Avaya Solution & Interoperability Test Lab

Application Notes for Avaya Aura® Communication Manager 6.3, Avaya Aura® Session Manager 6.3, and Avaya Session Border Controller for Enterprise 6.2 with AT&T IP Transfer Connect Service – Issue 1.0

Abstract

These Application Notes describe the steps for configuring Avaya Aura® Communication Manager 6.3 with SIP Network Call Redirection (NCR), Avaya Aura® Session Manager 6.3, and the Avaya Session Border Controller for Enterprise 6.2 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 (UUI) features can be utilized together, in conjunction with the Data Forwarding option of the AT&T IP Transfer Connect service, to transmit UUI 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 6.3, Avaya Aura® Session Manager 6.3 and Avaya Session Border Controller for Enterprise 6.2 with AT&T IP Toll Free SIP Trunk Service 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 6.3 with SIP Network Call Redirection (NCR), Avaya Aura® Session Manager 6.3, and the Avaya Session Border Controller for Enterprise 6.2, (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 connections[1]. 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[2] 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, Communication Managers 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 Communication Manager systems.

Note – These Application Notes are intended to supplement the separate document: *Applications Notes for Avaya Aura® Communication Manager 6.3, Avaya Aura® Session Manager 6.3 and Avaya Session Border Controller for Enterprise 6.2 with AT&T IP Toll Free SIP Trunk Service 1.0.* This document is listed in **Section 10** as reference document **[10]**. 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 System Manager, Session Manager, 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 to Session Manager (see **Section 3.2** for descriptions), subsequent routing 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 (e.g., 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.
- Verify transmission of SIP Multipart headers, including SDP and XML content.

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 ring back should not be specified in Communication Manager 302 redirection vectors (see **Section 6.3.1**).

2. **Communication Manager 18x responses and Refer** - Communication Managers SIP trunk form may be configured to send either a 180 Ringing message (default), or a 183 Session Progress message (see **Section 6.2**). The message type selected alters Communication Managers behavior upon receipt of a Notify from AT&T during Refer calls. If 180 Ringing is selected, then Communication Manager will issue a BYE upon receipt of the Notify/Ringing message from AT&T. If 183 Session Progress 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 (in some cases a network BYE may arrive before Communication Manager issues one). 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** - Communication Managers 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, 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 **[2]**). Therefore 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, should follow the procedures described in **Addendum 1** of this document.
4. **Avaya SIP endpoints may generate SIP messages containing Endpoint-View, AV-Correlation-ID, and Bandwidth headers that may cause AT&T network issues**. It has been observed that sending these headers may cause issues with AT&T IP Toll Free and Transfer Connect services. Therefore an Avaya SBCE Signaling Manipulation Rule is used to remove these headers. The Avaya SBCE provisioning, documented in **Section 8.3.9** of the IP Toll Free document **[10]**, also applies to the IP Transfer Connect service.
5. **Avaya SBCE removes XML portions of SIP Multipart headers.** One of the IP Transfer Connect features is to pass caller data via SIP Invite Multipart headers (containing both SDP and XML data). During testing it was found that the Avaya SBCE would remove the XML portion of these Multipart headers, after processing the Invite from the "outside" to the "inside" interface (toward Session Manager).
    - Avaya SBCE support notified and an MR was opened.
    - The Avaya SBCE team suggested a temporary workaround, which was successfully tested (see **Section 7**).
6. **Network Address Translation (NAT) cannot be used on the customer interface of the AT&T edge router when SIP Multipart headers are used.** In the reference configuration, NAT was used on the AT&T edge router to convert an AT&T public address to the private addressing used in the test environment. This caused the edge router to corrupt the Multipart header contents. The solution was not to use NAT on the AT&T edge router.

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:

- Session Manager provides core SIP routing and integration services that enables communication between disparate SIP-enabled entities, e.g., PBXs, SIP proxies, gateways, adjuncts, trunks, applications, etc. across the enterprise. Avaya SIP endpoints register to Session Manager.
- System Manager provides a common administration interface for centralized management of all Session Manager instances in an enterprise.
- Communication Manager 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 desk telephones are represented with Avaya 1603(H.323), 960x Series IP Telephones (running H.323 firmware), and 96x1 Series IP Telephones (running H.323 or SIP firmware), Avaya 6424 Digital Telephones, as well as Avaya one-X® Agent soft phone (H323).
- The Avaya SBCE provides SIP Session Border Controller (SBC) functionality, including address translation and SIP header manipulation between the AT&T IP Transfer Connect 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 in this sample configuration. Session Manager may use SIP over UDP, TCP, or TLS to communicate with SIP network elements, e.g., the Avaya SBCE and Communication Manager. In the reference configuration, Session Manager uses SIP over TCP to communicate with the Avaya SBCE, and SIP over TCP and TLS to communicate with Communication Manager. UDP transport protocol is used between the Avaya SBCE and the AT&T IP Transfer Connect service.
- 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 Session Manager, which routed the call to Communication Manager. Communication Manager terminated the call to the appropriate agent/phone or fax extension.

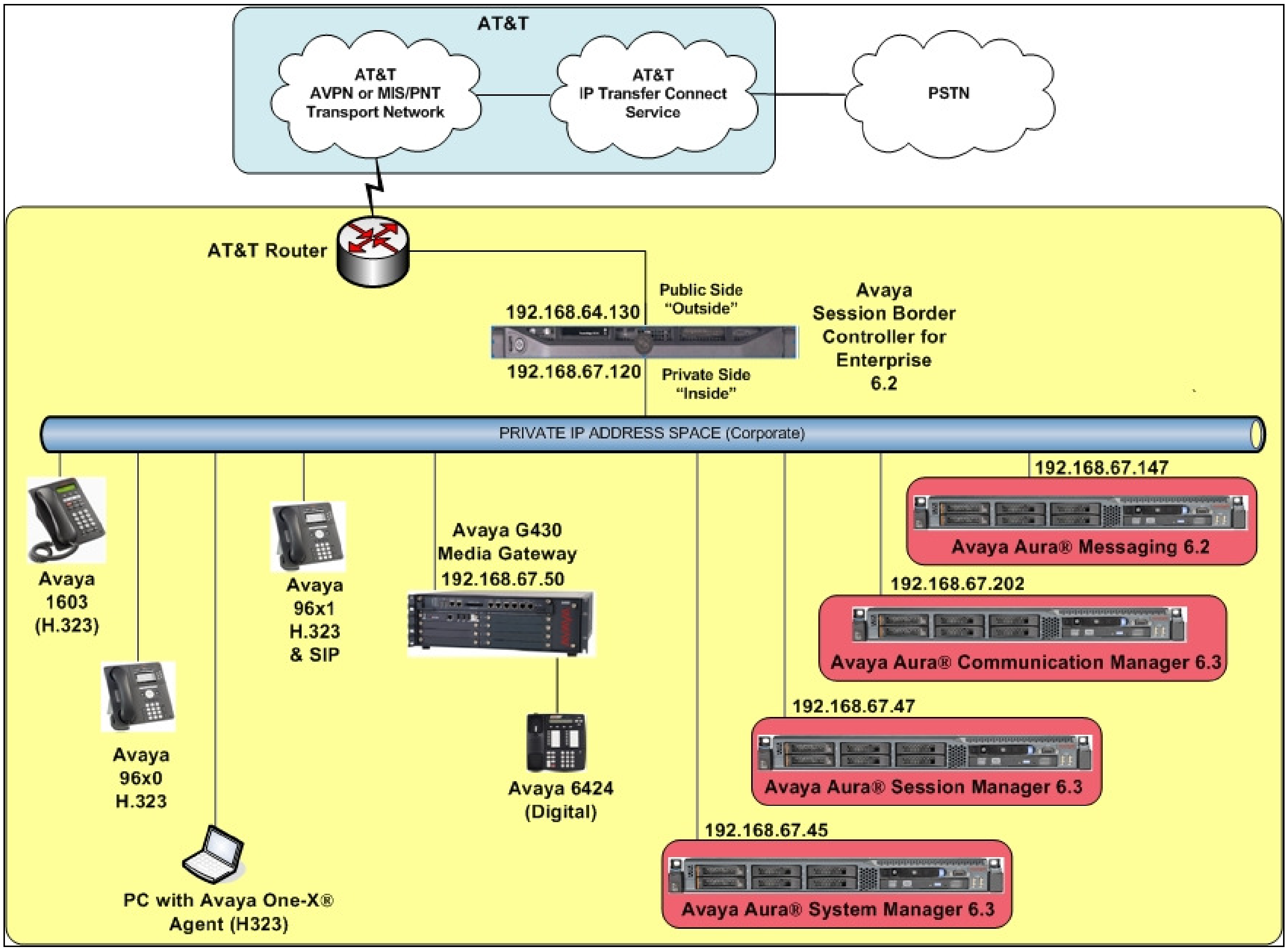


Figure 1: Reference configuration

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® System Manager	
IP Address	192.168.67.45
Avaya Aura® Session Manager	
Management IP Address	192.168.67.46
Network IP Address	192.168.67.47
Avaya Aura® Communication Manager	
IP Address	192.168.67.202
Avaya Aura® Communication Manager extensions	19xxx = Stations 4xxxx = VDNs
Voice Messaging Pilot Extension	36000
Avaya Session Border Controller for Enterprise (Avaya SBCE)	
IP Address of Outside (Public) Interface (to AT&T IP Toll Free Service)	192.168.64.130
IP Address of Inside (Private) Interface (connected to Avaya Aura® Session Manager)	192.168.67.120
Avaya Aura® Messaging	
IP Address	192.168.67.147
Messaging Mailboxes	19xxx

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 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 at the Avaya SBCE. IP address, domain, and header manipulations are applied and then forwarded to 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.

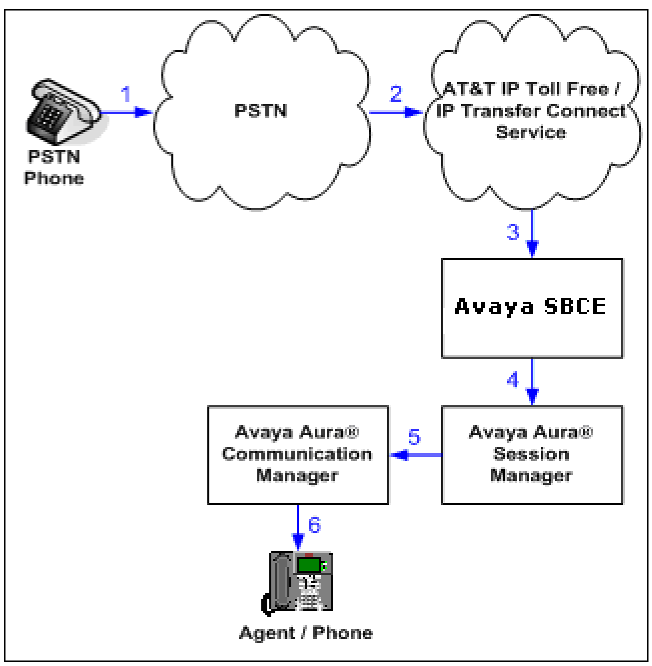


Figure 2: Inbound AT&T IP Transfer Connect Call – No Redirection

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 at the Avaya SBCE, is sent to 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 illustrated in **Figure 2**.
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).

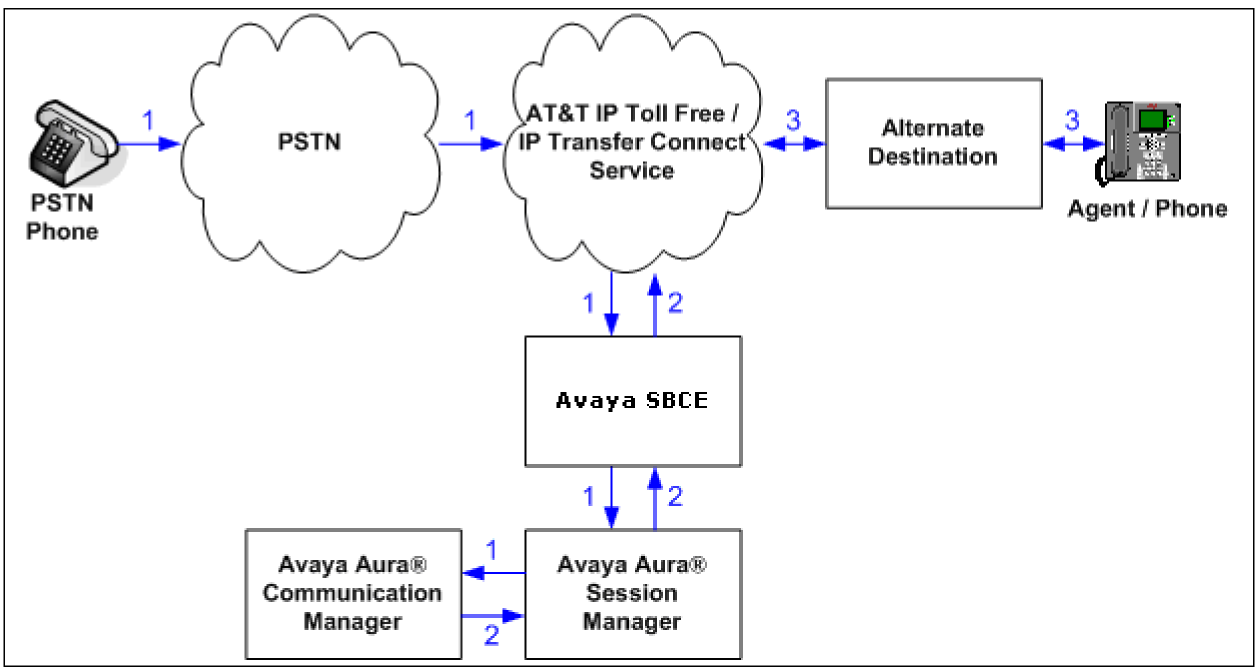


Figure 3: Inbound AT&T IP Transfer Connect Call – Pre-Answer SIP 302 Redirection

The third call scenario illustrated in **Figure 4** is an inbound AT&T IP Transfer Connect service call that arrives on the Avaya SBCE, then is transferred to 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 illustrated in **Figure 2**.
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).

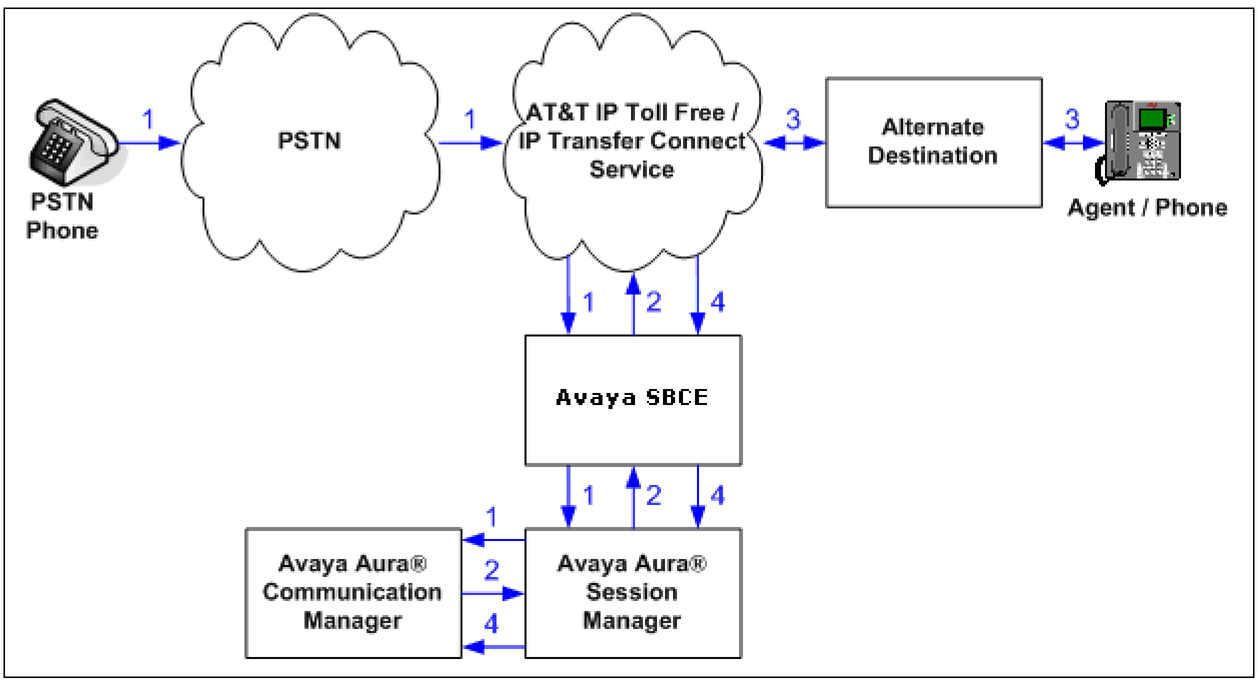


Figure 4: Inbound AT&T IP Transfer Connect Call – Post-Answer SIP REFER Redirection

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 first five steps from the first call scenario illustrated in **Figure 2**.
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), 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.

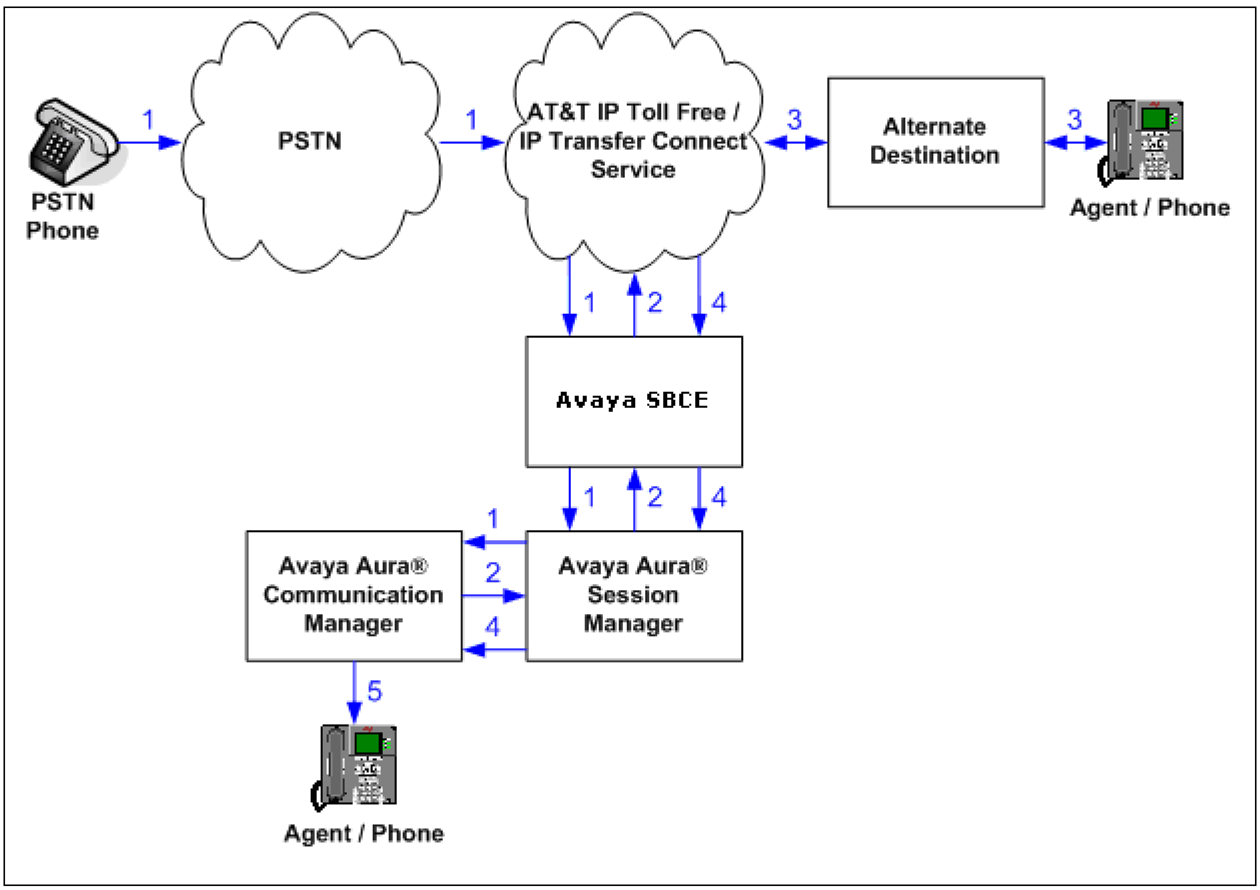


Figure 5: Inbound AT&T IP Transfer Connect Call - Post-Answer SIP REFER Redirection. Refer is unsuccessful, Communication Manager re-routes to Agent.

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 • System Platform • Avaya Aura® System Manager	 • 6.3.0.0.18002 • 6.3.2.4 with SP1 (r1212) and patch 2 (r1451)
IBM 8800 server • Avaya Aura® Session Manager	 • 6.3 SP2 (6.3.2.632023)
IBM 8800 server • System Platform • Avaya Aura® Communication Manager	 • 6.3.0.0.18002 • 6.3 SP0 (06.3-03.0.124.0-20553)
Dell R610 • System Platform • Avaya Aura® Messaging	 • 6.2.1.0.9 • 6.2 SP3 (MSG-02.0.823.0-109_0304)
Avaya G430 Media Gateway • MM712 Digital card	• 33.13.0 • HW7 FW11
Dell R310 • Avaya Session Border Controller for Enterprise	 • 6.2 Q48
Avaya 96x0 IP Telephone	H.323 Version S3.2
Avaya 96x1 IP Telephone	H.323 Version S6 2408 SIP Version 6.2.2.17
Avaya 9601 IP Telephone	SIP version 6.1.5.12
Avaya one-X® Communicator	H323 6.1 SP8 (6.1.8.06)
Avaya 1603 IP Telephone	H323 (ha1603ua1_3200.bin)
Avaya Flare® Experience for A175	SIP A175-IPT-SIP-R1_1_3-021913
Avaya Flare® Experience for Windows	SIP 1.1.2.11
Avaya 6424 Digital telephone	-
Ventafax Home Version (Windows based Fax device)	6.1.59.144

Table 2: Equipment and Software Versions

5. Avaya Aura® Session Manager

Session Manager administration for interaction with the AT&T IP Toll Free/IP Transfer Connect services is described in document **[10]**. This section describes the additional administration steps on Session Manager necessary for supporting interaction with the AT&T IP Transfer Connect service.

5.1. Dial Patterns

Provision additional AT&T IP Transfer Connect service DNIS numbers (digits delivered in the Request URIs of inbound Invites), so they can be converted to Communication Manager VDN, Agent, or station extensions, according to the procedures described in **Section 5.8** of document **[10]**.

6. Avaya Aura® Communication Manager

Communication Manager administration for interaction with the AT&T IP Toll Free service is described in document **[10]** and is also applicable for the AT&T IP Transfer Connect service. 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 Communication Managers 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.

6.1. System Parameters

This section reviews 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**, **EAD**, 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.)
```

6.2. 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 document **[10]** for inbound AT&T IP Toll Free service calls (e.g., trunk 2). 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 2                                            Page   4 of  21
                              PROTOCOL VARIATIONS

                                       Mark Users as Phone? n
Prepend '+' to Calling/Alerting/Diverting/Connected Number? n
                       Send Transferring Party Information? n
                                  Network Call Redirection? y
          Build Refer-To URI of REFER From Contact For NCR? n
                                     Send Diversion Header? y
                                   Support Request History? y
                               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
                 Accept Redirect to Blank User Destination? n
                                             Enable Q-SIP? n
```

6.3. 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 Communication Managers SIP Network Call Redirection (NCR) functionality.

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 in **Section 3.1**.

These Application Notes provide rudimentary vector definitions to demonstrate and test the SIP NCR and UUI 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 UUI. The definition and documentation of those complex applications and associated vectors are beyond the scope of these Application Notes. Consult documents **[7]** and **[8]** for further information.

6.3.1. 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 ring back (see **Section 2.2.1**, **item 1**). The vector does the following:

1. Assigns the data "**1234567890123456**" to ASAI UUI variable "**A**" (vector step **02**).
   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 **05**). Note that since this vector did not answer the call, the presence of the "**~**" in the "**route-to number**" line 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   UuiTest1                        asaiuui L     16     1
B
C
```

Figure 6: Change Variables Form

```
display vdn 19020                                              Page   1 of   3
                            VECTOR DIRECTORY NUMBER

                                 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                                              Page   1 of   6
                                  CALL VECTOR
    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 n if unconditionally
09 stop
10                                         09
```

Figure 8: Sample Vector for Pre-Answer Redirection (302)

6.3.2. 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 steps **02**). **Note**: The parameters for UUI variable "**A**" and other vector variables are defined using the **change variables** command (see **Figure 6**).
2. Answers the call to play an announcement (vector step **05**).
3. Attempts to redirect the call to the number "**1012**" (vector step **08**). Note that since this vector answered the call, the presence of the "**~**" in the "**route-to number**" line 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 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*:
* Follows VDN Override Rules
```

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
04 #    Play Refer announcement
05 announcement 42008
06
07 #    Refer occurs since this is post answer
08 route-to     number ~r1012           with cov n if unconditionally
09 #    If Refer fails play announcement and disconnect
10 disconnect   after announcement 42009
11
12
```

Figure 10: Sample Vector for Post-Answer Redirection (Refer)

7. Avaya Session Border Controller for Enterprise

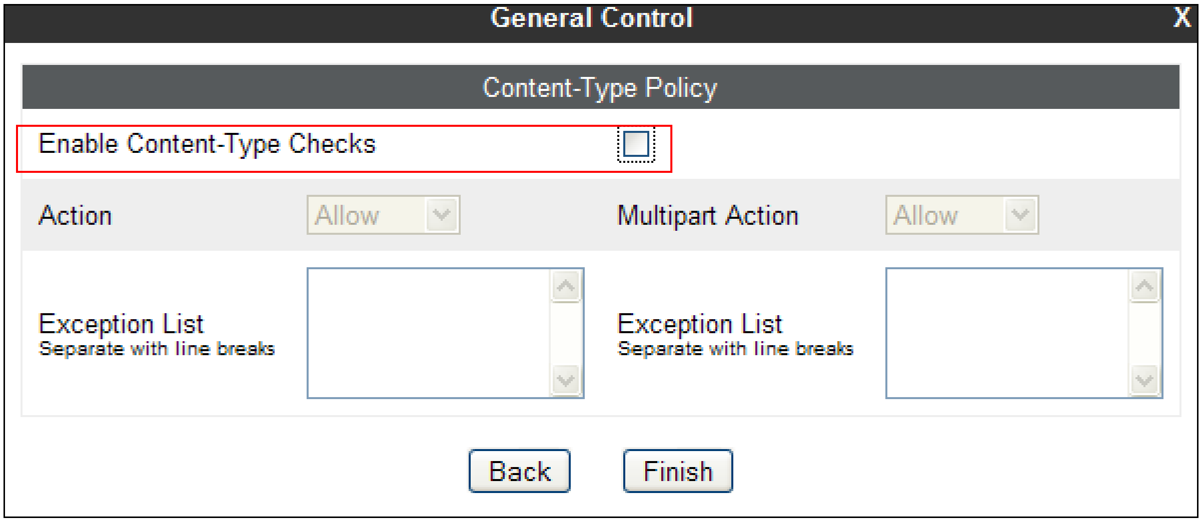
Avaya SBCE configuration for interaction with the AT&T IP Toll Free service provided in document **[10]** should also be followed for interoperability with the AT&T IP Transfer Connect service.

However the Avaya SBCE team provided the following workaround for the Multipart header issue described in **Section 2.2.1, Item 5**. This workaround calls for the Signaling Rule *Enable Content-Type Checks* option to be disabled.

7.1. Avaya Signaling Rules

Use the following steps to disable the *Enable Content-Type Checks* option:

1. Select **Domain Policies** from the menu on the left-hand side menu (not shown).
2. Select the **Signaling Rules** (not shown).
3. The Signaling Rules window will open (not shown). From the Signaling Rules menu, select the **Avaya_with_SM** rule created in **[10]**.
4. Select the **General** tab, and scroll to the bottom of the window and select **Edit** (not shown).
5. The General form will be displayed. Scroll to the bottom of the window and click on **Next** (not shown).
6. The Content-Type Policy window is displayed. Unselect the **Enable Content-Type Checks** option.
7. Click on **Finish**.



7.2. AT&T Signaling Rules

Using the same procedures shown in **Section 7.1**, disable the *Enable Content-Type Checks* option for the AT&T Signaling Rule **ATT_SR** created in **[10]**.

8. Verification Steps

8.1. Call Verification Tests

The call verification steps and troubleshooting tools described for the AT&T Toll Free service described in document **[10]**, also apply to the AT&T IP Transfer Connect service.

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.

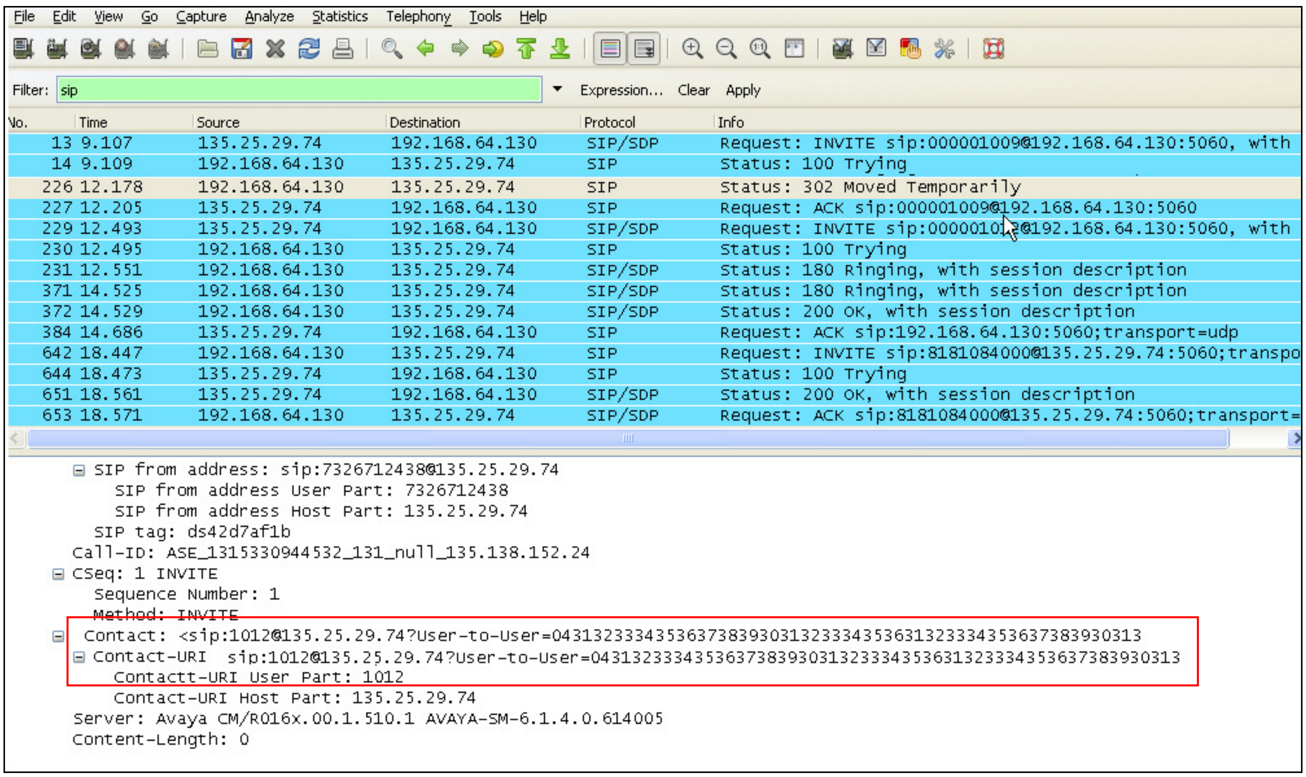
8.2. Protocol Traces

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.

8.2.1. 302 Redirection

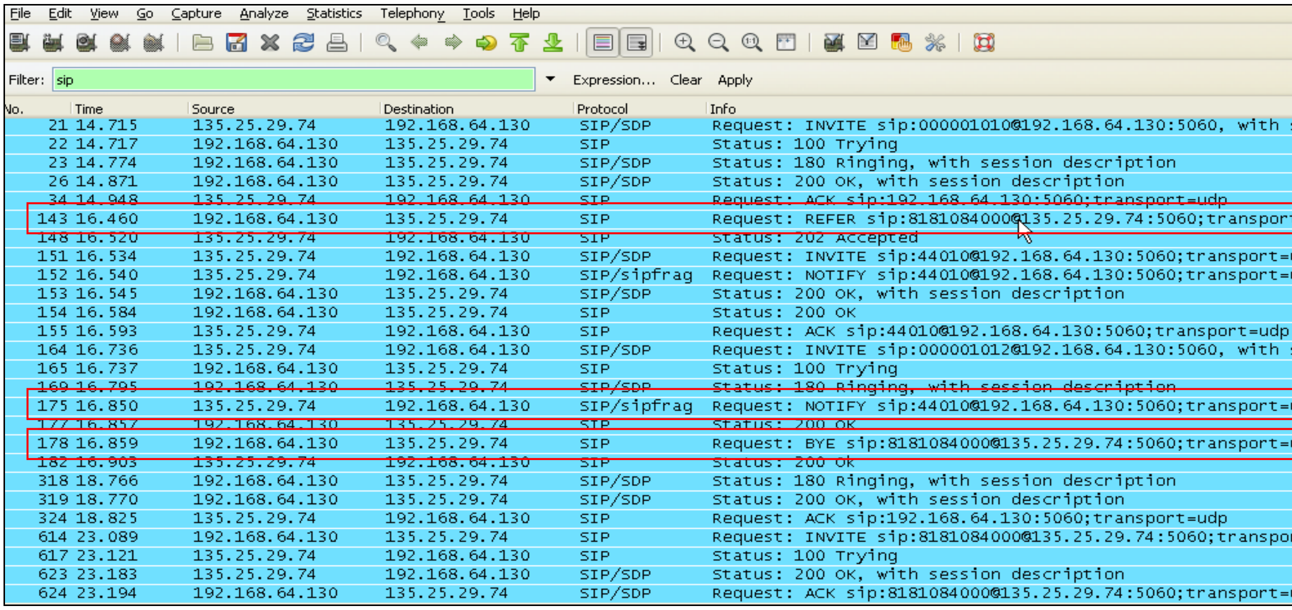
The following is an example of a 302 redirection call filtering on the SIP protocol. Note the following:

- The Contact header contains the new called number (1012) as defined in vector 22 (see **Section 6.3.1**).
- The User-to-User Information (UUI) defined in vector 22.
- 180 Ringing is not sent prior to the 302 (see **Section 6.3.1**).

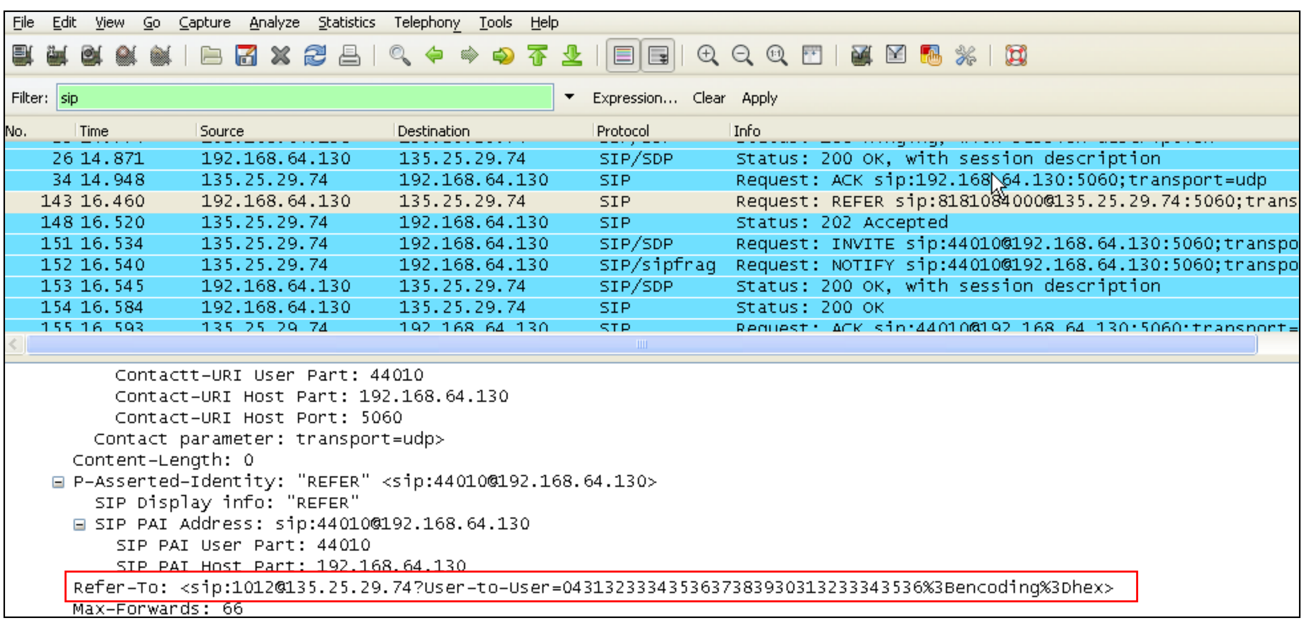


8.2.2. Refer with 180

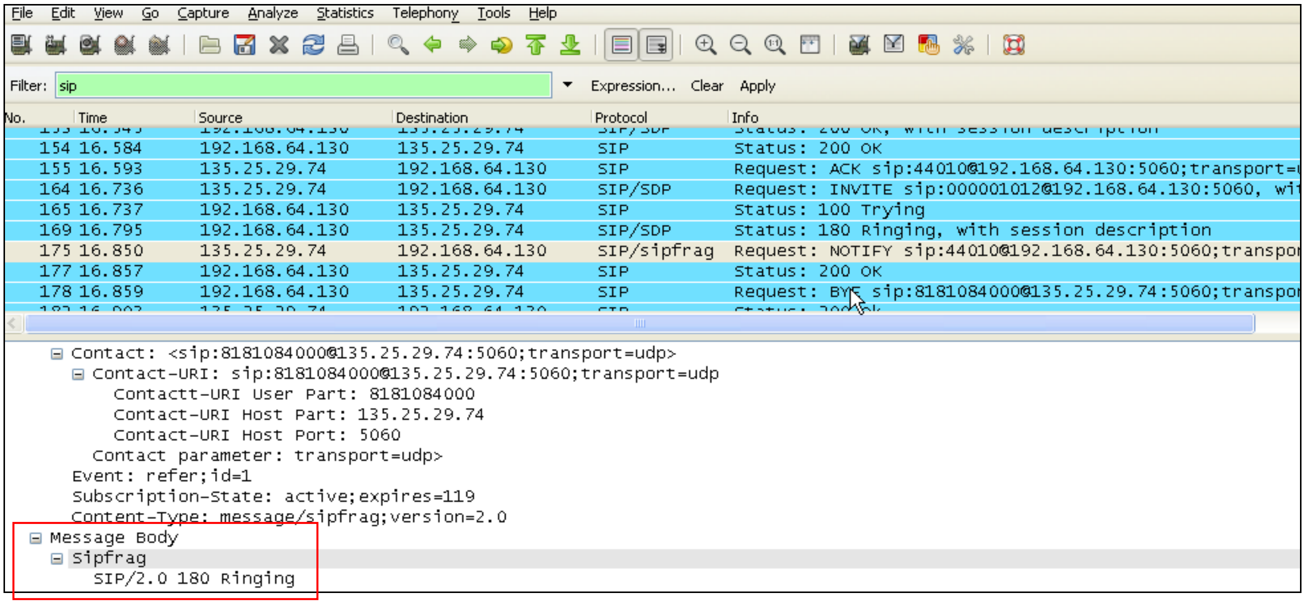
The following is an example of a Refer (frame 143) redirection call filtering on the SIP protocol. Note that Communication Manager is sending 180 Ringing. As described in **Section 2.2.1**, **item 2**, this causes Communication Manager to send a BYE (frame 178) upon receipt of the Notify/180 Ringing sent by AT&T in frame 175.



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 6.3**. Also note the UUI information also defined on vector 15.

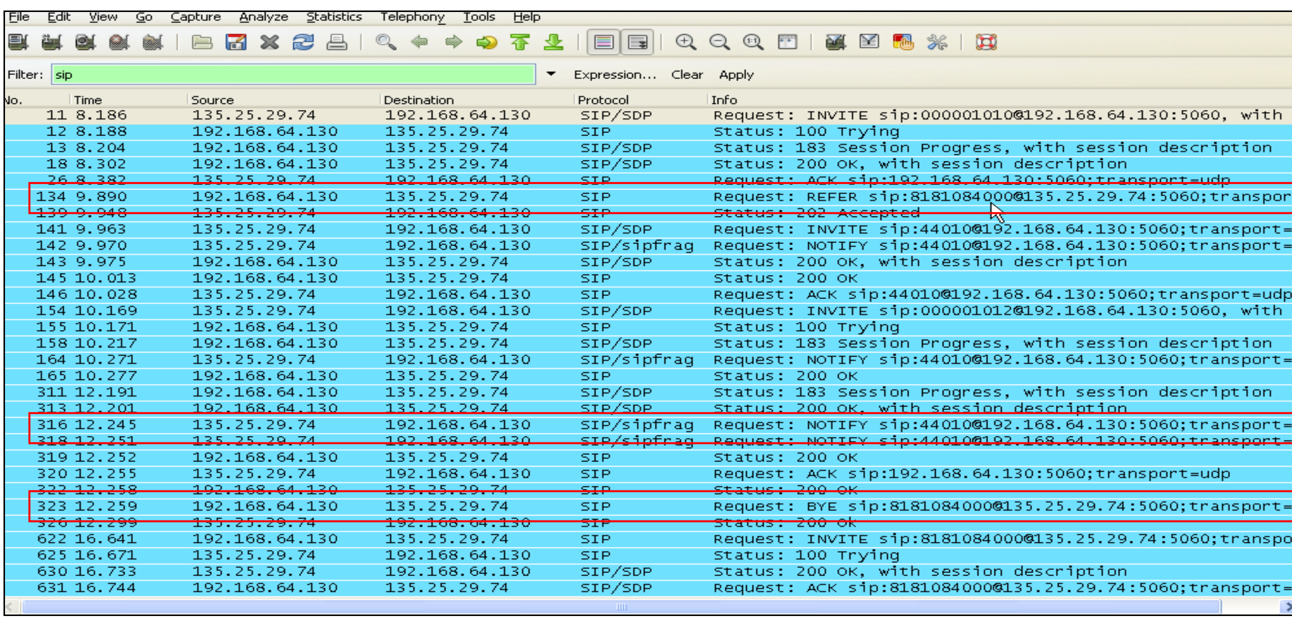


This screen shows the Notify/180 Ringing sent by AT&T in frame 175. Note that this Notify is in response to the 180 Ringing sent in frame 169.

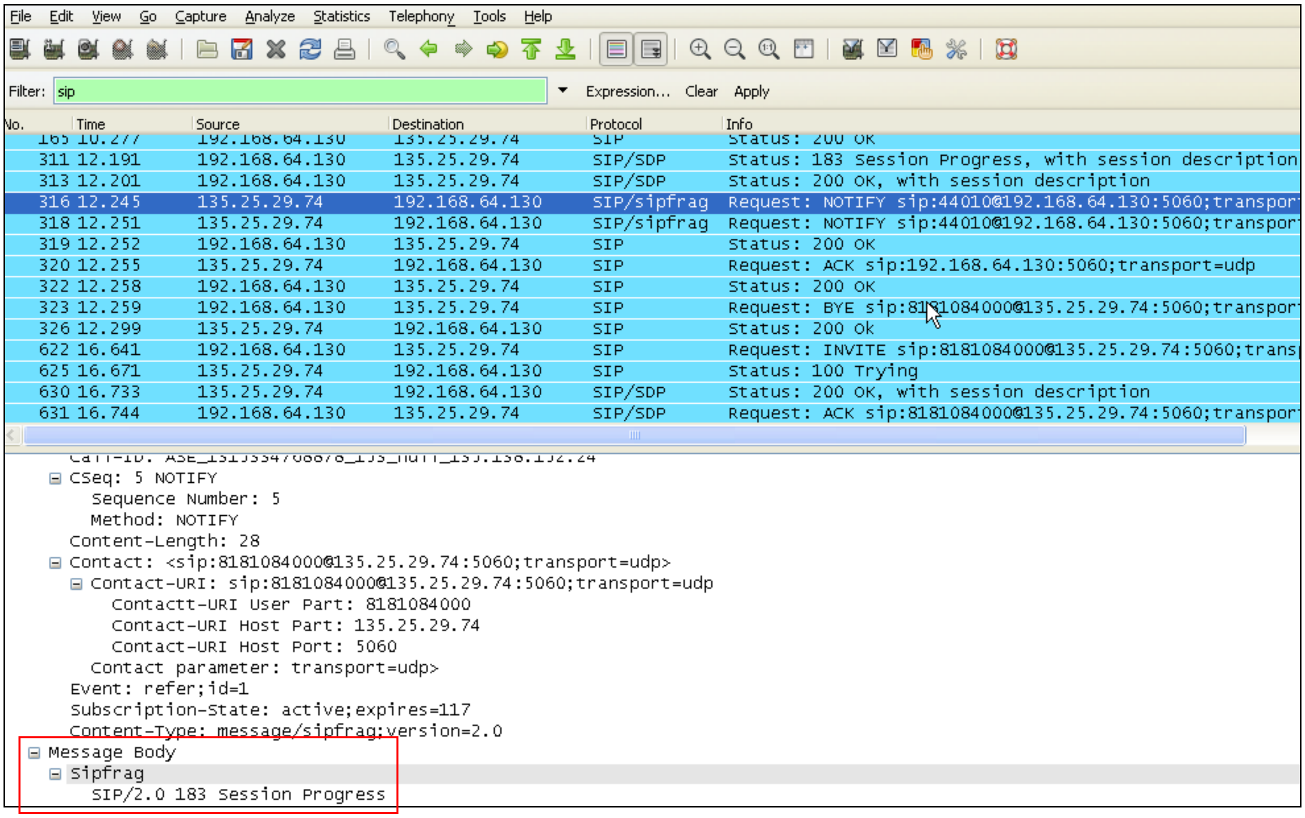


8.2.3. Refer with 183

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.



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.



8.2.4. Multipart Headers

When the associated IP Transfer Connect features generate SIP Multipart headers, their transmission may be confirmed by monitoring the "inside" (A1) interface of the Avaya SBCE. The Multipart headers are distinguished by their "Boundary" statements, which mark the beginning and end of each Multipart section. In the screenshot below the Message Body section of the AT&T Invite is displayed. Verify that the regular SDP, and the special IP Transfer Connect XML user data, have been passed by the Avaya SBCE.

```
    Content-Length: 1068
Message Body
  MIME Multipart Media Encapsulation, Type: multipart/mixed, Boundary: "as-boundary"
    [Type: multipart/mixed]
    First boundary: --as-boundary\r\n
    Encapsulated multipart part:  (application/sdp)
      Content-Type: application/sdp\r\n\r\n
      Session Description Protocol
        Session Description Protocol Version (v): 0
        Owner/Creator, Session Id (o): Sonus_UAC 11561 9497 IN IP4 192.168.67.120
          Owner Username: Sonus_UAC
          Session ID: 11561
          Session Version: 9497
          Owner Network Type: IN
          Owner Address Type: IP4
          Owner Address: 192.168.67.120
        Session Name (s): SIP
        Connection Information (c): IN IP4 192.168.67.120
          Connection Network Type: IN
          Connection Address Type: IP4
          Connection Address: 192.168.67.120
        Time Description, active time (t): 0 0
          Session Start Time: 0
          Session Stop Time: 0
        Media Description, name and address (m): audio 16420 RTP/AVP 18 0 2 100
          Media Type: audio
          Media Port: 16420
          Media Protocol: RTP/AVP
          Media Format: ITU-T G.729
          Media Format: ITU-T G.711 PCMU
          Media Format: ITU-T G.721
          Media Format: DynamicRTP-Type-100
        Media Attribute (a): rtpmap:18 G729/8000
          Media Attribute Fieldname: rtpmap
          Media Format: 18
          MIME Type: G729
          Sample Rate: 8000
        Media Attribute (a): fmtp:18 annexb=no
          Media Attribute Fieldname: fmtp
          Media Format: 18 [G729]
          Media format specific parameters: annexb=no
        Media Attribute (a): rtpmap:0 PCMU/8000
          Media Attribute Fieldname: rtpmap
          Media Format: 0
          MIME Type: PCMU
          Sample Rate: 8000
        Media Attribute (a): rtpmap:2 G726-32/8000
          Media Attribute Fieldname: rtpmap
          Media Format: 2
          MIME Type: G726-32
          Sample Rate: 8000
        Media Attribute (a): rtpmap:100 telephone-event/8000
          Media Attribute Fieldname: rtpmap
          Media Format: 100
          MIME Type: telephone-event
          Sample Rate: 8000
        Media Attribute (a): fmtp:100 0-15
          Media Attribute Fieldname: fmtp
          Media Format: 100 [telephone-event]
          Media format specific parameters: 0-15
        Media Attribute (a): sendrecv
        Media Attribute (a): maxptime:30
          Media Attribute Fieldname: maxptime
          Media Attribute Value: 30
    Boundary: \r\n--as-boundary\r\n
    Encapsulated multipart part:  (application/pidf+xml)
      Content-Type: application/pidf+xml\r\n
      Content-Transfer-Encoding: 7bit\r\n
      Content-Disposition: session; handling=required\r\n\r\n
      eXtensible Markup Language
        Content-ID:
        [ ERROR: Unrecognized text ]
        <presence
          xmlns="urn:ietf:params:xml:ns:pidf"
          xmlns:dm="urn:ietf:params:xml:ns:pidf:data-model"
          xmlns:gp="urn:ietf:params:xml:ns:pidf:geopriv10"
          xmlns:gml="http://www.opengis.net/gml"
          xmlns:gs="http://www.opengis.net/pidflo/1.0"
          xmlns:cl="urn:ietf:params:xml:ns:pidf:geopriv10:civicAddr"
          xmlns:tf="http://www.att.com/iptf"
          entity="pres:tfas1@att.net">
          <tf:dataresponse
            status="available"/>
          <dm:device
            id="194155685">
            <tf:smsAvailable>
              True
              </tf:smsAvailable>
            <tf:model>
              iPhone_4s
              </tf:model>
            <tf:vendor>
              APL
              </tf:vendor>
            </dm:device>
          </presence>
    Last boundary: \r\n--as-boundary--\r\n
```

9. Conclusion

As illustrated in these Application Notes, Avaya Aura® Session Manager 6.3, Avaya Aura® Communication Manager 6.3, and the Avaya Session Border Controller for Enterprise 6.2 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 (UUI) features can work in complement with the AT&T implementations of SIP NCR and UUI to support call redirection over SIP trunks while preserving initiating caller information. The transmission of Multipart headers was also confirmed. 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 UUI-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.

10. References

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

Avaya Aura® Session Manager/System Manager

1. *Administering Avaya Aura® Session Manager,* Release 6.3, December, 2012
2. *Implementing Avaya Aura® Session Manager,* Release 6.3, March, 2013
3. *Implementing Avaya Aura® System Manager,* Release 6.3, Issue 1, December, 2012
4. *Administering Avaya Aura® System Manager,* Release 6.3, Issue 1.0, December, 2012

Avaya Aura® Communication Manager

5. *Administering Avaya Aura® Communication Manager*, Release 6.3 03-300509, Issue 8, May 2013
6. *Programming Call Vectors in Avaya Aura® Call Center*, 6.0, June 2010

Avaya Aura® Messaging

7. *Administering Avaya Aura® Messaging,* Release 6.2, Issue 2.1, February, 2013

Avaya Session Border Controller for Enterprise

8. *Installing Avaya Session Border Controller for Enterprise*, Release 6.2, Issue 3, June 20
9. *Administering Avaya Session Border Controller for Enterprise*, Release 6.2, Issue 2, March 2013

Avaya Application Notes

10. *Applications Notes for Avaya Aura® Communication Manager 6.3, Avaya Aura® Session Manager 6.3 and Avaya Session Border Controller for Enterprise 6.2 with AT&T IP Toll Free SIP Trunk Service 1.0 –* https://devconnect.avaya.com/public/download/dyn/SM63CM63SBCETF.pdf

11. Addendum 1 – Additional Considerations for Customers that Subscribe to AT&T IP Transfer Connect and AT&T IP Flexible Reach - Enhanced Features

As described in **Section 2.2.1**, **Item 3**, 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 AT&T IP Flexible Reach - Enhanced Features service also supports the use of Refer for specific call scenarios, and therefore NCR is enabled for that service as well. However the *SendOnly* SIP parameter is not currently supported by the AT&T IP Flexible Reach - Enhanced Features service. For this, and other reasons, the AT&T IP Flexible Reach - Enhanced Features service Application Notes [**2]**, defines the use of a separate NCR enabled SIP trunk specifically for Refer based features (a SIP trunk with NCR disabled is defined for all other AT&T IP Flexible Reach traffic).

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, should use the NCR enabled trunk(s) for both the AT&T IP Flexible Reach - Enhanced Features that support Refer, as well as for the AT&T IP Transfer Connect features that support Refer and 302 call redirection.

Please e-mail any questions or comments pertaining to these Application Notes along with the full title name and filename, located in the lower right corner, directly to the Avaya DevConnect program at devconnect@avaya.com.