



Avaya Solution & Interoperability Test Lab

Application Notes for Avaya Aura™ Session Manager, Avaya Aura™ Communication Manager SIP Network Call Redirection, and Acme Packet Net-Net Session Director with AT&T IP Transfer Connect Service – Issue 1.0

Abstract

These Application Notes describe the steps for configuring Avaya Aura™ Session Manager, Avaya Aura™ Communication Manager SIP Network Call Redirection (NCR), and the Acme Packet Net-Net Session Director with the AT&T IP Transfer Connect service. 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 separate Application Notes covering Avaya Aura™ Session Manager, Avaya Aura™ Communication Manager, and Acme Packet Net-Net Session Director interaction with the AT&T IP Toll Free service (https://devconnect.avaya.com/public/download/dyn/ASMCM_IPTF.pdf).

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 at the Avaya Solution and Interoperability Test Lab.

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1. Introduction

These Application Notes describe the steps for configuring Avaya Aura™ Session Manager, Avaya Aura™ Communication Manager SIP Network Call Redirection (NCR), and the Acme Packet Net-Net Session Director with the AT&T IP Transfer Connect service. 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 alternate¹ 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.

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 separate Application Notes covering Avaya Aura™ Session Manager, Avaya Aura™ Communication Manager, and Acme Packet Net-Net Session Director interaction with the AT&T IP Toll Free service (see [1]).

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.

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

¹ Note that this is NOT the same as the “Alternate Destination Routing (ADR)” service option available with the AT&T IP Toll Free service.

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. For inbound calls that arrive on an AT&T IP Transfer Connect service line enabled with the IP Courtesy Transfer feature, Avaya Aura™ Communication Manager sends a SIP REFER message (via Avaya Aura™ Session Manager and the Acme Packet SBC) to the AT&T IP Transfer Connect service to invoke a network transfer. The transfer-to (target) number or speed dial code is specified in the Refer-To header of the SIP REFER message. However, if the AT&T IP Transfer Connect service deems the transfer-to number invalid, the network responds with a SIP 404 message. The Acme Packet SBC considers the SIP 404 response to be an indication of an invalid Request-URI in the SIP REFER request, and thereby drops the call about five seconds later. A workaround has been provided in the Acme Packet SBC (see Section 6), though the issue is still under investigation to determine the appropriate response for an invalid transfer-to number.
2. Although Avaya Aura™ Communication Manager release 5.2 supports the possibility of using SIP phones as agent stations, SIP phones were not tested as part of the configuration used to validate this solution.
3. Avaya Aura™ Communication Manager interaction with AT&T IP Transfer Connect service lines enabled with the Unattended IP Courtesy Transfer feature is not supported in this solution.

2. Reference Configuration

The sample 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 this sample configuration, Avaya Aura™ Communication Manager runs on an Avaya S8720 Server. 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 this sample configuration, an Avaya G650 Media Gateway is used. This solution is extensible to other Avaya Media Gateways.
- Avaya “office” phones are represented with Avaya 4600 and 9600 Series IP Telephones running H.323 software, as well as Avaya 6400 Series Digital Telephones.

- The Acme Packet Net-Net Session Director (SD) 3800 provides SIP Session Border Controller (SBC) functionality between the AT&T IP Toll Free service and the enterprise internal network. For brevity, the Acme Packet Net-Net SD 3800 will be referred to as the Acme Packet SBC through the remainder of these Application Notes.
- Avaya Modular Messaging (in MultiSite mode in this sample configuration) provides the corporate voice messaging capabilities for enterprise users.

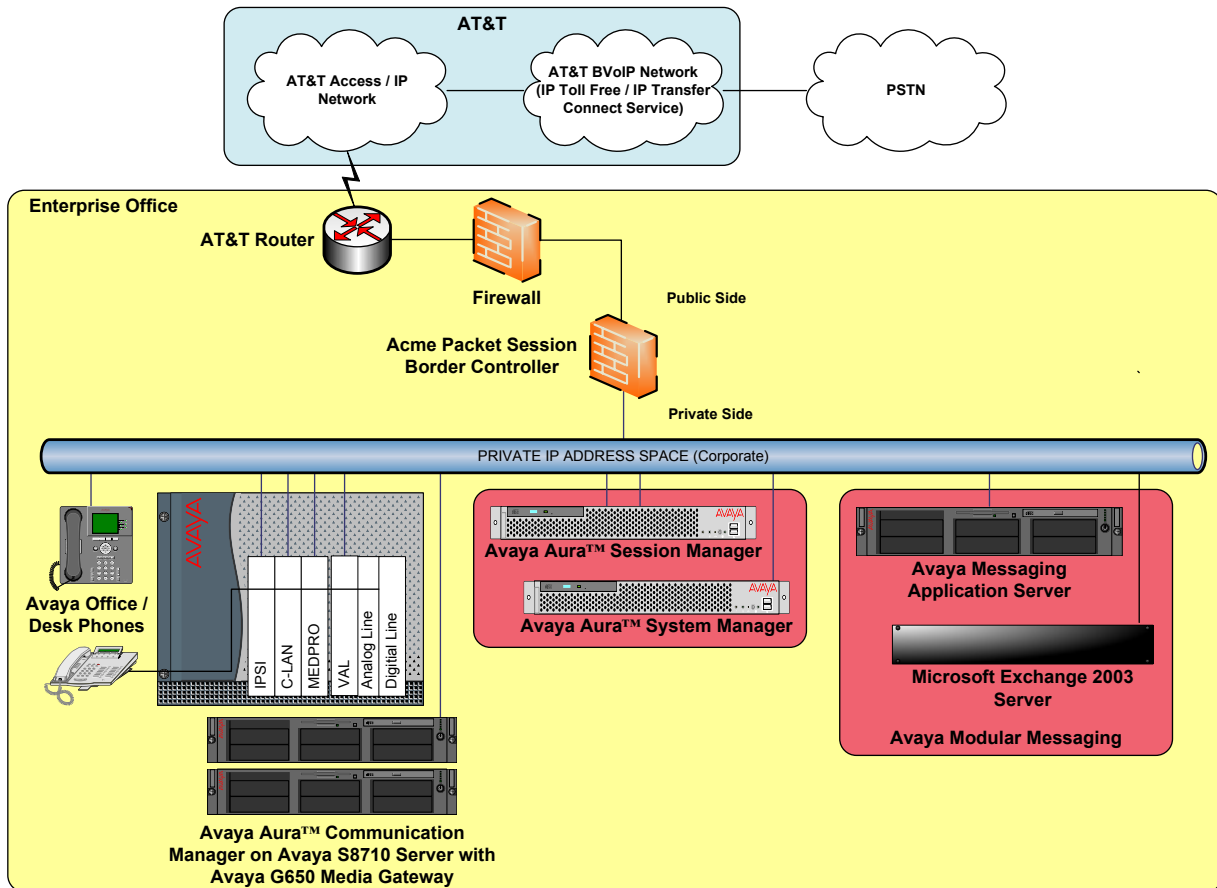


Figure 1: Sample Configuration

2.1. Illustrative Configuration Information

The specific values listed in **Table 1** below and in subsequent sections are used in the sample 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.

Component	Illustrative Value in these Application Notes
Avaya Aura™ System Manager	
Management IP Address	10.160.183.197
Avaya Aura™ Session Manager	
Management IP Address	10.160.183.207
SM100 Card IP Address	10.160.183.209

Component	Illustrative Value in these Application Notes
Avaya Aura™ Communication Manager	
C-LAN IP Address	10.160.179.110
Vector Directory Number (VDN) Extensions / Associated 10-digit Numbers	31xxx / 7328531xxx
Skill (Hunt Group) Extensions / Associated 10-digit Numbers	51xxx / 7328551xxx
Agent Extensions / Associated 10-digit Numbers	32xxx / 7328532xxx
Phone Extensions / Associated 10-digit Numbers	30xxx / 7328530xxx
Announcement Extensions / Associated 10-digit Numbers	52xxx / 7328552xxx
Voice Messaging Pilot Extension	30900
Avaya Modular Messaging	
Messaging Application Server (MAS) IP Address	10.160.183.220
Pilot Number	9089530000
Acme Packet SBC	
IP Address of “Outside” (Public) Interface (connected to AT&T IP Toll Free Service)	10.160.177.210 (active) 10.160.177.211 (primary) 10.160.177.212 (secondary)
IP Address of “Inside” (Private) Interface (connected to Avaya elements)	10.160.183.219 (active) 10.160.183.217 (primary) 10.160.183.218 (secondary)
AT&T IP Toll Free / IP Transfer Connect Service	
Border Element IP Address	10.242.225.200
Digits Passed in SIP Request-URI	0000010xx
IP Transfer Connect Service Lines	- Redirect with Data Forwarding - Attended IP Courtesy Transfer with Data Forwarding - Call Information Delivery

Table 1: Illustrative Values Used in these Application Notes

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 to 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.

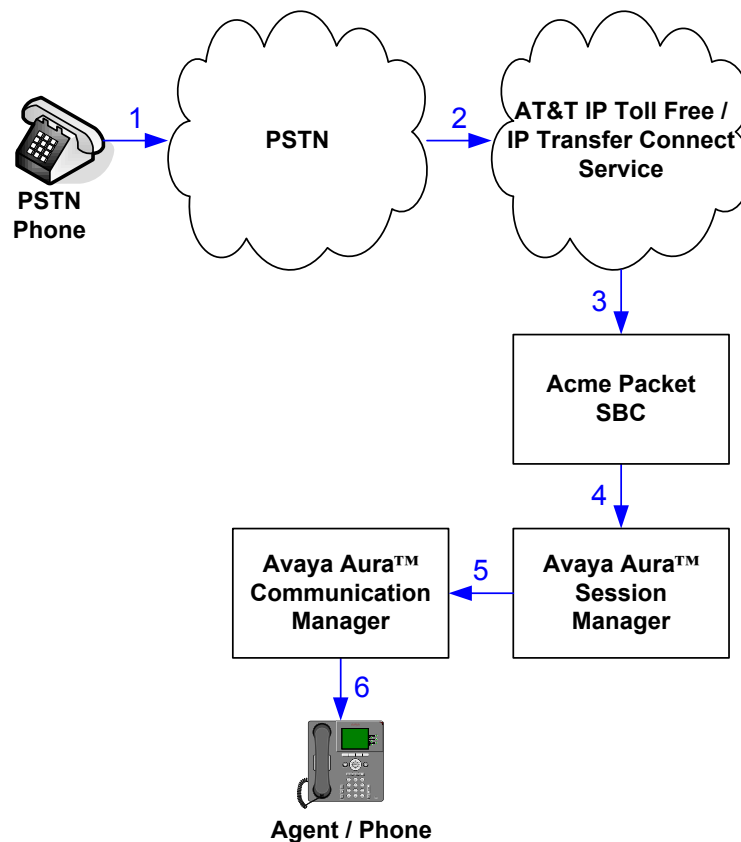


Figure 2: Inbound AT&T IP Transfer Connect Call – No Redirection

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).

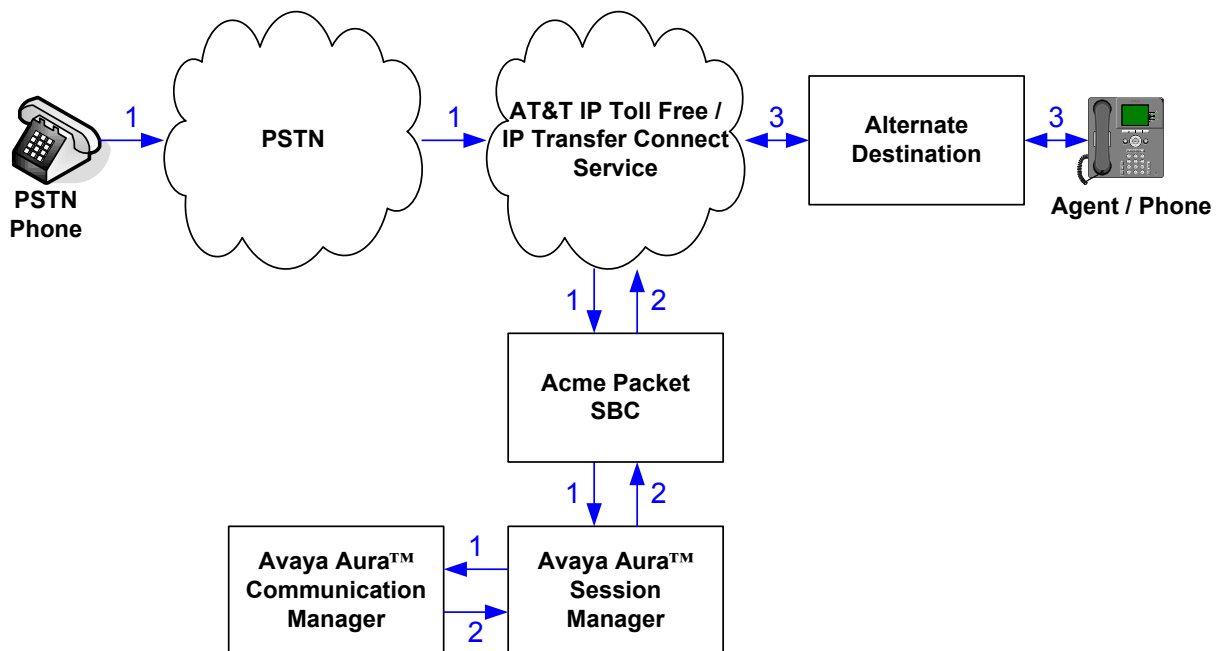


Figure 3: Inbound AT&T IP Transfer Connect Call – Pre-Answer SIP 302 Redirection

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).

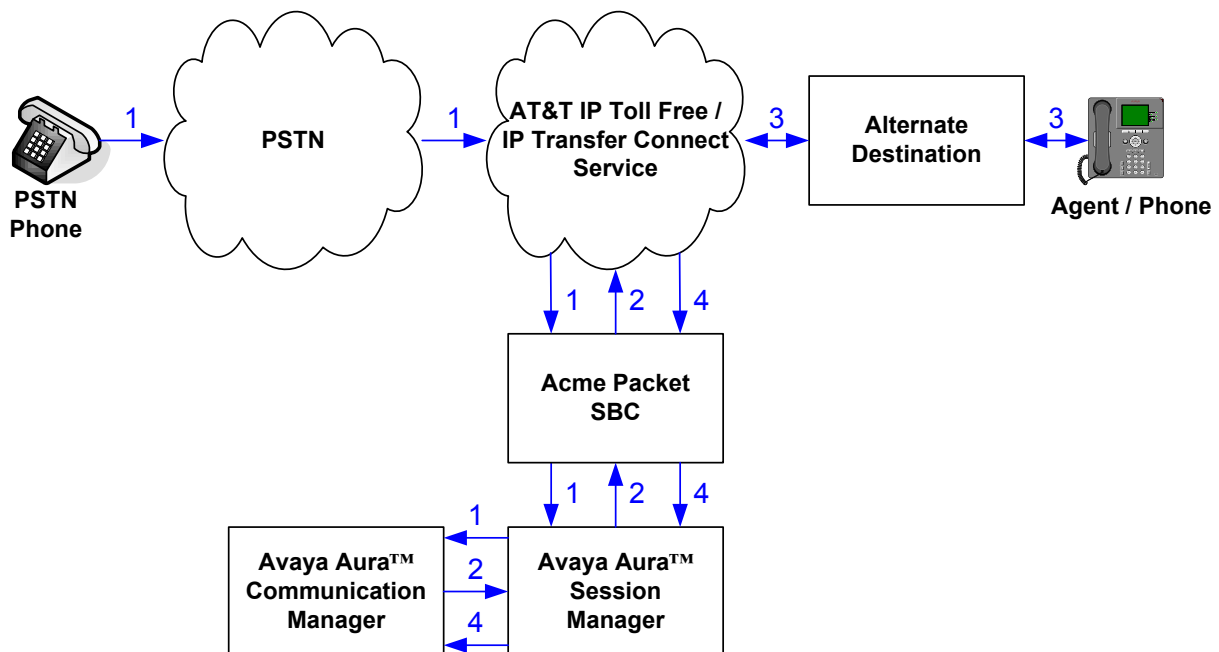


Figure 4: Inbound AT&T IP Transfer Connect Call – Post-Answer SIP REFER Redirection Successful

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.

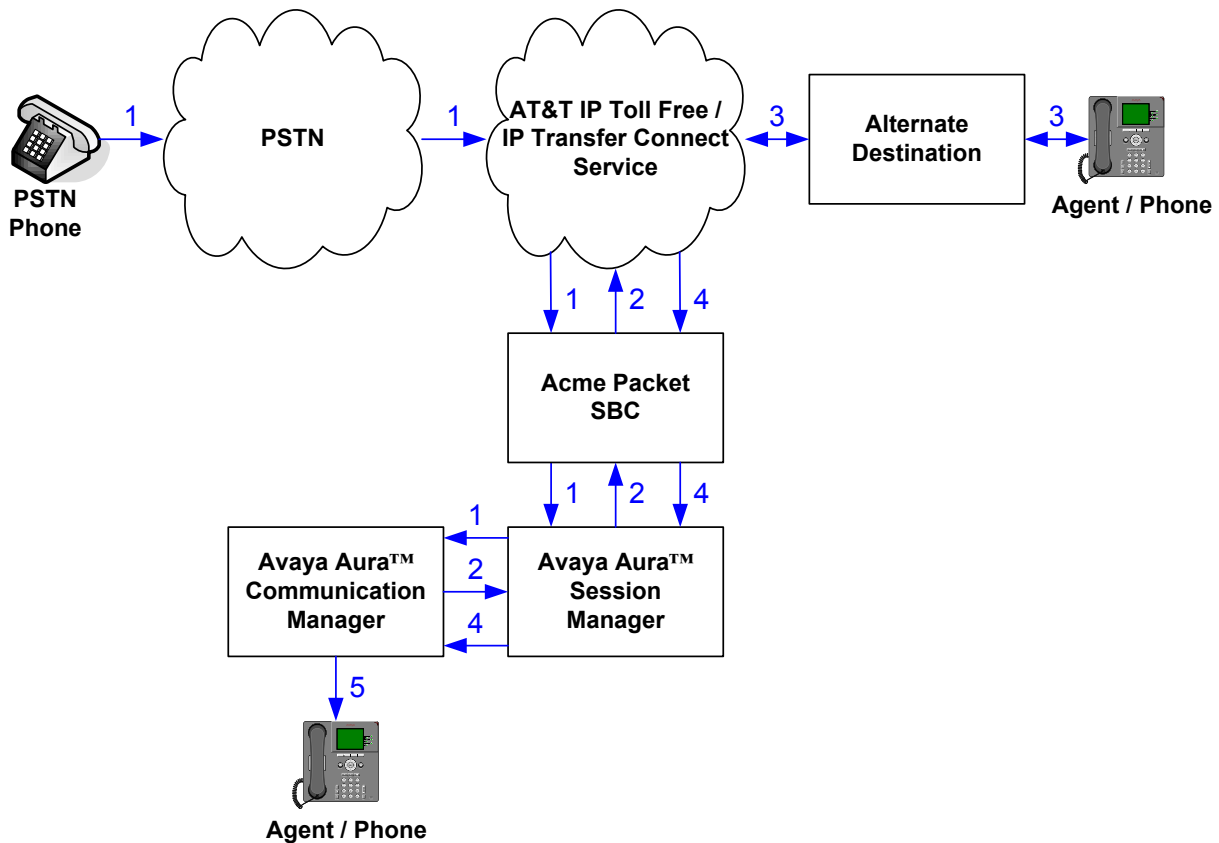


Figure 5: Inbound AT&T IP Transfer Connect Call – Post-Answer SIP REFER Redirection Unsuccessful

3. Equipment and Software Validated

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

Component	Version
Avaya S8510 Server	Avaya Aura™ System Manager 1.1 SP2 (1.1.5.0.112003)
Avaya S8510 Server	Avaya Aura™ Session Manager 1.1 SP2 (1.1.5.0.112003)
SM100 Card	-
Avaya S8710 Server	Avaya Aura™ Communication Manager 5.2 with Service Pack 2.01 (R015x.02.0.947.3 with update 17534)
Avaya G650 Media Gateway	
TN2312BP IP Server Interface (IPSI)	HW03 FW046
TN799DP Control-LAN (C-LAN)	HW01 FW032
TN2302AP IP Media Processor (MedPro)	HW20 FW120
TN2602AP IP Media Resource 320 (MedPro)	HW02 FW047
TN2501AP VAL-ANNOUNCEMENT	HW02 FW021
TN2224CP Digital Line	HW08 FW015
TN793B Analog Line	000005
Avaya 9630 IP Telephone	Avaya one-X™ Deskphone Edition H.323 Release 3.0 Service Pack 1
Avaya 9640 IP Telephone	Avaya one-X™ Deskphone Edition H.323 Release 3.0 Service Pack 1
Avaya 4610SW IP Telephone	2.9 SP1 (2.9.1)
Avaya 6416D+ Digital Telephone	-
Avaya S3500 Server	Avaya Modular Messaging 5.1 with Patch 8 (9.0.370.18)
Microsoft Exchange 2003 Server on Microsoft Windows Server 2003 R2 Enterprise Edition Service Pack 2	6.5.7638.1
Fax	-
Acme Packet Net-Net Session Director 3800	SCX6.1.0 MR-2 GA (Build 398)
AT&T IP Transfer Connect Service	VNI 14

Table 2: Equipment and Software Versions

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]. 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.

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 bolded fields in the following screenshots are 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

- On Page 6 of the **system-parameters customer-options** form, verify that the bolded fields in the following screenshots 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 4

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.

- Enter the **change signaling-group s** command, where **s** is the number of the signaling group administered in [1] for inbound AT&T IP Toll Free service calls. For **Far-end Domain**, enter the IP address of the AT&T Border Element. This IP address will be used as the host/domain part of the URI in the SIP 302 message Contact header and SIP REFER message Refer-To header.

```

change signaling-group 61                                     Page 1 of 1

Group Number: 61          Group Type: sip
                        Transport Method: tls

IMS Enabled? n

Near-end Node Name: clan-01a09          Far-end Node Name: ASM1
Near-end Listen Port: 5061             Far-end Listen Port: 5061
Far-end Network Region: 61
Far-end Domain: 10.242.225.200

Bypass If IP Threshold Exceeded? n

DTMF over IP: rtp-payload          Direct IP-IP Audio Connections? y
Session Establishment Timer(min): 3      IP Audio Hairpinning? n
Enable Layer 3 Test? n              Direct IP-IP Early Media? n
H.323 Station Outgoing Direct Media? n  Alternate Route Timer(sec): 6

```

Figure 8: Signaling-Group Form for Inbound AT&T IP Toll Free / IP Transfer Connect Calls

2. 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”.

```

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 9: 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 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 from Section 2.1.

These Application Notes provide rudimentary vector definitions simply necessary 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 11**, which invokes the vector shown in **Figure 12**. 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 10**).
- 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> (the host/domain is populated with the Far-end Domain value administered in the signaling group on which the inbound call arrived; see Section 5.2 Step 1) to the AT&T IP Transfer Connect service (via Avaya Aura™ Session Manager and the Acme Packet Net-Net Session Director).


```
change variables
```

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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 10: 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 11: 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 12: 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 13**, which invokes the vector shown in **Figure 14**. 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 10**).
- 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> (the host/domain is populated with the Far-end Domain value administered in the signaling group on which the inbound call arrived; see Section 5.2 Step 1) 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 13: 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 14: 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] as a reference. 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. The notable settings are highlighted in bold and brief annotations are provided on the pertinent settings. Consult with Acme Packet Support [8] for further details and explanations on the configuration below.

ANNOTATION: An existing SIP manipulation provided in [1] is modified below to change certain SIP 404 messages received from the AT&T IP Transfer Connect service to SIP 499 messages, specifically SIP 404 messages that are responses to SIP REFER messages sent by the Acme Packet SBC to the AT&T IP Transfer Connect service. The SIP 499 message is a dummy SIP 4xx code that is undefined, and is used here to allow the Acme Packet SBC to process the message rather than dropping the dialog. Note that from [1], this SIP manipulation is already applied on ingress from the AT&T IP Transfer Connect service, i.e., the realm configuration representing the network on which the AT&T IP Toll Free (IP Transfer Connect) service resides. Thus no modification of that realm configuration is necessary.

sip-manipulation

name	RemoveUPDATE
description	Strip Update from Allow list
header-rule	
name	EditAllow
header-name	Allow
action	manipulate
comparison-type	pattern-rule
match-value	
msg-type	any
new-value	
methods	
element-rule	
name	StripUPDATE
parameter-name	
type	header-value
action	find-replace-all
match-val-type	any
comparison-type	pattern-rule
match-value	(,\s*UPDATE UPDATE\s*,)
new-value	
header-rule	
name	MapRefer404
header-name	@status-line
action	manipulate
comparison-type	case-sensitive
match-value	
msg-type	reply
new-value	
methods	REFER
element-rule	
name	StatusCode
parameter-name	

type	status-code
action	replace
match-val-type	any
comparison-type	case-sensitive
match-value	404
new-value	499
last-modified-by	admin@console
last-modified-date	2009-08-25 17:38:29

ANNOTATION: The additional SIP manipulation below changes the SIP 499 messages back to SIP 404 messages.

```

sip-manipulation
  name                Remap499to404
  description
  header-rule
    name              MapRefer499
    header-name       @status-line
    action            manipulate
    comparison-type   case-sensitive
    match-value
    msg-type          reply
    new-value
    methods           REFER
    element-rule
      name            MapStatusCode
      parameter-name
      type            status-code
      action          replace
      match-val-type  any
      comparison-type case-sensitive
      match-value     499
      new-value       404
  last-modified-by   admin@console
  last-modified-date 2009-08-25 20:47:41

```

ANNOTATION: An existing realm configuration provided in [1] is modified to apply the "Remap499to404" SIP manipulation above on egress to Avaya Aura™ Session Manager.

```

realm-config
  identifier          INSIDE-SM
  description
  addr-prefix         0.0.0.0
  network-interfaces
  mm-in-realm         s0p1:0
  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	
in-translationid	
out-translationid	
in-manipulationid	
out-manipulationid	Remap499to404
manipulation-string	
class-profile	
average-rate-limit	0
access-control-trust-level	high
invalid-signal-threshold	0
maximum-signal-threshold	0
untrusted-signal-threshold	0
nat-trust-threshold	0
deny-period	30
ext-policy-svr	
symmetric-latching	disabled
pai-strip	disabled
trunk-context	
early-media-allow	
enforcement-profile	
additional-prefixes	
restricted-latching	none
restriction-mask	32
accounting-enable	enabled
user-cac-mode	none
user-cac-bandwidth	0
user-cac-sessions	0
icmp-detect-multiplier	0
icmp-advertisement-interval	0
icmp-target-ip	
monthly-minutes	0
net-management-control	disabled
delay-media-update	disabled
refer-call-transfer	disabled
codec-policy	
codec-manip-in-realm	disabled
constraint-name	
call-recording-server-id	
stun-enable	disabled
stun-server-ip	0.0.0.0
stun-server-port	3478
stun-changed-ip	0.0.0.0
stun-changed-port	3479
match-media-profiles	
qos-constraint	
last-modified-by	admin@console
last-modified-date	2009-08-25 20:48:36

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 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 of Section 7 with limitations as noted in Section 1.3 were verified.

8. Verification Steps

8.1. Verification Tests

The following steps may be used to verify the configuration:

1. Verify the call routing administration on Avaya Aura™ Session Manager. In the left pane of the Avaya Aura™ System Manager Common Console, under **Session Manager**, click on “**Call Routing Test**”. In the **Call Routing Test** page, enter the appropriate parameters of the test call. **Figure 15** shows a routing test for an inbound call arriving from the Acme Packet SBC (note the IP address “**10.160.183.219**” in the **Calling Party Address** field) to the number “**000001010**” in the SIP domain “**spdevcon.com**” (note the **Called Party URI** field). Click on “**Execute**”.

Figure 15: Call Routing Test Page

- Verify that the test results in the **Routing Decisions** and **Routing Decision Process** are consistent with the expected results of the routing administration administered on Avaya Aura™ Session Manager in Section 4.

Figure 16: Call Routing Test Page – Test Results

3. 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.
4. 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.
5. Verify that when Avaya Aura™ Communication Manager is the transfer target of redirected calls, the calls are answered with two-way talkpath. Verify that the calls remain stable for several minutes and disconnect properly.

8.2. Troubleshooting Tools

The Avaya Aura™ Communication Manager “list trace vector”, “list trace vdn”, “list trace tac”, and/or “status trunk-group” commands are helpful diagnostic tools to verify correct operation and to troubleshoot problems. MST (Message Sequence Trace) diagnostic traces (performed by Avaya Support) can be helpful in understanding the specific interoperability issues.

The logging and reporting functions within the Avaya Aura™ System Manager Common Console may be used to examine the details of Avaya Aura™ Session Manager calls. In addition, if port monitoring is available, a SIP protocol analyzer such as Wireshark (a.k.a. Ethereal) can be used to capture SIP traces at the various interfaces. SIP traces can be instrumental in understanding SIP protocol issues resulting from configuration problems.

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 (UII) features can work in complement with the AT&T implementations of SIP NCR and UII to support call redirection over SIP trunks while preserving initiating caller information. This solution provides contact center users of Avaya Aura™ Communication Manager the ability to redirect inbound AT&T IP Transfer Connect service calls to alternate destinations, and deliver UII-encoded customer information to those alternate destinations for the purposes of invoking contact center applications, e.g., triggering agent screen pop-ups with caller information, etc.

The sample configuration shown in these Application Notes is representative of a basic enterprise customer configuration and is intended to provide configuration guidance to supplement other Avaya product documentation. It is based upon formal interoperability compliance testing as part of the Avaya DevConnect Service Provider program.

10. References

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

- [1] *Application Notes for Avaya Aura™ Session Manager, Avaya Aura™ Communication Manager, and Acme Packet Net-Net Session Director with AT&T IP Toll Free Service*, Issue 1.0.
https://devconnect.avaya.com/public/download/dyn/ASMCM_IPTF.pdf
- [2] *Avaya Aura™ Session Manager Overview*, Issue 1, Release 1.1, May 2009, Document Number 03-603323
- [3] *Installing and Administering Avaya Aura™ Session Manager*, Issue 1.1, Release 1.1, June 2009, Document Number 03-603324
- [4] *Administering Avaya Aura™ Communication Manager*, Issue 5.0, Release 5.2, May 2009, Document Number 03-300509
- [5] *Avaya Aura™ Communication Manager Feature Description and Implementation*, Issue 7, Release 5.2, May 2009, Document Number 555-245-205
- [6] *Avaya Aura™ Call Center 5.2 Call Vectoring and Expert Agent Selection (EAS) Reference*, Release 5.2, April 2009, Document Number 07-600780
- [7] *Avaya Aura™ Call Center 5.2 Automatic Call Distribution Reference*, Release 5.2, April 2009, Document Number 07-602568

Acme Packet Support (login required):

- [8] <http://support.acmepacket.com>

AT&T IP Toll Free Service Descriptions:

- [9] *AT&T IP Toll Free*
<http://www.business.att.com/enterprise/Service/business-voip-enterprise/network-based-voip-enterprise/ip-toll-free-enterprise/>

11. Change History

Issue	Date	Reason
1.0		Initial issue.

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