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 6.3 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, Avaya Aura® Session Manager 6.3, and the Avaya Session Border Controller for Enterprise 6.3 with 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.

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 6.3 with AT&T IP Toll Free SIP Trunk Service 1.0*.

Readers should pay attention to **Section 2**, in particular the scope of testing as outlined in **Section 2.1** as well as the observations noted in **Section 2.2**, to ensure that their own use cases are adequately covered by this scope and results.

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

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1. Introduction

These Application Notes describe the steps for configuring Avaya Aura® Communication Manager 6.3 (*Communication Manager*), Avaya Aura® Session Manager 6.3 (*Session Manager*), and the Avaya Session Border Controller for Enterprise 6.3, (*Avaya SBCE*), with the AT&T IP Transfer Connect service (*IPTC*) 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 (IPTF) 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 Avaya Aura® Communication Manager SIP NCR feature, 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 Manager SIP User-to-User Information (UUI) feature can be utilized with the SIP NCR feature to transmit UUI within SIP signaling messages to the alternate destinations. This capability is used in conjunction with the Data Forwarding option of the AT&T IP Transfer Connect service to transmit a limited amount of call-related data between call centers to support enhanced, customer-friendly applications and/or support efficient use of call center resources. Examples of UUI data might include a customer account number obtained during a database query and the best service routing data exchanged between Communication Manager systems.

2. General Test Approach and Test Results

DevConnect Compliance Testing is conducted jointly by Avaya and DevConnect members. The jointly-defined test plan focuses on exercising APIs and/or standards-based interfaces pertinent to the interoperability of the tested products and their functionalities. DevConnect Compliance Testing is not intended to substitute full product performance or feature testing performed by DevConnect members, nor is it to be construed as an endorsement by Avaya of the suitability or completeness of a DevConnect member's solution.

The interoperability compliance testing focused on verifying call flows between IPTF and the Customer Premises Equipment (CPE) containing Communication Manager, Session Manager, and the Avaya SBCE (see **Section 3.2** for call flow examples). The test environment consisted of:

- A simulated enterprise with Communication Manager, Session Manager, System Manager (for Session Manager provisioning), Avaya SBCE, and various Avaya IP telephones (see **Section 3**).
- 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 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, to the Avaya CPE. The following SIP trunking VoIP features were tested with the IPTC service:

- Inbound AT&T IP Transfer Connect service calls to Communication Manager VDNs, agents/telephones; utilizing G.729A codec (IPTC preferred codec). However G.729B and G.711 codecs were tested as well.
- Avaya 96x1 IP telephones (H.323 and SIP), and Avaya one-X® Agent (H.323 softphone) were used in Agent mode on Communication Manager.
- Inbound AT&T IP Transfer Connect service calls that are immediately redirected by a Communication Manager vector (pre-answer redirection) back to IPTC for redirection to an alternate destination.
- Inbound IPTC calls that are answered by a Communication Manager vector and then redirected (post-answer redirection) back to IPTC for redirection to an alternate destination.
- Redirected IPTC 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.
- Verify reception of IPTC SIP Multipart/NSS headers, including SDP and XML content.
- Communication Manager features such as hold, resume, and local transfer.
- SIP OPTIONS messages used to monitor the health of the SIP trunks between the CPE and AT&T.
- Communication Manager NCR and SIP User-to-User Information (UUI) capability, 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.

2.2. Test Results

The test objectives stated in **Section 2.1**, with limitations as noted below, were verified.

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** –The Communication Manager 6.3 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 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 (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. **Use of Network Address Translation (NAT) on the customer interface of the AT&T CE router when SIP Multipart headers are used.** Previously, use of NAT on the AT&T CE (Cisco) router would corrupt the Multipart header contents, including the SDP. The solution was not to use NAT on the CE router.
    - **UPDATE** – Recent testing with Cisco IOS *c2900-universalk9-mz.SSA-**eng-sp-153-3.M1**.bin*, **in addition to specifying the router command *ip nat service allow-multipart***, showed that NAT can be used on the CE router without corrupting the Multipart headers (Mobility and NSS). Note that Cisco IOS *c2900-universalk9-mz.SPA.**154-3.M1**.bin*, **in addition to specifying the router command *ip nat service allow-multipart***, was tested successfully as well.

4. **IP Transfer Connect service Landline/Mobility test cases could not be executed**. The AT&T supplied IP Transfer Connect test plan specifies test cases to verify the transmission of Landline/Mobility data by the IP Transfer Connect service. Due to network access issues, these test cases could not be executed.

2.3. Support

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

Avaya customers may obtain documentation and support for Avaya products by visiting http://support.avaya.com. In the United States, (866) GO-AVAYA (866-462-8292) provides access to overall sales and service support menus.

3. Reference Configuration

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

Note – These Application Notes are intended to supplement the separate document: *Application Notes for Avaya Aura® Communication Manager 6.3, Avaya Aura® Session Manager 6.3 and Avaya Session Border Controller for Enterprise 6.3 with AT&T IP Toll Free SIP Trunk Service – Issue 1.0, March 2015.* This document is listed in **Section 10** as reference document **[11]**. 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.

The reference configuration used in these Application Notes is shown in **Figure 1** and consists of several components:

- Communication Manager 6.3, System Manager 6.3, Session Manager 6.3, and the Avaya SBCE 6.3 are used in the reference configuration.
- 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.
- An 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 used are Avaya 96x1 Series IP Telephones (H.323 and SIP), and Avaya one-X® Agent soft phone (H.323).
- The Avaya SBCE provides SIP Session Border Controller (SBC) functionality, including address translation and SIP header manipulation between the IPFR-EF service and the enterprise internal network.
- The IPTC service uses SIP over UDP to communicate with enterprise edge SIP devices, e.g., the Avaya SBCE. 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.
- Inbound calls were placed from PSTN via the IPTC service, through the Avaya SBCE to Session Manager, which routes the call to Communication Manager. Communication Manager terminates the call to the appropriate agent/phone extension, or redirects the call via 302 or Refer.

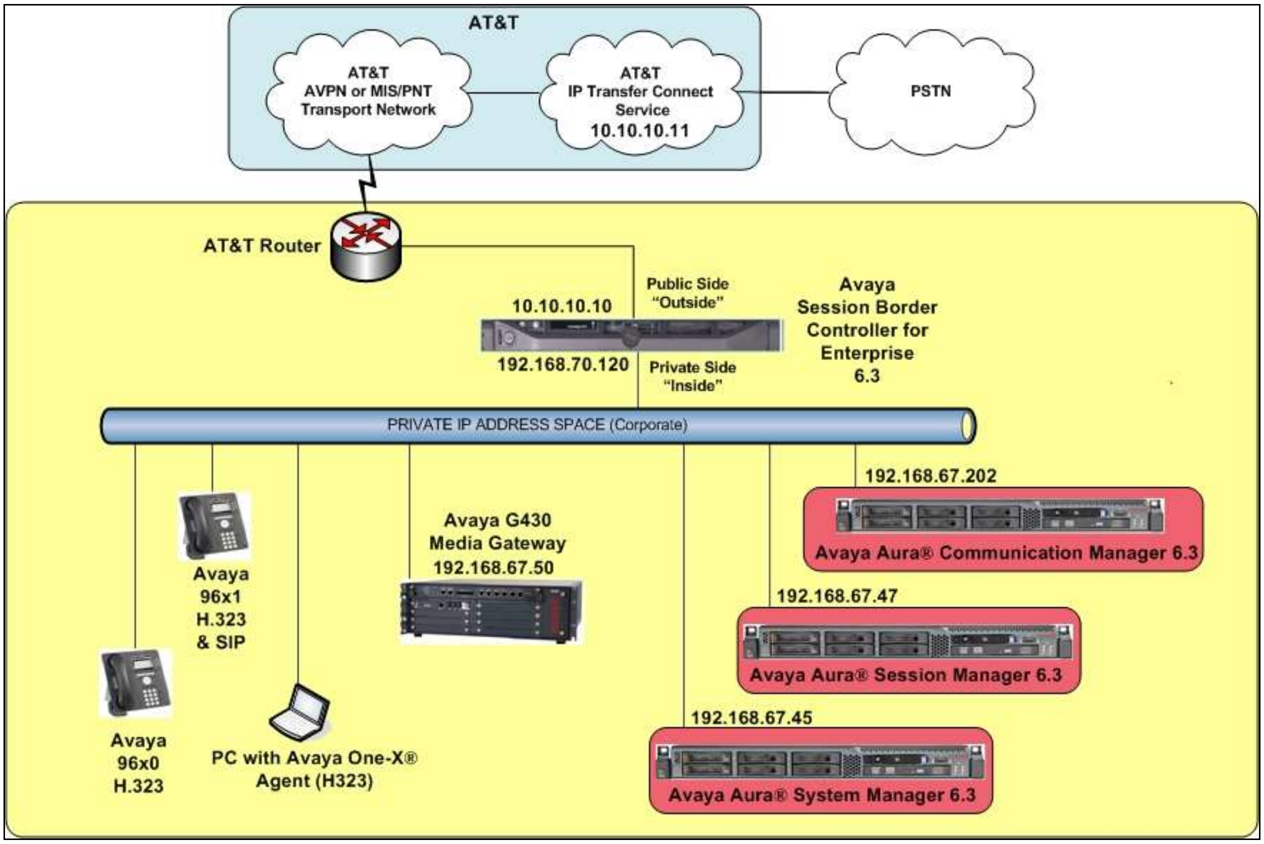


Figure 1: Reference configuration

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 configurations.

Note – The IPTC service Border Element IP address and DID/DNIS digits are shown in this document as examples. AT&T Customer Care will provide the actual IP addresses and DID/DNIS digits as part of the IPTC provisioning process. In addition, the dialed IPTC DID numbers may not be the same as the DNIS numbers that IPTC delivers in the SIP Request URIs. *The provisioning in the following sections is based on the DNIS digits that IPTC delivers,* ***not*** *the dialed DID numbers.*

Component	Illustrative Value in these Application Notes
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
Avaya Aura® System Manager	
IP Address	192.168.67.45
Avaya Session Border Controller for Enterprise (SBCE)	
IP Address of Outside (Public) Interface	10.10.10.10 (see note below)
AT&T IP Transfer Connect Border Element	
IP Address	10.10.10.11 (see note below)

Table 1: Illustrative Values Used in these Application Notes

NOTE – The Avaya SBCE Outside interface communicates with AT&T Border Elements (BEs) located in the AT&T IP Flexible Reach network. For security reasons, the IP addresses of the AT&T BEs are not included in this document. However as placeholders in the following configuration sections, the IP address of **10.10.10.10** (Avaya SBCE public interface), and **10.10.10.11** (AT&T BE IP address), are specified.

3.2. Call Flows

To understand how inbound IPTC service calls are handled by the Avaya SBCE, Session Manager, and Communication Manager, four general call flows are described in this section.

3.2.1. Basic Inbound Call

The first call scenario illustrated in **Figure 2** is an inbound IPTC service call that arrives at 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.

1. A PSTN phone originates a call via the IPTC service.
2. The PSTN routes the call to the IPTC service network.
3. The IPTC service routes the call to the Avaya SBCE.
4. The Avaya SBCE performs SIP Network Address Translation (NAT) and any necessary SIP header modifications, and routes the call to 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.

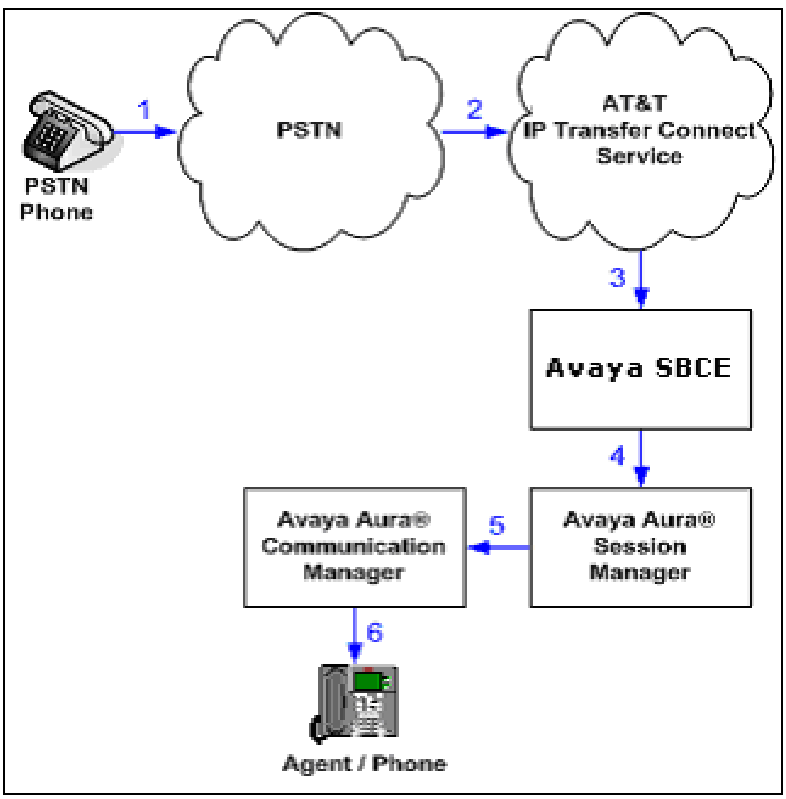


Figure 2: Inbound IPTC Call – No Redirection

3.2.2. 302 Call Redirection

Note: In the call scenarios that follow, the term "alternate destination" does NOT refer to the "Alternate Destination Routing (ADR)" service option of the AT&T IP Toll Free service. ADR and the IPTC service are unrelated.

The second call scenario illustrated in **Figure 3** is an inbound IPTC service call that arrives at 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 immediately redirects the call back to the IPTC 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 IPTC service places a call to the alternate destination and upon answer, connects the calling party to the target party (alternate destination).

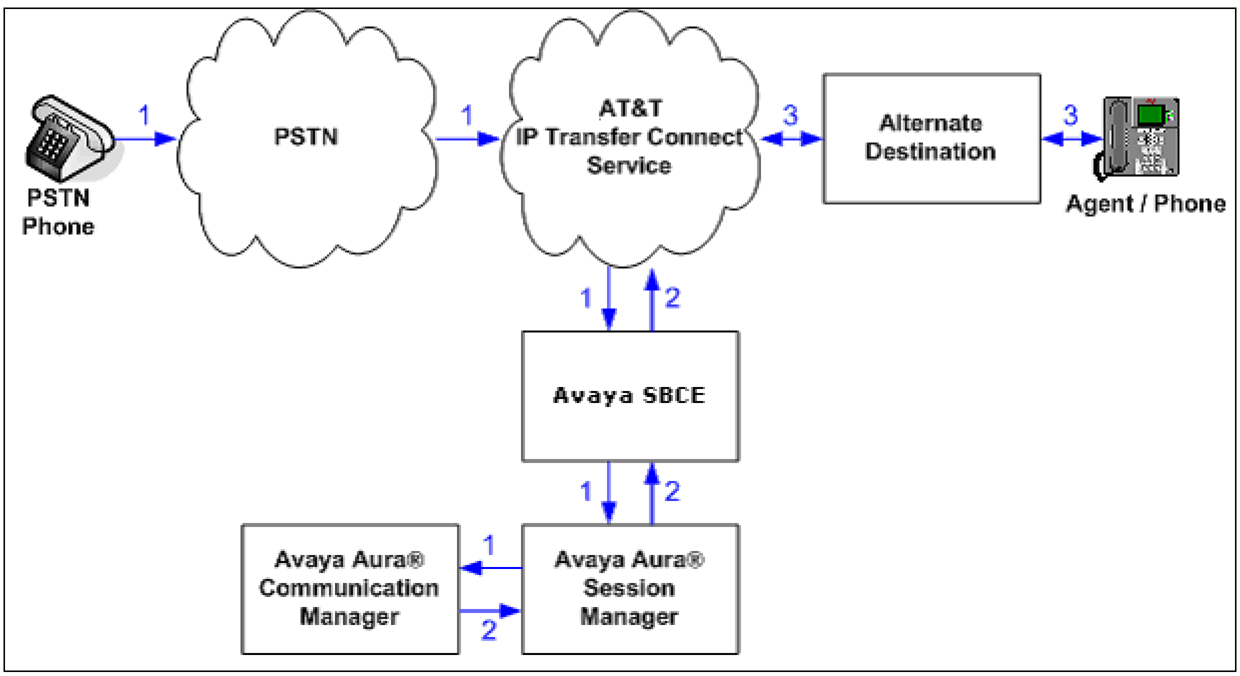


Figure 3: Inbound IPTC Call –SIP 302 Redirection

3.2.3. Refer Call Redirection

The third call scenario illustrated in **Figure 4** is an inbound IPTC service call that 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 IPTC 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 (without a *Replaces* header), 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 IPTC service network.
3. The IPTC service places a call to the target party (alternate destination) and upon answer, connects the calling party to the target party.
4. The IPTC service clears the call on the redirecting/referring party (Communication Manager).

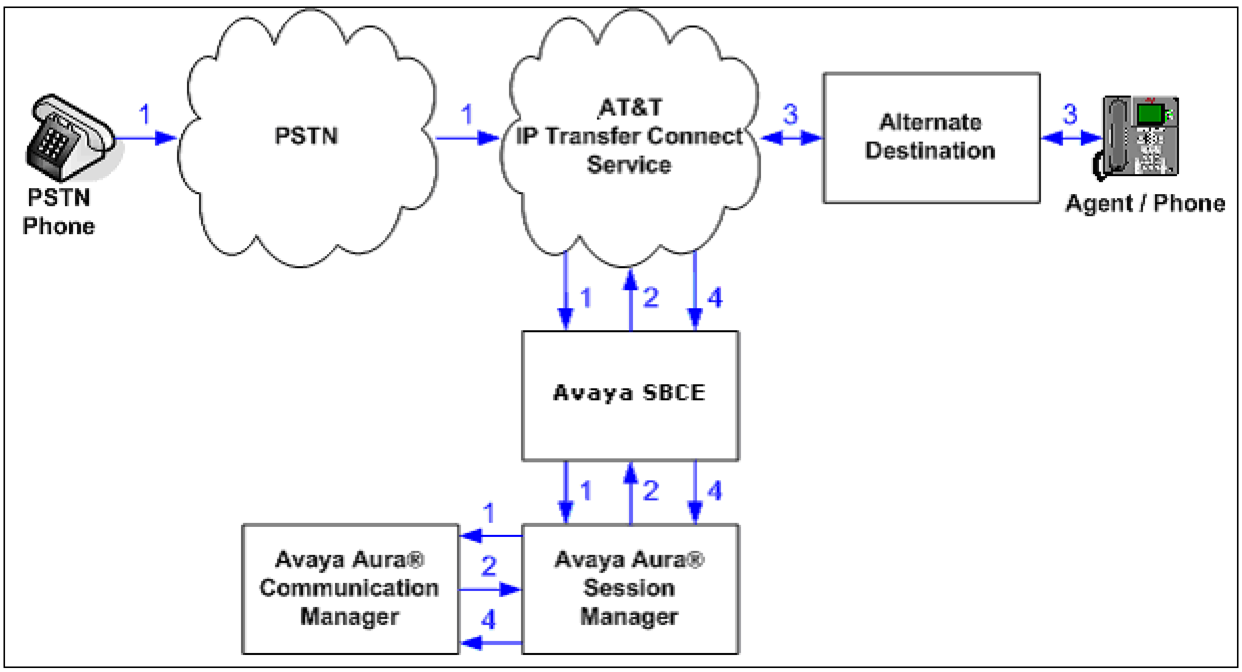


Figure 4: Inbound IPTC Call –SIP REFER Redirection

3.2.4. Refer Call Redirection – Error Handling

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 IPTC service network.
3. The IPTC service places a call to the target party (alternate destination), but the target party is busy or otherwise unavailable.
4. The IPTC 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 IPTC service lines enabled with the IPTC Attended IP Courtesy Transfer feature.

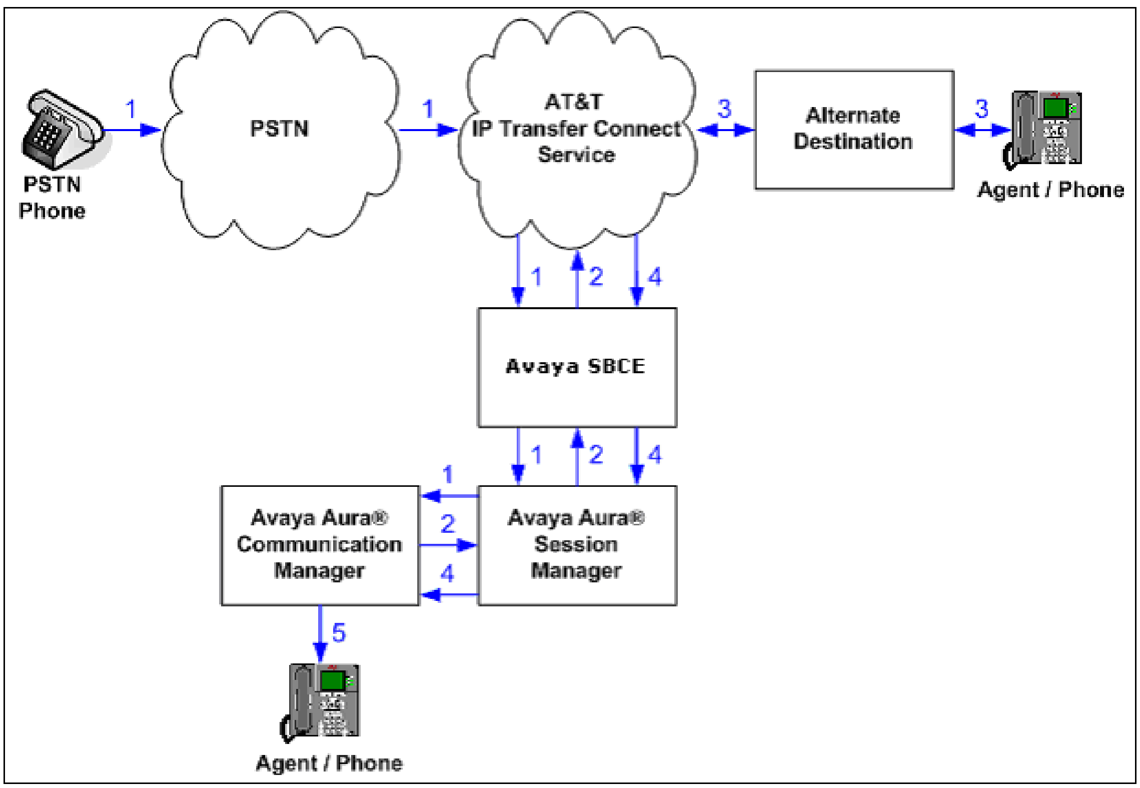


Figure 5: Inbound IPTC Call - Unsuccessful SIP REFER Redirection/Reroute.

4. Equipment and Software Validated

Note – See **Section 2.2, Item 4** regarding the Cisco IOS used in the customer premises AT&T CE router.

Equipment/Software	Release/Version
HP Proliant DL360 G7 server • System Platform • Avaya Aura® System Manager	 • 6.3.5.01003.0 • 6.3 SP 12 (6.3.12_r4903022)
Avaya 8800 server • Avaya Aura® Session Manager	 • 6.3 SP12 (6.3.12.0.631208)
Avaya 8800 server • System Platform • Avaya Aura® Communication Manager	 • 6.3.5.01003.0 • 6.3 SP10 (03.0.124.0-22147)
Avaya G430 Media Gateway	• g430_sw_36_9_0HW7 FW15
Dell R210 • Avaya Session Border Controller for Enterprise	 • 6.3 SP 1 (6.3.1-22-4653)
Avaya 96x1 IP Telephone	• H.323 Version 6.4014 • SIP Version 6.4.125
Avaya one-X® Agent (H.323)	• 2.5.5

Table 2: Equipment and Software Versions

5. Avaya Aura® Session Manager

Session Manager administration for interaction with the AT&T IP Toll Free service is described in document **[11]** and is applicable for the IPTC service as well. This section describes the additional administration steps on Session Manager necessary for supporting interaction with the IPTC service.

5.1. Dial Patterns

Provision additional IPTC service DNIS numbers (digits delivered in the Request URIs of inbound IPTC INVITEs), so they can be converted to Communication Manager VDN, Agent, or station extensions. Follow the procedures described in **Section 5.8** of document **[11]**.

6. Avaya Aura® Communication Manager

Communication Manager administration for interaction with the AT&T IP Toll Free service is described in document **[11]** and is applicable for the IPTC service as well. This section describes the additional administration steps on Communication Manager necessary for supporting interaction with the IPTC service. The steps shown below are performed using 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.

6.1. System Parameters

This section reviews additional Communication Manager licenses and features that are required for supporting the interaction with the IPTC 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.)
```

6.2. Trunks

This section describes the steps for modifying the SIP trunk to Session Manager to support the interaction with the IPTC service.

1. Enter the **change trunk-group x** command, where **x** is the number of the trunk group administered in document **[11]** 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 may alter the Refer NOTIFY response behavior (see **Section 2.2**, **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: From
            Block Sending Calling Party Location in INVITE? n
                 Accept Redirect to Blank User Destination? n
                                              Enable Q-SIP? n
          Interworking of ISDN Clearing with In-Band Tones: keep-channel-active
```

6.3. Inbound Call Routing

This section describes the steps for routing inbound IPTC service calls to reach Vector Directory Numbers (VDNs) with corresponding programmable vectors. These vectors contain steps that were used to 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 IPTC service call that invokes SIP NCR (using a SIP 302 message) prior to the call being answered.
- Post-Answer Redirection - An inbound IPTC service call that invokes SIP NCR (using a SIP REFER message) after the call has been answered by a vector and an announcement is played to the caller.

Note - These Application Notes provide rudimentary vector definitions to demonstrate and test the Communication Manager NCR and UUI functionalities. More complex vector functionality would be used by call centers that is tailored to their individual needs. Alternatively, call centers could use dedicated Interactive Voice Response (IVR) applications to trigger embedded, and/or Communication Manager, call redirection functionality. The definition and documentation of those IVR applications, or associated vector programming, are beyond the scope of these Application Notes.

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**, **item 1**). The vector does the following:

1. Assigns the data "**1234567890123456**" to ASAI UUI variable "**A**" (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)**. 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@<IP/domain>), to the IPTC 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
```

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@<IP/domain>)* to the IPTC 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 **[11]** should also be followed for interoperability with the IPTC service.

This section describes the additional administration steps on the Avaya SBCE necessary for supporting interaction with the IPTC service.

7.1. Avaya – Signaling Rules

As described in **Section 7.3.3** of **[11]**, in an effort to reduce overall packet sizes, CPE generated headers not required by AT&T are removed by the Avaya SBCE. Therefore, in an IPTC environment, the 302 messages generated by Communication Manager contain AV-Global-Session-ID and P-Location headers. These headers need to be removed as well.

Step 1 - Select **Domain Policies → Signaling Rules** from the left-hand side menu (not shown).
Step 2 - The Signaling Rules window will open (not shown). From the Signaling Rules menu, select the **Avaya_SR** rule created for IPTF in **[11]**.

7.1.1. Avaya – Signaling Rule Response Headers Tab

Step 1 - Using the same procedures shown in **Section 7.3.3.1.2** of **[11]**, remove the following headers:

- **P-Location header from 3xx responses:**
  - Select the **Response Headers** tab (not shown).
  - Click the **Edit** button and the **Edit Header Control** window will open.
  - Check the **Proprietary Request Header** box.
  - In the **Header Name** field, enter **P-Location**.
  - From the **Response Code** menu select **3xx**.
  - From the **Method Name** menu select **Invite**.
  - For **Header Criteria** select **Forbidden**.
  - From the **Presence Action** menu select **Remove Header**.
  - Click **Finish**.

- **AV-Global-Session-ID header from 3xx responses**.
  - In the **Header Name** field, enter **Endpoint-View**.
  - From the **Response Code** menu select **3xx**.
  - From the **Method Name** menu select **ALL**.
  - Click **Finish**.

The completed Response Headers form is shown below. Note that the screen shot also includes the headers removed in **[11]**.

General | Requests | Responses | Request Headers | Response Headers | Signaling QoS | UCID

Add In Header Control | Add Out Header Control

Row	Header Name	Response Code	Method Name	Header Criteria	Action	Proprietary	Direction		
1	AV-Global-Session-ID	1XX	ALL	Forbidden	Remove Header	Yes	IN	Edit	Delete
2	AV-Global-Session-ID	2XX	ALL	Forbidden	Remove Header	Yes	IN	Edit	Delete
3	AV-Global-Session-ID	3XX	ALL	Forbidden	Remove Header	Yes	IN	Edit	Delete
4	Endpoint-View	1XX	INVITE	Forbidden	Remove Header	Yes	IN	Edit	Delete
5	Endpoint-View	2XX	INVITE	Forbidden	Remove Header	Yes	IN	Edit	Delete
6	P-AV-Message-Id	1XX	ALL	Forbidden	Remove Header	Yes	IN	Edit	Delete
7	P-AV-Message-Id	2XX	ALL	Forbidden	Remove Header	Yes	IN	Edit	Delete
8	P-Location	1XX	ALL	Forbidden	Remove Header	Yes	IN	Edit	Delete
9	P-Location	2XX	ALL	Forbidden	Remove Header	Yes	IN	Edit	Delete
10	P-Location	3XX	ALL	Forbidden	Remove Header	Yes	IN	Edit	Delete
11	Remote-Party-ID	1XX	ALL	Forbidden	Remove Header	No	IN	Edit	Delete
12	Remote-Party-ID	2XX	ALL	Forbidden	Remove Header	No	IN	Edit	Delete

8. Verification Steps

8.1. Call Verification

The call verification steps and troubleshooting tools described for the AT&T Toll Free service described in document **[11]**, also apply to the IPTC service. In addition:

1. Place an inbound call to an IPTC service line enabled with Redirect features. Verify that an appropriate Communication Manager vector immediately redirects the call back to the IPTC service for delivery 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 IPTC service number enabled with Refer features. Verify that an appropriate Communication Manager vector answers the call and then redirects the call back to the IPTC service for delivery to an alternate destination using Refer. Verify two-way talk path and transmission of UUI information as appropriate.

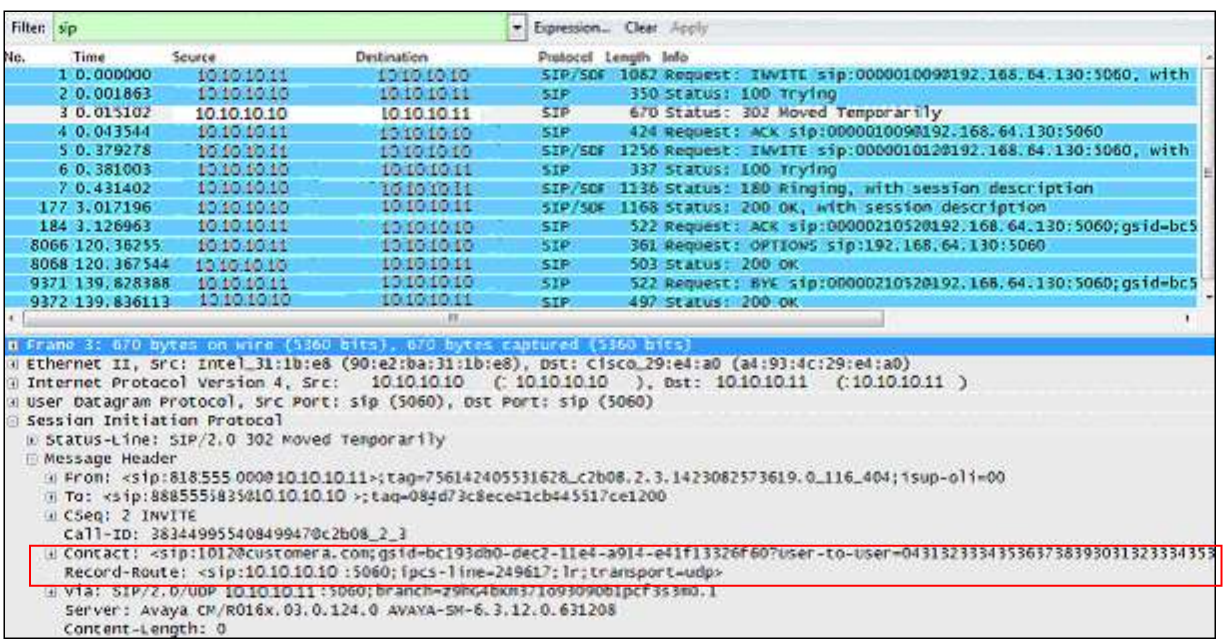
8.2. Protocol Traces

Use a SIP protocol analyzer (e.g. Wireshark), to monitor the SIP traffic at the locations specified below.

8.2.1. 302 Redirection

Monitor the SIP traffic at the Avaya SBCE public “outside” interface connection to the IPTC service. Below is an example of a 302 redirection call filtering on the SIP protocol. Note the following:

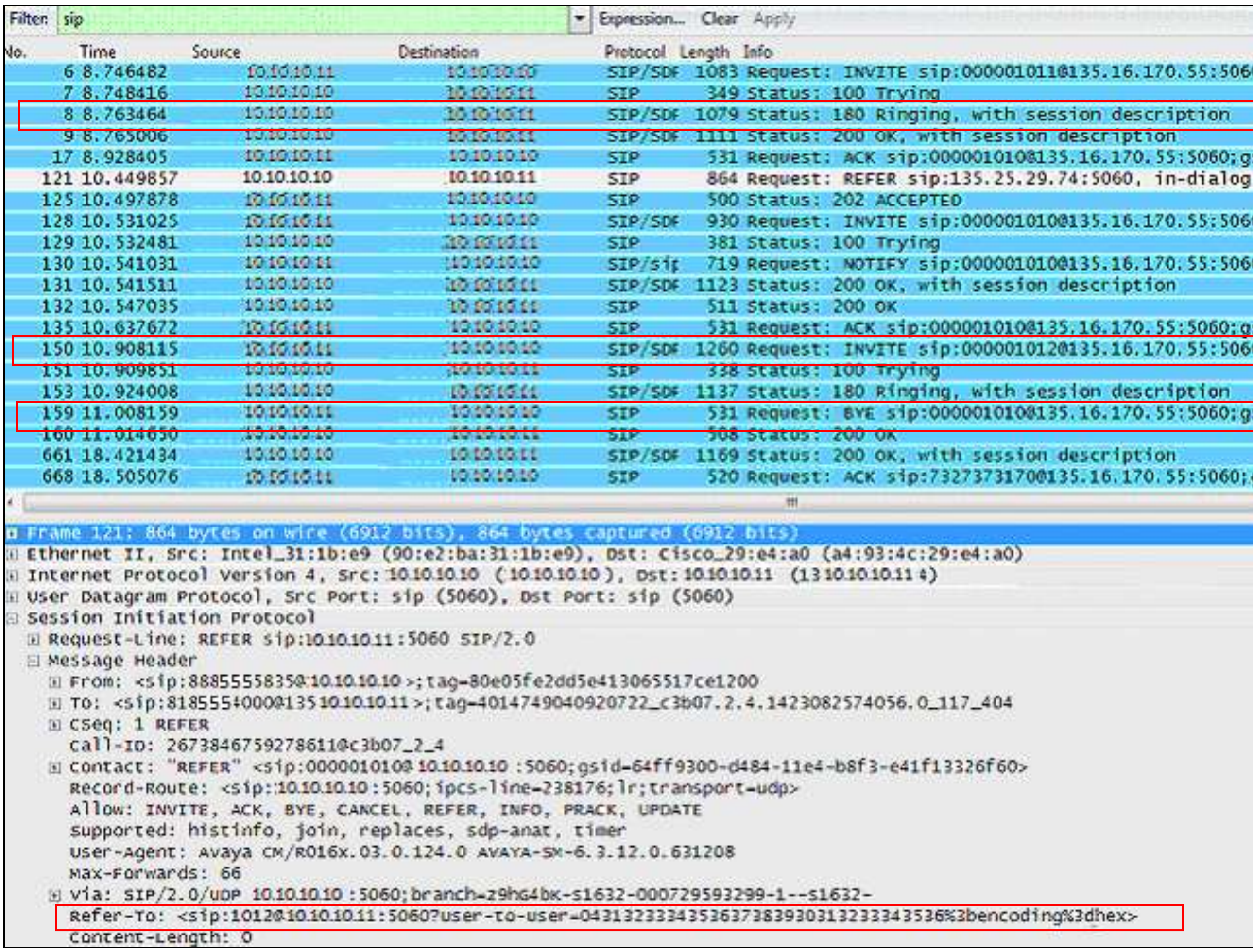
- The Contact header contains the new called number (1012) as defined in vector 22 (see **Section 6.3.1)**.
- The User-to-User Information (UUI) defined in vector 22.
- 180 Ringing is not sent prior to the 302 (see **Sections 2.2, Item 1** and **Section 6.3.1**).



8.2.2. Refer

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

- Communication Manager is sending 180 Ringing in frame 8 (see the comment in **Section 8.2.1**, above).
- The Refer-To header (shown at the bottom of the screen shot) specifies the new called number (1012) as defined in vector 15 (see **Section 6.3.2**). Also note the UUI information also defined on vector 15.
- In frame 150 the IPTC network generates the new INVITE to the 1012 number.
- In frame 159 the IPTC network terminates the initial call leg.



8.2.3. Multipart Headers

The IPTC network may generate SIP Multipart headers. The successful transmission may be confirmed by monitoring the "inside" (A1) interface of the Avaya SBCE (see **Section 2.2, Item 4**). The Multipart headers are distinguished by their "Boundary" statements, which mark the beginning and end of each Multipart section. In addition the SDP information is also contained within the Multipart data. In the example below, the Message Body section of the IPTC INVITE is displayed. Verify that the regular SDP, and the special IPTC XML user data, have been passed by the Avaya SBCE.

```
Filter: sip                                   Expression...  Clear  Apply

No.   Time       Source           Destination      Protocol Length Info
  126 14.291125  192.168.70.120   192.168.67.47    SIP/SDF  1460 Request: INVITE sip:0000055
  128 14.292522  192.168.67.47    192.168.70.120   SIP       378 Status: 100 Trying

⊞ Frame 126: 1460 bytes on wire (11680 bits), 1460 bytes captured (11680 bits)
⊞ Ethernet II, Src: Avaya_3b:75:43 (00:04:0d:3b:75:43), Dst: Ibm_40:56:90 (e4:1f:13:40:56:90)
⊞ Internet Protocol Version 4, Src: 192.168.70.120 (192.168.70.120), Dst: 192.168.67.47 (192.168
⊞ Transmission Control Protocol, Src Port: 12277 (12277), Dst Port: sip (5060), Seq: 2, Ack: 1,
⊟ Session Initiation Protocol
  ⊞ Request-Line: INVITE sip:0000055555415350105O@customera.com SIP/2.0
  ⊞ Message Header
  ⊟ 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 31392 6206 IN IP4 192.168.70.120
            Session Name (s): SIP
          ⊞ Connection Information (c): IN IP4 192.168.70.120
          ⊞ Time Description, active time (t): 0 0
          ⊞ Media Description, name and address (m): audio 17448 RTP/AVP 18 0 97 100
          ⊞ Media Attribute (a): rtpmap:18 G729/8000
          ⊞ Media Attribute (a): fmtp:18 annexb=no
          ⊞ Media Attribute (a): rtpmap:0 PCMU/8000
          ⊞ Media Attribute (a): rtpmap:97 G726-32/8000
          ⊞ Media Attribute (a): rtpmap:100 telephone-event/8000
          ⊞ Media Attribute (a): fmtp:100 0-15
            Media Attribute (a): sendrecv
          ⊞ Media Attribute (a): maxptime:30
        Boundary: \r\n--as-boundary\r\n
      ⊟ Encapsulated multipart part:  (application/nss)
          Content-Type: application/nss\r\n
          Content-Transfer-Encoding: 7bit\r\n
          Content-Disposition: session; handling=required\r\n\r\n
        ⊟ Media Type
            Media Type: application/nss (41 bytes)
        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 6.3 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

Avaya Aura® Session Manager/System Manager

1. Deploying Avaya Aura® Session Manager, Release 6.3, Issue 6, November 2014

2. Administering Avaya Aura® Session Manager, Release 6.3, Issue 7, September 2014

3. Deploying Avaya Aura® System Manager on System Platform, Release 6.3, Issue 4, June 2014

4. Administering Avaya Aura® System Manager for Release 6.3.10, Release 6.3, Issue 6, February 2015

Avaya Aura® Communication Manager

5. Deploying Avaya Aura® Communication Manager on System Platform, Release 6.3, 18-604394, Issue 6, June 2014

6. Administering Avaya Aura® Communication Manager, Release 6.3, 03-300509, Issue 10, June 2014

7. Administering Avaya G430 Branch Gateway, Release 6.3, 03-603228, Issue 5, October 2013

8. Programming Call Vectors in Avaya Aura® Call Center, 6.0, June 2010

Avaya Session Border Controller for Enterprise

9. Administering Avaya Session Border Controller for Enterprise, Release 6.3, Issue 4, October 2014

10. Deploying Avaya Session Border Controller for Enterprise, Release 6.3, Issue 4, October 2014

Avaya Application Notes

11. *Application Notes for Avaya Aura® Communication Manager 6.3, Avaya Aura® Session Manager 6.3 and Avaya Session Border Controller for Enterprise 6.3 with AT&T IP Toll Free SIP Trunk Service – Issue 1.0, March 2015*: https://www.devconnectprogram.com/fileMedia/download/b5f6b53a-ba8e-45ca-b833-a03d6d4d2923

AT&T IP Transfer Connect Service:

The AT&T Transfer Connect service is part of the AT&T IP Toll Free offer.

- AT&T IP Toll Free Service description - http://www.corp.att.com/gov/solution/network_services/voice_services/tollfreeserv.html

- AT&T IP Toll Free service support: (800) 325-5555.

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.