Avaya Solution & Interoperability Test Lab

# Application Notes for Avaya Aura® Communication Manager 6.0, Avaya Aura® Session Manager 6.0, and Acme Packet Net-Net Session Director 6.2.0 with AT&T IP Transfer Connect Service – Issue 1.1

## Abstract

These Application Notes describe the steps for configuring Avaya Aura® Communication Manager with SIP Network Call Redirection (NCR), Avaya Aura® Session Manager, and the Acme Packet Net-Net Session Director (models 3800, 4250, 4500, and 9200) with the AT&T IP Transfer Connect service using **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. Avaya Aura® Session Manager is a core SIP routing and integration engine that connects disparate SIP devices and applications within an enterprise. Note that these Application Notes are intended to supplement the separate document: *Applications Notes for Avaya Aura® Communication Manager 6.0, Avaya Aura® Session Manager 6.0 and Acme Packet Net-Net Session Director 6.1.0 with AT&T IP Toll Free SIP Trunk Service – Issue 1.0.*

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

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# 1. Introduction

These Application Notes describe the steps for configuring Avaya Aura® Communication Manager SIP Network Call Redirection (NCR), Avaya Aura® Session Manager, and the Acme Packet Net-Net Session Director (models 3800, 4250, 4500, and 9200) Session Border Controller (SBC) with the AT&T IP Transfer Connect service using **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. Both intra-site and IP Transfer Connect call scenarios were tested.

**Note**: The AT&T IP Transfer Connect service does not support rerouting of inbound toll free calls to international destinations. Please contact AT&T for service availability in your area.

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

**Note that these Application Notes are intended to supplement the separate document** *Applications Notes for Avaya Aura® Communication Manager 6.0, Avaya Aura® Session Manager 6.0 and Acme Packet Net-Net Session Director 6.1.0 with AT&T IP Toll Free SIP Trunk Service – Issue 1.0*

## 1.1. Interoperability Compliance Testing

The interoperability compliance testing focused on verifying inbound call flows (see **Section 2.2** for descriptions) to Avaya Aura® Session Manager and subsequent routing to Avaya Aura® Communication Manager, and subsequent redirection messages to AT&T for rerouting to alternate destinations.

## 1.2. Support

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

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[1] MIS/PNT does not support compressed RTP (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.

Avaya customers may obtain documentation and support for Avaya products by visiting http://support.avaya.com. The "Connect with Avaya" section provides the worldwide support directory. 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.

## 1.3. Known Limitations

1. An issue was found with the Acme Packet SBC and its behavior responding to a NOTIFY/403 during Refer processing. However an Acme Packet SBC sip-manipulation alleviated the issue pending a new software fix from Acme Packet. See **Section 6**.
2. Avaya Aura® Communication Manager 6.0 issues a BYE upon receipt of a Notify/200OK during Refer calls (expected behavior was for the Avaya CPE to wait for AT&T IP Transfer Connect service to issue the BYE). However no issues were encountered due to this behavior.
3. Shuffling must be disabled on the Avaya Aura® Communication Manager "local" SIP trunk due to codec negotiation issues with Avaya SIP telephones (see [1] for more information).
4. Avaya Aura® Communication Manager 6.0 inserts a leading plus sign to calling number headers by default (e.g. Update, From, PAI, Contact). The AT&T IP Transfer Connect service does not support the use of digit strings with a leading plus sign ("+") in headers containing calling numbers. See the addendum in the AT&T IP Toll Free document [1] that describes an alternate method, utilizing the Acme Packet Net-Net Session Director, to remove the plus signs inserted by Avaya Aura® Communication Manager.
5. The Avaya Aura® Communication Manager Network Call Redirection (NCR) feature is required to enable Refer and 302 call redirection with the AT&T IP Transfer Connect Service (see **Section 5.1**). With this feature enabled, Avaya Aura® Communication Manager will also use the SIP parameter *SendOnly* to signal any hold call conditions. The *SendOnly* SIP parameter is not supported by the AT&T Flexible Reach service. Any customers that access both AT&T IP Transfer Connect and AT&T IP Flexible Reach services via the same Avaya Aura® Communication Manager environment, must use the procedures described in **Addendum 1** of this document that describes having the Acme Packet SBC replace the *SendOnly* parameter with the *SendRecv* parameter that the AT&T Flexible Reach service does support.

# 2. Reference Configuration

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

- Avaya Aura® Session Manager provides core SIP routing and integration services that enables communications between disparate SIP-enabled entities, e.g., PBXs, SIP proxies, gateways, adjuncts, trunks, applications, etc. across the enterprise. Avaya Aura® Session Manager allows enterprises to implement centralized and policy-based routing, centralized yet flexible dial plans, consolidated trunking, and centralized access to adjuncts and applications.

- Avaya Aura® System Manager provides a common administration interface for centralized management of all Avaya Aura® Session Manager instances in an enterprise.
- Avaya Aura® Communication Manager provides the voice communications services for a particular enterprise site. In the reference configuration, Avaya Aura® Communication Manager runs on an Avaya S8800 Server in a Processor Ethernet (Procr) configuration. This solution is extensible to other Avaya S8xxx Servers.
- The Avaya Media Gateway provides the physical interfaces and resources for Avaya Aura® Communication Manager. In the reference configuration, an Avaya G450 Media Gateway is used. This solution is extensible to other Avaya Media Gateways.
- Avaya "desk" phones are represented with Avaya 4600 and 9600 Series IP Telephones running H.323 software, 9600 Series IP Telephones running SIP software, Avaya 6211 Series Analog Telephones, as well as Avaya one-X® Communicator and Avaya one-X® Agent, PC based H323 softphones.
- The Acme Packet Net-Net Session Director (SD) 3800[3] 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. UDP transport protocol is used between the Acme Packet Net-Net SD and the AT&T IP Transfer Connect service.
- An existing Avaya Modular Messaging system (in Multi-Site mode in this reference configuration) provides the corporate voice messaging capabilities in the reference configuration. The provisioning of Modular Messaging is beyond the scope of this document.
- Inbound calls were placed from PSTN via the AT&T IP Transfer Connect service, through the Acme Packet Session Director to the Session Manager which routed the call to Avaya Aura® Communication Manager. Avaya Aura® Communication Manager terminated the call to the appropriate agent/phone or fax extension. The H.323 phones on the enterprise side registered to the Avaya Aura® Communication Manager Processor Ethernet interface (Procr, see [1]) . The SIP phones on the enterprise side registered to the Avaya Aura® Session Manager.

[3] Although an Acme Packet Net-Net SD 3800 was used in the reference configuration, the 4250, 4500 and 9200 platforms are also supported.

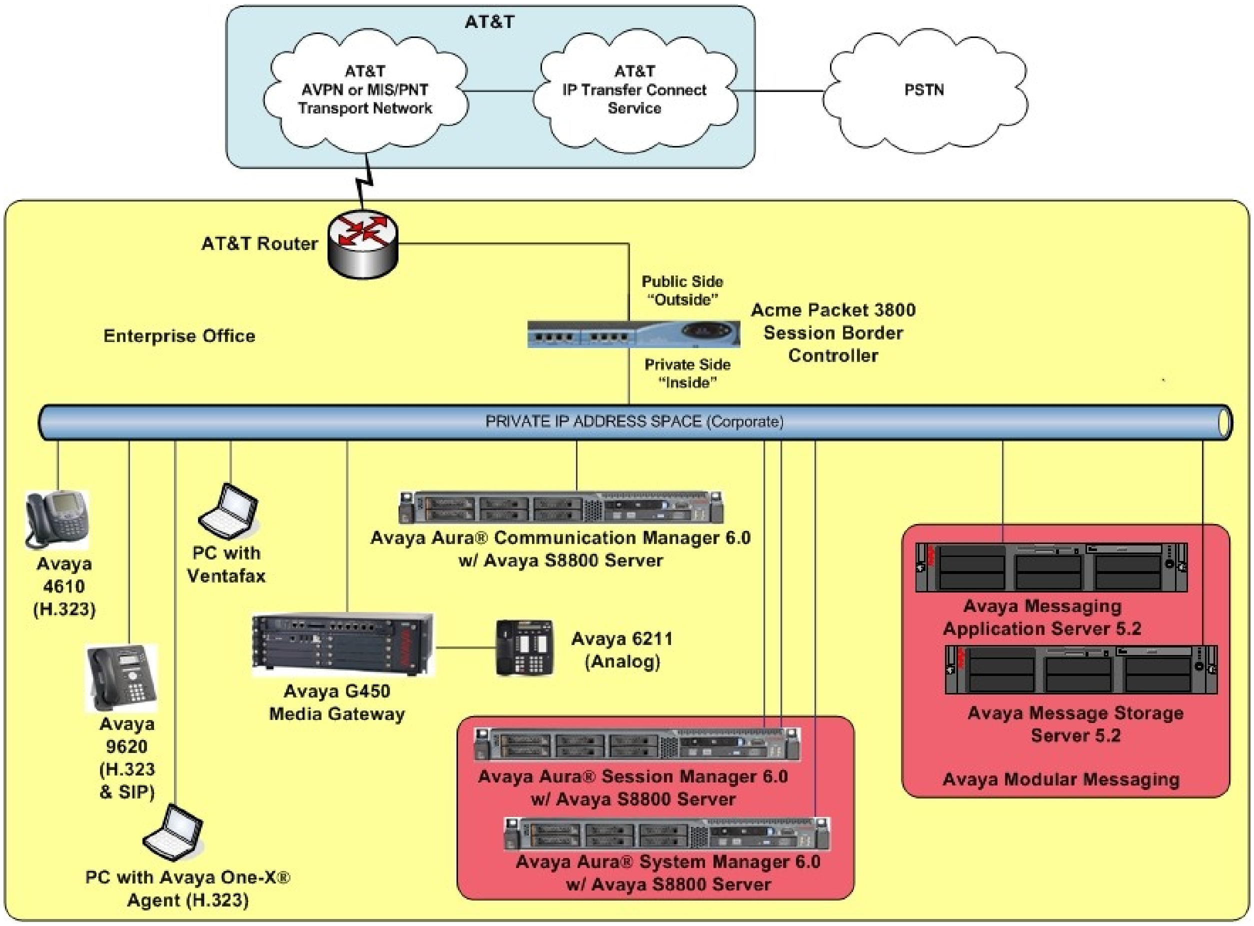


**Figure 1: Reference Configuration**

## 2.1. 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), shown in this document are examples. AT&T Customer Care will provide the actual IP addresses and DNIS digits as part of the IP Transfer Connect provisioning process.

| Component | Illustrative Value in these Application Notes |
|---|---|
| **Avaya Aura® System Manager** | |
| Management IP Address | 192.168.67.207 |
| **Avaya Aura® Session Manager** | |
| Management IP Address | 192.168.67.209 |
| Network IP Address | 192.168.67.210 |
| **Avaya Aura® Communication Manager** | |
| Procr IP Address | 192.168.67.202 |
| Avaya CPE local dial plan | 4xxxx |
| Avaya Aura® Communication Manager extensions | 40xxx = H323 and Analog<br>41xxx = SIP |
| Vector Directory Number (VDN) Extensions | 44xxx |
| Skill (Hunt Group) Extensions | 43xxx |
| Agent Extensions | 47000 |
| Voice Messaging Pilot Extension | 46000 |
| **Avaya Modular Messaging** | |
| Messaging Application Server (MAS) IP Address | 192.168.67.141 |
| Messaging Server (MSS) IP Address | 192.168.67.140 |
| Modular Messaging Dial Plan | 1723114xxxx |
| **Acme Packet SBC** | |
| IP Address of “Outside” (Public) Interface (connected to AT&T Access Router/IP Transfer Connect Service) | 192.168.64.130 (active) |
| IP Address of “Inside” (Private) Interface (connected to Avaya Aura® Session Manager) | 192.168.67.130 (active) |
| **AT&T IP Transfer Connect Service** | |
| Border Element IP Address | 135.25.29.74 |
| AT&T Access router interface (to Acme Packet outside) | 192.168.64.254 |
| AT&T Access Router NAT address (Acme Packet outside address) | 135.16.170.55 |

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

## 2.2. Call Flows

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

The first call scenario illustrated in **Figure 2** is an inbound AT&T IP Transfer Connect service call that arrives on Avaya Aura® Session Manager and is subsequently routed to Avaya Aura® 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 Acme Packet SBC.
4. The Acme Packet SBC performs SIP Network Address Translation (NAT) and any necessary SIP header modifications, and routes the call to Avaya Aura® Session Manager.
5. Avaya Aura® 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, Avaya Aura® Session Manager routes the call to Avaya Aura® Communication Manager.
6. Depending on the called number, Avaya Aura® Communication Manager routes the call to a) a vector, which in turn, routes the call to an agent or phone, or b) directly to an agent or phone.

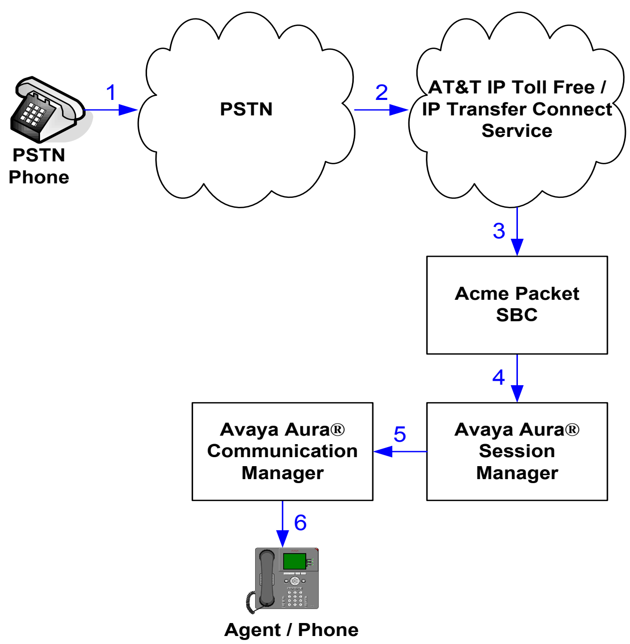


**Figure 2: Inbound AT&T IP Transfer Connect Call – No Redirection**

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

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

1. Same as the first five steps from the first call scenario.
2. Avaya Aura® 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 Avaya Aura® Session Manager and then the Acme Packet SBC to the AT&T IP Transfer Connect service network. Since the SIP 302 message is a final response, the redirecting party (Avaya Aura® 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).

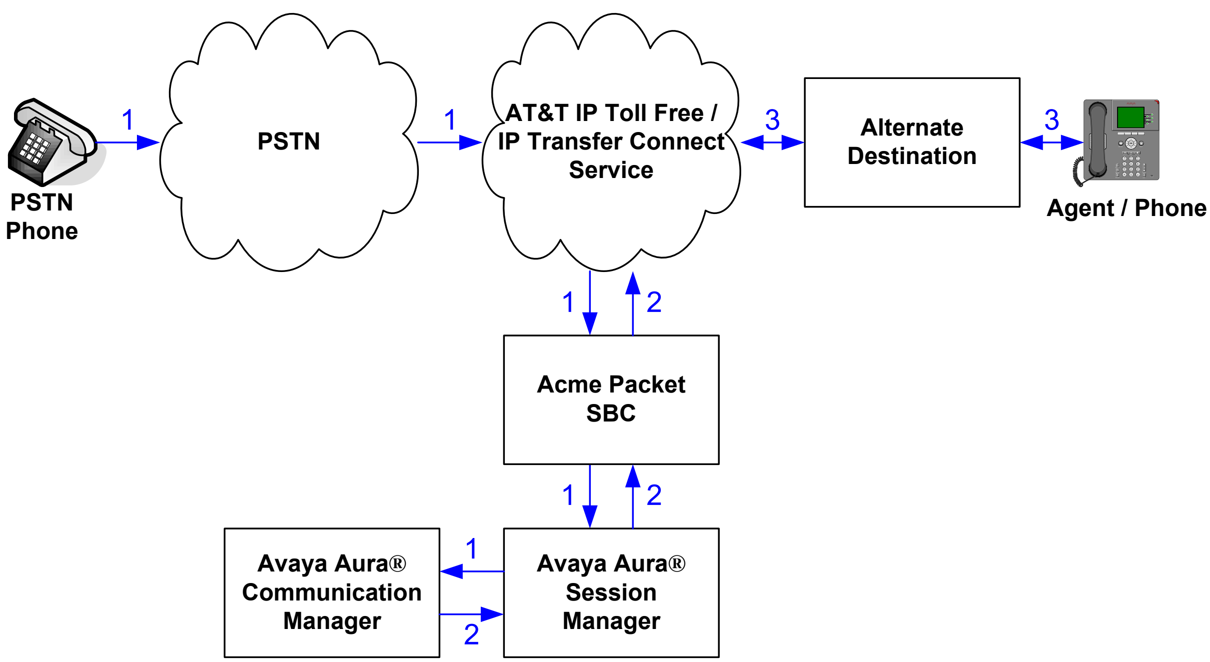


**Figure 3: Inbound AT&T IP Transfer Connect Call – Pre-Answer SIP 302 Redirection**

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

1. Same as the first five steps from the first call scenario.
2. Avaya Aura® 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 Avaya Aura® Session Manager and then the Acme Packet SBC 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 (Avaya Aura® Communication Manager).

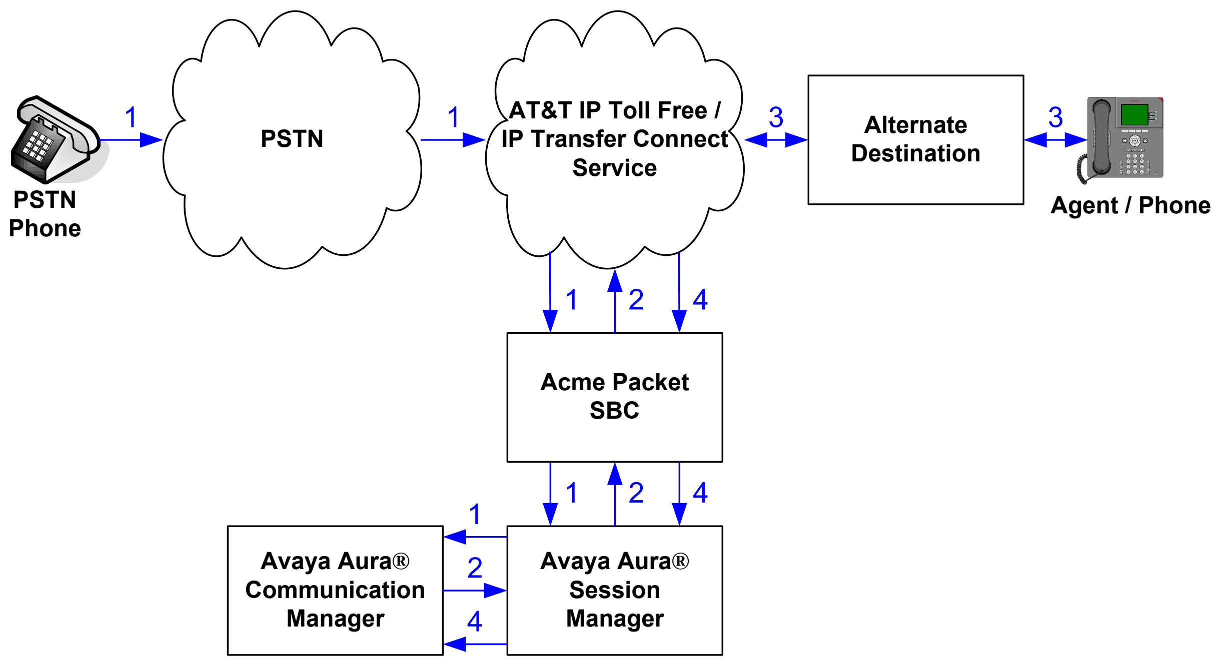


**Figure 4: Inbound AT&T IP Transfer Connect Call – Post-Answer SIP REFER Redirection Successful**

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

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

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

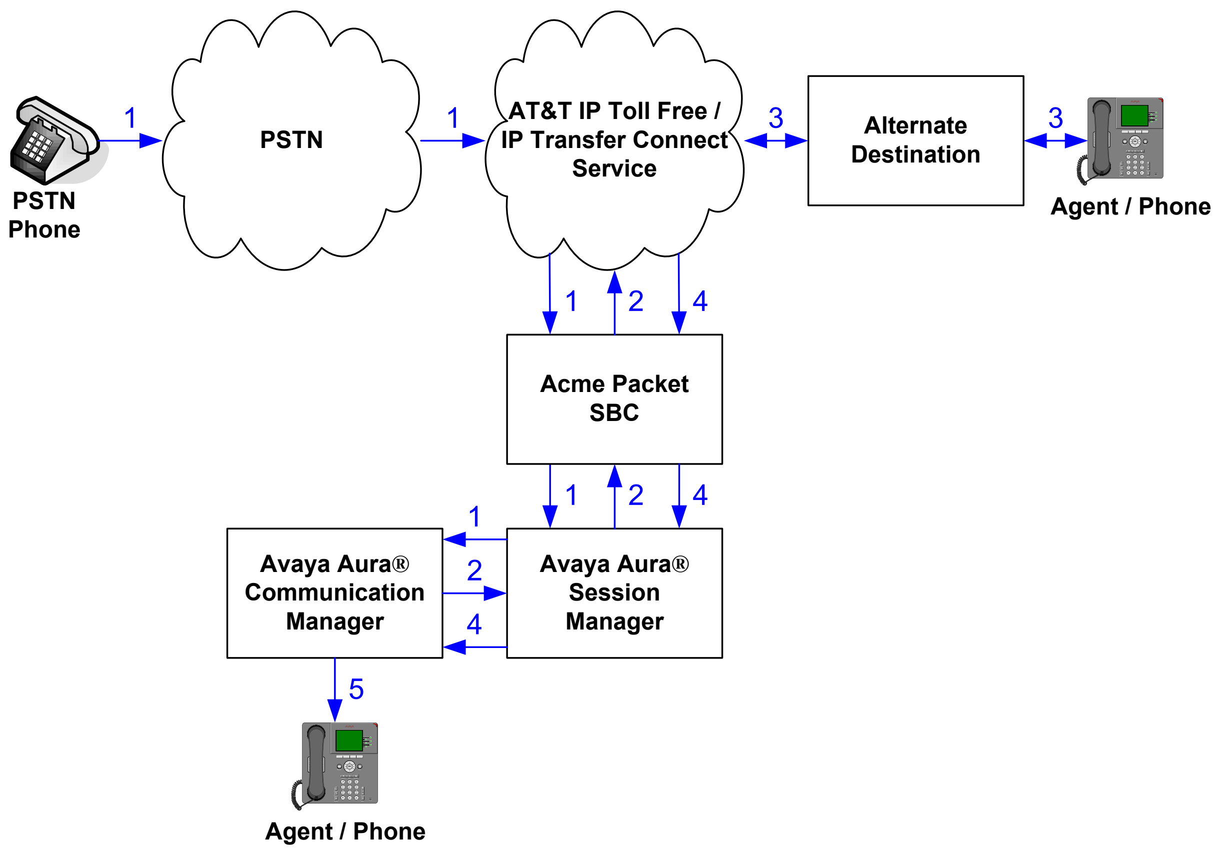


**Figure 5: Inbound AT&T IP Transfer Connect Call – Post-Answer SIP REFER Redirection Unsuccessful**

# 3. Equipment and Software Validated

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

| Component | Version |
|---|---|
| Avaya S8800 Server | Avaya Aura® System Manager 6.0 (6.0.0.0.556-3.0.6.1) |
| Avaya S8800 Server | Avaya Aura® Session Manager 6.0 (6.0.0.0.600020 ) |
| Avaya S8800 Server | Avaya Aura® Communication Manager 6.0 (R016x.00.0.345.0 ) with patch 18246 |
| Avaya G450 Media Gateway | 30.13.2 |
| MM711 Analog | HW31 FW094 |
| Avaya 9630 IP Telephone | Avaya one-X® Deskphone Edition H.323 Version S3.110b (ha96xxua3_11.bin) |
| Avaya 9640 IP Telephone | Avaya one-X® Deskphone Edition SIP Version 2.6.0 (sip96xx_2_6_0_0.bin) |
| Avaya one-X® Communicator | 5.2.0.14 |
| Avaya 4610SW IP Telephone | a10d01b2_9_1.bin |
| Avaya 6211 Analog phone | - |
| Avaya S3500 Server | Avaya Modular Messaging 5.1-4.0 (9.0.424.1.013) |
| Fax device | Ventafax Home Version 6.1.59.144 |
| Acme Packet Net-Net Session Director 3800 | SCX6.2.0m3 |
| AT&T IP Transfer Connect Service using MIS/PNT transport service connections. | VNI 18 |

**Table 2: Equipment and Software Versions**

Note - The solution integration validated in these Application Notes should be considered valid for deployment with Avaya Aura® Communication Manager release 6.0.1 and Avaya Aura® Session Manager release 6.1. Avaya agrees to provide service and support for the integration of Avaya Aura® Communication Manager release 6.0.1 and Avaya Aura® Session Manager release 6.1 with the AT&T IP Transfer Connect service offer, in compliance with existing support agreements for Avaya Aura® Communication Manager release 6.0 and Avaya Aura® Session Manager 6.0, and in conformance with the integration guidelines as specified in the body of this document.

# 4. Avaya Aura® Session Manager

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

## 4.1. Dial Patterns

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

# 5. Avaya Aura® Communication Manager

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

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

## 5.1. System Parameters

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

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

```
display system-parameters customer-options                     Page   4 of  11
                                OPTIONAL FEATURES

   Emergency Access to Attendant? y                            IP Stations? y
           Enable 'dadmin' Login? y
           Enhanced Conferencing? y                     ISDN Feature Plus? y
                  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? n            Local Survivable Processor? n
          Extended Cvg/Fwd Admin? y                  Malicious Call Trace? n
     External Device Alarm Admin? n              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? n              Multifrequency Signaling? y
      Global Call Classification? n       Multimedia Call Handling (Basic)? y
             Hospitality (Basic)? y    Multimedia Call Handling (Enhanced)? y
 Hospitality (G3V3 Enhancements)? n            Multimedia IP SIP Trunking? n
                       IP Trunks? y

           IP Attendant Consoles? n
```

**Figure 6: System-Parameters Customer-Options Form – Page 4**

2. On Page 6 of the **system-parameters customer-options** form, verify that the vectoring features shown in **Figure 7** are set to "**y**".

```
display system-parameters customer-options                  Page   6 of  11
                         CALL CENTER OPTIONAL FEATURES

                          Call Center Release: 5.0

                                ACD? y                          Reason Codes? n
                       BCMS (Basic)? y               Service Level Maximizer? n
         BCMS/VuStats Service Level? n             Service Observing (Basic)? y
  BSR Local Treatment for IP & ISDN? n   Service Observing (Remote/By FAC)? n
                  Business Advocate? n            Service Observing (VDNs)? n
                    Call Work Codes? n                            Timed ACW? n
      DTMF Feedback Signals For VRU? n                    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? n   Vectoring (G3V4 Advanced Routing)? y
          Lookahead Interflow (LAI)? n                    Vectoring (CINFO)? n
Multiple Call Handling (On Request)? n    Vectoring (Best Service Routing)? y
    Multiple Call Handling (Forced)? n                 Vectoring (Holidays)? n
  PASTE (Display PBX Data on Phone)? n                Vectoring (Variables)? y
```

**Figure 7: System-Parameters Customer-Options Form – Page 6**

## 5.2. Trunks

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

1. Enter the **change trunk-group t** command, where **t** is the number of the trunk group administered in [1] for inbound AT&T IP Toll Free service calls. On Page 4 of the **trunk-group** form, set **Network Call Redirection** to "**y**" (see **item 5** in **Section 1.3**).

```
change trunk-group 61                                       Page   4 of  21
                              PROTOCOL VARIATIONS

                      Mark Users as Phone? n
            Prepend '+' to Calling Number? n
      Send Transferring Party Information? n
                 Network Call Redirection? y
                   Send Diversion Header? n
                 Support Request History? y
             Telephone Event Payload Type:
```

**Figure 8: Trunk-Group Form for Inbound AT&T IP Toll Free / IP Transfer Connect Calls – Page 4**

## 5.3. Inbound Call Routing

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

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

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

The following inbound call treatment information is defined using the information provided by AT&T in **Section 2.1**.

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 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 [6] and [7] for further information.

### 5.3.1. Pre-Answer Redirection

This section provides an example of Pre-Answer Redirection. In this example, the inbound call is routed to the VDN shown in **Figure 10**, which invokes the vector shown in **Figure 11**. The vector does the following:

- Plays ringback for 2 seconds (vector step **02**).
- Assigns the data "**1234567890123456**" to ASAI UUI variable "**A**" and "**7890123456789012**" to ASAI UUI variable "**B**" (vector steps **05** and **06**).
  **Note**: The parameters for ASAI UUI variables "**A**" and "**B**", and other vector variables are defined using the **change variables** command (see **Figure 9**).
- 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**" instructs Avaya Aura® 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 Avaya Aura® Session Manager and the Acme Packet Net-Net Session Director).

```
change variables                                                      Page   1 of  39
                              VARIABLES FOR VECTORS

Var Description                     Type    Scope Length Start Assignment      VAC
A   UuiTest1                        asaiuui L     16     1
B   UuiTest2                        asaiuui L     16     17
C
D
E
F
G
H
I
J
K
L
M
N
O
P
Q
R
```

**Figure 9: Change Variables Form**

```
display vdn 31009                                                     Page   1 of   3
                            VECTOR DIRECTORY NUMBER

                             Extension: 31009
                                 Name*: NCR Ringback 302 UUI
                           Destination: Vector Number        1009
                   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 10: Sample VDN for Pre-Answer Redirection**

```
display vector 1009                                            Page   1 of   6
                                  CALL VECTOR

    Number: 1009             Name: NcrRedir_wUui
Multimedia? n      Attendant Vectoring? n    Meet-me Conf? n          Lock? n
     Basic? y   EAS? y   G3V4 Enhanced? y   ANI/II-Digits? y  ASAI Routing? y
 Prompting? y   LAI? n  G3V4 Adv Route? y   CINFO? n   BSR? y  Holidays? n
 Variables? y   3.0 Enhanced? y
01 #    NCR Redirection with ringback and uui forwarding
02 wait-time    2   secs hearing ringback
03
04 #    Define UUI variable to send
05 set          A      = none   CATR  1234567890123456
06 set          B      = none   CATR  7890123456789012
07 #    Immediate redirect to AT&T speed dial number
08 route-to     number ~r1012           with cov n if unconditionally
09
10 #    Play this announcement only on redirect failure
11 announcement 59120
12
```

**Figure 11: Sample Vector for Pre-Answer Redirection**

### 5.3.2. Post-Answer Redirection

This section provides an example of Post-Answer Redirection. In this example, the inbound call is routed to the VDN shown in **Figure 12**, which invokes the vector shown in **Figure 13**. The vector does the following:

- Plays ringback for 2 seconds (vector step **02**).
- Assigns the data "**1234567890123456**" to ASAI UUI variable "**A**" and "**7890123456789012**" to ASAI variable "**B**" (vector steps **05** and **06**).
  **Note**: The parameters for UUI variable "**A**" and other vector variables are defined using the **change variables** command (see **Figure 9**).
- Answers the call to play an announcement (vector step **08**).
- Attempts to redirect the call to the number "**1012**" (vector step **09**). Note that since this vector answered the call, the presence of the "~" in the "**route-to number**" instructs Avaya Aura® 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 Avaya Aura® Session Manager and the Acme Packet Net-Net Session Director).

```
display vdn 31010                                           Page   1 of   3
                            VECTOR DIRECTORY NUMBER

                             Extension: 31010
                                 Name*: NCR Ringback REFER UUI
                           Destination: Vector Number        1010
                   Attendant Vectoring? n
                  Meet-me Conferencing? n
                    Allow VDN Override? n
                                   COR: 1
                                   TN*: 1
                              Measured: none


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



* Follows VDN Override Rules
```

**Figure 12: Sample VDN for Post-Answer Redirection**

```
display vector 1010                                         Page   1 of   6
                                  CALL VECTOR

    Number: 1010             Name: NcrRefer_wUui
Multimedia? n      Attendant Vectoring? n    Meet-me Conf? n          Lock? n
     Basic? y   EAS? y   G3V4 Enhanced? y   ANI/II-Digits? y  ASAI Routing? y
 Prompting? y   LAI? n  G3V4 Adv Route? y   CINFO? n   BSR? y  Holidays? n
 Variables? y   3.0 Enhanced? y
01 #    NCR Refer with ringback and uui forwarding
02 wait-time    2   secs hearing ringback
03
04 #    Define UUI variable to send
05 set          A      = none   CATR  1234567890123456
06 set          B      = none   CATR  7890123456789012
07 #    Refer to AT&T speed dial number
08 announcement 59113
09 route-to     number ~r1012           with cov n if unconditionally
10 #    Play this announcement only on redirect failure
11 disconnect   after announcement 52220
12
```

**Figure 13: Sample Vector for Post-Answer Redirection**

# 6. Acme Packet SBC

The Acme Packet SBC configuration for interaction with the AT&T IP Toll Free service is provided in [1]. The additional configuration on the Acme Packet SBC necessary for supporting interaction with the AT&T IP Transfer Connect service is provided below as a reference.

1. An issue was found during SIP Refer call processing where an invalid route-to number was purposely sent to the AT&T IP Transfer Connect service in the Refer. The expected response from AT&T was a Notify/403 message. The Acme Packet is supposed to send this message on to the Avaya CPE for additional call processing. Instead the Acme Packet responded to the Notify/403 and sent a BYE message to both the AT&T IP Transfer Connect service and the Avaya CPE. Initially Acme Packet support provided a workaround by the use of a *sip-manipulation* that would alleviate the problem. This configuration is shown below. Consult with Acme Packet Support [11] for further details of these parameters.

```
sip-manipulation
      name                            ChangeSubState
      description                     IPTC_NOTIFY
      header-rule
            name                            Subscription-State
            header-name                     Subscription-State
            action                          manipulate
            comparison-type                 case-sensitive
            match-value
            msg-type                        request
            new-value
            methods                         NOTIFY
            element-rule
                  name                            value
                  parameter-name
                  type                            header-value
                  action                          replace
                  match-val-type                  any
                  comparison-type                 pattern-rule
                  match-value                     terminated
                  new-value                       active;expires=1
```

This new sip-manipulation is then applied to the *in-manipulationid* of the existing *OUTSIDE realm-config* described in [1].            .

```
realm-config
      identifier                      OUTSIDE
      description
      addr-prefix                     0.0.0.0
      network-interfaces
                                      s0p0:0
      mm-in-realm                     enabled
      mm-in-network                   enabled
```

```
mm-same-ip                      enabled
mm-in-system                    enabled
bw-cac-non-mm                   disabled
msm-release                     disabled
generate-UDP-checksum           disabled
max-bandwidth                   0
fallback-bandwidth              0
max-priority-bandwidth          0
max-latency                     0
max-jitter                      0
max-packet-loss                 0
observ-window-size              0
parent-realm
dns-realm
media-policy
media-sec-policy
in-translationid
out-translationid
in-manipulationid               ChangeSubState
out-manipulationid              NAT_IP
manipulation-string
manipulation-pattern
```

2. The AT&T IP Transfer Connect service requires that SIP Refer and 302 call redirection messages contain the AT&T Border Element IP address in the Refer-to header of a Refer call and the Contact header of a 302 call. The following sections describe the Acme Packet header-rule provisioning. The header-rules described below were added to the existing sip-manipulation ***NAT_IP*** described in [1].

- **Refer-to header**

```
header-rule
        name                            modReferTo
        header-name                     Refer-To
        action                          manipulate
        comparison-type                 case-sensitive
        msg-type                        any
        methods                         REFER
        match-value
        new-value
        element-rule
            name                            modmline
            parameter-name
            type                            uri-host
            action                          find-replace-all
            match-val-type                  any
            comparison-type                 case-sensitive
            match-value                     customerb.com
            new-value                       $REMOTE_IP
```

- **302 Contact header**

```
header-rule
            name                                mod302
            header-name                         Contact
            action                              manipulate
            comparison-type                     case-sensitive
            msg-type                            Reply
            methods                             INVITE
            match-value
            new-value
            element-rule
                  name                                modmline
                  parameter-name
                  type                                uri-host
                  action                              find-replace-all
                  match-val-type                      any
                  comparison-type                     case-sensitive
                  match-value                         customerb.com
                  new-value                           $REMOTE_IP
```

The following shows the completed ***NAT_IP*** sip-manipulation with the additions described in item 2 above.

```
sip-manipulation
   name                           NAT_IP
   description                    Topology hiding for TO and FROM headers
   split-headers
   join-headers
   header-rule
         name                           manipFrom
         header-name                    From
         action                         manipulate
         comparison-type                case-sensitive
         msg-type                       request
         methods
         match-value
         new-value
         element-rule
               name                           FROM
               parameter-name
               type                           uri-host
               action                         replace
               match-val-type                 any
               comparison-type                case-sensitive
               match-value
               new-value                      $LOCAL_IP
   header-rule
         name                           manipTo
         header-name                    To
         action                         manipulate
         comparison-type                case-sensitive
```

```
        msg-type                        request
        methods
        match-value
        new-value
        element-rule
              name                            TO
              parameter-name
              type                            uri-host
              action                          replace
              match-val-type                  any
              comparison-type                 case-sensitive
              match-value
              new-value                       $REMOTE_IP
  header-rule
        name                            deletePSITE
        header-name                     P-Site
        action                          delete
        comparison-type                 pattern-rule
        msg-type                        request
        methods
        match-value
        new-value

  header-rule
        name                            modPAI
        header-name                     P-Asserted-Identity
        action                          manipulate
        comparison-type                 pattern-rule
        msg-type                        any
        methods                         INVITE
        match-value
        new-value
        element-rule
              name                            modVal
              parameter-name
              type                            uri-user
              action                          find-replace-all
              match-val-type                  any
              comparison-type                 case-sensitive
              match-value                     \+(.*)
              new-value                       $modPAI.$modVal.$1
  header-rule
        name                            modContact
        header-name                     Contact
        action                          manipulate
        comparison-type                 pattern-rule
        msg-type                        any
        methods                         INVITE
        match-value
        new-value
        element-rule
              name                            modVal
              parameter-name
              type                            uri-user
              action                          find-replace-all
```

```
            match-val-type                  any
            comparison-type                 case-sensitive
            match-value                     \+(.*)
            new-value                       $modContact.$modVal.$1
header-rule
      name                            modFrom
      header-name                     From
      action                          manipulate
      comparison-type                 pattern-rule
      msg-type                        any
      methods                         INVITE
      match-value
      new-value
      element-rule
            name                            modVal
            parameter-name
            type                            uri-user
            action                          find-replace-all
            match-val-type                  any
            comparison-type                 case-sensitive
            match-value                     \+(.*)
            new-value                       $modFrom.$modVal.$1
header-rule
      name                            modUpdate
      header-name                     Update
      action                          manipulate
      comparison-type                 pattern-rule
      msg-type                        any
      methods
      match-value
      new-value
      element-rule
            name                            modVal
            parameter-name
            type                            uri-user
            action                          find-replace-all
            match-val-type                  any
            comparison-type                 case-sensitive
            match-value                     \+(.*)
            new-value                       $modUpdate.$modVal.$1
header-rule
            name                            modReferTo
            header-name                     Refer-To
            action                          manipulate
            comparison-type                 case-sensitive
            msg-type                        any
            methods                         REFER
            match-value
            new-value
            element-rule
                  name                            modmline
                  parameter-name
                  type                            uri-host
                  action                          find-replace-all
                  match-val-type                  any
```

```
                  comparison-type               case-sensitive
                  match-value                   customerb.com
                  new-value                     $REMOTE_IP
header-rule
            name                          mod302
            header-name                   Contact
            action                        manipulate
            comparison-type               case-sensitive
            msg-type                      Reply
            methods                       INVITE
            match-value
            new-value
            element-rule
                  name                          modmline
                  parameter-name
                  type                          uri-host
                  action                        find-replace-all
                  match-val-type                any
                  comparison-type               case-sensitive
                  match-value                   customerb.com
                  new-value                     $REMOTE_IP
```

# 7. General Test Approach and Test Results

The test environment consisted of:

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

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

- Inbound AT&T IP Transfer Connect service calls to Avaya Aura® Communication Manager VDNs, agents, and phones.
- Inbound AT&T IP Transfer Connect service calls that are immediately redirected by an Avaya Aura® 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 an Avaya Aura® 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 Avaya Aura® Communication Manager VDNs, agents, and phones (i.e., Avaya Aura® 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 talkpath establishment between callers and Avaya Aura® Communication Manager agents/phones.

The above test objectives with limitations as noted in **Section 1.3** were verified.

# 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 [1], apply to the AT&T IP Transfer Connect service as well.

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

# 9. Conclusion

As illustrated in these Application Notes, Avaya Aura® Session Manager, Avaya Aura® Communication Manager Network Call Redirection, and the Acme Packet Net-Net Session Director 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. 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, 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.


[1] *Applications Notes for Avaya Aura® Communication Manager 6.0, Avaya Aura® Session Manager 6.0 and Acme Packet Net-Net Session Director 6.1.0 with AT&T IP Toll Free SIP Trunk Service – Issue 1.0* - https://devconnect.avaya.com/

[2] *Installing and Configuring Avaya Aura® Session Manager,* Doc ID 03-603473 Release 6.

[3] *Administering Avaya Aura® Session Manager*, Doc ID 03-603324, Release 6.0, June 2010

[4] *Installing and Configuring Avaya Aura® Communication Manager*, Doc ID 03-603558, Release 6.0 June, 2010

[5] *Avaya Aura® Communication Manager Feature Description and Implementation,* Release 6.0, 555-245-205, Issue 8.0, June 2010

[6] *Administering Avaya Aura® Call Center Features*, Release 6.0, June 2010

[7] *Programming Call Vectors in Avaya Aura® Call Center*, 6.0, June 2010

[8] *Modular Messaging Multi-Site Guide Release 5.1*, June 2009

[9] *Modular Messaging for Microsoft Exchange Release 5.1 Installation and Upgrades*, June 2009

[10] *Modular Messaging for the Avaya Message Storage Server (MSS) Configuration Release 5.1 Installation and Upgrades*, June 2009

[11] Acme Packet Support (login required) - http://support.acmepacket.com

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

The Avaya Aura® Communication Manager Network Call Redirection (NCR) feature is required to enable Refer and 302 call redirection with the AT&T IP Transfer Connect Service (see **Section 5.1**). With this feature enabled, Avaya Aura® Communication Manager will also use the SIP parameter *SendOnly* to signal any hold call conditions. The *SendOnly* SIP parameter is not supported by the AT&T Flexible Reach service. Any customers that access both AT&T IP Transfer Connect and AT&T IP Flexible Reach services via the same Avaya Aura® Communication Manager environment, must use the following procedures to have the Acme Packet SBC replace the *SendOnly* parameter with the *SendRecv* parameter that the AT&T Flexible Reach service does support.

**Note** – Though not described in these Application Notes, an alternative solution utilizing separate SIP trunks between Avaya Aura® Communication Manager and Avaya Aura® Session Manager for the AT&T IP Transfer Connect and IP Flexible Reach services (specifying different TCP ports for each), was also tested. The trunk for IP Transfer Connect would have NCR enabled and the trunk for IP Flexible Reach would have NCR disabled.

The header-rules described below were added to the existing sip-manipulation ***NAT_IP*** described in [1].

```
header-rule
        name                modsendonly
        header-name         Content-Type
        action              manipulate
        comparison-type     case-sensitive
        msg-type            any
        methods             INVITE
        match-value
        new-value
        element-rule
            name                modmline
            parameter-name      application/sdp
            type                mime
            action              find-replace-all
            match-val-type      any
            comparison-type     case-sensitive
            match-value         sendonly
            new-value           sendrecv
```

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.