IP Transfer Connect service for routing to an alternate destination.

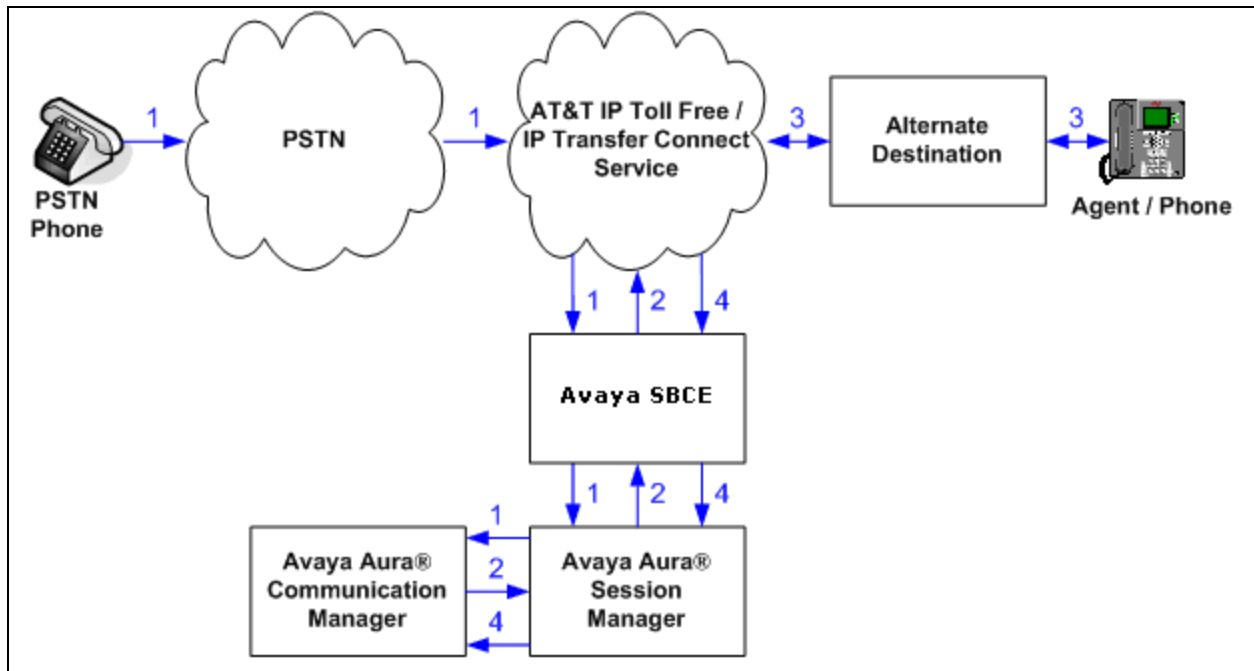
1. Same as the first five steps from the first call scenario.
2. Communication Manager routes the call to a vector, which redirects the call by sending a SIP 302 message back out on the SIP trunk on which the inbound call arrived. The SIP 302 message is routed back through Session Manager and then the Avaya SBCE sends the call to the AT&T IP Transfer Connect service network. Since the SIP 302 message is a final response, the redirecting party (Communication Manager) is no longer involved in the call whether the redirection succeeds or fails, and thereby releases the trunk.
3. The AT&T IP Transfer Connect service places a call to the alternate destination and upon answer, connects the calling party to the target party (alternate destination).



**Figure 3: Inbound AT&T IP Transfer Connect Call – Pre-Answer SIP 302 Redirection**

The third call scenario illustrated in **Figure 4** is an inbound AT&T IP Transfer Connect service call that arrives on Session Manager and is subsequently routed to Communication Manager, which in turn routes the call to a vector. The vector answers the call and then redirects the call back to the AT&T IP Transfer Connect service for routing to an alternate destination.

1. Same as the first five steps from the first call scenario.
2. Communication Manager routes the call to a vector, which answers the call and plays an announcement, and attempts to redirect the call by sending a SIP REFER message back out on the SIP trunk on which the inbound call arrived. The SIP REFER message specifies the alternate destination, and is routed back through Session Manager and then the Avaya SBCE sends the call to the AT&T IP Transfer Connect service network.
3. The AT&T IP Transfer Connect service places a call to the target party (alternate destination) and upon answer, connects the calling party to the target party.
4. The AT&T IP Transfer Connect service clears the call on the redirecting/referring party (Communication Manager).

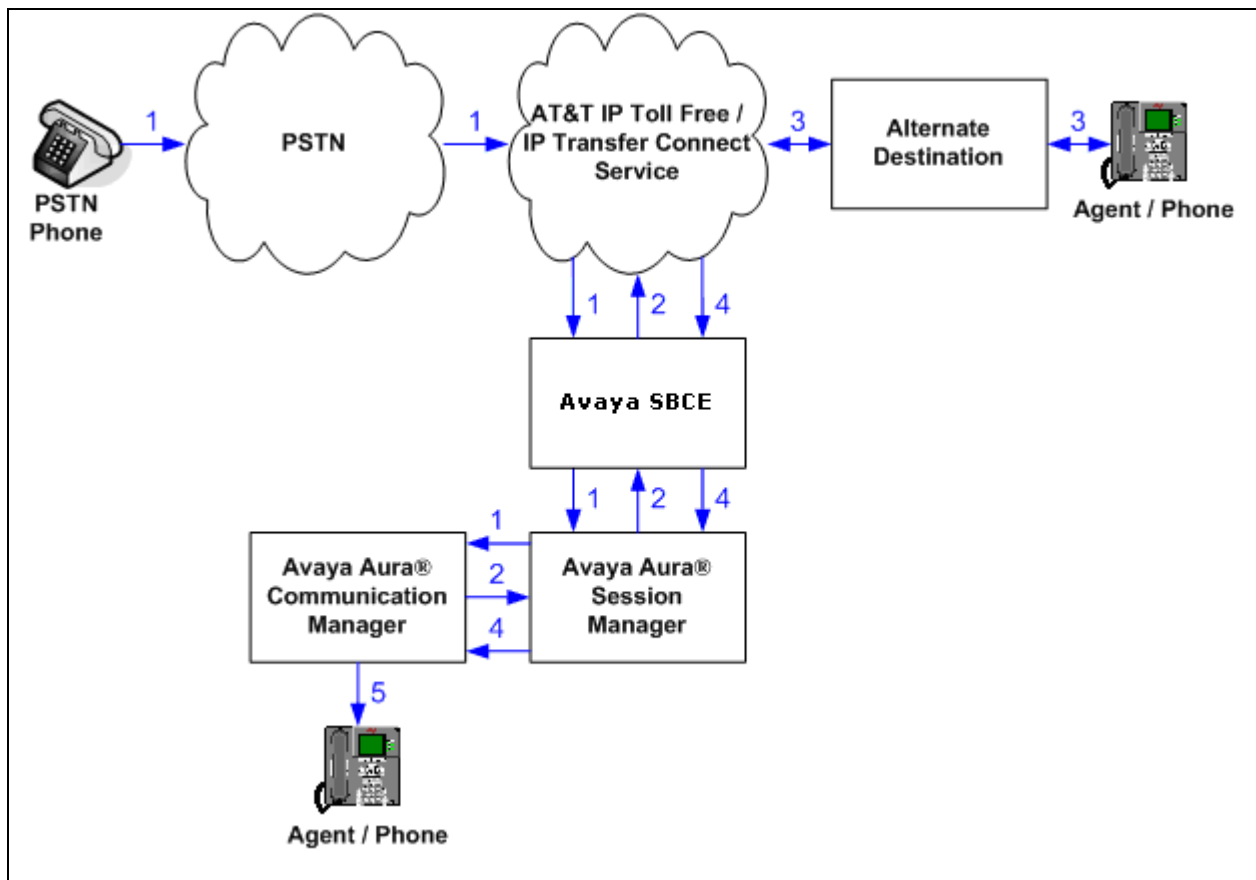


**Figure 4: Inbound AT&T IP Transfer Connect Call – Post-Answer SIP REFER Redirection Successful**

The fourth call scenario illustrated in **Figure 5** is similar to the fourth call scenario, except that the redirection is unsuccessful due to the alternate destination being busy or otherwise unavailable. As a result, Communication Manager “takes the call back” and routes the call to an agent/phone.

1. Same as the third call scenario.
2. Same as the third call scenario.
3. The AT&T IP Transfer Connect service places a call to the target party (alternate destination), but the target party is busy or otherwise unavailable.
4. The AT&T IP Transfer Connect service notifies the redirecting/referring party (Communication Manager) of the error condition.
5. Communication Manager routes the call to a local agent or phone.

**Note:** This “error handling” scenario occurs only with AT&T IP Transfer Connect service lines enabled with the Attended IP Courtesy Transfer feature.



**Figure 5: Inbound AT&T IP Transfer Connect Call – Post-Answer SIP REFER Redirection Unsuccessful**

## 5. Equipment and Software Validated

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

Component/Software	Release/Version
Avaya S8800 Server	Avaya Aura® System Manager 6.1 SP6 (System Manager 6.1.5.0 Build Number 6.1.0.0.7345 Patch 6.1.5.606 Build Number 6.1.10.1.1774) System Platform 6.0.3.6.3
Avaya S8800 Server	Avaya Aura® Session Manager 6.1 (6.1.6.0.616008)
Avaya S8800 Server	Avaya Aura® Communication Manager 6.0.1 SP7 (R016x.00.1.510.1-19528) System Platform 6.0.3.6.3
Dell R610	Avaya Aura® Messaging 6.1 with SP0 (00.1.510.1-115_0006) System Platform 6.0.3.6.3
Avaya G450 Media Gateway	31.20.1
MM711 Analog card	HW31 FW094
Dell R310	Avaya Session Border Controller for Enterprise 4.0.5.Q09
Avaya 9630 IP Telephone	H.323 Version S3.103S (ha96xxua3_1_03_S.bin)
Avaya 9611 IP Telephone	S6.020S
Avaya one-X® Agent	2.5.00467.0
Avaya 1603 IP Telephone	H323 (ha1603ua1_3100.bin)
AT&T IP Transfer Connect Service using AVPN/MIS-PNT transport service connection	VNI 22

**Table 2: Equipment and Software Versions**

## **6. Avaya Aura® Session Manager**

The Avaya Aura® Session Manager administration for interaction with the AT&T IP Toll Free / IP Transfer Connect service is described in [1]. This section describes the additional administration steps on Session Manager necessary for supporting interaction with the AT&T IP Transfer Connect service.

### **6.1. Dial Patterns**

If the dial pattern(s) provisioned in [1] for matching inbound AT&T IP Toll Free service calls are insufficient for matching inbound AT&T IP Transfer Connect service calls, then provision additional dial patterns according to the procedures described in [1] as necessary.

## 7. Avaya Aura® Communication Manager

The Avaya Aura® Communication Manager administration for interaction with the AT&T IP Toll Free service is described in [1] and are applicable for the AT&T IP Transfer Connect service as well. This section describes the additional administration steps on Communication Manager necessary for supporting interaction with the AT&T IP Transfer Connect service. The steps are performed from the Communication Manager System Access Terminal (SAT) interface.

**Note** – In the following sections, only the **highlighted** parameters are applicable to these Application Notes. Other parameters shown should be considered informational.

### 7.1. System Parameters

This section reviews the additional Communication Manager licenses and features that are required for supporting the interaction with the AT&T IP Transfer Connect service. For required licenses that are not enabled in the steps that follow, contact an authorized Avaya account representative to obtain the licenses.

1. Enter the **display system-parameters customer-options** command. On Page 4 of the **system-parameters customer-options** form, verify that the **ISDN/SIP Network Call Redirection?** feature is set to “y”.

display system-parameters customer-options		Page	4 of	11
OPTIONAL FEATURES				
Emergency Access to Attendant? y		IP Stations? y		
Enable 'dadmin' Login? y				
Enhanced Conferencing? y		ISDN Feature Plus? n		
Enhanced EC500? y	<b>ISDN/SIP Network Call Redirection? y</b>			
Enterprise Survivable Server? n		ISDN-BRI Trunks? y		
Enterprise Wide Licensing? n		ISDN-PRI? y		
ESS Administration? y		Local Survivable Processor? n		
Extended Cvg/Fwd Admin? y		Malicious Call Trace? y		
External Device Alarm Admin? y		Media Encryption Over IP? n		
Five Port Networks Max Per MCC? n	Mode Code for Centralized Voice Mail? n			
Flexible Billing? n				
Forced Entry of Account Codes? y		Multifrequency Signaling? y		
Global Call Classification? y		Multimedia Call Handling (Basic)? y		
Hospitality (Basic)? y		Multimedia Call Handling (Enhanced)? y		
Hospitality (G3V3 Enhancements)? y		Multimedia IP SIP Trunking? y		
IP Trunks? y				
IP Attendant Consoles? y				
(NOTE: You must logoff & login to effect the permission changes.)				

2. On Page 6 of the **system-parameters customer-options** form, verify that the vectoring features are set to “y”.

display system-parameters customer-options		Page 6 of 11
CALL CENTER OPTIONAL FEATURES		
Call Center Release: 6.0		
<b>ACD? y</b>		Reason Codes? y
BCMS (Basic)? y		Service Level Maximizer? n
BCMS/VuStats Service Level? y		Service Observing (Basic)? y
BSR Local Treatment for IP & ISDN? y	Service Observing (Remote/By FAC)? y	
Business Advocate? n	Service Observing (VDNs)? y	
Call Work Codes? y	Timed ACW? y	
DTMF Feedback Signals For VRU? y	<b>Vectoring (Basic)? y</b>	
Dynamic Advocate? n	<b>Vectoring (Prompting)? y</b>	
Expert Agent Selection (EAS)? y	<b>Vectoring (G3V4 Enhanced)? y</b>	
EAS-PHD? y	<b>Vectoring (3.0 Enhanced)? y</b>	
Forced ACD Calls? n	Vectoring (ANI/II-Digits Routing)? y	
Least Occupied Agent? y	<b>Vectoring (G3V4 Advanced Routing)? y</b>	
Lookahead Interflow (LAI)? y	Vectoring (CINFO)? y	
Multiple Call Handling (On Request)? y	<b>Vectoring (Best Service Routing)? y</b>	
Multiple Call Handling (Forced)? y	Vectoring (Holidays)? y	
PASTE (Display PBX Data on Phone)? y	Vectoring (Variables)? y	
(NOTE: You must logoff & login to effect the permission changes.)		



## 7.2. Trunks

This section describes the steps for modifying the SIP trunk to Session Manager to support the interaction with the AT&T IP Transfer Connect service.

1. Enter the **change trunk-group x** command, where **x** is the number of the trunk group administered in [1] for inbound AT&T IP Toll Free service calls (e.g. 2). On **Page 4** of the **trunk-group** form, set **Network Call Redirection** to “y” (see **item 3** in **Section 2.2.1** and **Addendum 1**).
2. Note whether the setting for **Convert 180 to 183 for Early Media?** is “n” (default) or “y” (based on customer configuration). The value defined will alter the Refer NOTIFY response behavior (see **item 1** in **Section 2.2.1**).

<b>change trunk-group 2</b>	Page 4 of 21
PROTOCOL VARIATIONS	
Mark Users as Phone? n	
Prepend '+' to Calling Number? n	
Send Transferring Party Information? n	
<b>Network Call Redirection? y</b>	
Send Diversion Header? n	
Support Request History? y	
Telephone Event Payload Type: 100	
<b>Convert 180 to 183 for Early Media? n</b>	
Always Use re-INVITE for Display Updates? n	
Identity for Calling Party Display: P-Asserted-Identity	
Enable Q-SIP? n	

## 7.3. Inbound Call Routing

This section describes the steps for routing inbound AT&T IP Transfer Connect service calls to reach Vector Directory Numbers (VDNs) with corresponding programmable vectors. These vectors contain steps that invoke the Communication Manager SIP Network Call Redirection (NCR) functionality. The routing of inbound AT&T IP Transfer Connect service calls that do not invoke the SIP NCR functionality is addressed in [1].

Two different inbound call routing scenarios are described in these Application Notes:

- Pre-Answer Redirection - An inbound AT&T IP Transfer Connect service call that invokes SIP NCR (using a SIP 302 message) prior to the call being answered.
- Post-Answer Redirection - An inbound AT&T IP Transfer Connect service call that invokes SIP NCR (using a SIP REFER message) after the call has been answered by a vector.

The following inbound call treatment information is defined using the inbound number information provided by AT&T

These Application Notes provide rudimentary vector definitions to demonstrate and test the SIP NCR and UII functionalities. In general, call centers will use vector functionality that is more complex and tailored to their individual needs. Call centers may also use customer hosts running applications used in conjunction with Avaya Aura® Application Enablement Services (AES) to define call routing and provide associated UII. The definition and documentation of those complex applications and associated vectors are beyond the scope of these Application Notes. Consult [6] and [7] for further information.

### 7.3.1. Pre-Answer Redirection

This section provides an example of Pre-Answer Redirection. In this example, the inbound call is routed to the VDN shown in **Figure 7**, which invokes the vector shown in **Figure 8**. Note that the vector does not specify ring back (see **Section 2.2.1**). The vector does the following:

- Assigns the data “**1234567890123456**” to ASAI UII variable “**A**” (vector step **02**).  
**Note:** The parameters for ASAI UII variables “**A**” and “**B**”, and other vector variables are defined using the **change variables** command (see **Figure 6**).
- Redirects the call to the number “**1012**” (vector step **05**). Note that since this vector did not answer the call, the presence of the “~” in the “**route-to number**” instructs Communication Manager to send a SIP 302 message with the number “**1012**” in the user part of the Contact header URI, e.g., 1012@<host/domain>, to the AT&T IP Transfer Connect service (via Session Manager and the Avaya SBCE).

change variables							Page	1 of 39
VARIABLES FOR VECTORS								
Var	Description	Type	Scope	Length	Start	Assignment	VAC	
A	UiTest1	asaiuii	L	16	1			
B								
C								
D								
E								
F								
G								
H								
I								
J								
K								
L								
M								
N								
O								
P								
Q								
R								

**Figure 6: Change Variables Form**

```

display vdn 44020                                     Page 1 of 3
                                                    VECTOR DIRECTORY NUMBER

      Extension: 44020
      Name*: 302
      Destination: Vector Number      22
    Attendant Vectoring? n
    Meet-me Conferencing? n
      Allow VDN Override? n
        COR: 1
        TN*: 1
        Measured: none

    VDN of Origin Annc. Extension*:
        1st Skill*:
        2nd Skill*:
        3rd Skill*:

* Follows VDN Override Rules

```

**Figure 7: Sample VDN for Pre-Answer Redirection**

```

display vector 22                                     Page 1 of 6
                                                    CALL VECTOR

    Number: 22                      Name: 302RingUII
Multimedia? n      Attendant Vectoring? n      Meet-me Conf? n      Lock? n
  Basic? y      EAS? y      G3V4 Enhanced? y      ANI/II-Digits? y      ASAI Routing? y
  Prompting? y      LAI? y      G3V4 Adv Route? y      CINFO? y      BSR? y      Holidays? y
  Variables? y      3.0 Enhanced? y
01# Define UII variable
02 set      A      = none      CATR 1234567890123456
03
04 # Redirect
05 route-to      number ~r1012      with cov n if unconditionally
06
07 stop
08
09

```

**Figure 8: Sample Vector for Pre-Answer Redirection**

### 7.3.2. Post-Answer Redirection

This section provides an example of Post-Answer Redirection. In this example, the inbound call is routed to the VDN shown in **Figure 9**, which invokes the vector shown in **Figure 10**. The vector does the following:

- Plays ringback for 2 seconds (vector step **02**).
- Assigns the data “**1234567890123456**” to ASAI UUI variable “**A**” (vector steps **05**).  
**Note:** The parameters for UUI variable “**A**” and other vector variables are defined using the **change variables** command (see **Figure 6**).
- Answers the call to play an announcement (vector step **08**).
- Attempts to redirect the call to the number “**1012**” (vector step **09**). Note that since this vector answered the call, the presence of the “~” in the “**route-to number**” instructs Communication Manager to send a SIP REFER message with the number “**1012**” in the user part of the Refer-To header URI, e.g., 1012@<host/domain> to the AT&T IP Transfer Connect service (via Session Manager and the Avaya SBCE).

```
display vdn 31010                                     Page 1 of 3
VECTOR DIRECTORY NUMBER
Extension: 31010
Name*: NCR Ringback REFER UUI
Destination: Vector Number 1010
Attendant Vectoring? n
Meet-me Conferencing? n
Allow VDN Override? n
COR: 1
TN*: 1
Measured: none
VDN of Origin Annc. Extension*:
1st Skill*:
2nd Skill*:
3rd Skill*:
* Follows VDN Override Rules
```

**Figure 9: Sample VDN for Post-Answer Redirection**

display vector 1010

Page 1 of 6

CALL VECTOR

```
Number: 1010           Name: NcrRefer_wUui
Multimedia? n      Attendant Vectoring? n      Meet-me Conf? n      Lock? n
Basic? y    EAS? y    G3V4 Enhanced? y    ANI/II-Digits? y    ASAI Routing? y
Prompting? y    LAI? n    G3V4 Adv Route? y    CINFO? n    BSR? y    Holidays? n
Variables? y    3.0 Enhanced? y
01 #    NCR Refer with ringback and uui forwarding
02 wait-time    2    secs hearing ringback
03
04 #    Define UUI variable to send
05 set          A      = none    CATR 1234567890123456
06
07 #    Refer to AT&T speed dial number
08 announcement 59113
09 route-to      number ~r1012      with cov n if unconditionally
10 #    Play this announcement only on redirect failure
11 disconnect    after announcement 52220
12
```

**Figure 10: Sample Vector for Post-Answer Redirection**

## 8. Avaya Session Border Controller for Enterprise

The Avaya SBCE configuration for interaction with the AT&T IP Toll Free service provided in [1] should also be followed for interoperability with the AT&T IP Transfer Connect service, with the following addition.

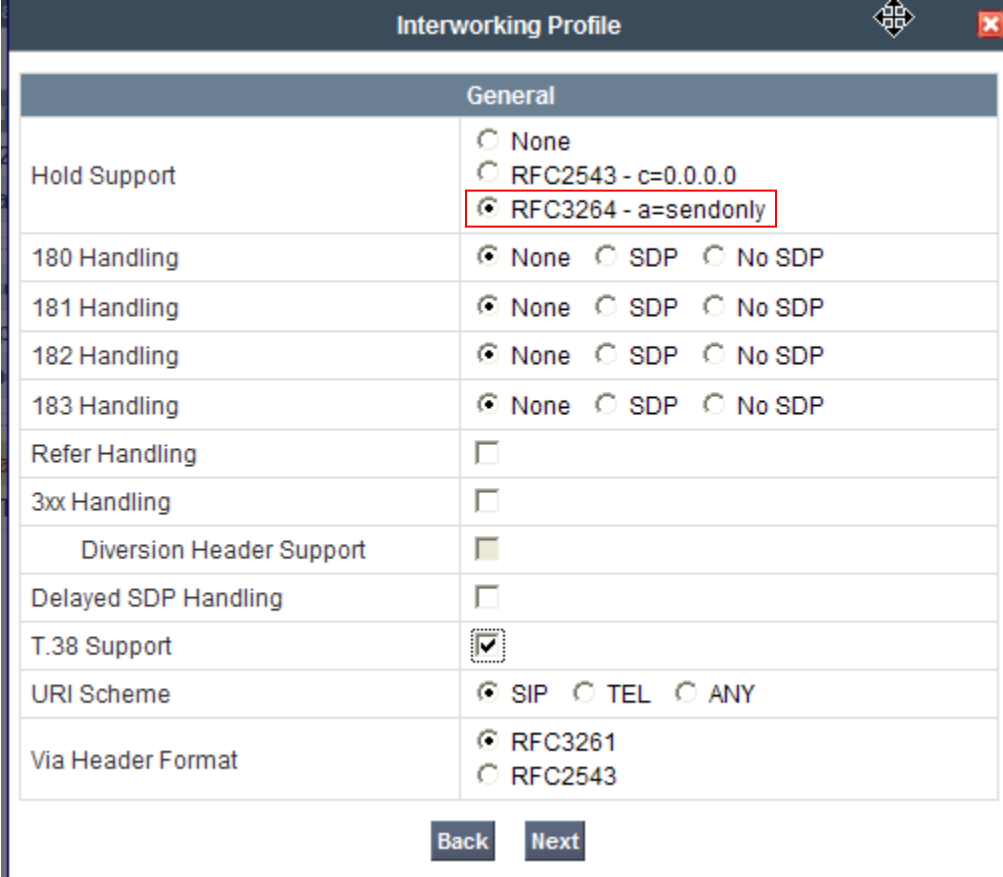
Enabling Network Call Redirection (NCR) on Communication Manager (see **Section 7.2**), also enables Communication Manager's use of SendOnly/RecvOnly to signal hold conditions. The AT&T IP Transfer Connect service also uses SendOnly to signal hold. Therefore the Avaya SBCE must be set to use RFC3264 to support this signaling.

**Step 1** – Following the procedures shown in [1], log in the Avaya SBCE GUI interface.

**Step 2** – Navigate to **Global Profiles → Server Interworking** and select the profile **Avaya** created in [1].

**Step 3** – In the **General** tab, scroll down and click on **Edit**. The Editing Profile window will open.

**Step 4** – In the **Hold Support** row, select **RFC3264**, and then click on **Next**.



General	
Hold Support	<input type="radio"/> None <input type="radio"/> RFC2543 - c=0.0.0.0 <input checked="" type="radio"/> RFC3264 - a=sendonly
180 Handling	<input checked="" type="radio"/> None <input type="radio"/> SDP <input type="radio"/> No SDP
181 Handling	<input checked="" type="radio"/> None <input type="radio"/> SDP <input type="radio"/> No SDP
182 Handling	<input checked="" type="radio"/> None <input type="radio"/> SDP <input type="radio"/> No SDP
183 Handling	<input checked="" type="radio"/> None <input type="radio"/> SDP <input type="radio"/> No SDP
Refer Handling	<input type="checkbox"/>
3xx Handling	<input type="checkbox"/>
Diversion Header Support	<input type="checkbox"/>
Delayed SDP Handling	<input type="checkbox"/>
T.38 Support	<input checked="" type="checkbox"/>
URI Scheme	<input checked="" type="radio"/> SIP <input type="radio"/> TEL <input type="radio"/> ANY
Via Header Format	<input checked="" type="radio"/> RFC3261 <input type="radio"/> RFC2543

Back Next

**Step 5** - The **Privacy** window will open. Make no changes to this window, and click on **Finish**.

The screenshot shows a window titled "Editing Profile: Avaya" with a close button in the top right corner. The window is divided into two main sections: "Privacy" and "DTMF".

**Privacy Section:**

Privacy Enabled	<input type="checkbox"/>
User Name	<input type="text"/>
P-Asserted-Identity	<input type="text"/>
P-Preferred-Identity	<input type="text"/>
Privacy Header	<input type="text"/>

**DTMF Section:**

DTMF Support	<input checked="" type="radio"/> None <input type="radio"/> SIP NOTIFY <input type="radio"/> SIP INFO
--------------	---

At the bottom of the window, there are two buttons: "Back" and "Finish".

**Note** – Due to interoperability issues described in **Section 2.2.1** (items **3** and **6**), additional Avaya SBCE provisioning may be required as temporary workarounds for these issues. These additional procedures are shown in **Addendums 1** and **2**.

## 9. Verification Steps

### 9.1. Call Verification Tests

The call verification steps and troubleshooting tools described for the AT&T Toll Free service described in [1], apply to the AT&T IP Transfer Connect service as well.

1. Place an inbound call to an AT&T IP Transfer Connect service line enabled with Redirect features. Verify that an appropriate Communication Manager vector immediately redirects the call back to the AT&T IP Transfer Connect service for redirection to an alternate destination using 302. Verify two-way talk path and transmission of UUI information as appropriate.
2. Place an inbound call to an AT&T IP Transfer Connect service line enabled with Refer features. Verify that an appropriate Communication Manager vector answers the call and then redirects the call back to the AT&T IP Transfer Connect service for redirection to an alternate destination using Refer. Verify two-way talk path and transmission of UUI information as appropriate.
3. Verify that when Communication Manager is the transfer target of redirected calls, the calls are answered with two-way talk path. Verify that the calls remain stable for several minutes and disconnect properly.

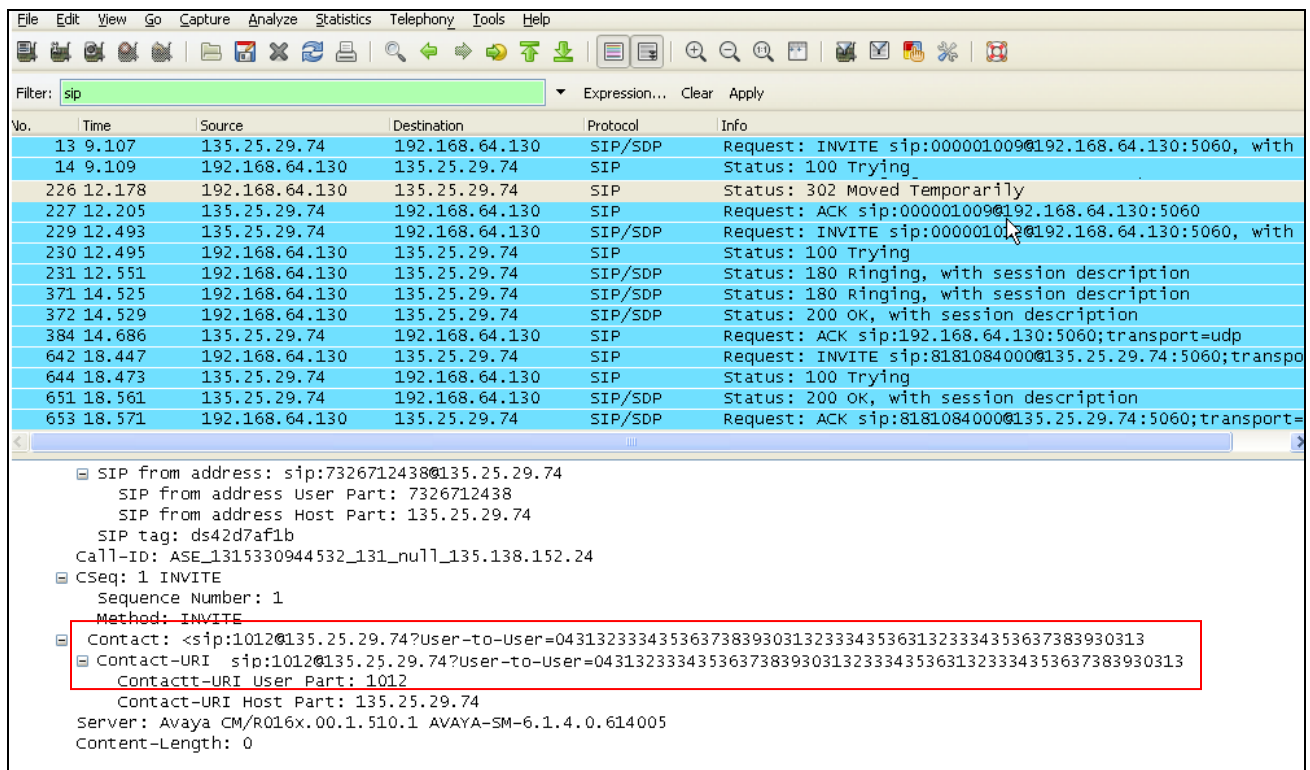


## 9.2. Protocol Traces

Using a SIP protocol analyzer (e.g. Wireshark), monitor the SIP traffic at the Avaya SBCE public “outside” interface connection to the AT&T IP Transfer Connect service.

### 9.2.1. 302 Redirection

The following is an example of a 302 redirection call filtering on the SIP protocol. Note that the Contact header contains the new called number (1012) as defined in **Section 6.3**. Also note the UUI information.



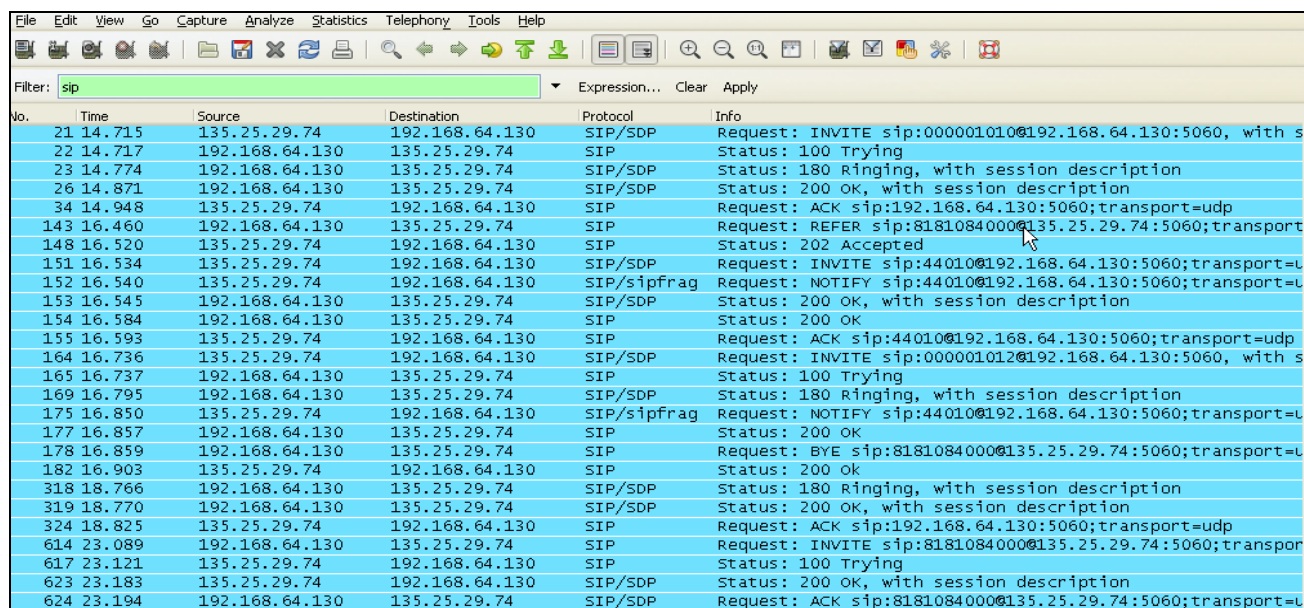
No.	Time	Source	Destination	Protocol	Info
13	9.107	135.25.29.74	192.168.64.130	SIP/SDP	Request: INVITE sip:000001009@192.168.64.130:5060, with
14	9.109	192.168.64.130	135.25.29.74	SIP	Status: 100 Trying
226	12.178	192.168.64.130	135.25.29.74	SIP	Status: 302 Moved Temporarily
227	12.205	135.25.29.74	192.168.64.130	SIP	Request: ACK sip:000001009@192.168.64.130:5060
229	12.493	135.25.29.74	192.168.64.130	SIP/SDP	Request: INVITE sip:000001012@192.168.64.130:5060, with
230	12.495	192.168.64.130	135.25.29.74	SIP	Status: 100 Trying
231	12.551	192.168.64.130	135.25.29.74	SIP/SDP	Status: 180 Ringing, with session description
371	14.525	192.168.64.130	135.25.29.74	SIP/SDP	Status: 180 Ringing, with session description
372	14.529	192.168.64.130	135.25.29.74	SIP/SDP	Status: 200 OK, with session description
384	14.686	135.25.29.74	192.168.64.130	SIP	Request: ACK sip:192.168.64.130:5060;transport=udp
642	18.447	192.168.64.130	135.25.29.74	SIP	Request: INVITE sip:8181084000@135.25.29.74:5060;transport=
644	18.473	135.25.29.74	192.168.64.130	SIP	Status: 100 Trying
651	18.561	135.25.29.74	192.168.64.130	SIP/SDP	Status: 200 OK, with session description
653	18.571	192.168.64.130	135.25.29.74	SIP/SDP	Request: ACK sip:8181084000@135.25.29.74:5060;transport=

<div>SIP from address: sip:7326712438@135.25.29.74 SIP from address User Part: 7326712438 SIP from address Host Part: 135.25.29.74 SIP tag: ds42d7af1b Call-ID: ASE_1315330944532_131_nu11_135.138.152.24</div> <div>CSeq: 1 INVITE Sequence Number: 1 Method: INVITE</div> <div>Contact: &lt;sip:1012@135.25.29.74?User-to-User=043132333435363738393031323334353631323334353637383930313 Contact-URI sip:1012@135.25.29.74?User-to-User=043132333435363738393031323334353631323334353637383930313 Contact-URI User Part: 1012 Contact-URI Host Part: 135.25.29.74 Server: Avaya CM/R016x.00.1.510.1 AVAYA-SM-6.1.4.0.614005 Content-Length: 0</div>
---

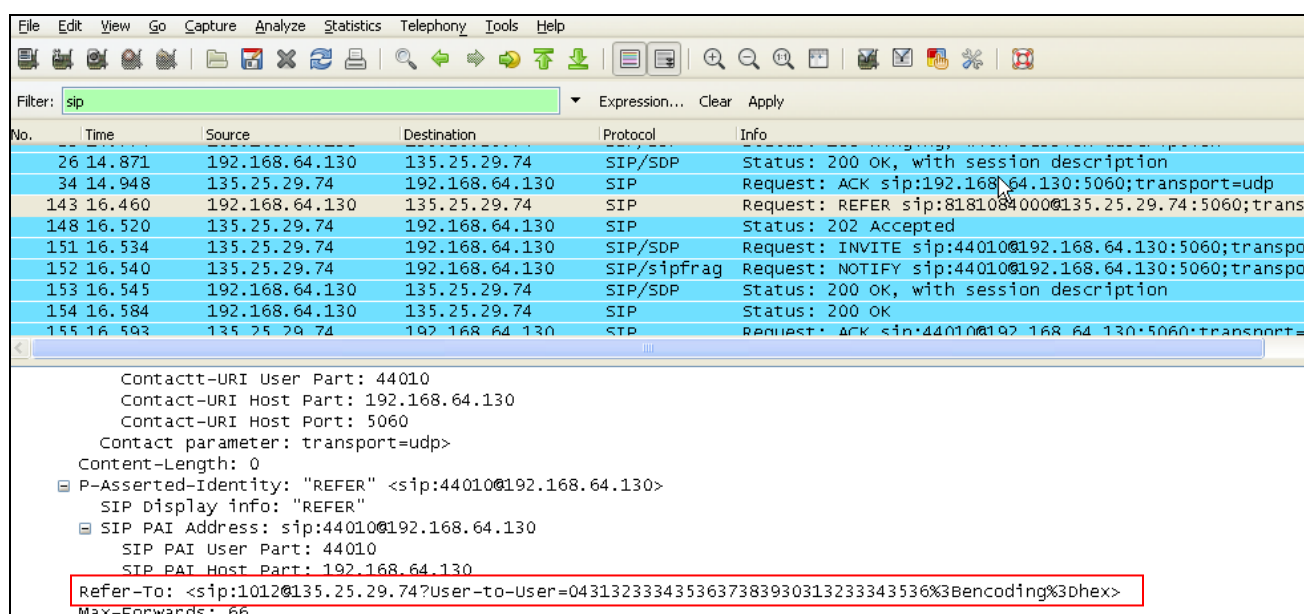
## 9.2.2. Refer with 180

The following is an example of a Refer (frame 143) redirection call filtering on the SIP protocol. Note that Communication Manager is sending 180 Ringing. As described in **Section 2.2.1**, this causes Communication Manager to send a BYE (frame 178) upon receipt of the Notify/180 Ringing sent by AT&T in frame 175.



No.	Time	Source	Destination	Protocol	Info
21	14.715	135.25.29.74	192.168.64.130	SIP/SDP	Request: INVITE sip:000001010@192.168.64.130:5060, with s
22	14.717	192.168.64.130	135.25.29.74	SIP	Status: 100 Trying
23	14.774	192.168.64.130	135.25.29.74	SIP/SDP	Status: 180 Ringing, with session description
26	14.871	192.168.64.130	135.25.29.74	SIP/SDP	Status: 200 OK, with session description
34	14.948	135.25.29.74	192.168.64.130	SIP	Request: ACK sip:192.168.64.130:5060;transport=udp
143	16.460	192.168.64.130	135.25.29.74	SIP	Request: REFER sip:8181084000@135.25.29.74:5060;transport=
148	16.520	135.25.29.74	192.168.64.130	SIP	Status: 202 Accepted
151	16.534	135.25.29.74	192.168.64.130	SIP/SDP	Request: INVITE sip:44010@192.168.64.130:5060;transport=U
152	16.540	135.25.29.74	192.168.64.130	SIP/sipfrag	Request: NOTIFY sip:44010@192.168.64.130:5060;transport=U
153	16.545	192.168.64.130	135.25.29.74	SIP/SDP	Status: 200 OK, with session description
154	16.584	192.168.64.130	135.25.29.74	SIP	Status: 200 OK
155	16.593	135.25.29.74	192.168.64.130	SIP	Request: ACK sip:44010@192.168.64.130:5060;transport=udp
164	16.736	135.25.29.74	192.168.64.130	SIP/SDP	Request: INVITE sip:000001012@192.168.64.130:5060, with s
165	16.737	192.168.64.130	135.25.29.74	SIP	Status: 100 Trying
169	16.795	192.168.64.130	135.25.29.74	SIP/SDP	Status: 180 Ringing, with session description
175	16.850	135.25.29.74	192.168.64.130	SIP/sipfrag	Request: NOTIFY sip:44010@192.168.64.130:5060;transport=U
177	16.857	192.168.64.130	135.25.29.74	SIP	Status: 200 OK
178	16.859	192.168.64.130	135.25.29.74	SIP	Request: BYE sip:8181084000@135.25.29.74:5060;transport=U
182	16.903	135.25.29.74	192.168.64.130	SIP	Status: 200 OK
318	18.766	192.168.64.130	135.25.29.74	SIP/SDP	Status: 180 Ringing, with session description
319	18.770	192.168.64.130	135.25.29.74	SIP/SDP	Status: 200 OK, with session description
324	18.825	135.25.29.74	192.168.64.130	SIP	Request: ACK sip:192.168.64.130:5060;transport=udp
614	23.089	192.168.64.130	135.25.29.74	SIP	Request: INVITE sip:8181084000@135.25.29.74:5060;transpor
617	23.121	135.25.29.74	192.168.64.130	SIP	Status: 100 Trying
623	23.183	135.25.29.74	192.168.64.130	SIP/SDP	Status: 200 OK, with session description
624	23.194	192.168.64.130	135.25.29.74	SIP/SDP	Request: ACK sip:8181084000@135.25.29.74:5060;transport=U

This screen shows the Refer in frame 143 in detail. The Refer-To header specifies the new called number (1012) as defined in **Section 6.3**. Also note the UUI information.



No.	Time	Source	Destination	Protocol	Info
26	14.871	192.168.64.130	135.25.29.74	SIP/SDP	Status: 200 OK, with session description
34	14.948	135.25.29.74	192.168.64.130	SIP	Request: ACK sip:192.168.64.130:5060;transport=udp
143	16.460	192.168.64.130	135.25.29.74	SIP	Request: REFER sip:8181084000@135.25.29.74:5060;trans
148	16.520	135.25.29.74	192.168.64.130	SIP	Status: 202 Accepted
151	16.534	135.25.29.74	192.168.64.130	SIP/SDP	Request: INVITE sip:44010@192.168.64.130:5060;transport=
152	16.540	135.25.29.74	192.168.64.130	SIP/sipfrag	Request: NOTIFY sip:44010@192.168.64.130:5060;transport=
153	16.545	192.168.64.130	135.25.29.74	SIP/SDP	Status: 200 OK, with session description
154	16.584	192.168.64.130	135.25.29.74	SIP	Status: 200 OK
155	16.593	135.25.29.74	192.168.64.130	SIP	Request: ACK sip:44010@192.168.64.130:5060;transport=

Contact-URI User Part: 44010
Contact-URI Host Part: 192.168.64.130
Contact-URI Host Port: 5060
Contact parameter: transport=udp>
Content-Length: 0
P-Asserted-Identity: "REFER" <sip:44010@192.168.64.130>
SIP Display info: "REFER"
SIP PAI Address: sip:44010@192.168.64.130
SIP PAI User Part: 44010
SIP PAI Host Part: 192.168.64.130
Refer-To: <sip:1012@135.25.29.74?User-to=User=0431323334353637383930313233343536%3Bencoding%3Dhex>
Max-Forwards: 66

This screen shows the Notify/180 Ringing sent by AT&T in frame 175. Note that this Notify is in response to the 180 Ringing sent in frame 169.

Filter: sip

No.	Time	Source	Destination	Protocol	Info
153	16.543	192.168.64.130	135.25.29.74	SIP/SDP	Status: 200 OK, with session description
154	16.584	192.168.64.130	135.25.29.74	SIP	Status: 200 OK
155	16.593	135.25.29.74	192.168.64.130	SIP	Request: ACK sip:44010@192.168.64.130:5060;transport=udp
164	16.736	135.25.29.74	192.168.64.130	SIP/SDP	Request: INVITE sip:000001012@192.168.64.130:5060, with session description
165	16.737	192.168.64.130	135.25.29.74	SIP	Status: 100 Trying
169	16.795	192.168.64.130	135.25.29.74	SIP/SDP	Status: 180 Ringing, with session description
175	16.850	135.25.29.74	192.168.64.130	SIP/sipfrag	Request: NOTIFY sip:44010@192.168.64.130:5060;transport=udp
177	16.857	192.168.64.130	135.25.29.74	SIP	Status: 200 OK
178	16.859	192.168.64.130	135.25.29.74	SIP	Request: BYE sip:8181084000@135.25.29.74:5060;transport=udp

Contact: <sip:8181084000@135.25.29.74:5060;transport=udp>  
 Contact-URI: sip:8181084000@135.25.29.74:5060;transport=udp  
 Contact-URI User Part: 8181084000  
 Contact-URI Host Part: 135.25.29.74  
 Contact-URI Host Port: 5060  
 Contact parameter: transport=udp  
 Event: refer;id=1  
 Subscription-State: active;expires=119  
 Content-Type: message/sipfrag;version=2.0  
 Message Body  
 sipfrag  
 SIP/2.0 180 Ringing

### 9.2.3. Refer with 183

The following is an example of a Refer (frame 134) redirection call filtering on the SIP protocol. Note that Communication Manager is sending 183 Session Progress. As described in **Section 2.2.1**, this causes Communication Manager to send a BYE (frame 323) upon receipt of the Notify/183 Session Progress sent by AT&T in frame 316.

Filter: sip

No.	Time	Source	Destination	Protocol	Info
11	8.186	135.25.29.74	192.168.64.130	SIP/SDP	Request: INVITE sip:000001010@192.168.64.130:5060, with session description
12	8.188	192.168.64.130	135.25.29.74	SIP	Status: 100 Trying
13	8.204	192.168.64.130	135.25.29.74	SIP/SDP	Status: 183 Session Progress, with session description
18	8.302	192.168.64.130	135.25.29.74	SIP/SDP	Status: 200 OK, with session description
26	8.382	135.25.29.74	192.168.64.130	SIP	Request: ACK sip:192.168.64.130:5060;transport=udp
134	9.890	192.168.64.130	135.25.29.74	SIP	Request: REFER sip:8181084000@135.25.29.74:5060;transport=udp
139	9.948	135.25.29.74	192.168.64.130	SIP	Status: 202 Accepted
141	9.963	135.25.29.74	192.168.64.130	SIP/SDP	Request: INVITE sip:44010@192.168.64.130:5060;transport=udp
142	9.970	135.25.29.74	192.168.64.130	SIP/sipfrag	Request: NOTIFY sip:44010@192.168.64.130:5060;transport=udp
143	9.975	192.168.64.130	135.25.29.74	SIP/SDP	Status: 200 OK, with session description
145	10.013	192.168.64.130	135.25.29.74	SIP	Status: 200 OK
146	10.028	135.25.29.74	192.168.64.130	SIP	Request: ACK sip:44010@192.168.64.130:5060;transport=udp
154	10.169	135.25.29.74	192.168.64.130	SIP/SDP	Request: INVITE sip:000001012@192.168.64.130:5060, with session description
155	10.171	192.168.64.130	135.25.29.74	SIP	Status: 100 Trying
158	10.217	192.168.64.130	135.25.29.74	SIP/SDP	Status: 183 Session Progress, with session description
164	10.271	135.25.29.74	192.168.64.130	SIP/sipfrag	Request: NOTIFY sip:44010@192.168.64.130:5060;transport=udp
165	10.277	192.168.64.130	135.25.29.74	SIP	Status: 200 OK
311	12.191	192.168.64.130	135.25.29.74	SIP/SDP	Status: 183 Session Progress, with session description
313	12.201	192.168.64.130	135.25.29.74	SIP/SDP	Status: 200 OK, with session description
316	12.245	135.25.29.74	192.168.64.130	SIP/sipfrag	Request: NOTIFY sip:44010@192.168.64.130:5060;transport=udp
318	12.251	135.25.29.74	192.168.64.130	SIP/sipfrag	Request: NOTIFY sip:44010@192.168.64.130:5060;transport=udp
319	12.252	192.168.64.130	135.25.29.74	SIP	Status: 200 OK
320	12.255	135.25.29.74	192.168.64.130	SIP	Request: ACK sip:192.168.64.130:5060;transport=udp
322	12.258	192.168.64.130	135.25.29.74	SIP	Status: 200 OK
323	12.259	192.168.64.130	135.25.29.74	SIP	Request: BYE sip:8181084000@135.25.29.74:5060;transport=udp
326	12.299	135.25.29.74	192.168.64.130	SIP	Status: 200 OK
622	16.641	192.168.64.130	135.25.29.74	SIP	Request: INVITE sip:8181084000@135.25.29.74:5060;transport=udp
625	16.671	135.25.29.74	192.168.64.130	SIP	Status: 100 Trying
630	16.733	135.25.29.74	192.168.64.130	SIP/SDP	Status: 200 OK, with session description
631	16.744	192.168.64.130	135.25.29.74	SIP/SDP	Request: ACK sip:8181084000@135.25.29.74:5060;transport=udp

This screen shows the Notify/183 Session Progress sent by AT&T in frame 316. Note that this Notify is in response to the 183 Session Progress sent in frame 311.

The image shows a Wireshark packet capture of SIP messages. The packet list pane shows several frames, with frame 316 selected. The packet details pane for frame 316 shows the following structure:

- Call-ID: ASE\_1317554708878\_175\_1001\_135.138.132.24
- CSeq: 5 NOTIFY
  - Sequence Number: 5
  - Method: NOTIFY
  - Content-Length: 28
- Contact: <sip:8181084000@135.25.29.74:5060;transport=udp>
  - Contact-URI: sip:8181084000@135.25.29.74:5060;transport=udp
    - Contact-URI User Part: 8181084000
    - Contact-URI Host Part: 135.25.29.74
    - Contact-URI Host Port: 5060
    - Contact parameter: transport=udp
- Event: refer;id=1
- Subscription-State: active;expires=117
- Content-Type: message/sipfrag;version=2.0
- Message Body
  - Sipfrag
    - SIP/2.0 183 Session Progress

## 10. Conclusion

As illustrated in these Application Notes, Avaya Aura® Session Manager, Avaya Aura® Communication Manager, and the Avaya Session Border Controller for Enterprise can be configured to interoperate successfully with the AT&T IP Transfer Connect service. In addition, these Application Notes further demonstrate that the Avaya Aura® Communication Manager SIP Network Call Redirection (NCR) and User-to-User Information (UII) features can work in complement with the AT&T implementations of SIP NCR and UII to support call redirection over SIP trunks while preserving initiating caller information. This solution provides contact center users of Avaya Aura® Communication Manager the ability to redirect inbound AT&T IP Transfer Connect service calls to alternate destinations (using Refer and 302 redirection), and deliver UII-encoded customer information to those alternate destinations for the purposes of invoking contact center applications, e.g., triggering agent screen pop-ups with caller information, etc. Both intra-site and IP Transfer Connect call scenarios were tested.

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.

## 11. References

The Avaya product documentation is available at <http://support.avaya.com> unless otherwise noted.

- [1] *Applications Notes for Avaya Aura® Communication Manager 6.0.1, Avaya Aura® Session Manager 6.1 and Avaya Session Border Controller for Enterprise AT&T IP Toll Free SIP Trunk Service, Version 1.0*

### **Avaya Aura® Session Manager/System Manager**

- [2] *Administering Avaya Aura® Session Manager*, Doc ID 03-603324, Issue 4, May 2011
- [3] *Installing and Configuring Avaya Aura® Session Manager*, Doc ID 03-603473 Issue 2.2, April 2011
- [4] *Maintaining and Troubleshooting Avaya Aura® Session Manager*, Doc ID 03-603325, Issue 4.1, March 2011
- [5] *Administering Avaya Aura® System Manager*, Document Number 03-603324, June 2010

### **Avaya Aura® Communication Manager**

- [6] *Administering Avaya Aura® Communication Manager*, Release 6.003-300509, Issue 6.0, June 2010
- [7] *Administering Avaya Aura® Call Center Features*, Release 6.0, June 2010
- [8] *Programming Call Vectors in Avaya Aura® Call Center*, 6.0, June 2010

### **Avaya Session Border Controller for Enterprise**

Product documentation for UC-Sec can be obtained from Sipera using the link at <http://www.sipera.com>.

- [9] *E-SBC 1U Installation Guide, Release 4.0.5*, Part Number: 101-5225-405v1.00, Release Date: November 2011
- [10] *E-SBC Administration Guide, Release 4.0.5*, Part Number: 010-5424-405v1.00, Release Date: November 2011

## 12. Addendum 1 – Additional provisioning for customers that use both AT&T IP Transfer Connect and IP Flexible Reach services.

The Avaya Aura® Communication Manager Network Call Redirection (NCR) feature is required to enable Refer and 302 call redirection with the AT&T IP Transfer Connect Service (see **Section 6.2**). With this feature enabled, Avaya Aura® Communication Manager will also use the SIP parameter *SendOnly* to signal any hold call conditions. The *SendOnly* SIP parameter is not supported by the AT&T Flexible Reach service.

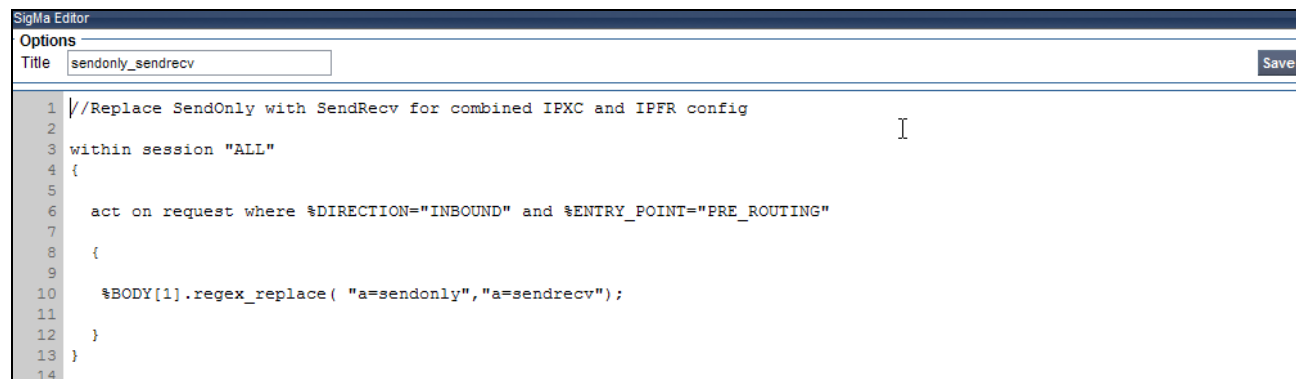
Any customers that access both AT&T IP Transfer Connect and AT&T IP Flexible Reach services via the same Avaya Aura® Communication Manager environment, must use the following procedures to have the Avaya SBCE replace the *SendOnly* parameter with the *SendRecv* parameter that the AT&T Flexible Reach service does support.

Refer to [1] for more information on configuring the Avaya SBCE. Refer to [8] and [9] for information on the Avaya SBE scripting language.

1. Select **Global Profiles** from the menu on the left-hand side.
2. Select **Signaling Manipulation**.
3. Click **Add Script** (not shown) and the script editor window will open.
4. Enter a name for the script in the **Title** box (e.g., **sendonly\_sendrecv**).
5. The following script is defined:

```
//Replace SendOnly with SendRecv for combined IPXC and IPFR config

within session "ALL"
{
    act on request where %DIRECTION="INBOUND" and %ENTRY_POINT="PRE_ROUTING"
    {
        %BODY[1].regex_replace( "a=sendonly","a=sendrecv");
    }
}
```



6. Click on **Save**. The script editor will test for any errors, and the editor window will close.

7. If changes are required, click on the **Edit** button.
8. Select **Sever Configuration** from the menu on the left-hand side.
9. Select the **AvayaSM** profile created in [1], **Section 8.4.5**.
10. Select the **Advanced** tab, and click on **Edit**. The edit window will open
11. In the **Interworking Profile** verify that **Avaya** is selected (defined in [1]).
12. In the **Signaling Manipulation Script** row, select the script created in **Step 4** above (e.g., **sendonly\_sendrecv**). Click on **Finish**.

Enable DoS Protection	<input type="checkbox"/>
Enable Grooming	<input type="checkbox"/>
Interworking Profile	Avaya
Signaling Manipulation Script	sendonly_sendrecv
TCP Connection Type	<input checked="" type="radio"/> SUBID <input type="radio"/> PORTID <input type="radio"/> MAPPING
UDP Connection Type	<input checked="" type="radio"/> SUBID <input type="radio"/> PORTID <input type="radio"/> MAPPING

**Finish**

UC-Sec Control Center

Welcome ucsec, you signed in as Admin. Current server time is 9:35:24 AM GMT

Alarms Incidents Statistics Logs Diagnostics Users Logout Help

System Management

- Global Parameters
- Global Profiles
  - Domain DoS
  - Fingerprint
  - Server Interworking
  - Phone Interworking
  - Media Forking
  - Routing
  - Server Configuration
  - Subscriber Profiles
  - Topology Hiding
  - Signaling Manipulation
- SIP Cluster
- Domain Policies
- Application Rules

Global Profiles > Server Configuration: AvayaSM

Add Profile Profile AvayaSM ATT

Rename Profile Clone Profile Delete Profile

General Authentication Heartbeat Advanced

Advanced	
Enable DoS Protection	<input type="checkbox"/>
Enable Grooming	<input type="checkbox"/>
Interworking Profile	Avaya
Signaling Manipulation Script	sendonly_sendrecv
TCP Connection Type	SUBID
UDP Connection Type	SUBID

**Edit**

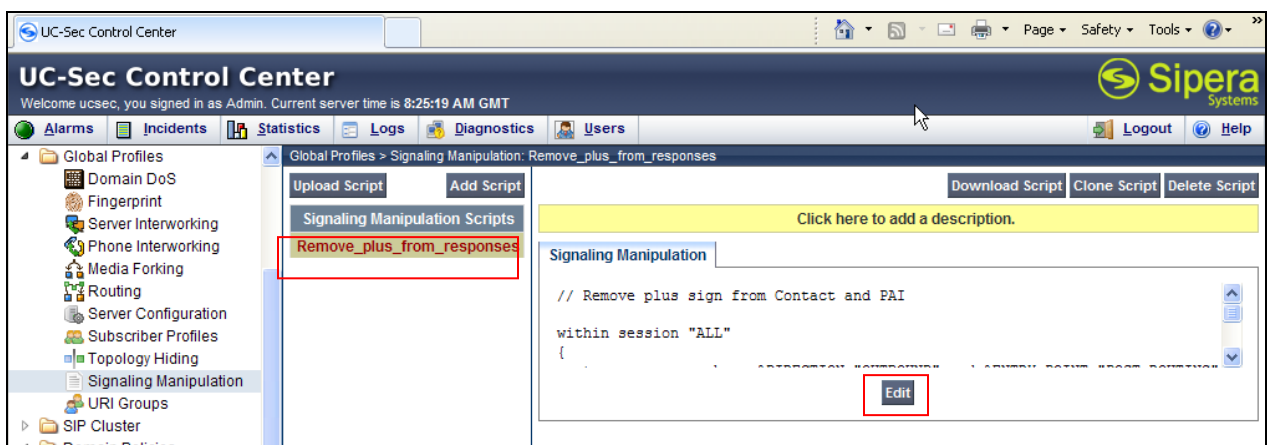


## 13. Addendum 2 – Workaround for Avaya SBCE Refer/401 Issue.

As noted in **Section 2.2.1**, an issue was found with the Avaya SBCE in regards to Refer and 401 Unauthorized call processing. A test case specifies that the CPE generate a Refer using the wrong DNIS access number. The AT&T IP Transfer Connect service then responds with 401 Unauthorized. This 401 has a Contact header containing an AT&T internal IP address (not routable from the CPE). When the CPE generated a subsequent ReInvite, the Avaya SBCE used the IP address defined in the 401 Contact header as the Internet Protocol Destination header IP address, and not the AT&T Border Element IP address. An MR was opened with Avaya SBCE support. Until a fix is delivered, the following workaround may be used to alleviate this issue.

The following addition scripting needs to be added to the existing **Remove\_plus\_from\_responses** script defined in **Section 8.4.9** of [1], (refer to [8] and [9] for information on the Avaya SBE Sigma scripting language).

1. Select **Global Profiles** from the menu on the left-hand side.
2. Select **Signaling Manipulation**.
3. Click on the existing **Remove\_plus\_from\_responses** script listing and then click on **Edit**.  
The Sigma editor window will open.



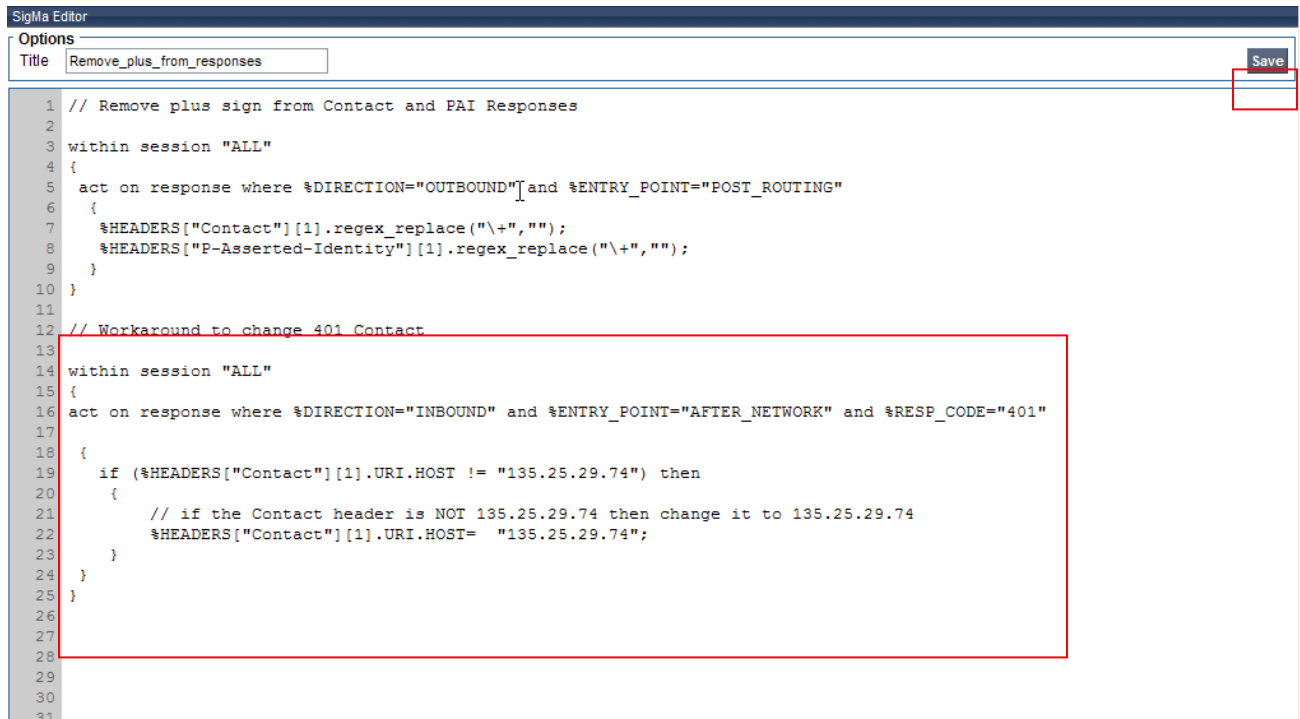


4. Enter the following additional lines below the existing script:

```
// Workaround to change 401 Contact

within session "ALL"
{
act on response where %DIRECTION="INBOUND" and %ENTRY_POINT="AFTER_NETWORK" and %RESP_CODE="401"
{
  if (%HEADERS["Contact"][1].URI.HOST != "135.25.29.74") then
  {
    // if the Contact header is NOT 135.25.29.74 then change it to 135.25.29.74
    %HEADERS["Contact"][1].URI.HOST= "135.25.29.74";
  }
}
}
```

5. Click on **Save** to update the existing script.



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