



Avaya Solution & Interoperability Test Lab

Application Notes for Avaya Aura® Communication Manager 6.2 and Avaya Session Border Controller for Enterprise with AT&T IP Transfer Connect Service – Issue 1.0

Abstract

These Application Notes describe the steps for configuring Avaya Aura® Communication Manager 6.2 with SIP Network Call Redirection (NCR) and the Avaya Session Border Controller for Enterprise 4.0.5, over the AT&T IP Transfer Connect service using AT&T 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 (UI) features can be utilized together, in conjunction with the Data Forwarding option of the AT&T IP Transfer Connect service, to transmit UI within SIP signaling messages to the alternate destinations.

Note that these Application Notes are intended to supplement the separate document: *Applications Notes for Avaya Aura® Communication Manager 6.2 and Avaya Session Border Controller for Enterprise with AT&T IP Toll Free SIP Trunk 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 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 6.2 with SIP Network Call Redirection (NCR) and the Avaya Session Border Controller for Enterprise 4.0.5, (referred to in the remainder of this document as *Avaya SBCE*), with the AT&T IP Transfer Connect service using AT&T **AVPN** or **MIS/PNT** transport connections¹. 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 alternate² 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 at 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 (UII) feature can be utilized with the SIP NCR feature to transmit UII 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 UII 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: *Application Notes for Avaya Aura® Communication Manager 6.2 and Avaya Session Border Controller for Enterprise with AT&T IP Toll Free SIP Trunk Service*. It is recommended that this AT&T IP Toll Free service document should be available as a reference during provisioning of the AT&T IP Transfer Connect service.

2. General Test Approach and Test Results

The test environment consisted of:

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

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.

2.1. Interoperability Compliance Testing

The interoperability compliance testing focused on verifying inbound call flows (see **Section 3.2** for descriptions) to Communication Manager, and subsequent redirection messages to AT&T for rerouting to alternate destinations.

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 local transfer.

Note – Documents used to provision the reference configuration are listed in **Section 10**. Specific references to these documents are indicated in the following sections by the notation [x], where x is the document reference number.

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. **302 redirections and 180 Ringing** - The IP Transfer Connect service specifies that 18x responses should not be used in conjunction with 302 redirection calls. Therefore ringback should not be specified in Communication Manager 302 redirection vectors (see **Section 5.3.1**).

2. **Communication Manager 18x responses and Refer** - The Communication Manager 6.2 SIP trunk form may be configured to send either a 180 Ringing message (default), or a 183 Session Progress message (see **Section 5.2**). The message type selected alters the Communication Manager behavior upon receipt of a Notify from 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. **Customers subscribing to AT&T IP Transfer Connect and IP Flexible Reach Enhanced Features services** - 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 5.2**). With this feature enabled, Communication Manager will also use the SIP parameter *SendOnly* to signal any hold call conditions (as well as the *RecvOnly* response as required). The *SendOnly* SIP parameter is not currently supported by the AT&T IP Flexible Reach - Enhanced Features service (see document [10]). Any customers subscribing to both AT&T IP Transfer Connect and AT&T IP Flexible Reach - Enhanced Features services via the same Communication Manager environment, must use the procedures described in **Section 6.1** of this document. That section describes having the Avaya SBCE replace the *SendOnly* parameter with the *SendRecv* parameter that the AT&T IP Flexible Reach - Enhanced Features service does support.

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:

- Communication Manager 6.2 provides the voice communication services for a particular enterprise site. Avaya H.323 endpoints register to Communication Manager.
- The Avaya Media Gateway provides the physical interfaces and resources for Communication Manager. In the reference configuration, an Avaya G430 Media Gateway is used. This solution is extensible to other Avaya Media Gateways.

- Avaya 1603 (H.323), 96x0 Series IP Telephones (running H.323 firmware), and 96x1 Series IP Telephones (running H.323 firmware) were tested, as well as Avaya 6211 Analog Telephones, and Avaya one-X® Agent soft phone (H323). Note that Avaya SIP endpoints are not supported in configurations without Avaya Aura® Session Manager.
- The Avaya SBCE provides SIP Session Border Controller (SBC) functionality, including address translation and SIP header manipulation between the AT&T IP Toll Free service and the enterprise internal network.
- The AT&T IP Transfer Connect service uses SIP over UDP to communicate with enterprise edge SIP devices, (e.g., the Avaya SBCE). Communication Manager may use SIP over TCP or TLS to communicate with SIP network elements, (e.g., the Avaya SBCE). In the reference configuration, Communication Manager uses SIP over TCP to communicate with the Avaya SBCE to facilitate network tracing.
- Avaya Aura® Messaging was used in the reference configuration to provide voice messaging capabilities. 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 Communication Manager. Communication Manager terminates the call to the appropriate agent/phone. The Avaya H.323 telephones in the enterprise register to the Communication Manager Processor Ethernet (Procr) interface.

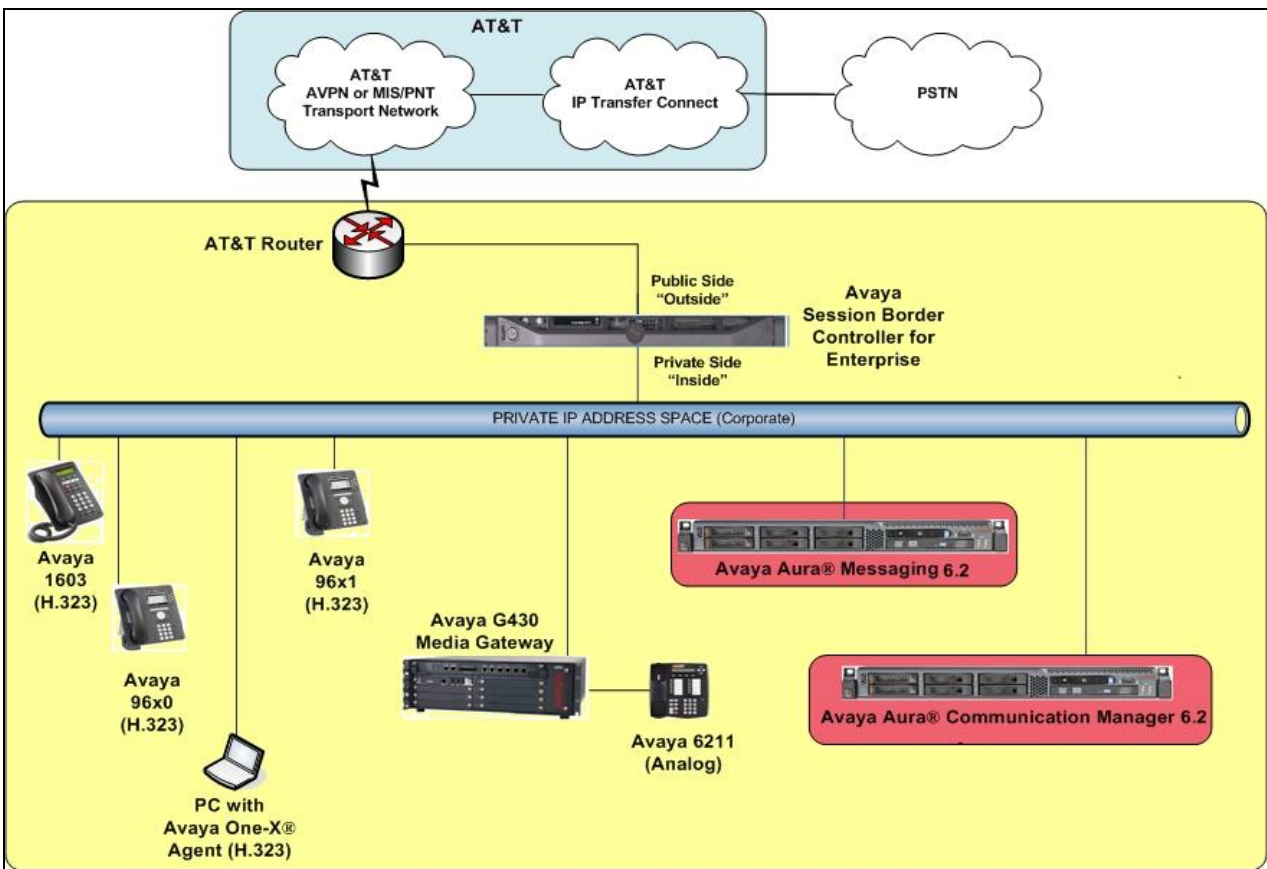


Figure 1: Reference configuration

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 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) are shown in this document as 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
Avaya Aura® Communication Manager	
IP Address	192.168.67.44
Avaya Aura® Communication Manager extensions	19xxx – Stations 4xxxx – Agents and VDNs
Voice Messaging Pilot Extension	36000
Avaya Session Border Controller for Enterprise (SBCE)	
IP Address of Outside (Public) Interface (to AT&T IP Flexible Reach-Enhanced Features Service)	192.168.64.130
IP Address of Inside (Private) Interface (connected to Avaya Aura® Communication Manager)	192.168.67.120
Avaya Aura Messaging	
IP Address	192.168.67.147
Messaging Mailboxes	19xxx & 4xxxx
AT&T IP Transfer Connect Service	
Border Element IP Address	135.25.29.74

Table 1: Illustrative Values Used in these Application Notes

3.2. Call Flows

To understand how inbound AT&T IP Transfer Connect service calls are handled by the Avaya SBCE 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 at the Avaya SBCE. IP address, domain, and header manipulations are applied and then forwarded 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 Communication Manager.
5. Depending on the called number, Communication Manager routes the call to a vector, which in turn routes the call to an agent or phone, or Communication Manager routes the call directly to an agent or phone.

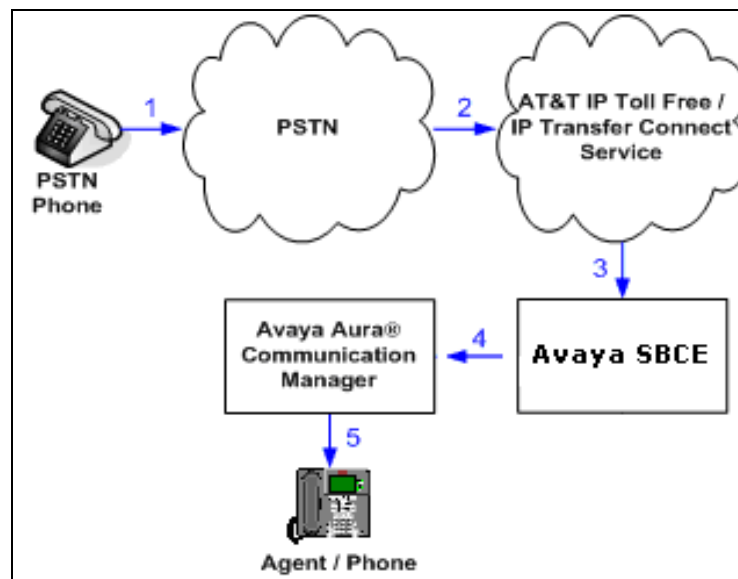


Figure 2: Inbound AT&T IP Transfer Connect Call – No Redirection

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 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 four 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 the SIP trunk on which the inbound call arrived. The SIP 302 message is routed through the Avaya SBCE, which sends the call to the AT&T IP Transfer Connect service. 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).

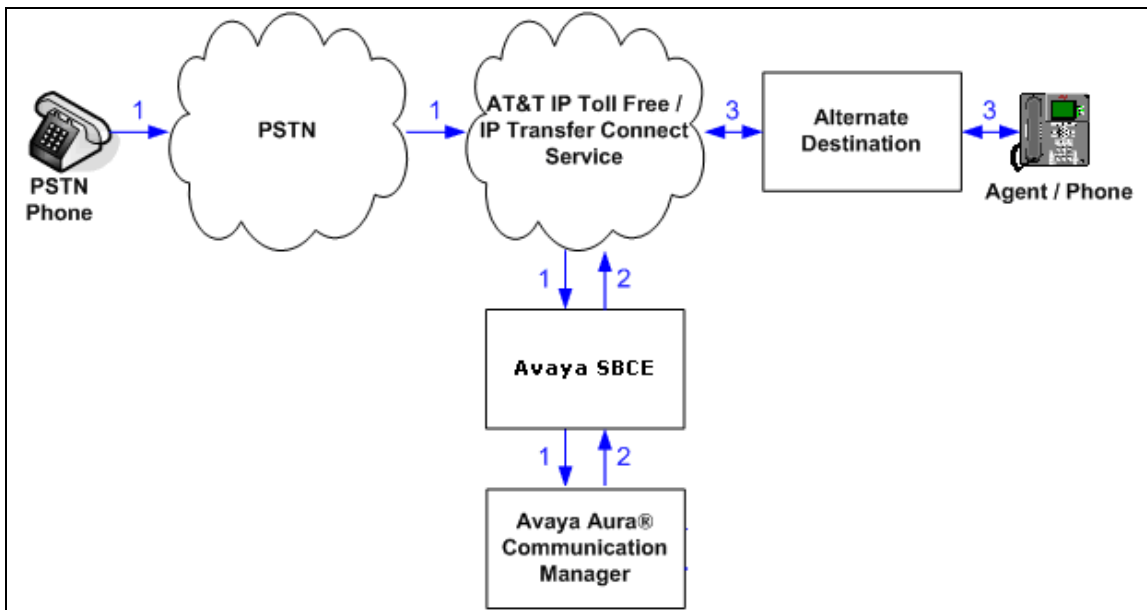


Figure 3: Inbound AT&T IP Transfer Connect Call – Pre-Answer SIP 302 Redirection

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

1. Same as the first four 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 number, and is routed back through the Avaya SBCE, which 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. Either Communication Manager or the AT&T IP Transfer Connect service clears the redirecting/referring party call.

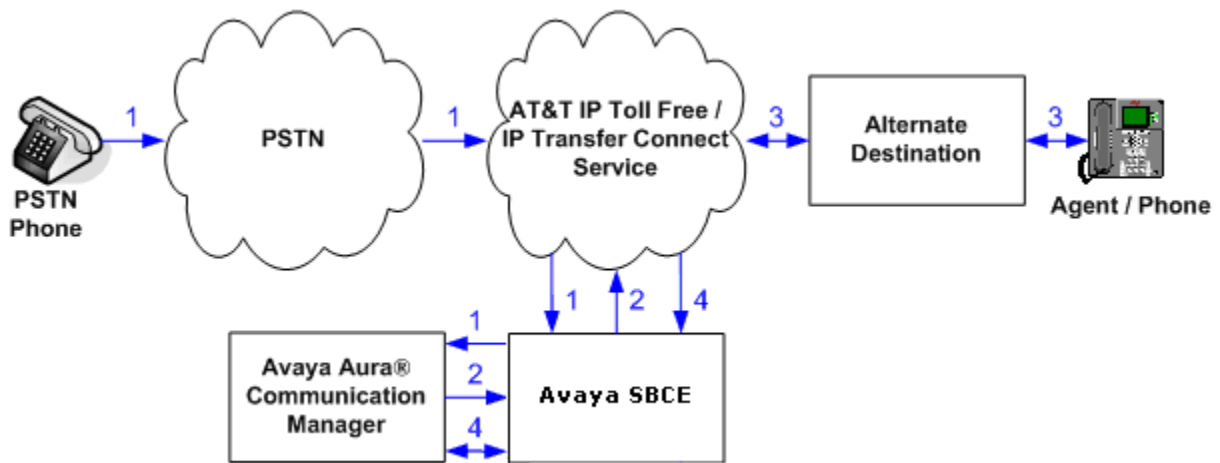


Figure 4: Inbound AT&T IP Transfer Connect Call – Post-Answer SIP Refer Redirection Successful

The fourth call scenario illustrated in **Figure 5** is similar to the third 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 a local 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.

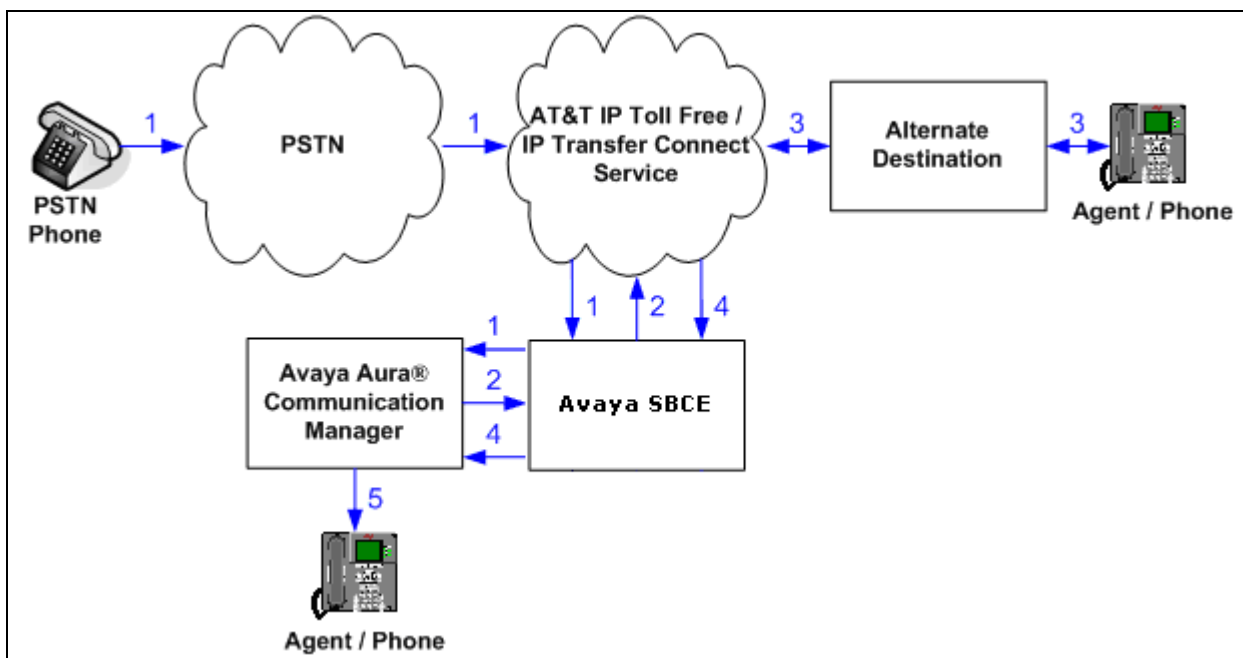


Figure 5: Inbound AT&T IP Transfer Connect Call - Post-Answer SIP Refer Redirection Unsuccessful, Communication Manager reroutes to Agent

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
HP Proliant DL360 G7 server/ <ul style="list-style-type: none"> • System Platform 6.2 • Avaya Aura® Communication Manager 6.2 	<ul style="list-style-type: none"> • System Platform 6.2.1.0.9 • 6.2 SP4 (02.0.823.0-20199)
Dell R610/Avaya Aura® Messaging 6.2	<ul style="list-style-type: none"> • System Platform 6.2.1.0.9 • 6.2 SP1 (MSG-02.0.823.0-109_0102)
Avaya G430 Media Gateway	31.20.0
MM711 Analog card	HW31 FW094
MM712 Digital card	HW07 FW011
Dell R210 V2/ Avaya Session Border Controller for Enterprise	4.0.5.Q19
Avaya 96x0 IP Telephone	H.323 Version S3.105S
Avaya 96x1 IP Telephone	H.323 Version S6.020S
Avaya one-X® Agent	2.5 SP1 Patch 1 (2.5.1072.11082)
Avaya 1603 IP Telephone	H323 (ha1603ua1_3200.bin)
Avaya 6424 Digital telephone	
Windows PC/ Ventafax Home Version (Fax device)	6.1.59.144
AT&T IP Transfer Connect service using AVPN/MIS-PNT transport service connection	VNI 26

Table 2: Equipment and Software Versions

5. Avaya Aura® Communication Manager

The Communication Manager administration for interaction with the AT&T IP Toll Free service is described in document [9] and is 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.

5.1. 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.

NOTE - For any required features that cannot be enabled in the steps that follow, contact an authorized Avaya account representative to obtain the necessary 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                                               ISDN/SIP Network Call Redirection? y
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 **ACD**, **EAS**, and **Vectoring** features are set to “y”.

```

display system-parameters customer-options                               Page 6 of 11
                                CALL CENTER OPTIONAL FEATURES
                                Call Center Release: 6.0
                                ACD? y                                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     Vectoring (Basic)? y
                                Dynamic Advocate? n                 Vectoring (Prompting)? y
                                Expert Agent Selection (EAS)? y      Vectoring (G3V4 Enhanced)? y
                                EAS-PHD? y                          Vectoring (3.0 Enhanced)? y
                                Forced ACD Calls? n                 Vectoring (ANI/II-Digits Routing)? y
                                Least Occupied Agent? y             Vectoring (G3V4 Advanced Routing)? y
                                Lookahead Interflow (LAI)? y        Vectoring (CINFO)? y
                                Multiple Call Handling (On Request)? y Vectoring (Best Service Routing)? y
                                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.)

```

5.2. Trunks

This section describes the steps for modifying the SIP trunk to the Avaya SBCE 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 document [9] for inbound AT&T IP Toll Free service calls (e.g., trunk 4).
 - On **Page 4** of the **trunk-group** form, set **Network Call Redirection** to **y**.
2. Note whether the setting for **Convert 180 to 183 for Early Media?** is **n** (default) or **y**. The value defined will alter the Refer NOTIFY response behavior (see **Section 2.2.1, item 2**).

```

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

```

5.3. 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 document [9].

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 vector generated SIP 302 message. In this case no audio announcement is generated by the vector.
- Post-Answer Redirection - An inbound AT&T IP Transfer Connect service call that invokes SIP NCR, using a vector generated SIP Refer message. In this case the vector does play an announcement back to the caller prior to the redirection.

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

NOTE - These Application Notes provide rudimentary vector and agent definitions to demonstrate and test the SIP NCR and UUI functionalities. In general, call centers will use vector and agent 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 UUI. The definition and documentation of those complex applications, and associated vectors and agents, are beyond the scope of these Application Notes. Consult documents [3] and [4] for further information.

5.3.1. Pre-Answer Redirection - 302

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 ringback (see **Section 2.2.1, item 1**), or generate an announcement. The vector does the following:

1. Assigns the data “**1234567890123456**” to ASAI UUI variable “**B**” (vector step **05**).
Note: The parameters for ASAI UUI variables “**A**” and “**B**”, and other vector variables are defined using the **change variables** command (see **Figure 6**).
2. Redirects the call to the number “**1012**” (vector step **08**), which is a predefined AT&T IP Transfer Connect access number. The presence of the “**~r**” in the “**route-to number**” triggers Communication Manager NCR processing 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 the Avaya SBCE).

change variables		VARIABLES FOR VECTORS					Page 1 of 39
Var	Description	Type	Scope	Length	Start	Assignment	VAC
A	UuiTest1	asaiuui	L	16	1		
B	UUI 16	asaiuui	L	16	1	C	

Figure 6: Change Variables Form

display vdn 19020		VECTOR DIRECTORY NUMBER					Page 1 of 3
Extension: 19020							
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		CALL VECTOR					Page 1 of 6
Number: 22		Name: 302NoRingUUI					
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							
02							
03							
04 #	Define UUI variable						
05 set	B	= none	CATR	1234567890123456			
06							
07 #	Redirect						
08 route-to	number ~r1012	with cov y if unconditionally					
09 stop							
10							

Figure 8: Sample Vector for Pre-Answer Redirection

5.3.2. Post-Answer Redirection - Refer

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:

1. Assigns the data “1234567890123456” to ASAI UUI variable “A” (vector step 02).
Note: The parameters for UUI variable “A” and other vector variables are defined using the **change variables** command (see **Figure 6**).
2. The vector plays an announcement, (vector step 04), which answers the call. This action of answering the call and sending media, distinguishes the Refer call processing from the 302 described in **Section 5.3.1**.
3. Attempts to redirect the call to the number “1012”, (vector step 06), which is a predefined AT&T IP Transfer Connect access number. The presence of the “~r” in the “**route-to number**” triggers Communication Manager NCR processing 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 the Avaya SBCE).

```
display vdn 19010                                     Page 1 of 3
              VECTOR DIRECTORY NUMBER
              Extension: 19010
              Name*: Refer
              Destination: Vector Number              15
              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*:
```

Figure 9: Sample VDN for Post-Answer Redirection

```
display vector 15                                     Page 1 of 6
              CALL VECTOR
              Number: 15                               Name: Refer UUI
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 #      Generate UUI
02 set          B          = none    CATR  1234567890123456
03 #      Play Refer announcement
04 announcement 42008
05 #      Refer occurs since this is post answer
06 route-to    number ~r1012          with cov y if unconditionally
07 stop
```

Figure 10: Sample Vector for Post-Answer Redirection

6. Avaya Session Border Controller for Enterprise

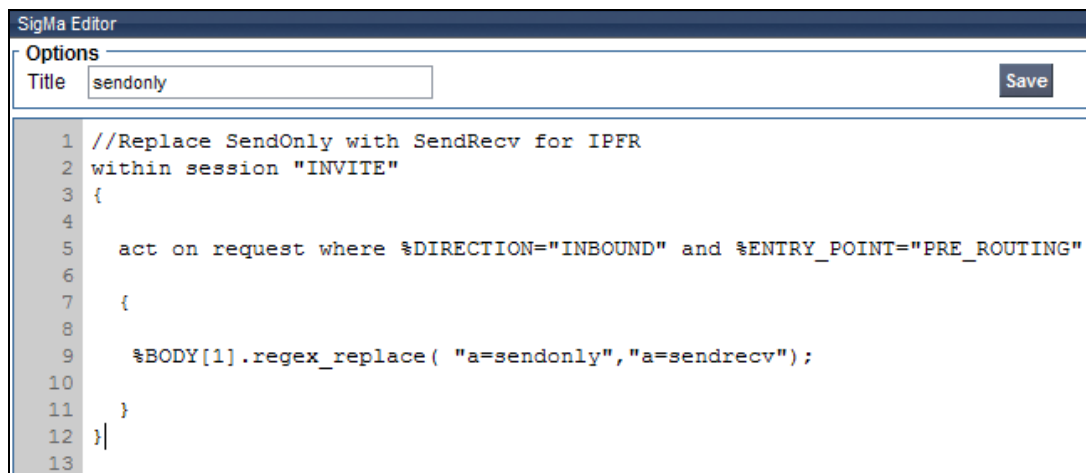
The Avaya SBCE configuration for interaction with the AT&T IP Toll Free service provided in document [9] should also be followed for interoperability with the AT&T IP Transfer Connect service. However additional Avaya SBCE provisioning may be required for interoperability with the AT&T IP Transfer Connect service if those customers also subscribe to the AT&T IP Flexible Reach – Enhanced Features service.

6.1. SIP Header Manipulation – Convert SendOnly to SendRecv

As described in **Section 2.2.1, Item 3**, the Communication Manager NCR feature is required to enable Refer and 302 call redirection with the AT&T IP Transfer Connect Service. 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 currently supported by the AT&T IP Flexible Reach - Enhanced Features service.

As a workaround, an Avaya SBCE SIP header manipulation script was written for the AT&T IP Flexible Reach - Enhanced Features service, which includes the conversion of *SendOnly* to *SendRecv*. That script, (named “**sendonly**”), is documented in **Section 12.1.3** of document [10], however it is also shown below as a reference.

Customers that subscribe to both the AT&T IP Transfer Connect and AT&T IP Flexible Reach - Enhanced Features services via the same Communication Manager environment, must use this script as well.



The screenshot shows the Sigma Editor interface. At the top, there is a title bar 'Sigma Editor' and an 'Options' section with a 'Title' field containing 'sendonly' and a 'Save' button. Below this is a code editor with the following script:

```
1 //Replace SendOnly with SendRecv for IPFR
2 within session "INVITE"
3 {
4
5     act on request where %DIRECTION="INBOUND" and %ENTRY_POINT="PRE_ROUTING"
6
7     {
8
9         %BODY[1].regex_replace( "a=sendonly","a=sendrecv");
10
11     }
12 }|
13
```

7. Verification Steps

7.1. Call Verification Tests

The call verification steps and troubleshooting tools described for the AT&T Toll Free service described in document [9], 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.

7.2. 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.

7.2.1. 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 vector 22 shown in **Section 5.3.1**. Also note the UUI information also defined in vector 22.

Filter: sip

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:81855584000@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:81855584000@135.25.29.74:5060;transport=

SIP from address: sip:7325552438@135.25.29.74
 SIP from address User Part: 7325552438
 SIP from address Host Part: 135.25.29.74
 SIP tag: ds42d7af1b
 Call-ID: ASE_1315330944532_131_null_135.138.152.24
 CSeq: 1 INVITE
 Sequence Number: 1
 Method: INVITE
 Contact: <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

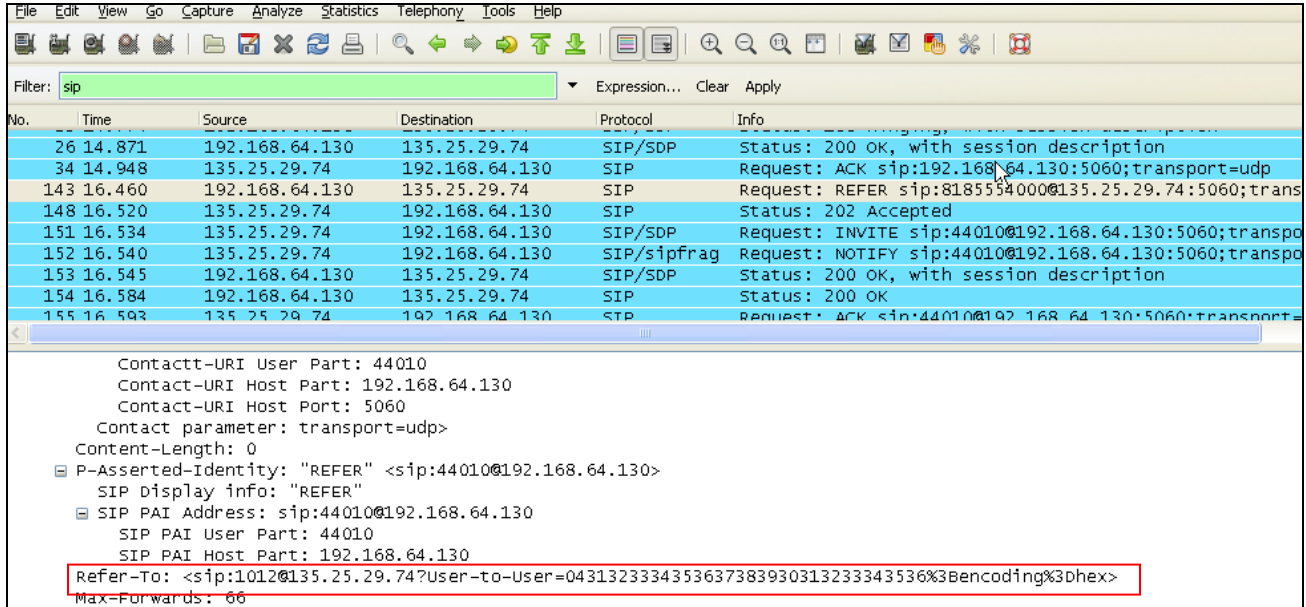
7.2.2. Refer with 180

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, item 2**, this causes Communication Manager to send a BYE (frame 178) upon receipt of the Notify/180 Ringing sent by AT&T in frame 175.

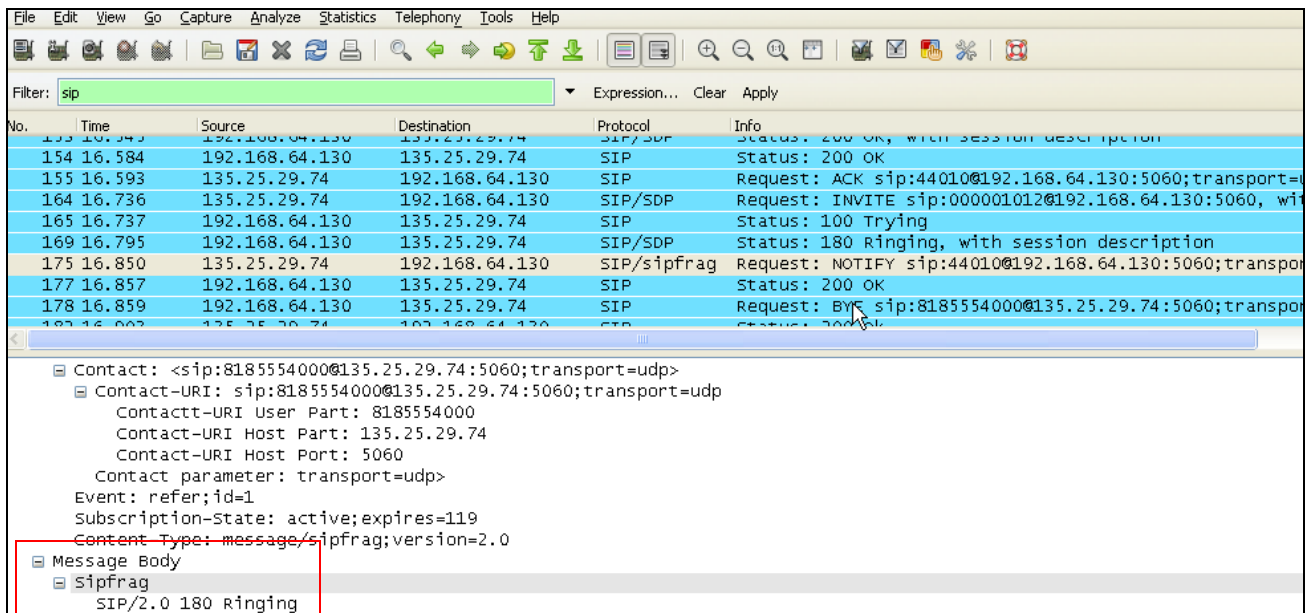
Filter: sip

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:8185554000@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=L
152	16.540	135.25.29.74	192.168.64.130	SIP/sipfrag	Request: NOTIFY sip:44010@192.168.64.130:5060;transport=L
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=L
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:8185554000@135.25.29.74:5060;transport=L
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:8185554000@135.25.29.74:5060;transport
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:8185554000@135.25.29.74:5060;transport=L

This screen shows the Refer in frame 143 in detail. The Refer-To header specifies the new called number (1012) as defined in vector 15 shown in **Section 5.3.2**. Also note the UII information defined in vector 15 (see Figure 10).

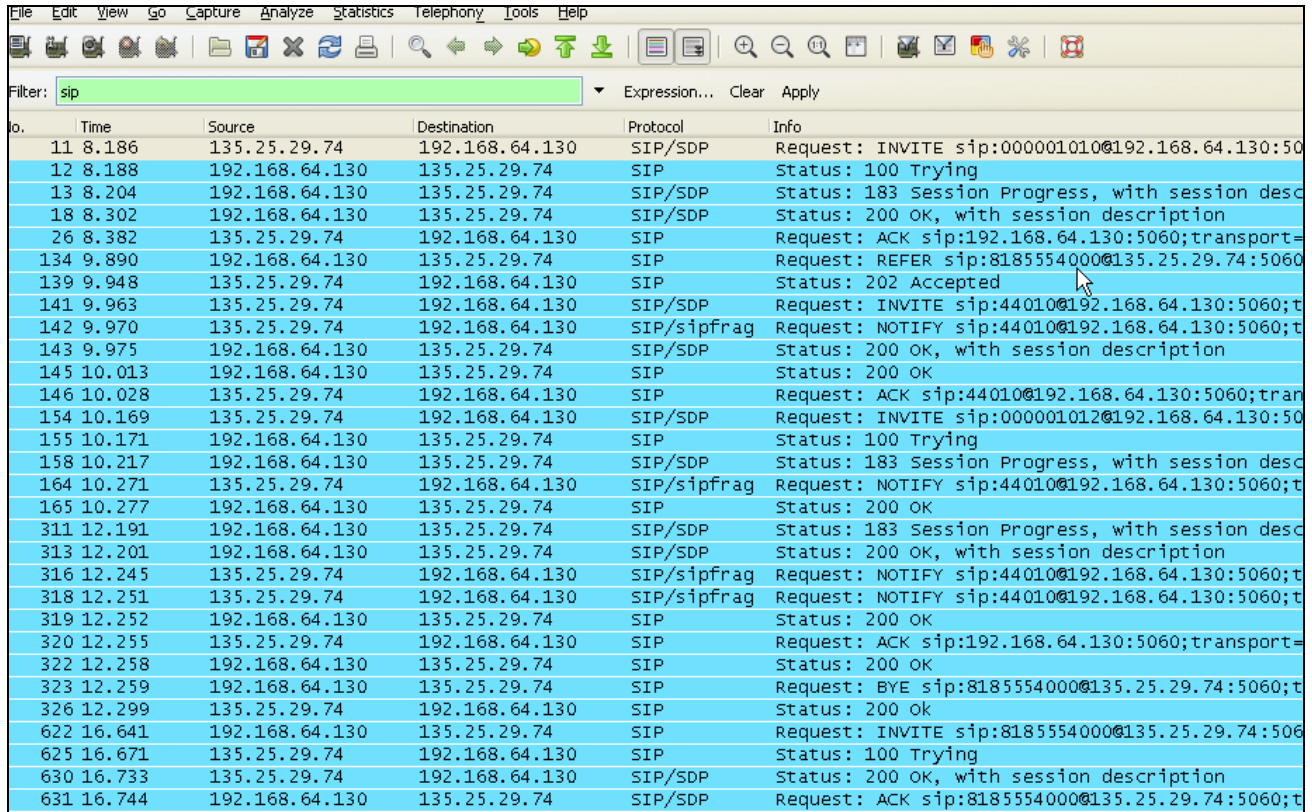


This screen shows the Notify/180 Ringing sent by AT&T in frame 175. Note that this Notify is in response to the 180 Ringing sent in frame 169.



7.2.3. 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, item 2**, 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.



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
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=tcp
134	9.890	192.168.64.130	135.25.29.74	SIP	Request: REFER sip:8185554000@135.25.29.74:5060;transport=tcp
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=tcp
142	9.970	135.25.29.74	192.168.64.130	SIP/sipfrag	Request: NOTIFY sip:44010@192.168.64.130:5060;transport=tcp
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=tcp
154	10.169	135.25.29.74	192.168.64.130	SIP/SDP	Request: INVITE sip:000001012@192.168.64.130:5060
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=tcp
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=tcp
318	12.251	135.25.29.74	192.168.64.130	SIP/sipfrag	Request: NOTIFY sip:44010@192.168.64.130:5060;transport=tcp
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=tcp
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:8185554000@135.25.29.74:5060;transport=tcp
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:8185554000@135.25.29.74:5060
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:8185554000@135.25.29.74:5060;transport=tcp

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 screenshot shows a network analysis tool interface. At the top, there is a menu bar with options: File, Edit, View, Go, Capture, Analyze, Statistics, Telephony, Tools, Help. Below the menu is a toolbar with various icons. A filter box contains the text 'sip'. Below the filter is a table of captured packets. The table has columns: No., Time, Source, Destination, Protocol, and Info. The packet at frame 316 is highlighted in blue. Below the table, a detailed view of the selected packet is shown. It is a NOTIFY message with the following details:

- Call-ID: ASE1317539708876_17539708876_135.138.132.24
- CSeq: 5 NOTIFY
- Sequence Number: 5
- Method: NOTIFY
- Content-Length: 28
- Contact: <sip:8185554000@135.25.29.74:5060;transport=udp>
- Contact-URI: sip:8185554000@135.25.29.74:5060;transport=udp
- Contact-URI User Part: 8185554000
- 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

8. Conclusion

As illustrated in these Application Notes, 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.

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.

9. References

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

Avaya Aura® Communication Manager

- [1] *Administering Avaya Aura® Communication Manager* Release 6.2, 03-300509, Issue 7.0, July 2012
- [2] *Implementing Avaya Aura® Communication Manager*, 03-603558, Issue 3, Release 6.2, July 2012
- [3] *Programming Call Vectors in Avaya Aura® Call Center*, 6.0, June 2010
- [4] *Administering Avaya Aura® Call Center Features*, Release 6.0, June 2010

Avaya Aura® Messaging

- [5] *Administering Avaya Aura® Messaging, 6.1*, CID: 151610, December 2011
- [6] *Implementing Avaya Aura® Messaging 6.1*, CID: 150976, October 2011

Avaya Session Border Controller for Enterprise

Product documentation for UC-Sec is provided on the installation CD provided with the device.

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

Avaya Application Notes

Avaya application notes are available at:

<https://www.devconnectmarketplace.com/at-t/at-t-sip-trunking>, then select *Resources*.

- [9] *Applications Notes for Avaya Aura® Communication Manager 6.2 and Avaya Session Border Controller for Enterprise with AT&T IP Toll Free SIP Trunk Service – Issue 1.0*
- [10] *Application Notes for Avaya Aura® Communication Manager 6.2, and Avaya Session Border Controller for Enterprise with AT&T IP Flexible Reach - Enhanced Features – Issue 1.1*

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