AVAYA

Avaya Solution & Interoperability Test Lab

Application Notes for Avaya Aura™ Session Manager 5.2, Avaya Aura™ Communication Manager 5.2.1 SIP Network Call Redirection, and Acme Packet Net-Net Session Director 6.1.0 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 the separate document: *Application Notes for Avaya Aura™ Session Manager 5.2, Avaya Aura™ Communication Manager 5.2.1, and Acme Packet Net-Net Session Director 6.1.0 with AT&T IP Toll Free 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™ 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[1] 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 the separate document *Application Notes for Avaya Aura™ Session Manager 5.2, Avaya Aura™ Communication Manager 5.2.1, and Acme Packet Net-Net Session Director 6.1.0 with AT&T IP Toll Free 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.

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

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[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. In an attended Refer call scenario, the initial call leg is supposed to be maintained until the refer-to call leg has been established. During REFER testing, it was found that Communication Manager sends a BYE upon receipt of a NOTIFY/Ringing message. The call completes if the refer-to party answers the call; however if the refer-to party does not answer the call, Communication Manager cannot resume control of the original call leg since it sent a BYE. This issue is being investigated by the Avaya product team and will be addressed in a future release.
2. Depending on how accurate CDRs need to be, reports may be off by the elapsed time between the 180 Ringing sipfrag NOTIFY and BYE messages from the AT&T network.
3. Although Avaya Aura™ Communication Manager release 5.2.1 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.
4. An issue was found with the Acme Packet SBC and its behavior responding to a NOTIFY/403 during Refer processing. However a new Acme Packet SBC software release alleviated the issue. See **Section 6**.
5. Note - Avaya Aura Communication Manager R5.2.1 will send a SIP UPDATE message for any display info changes.

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.
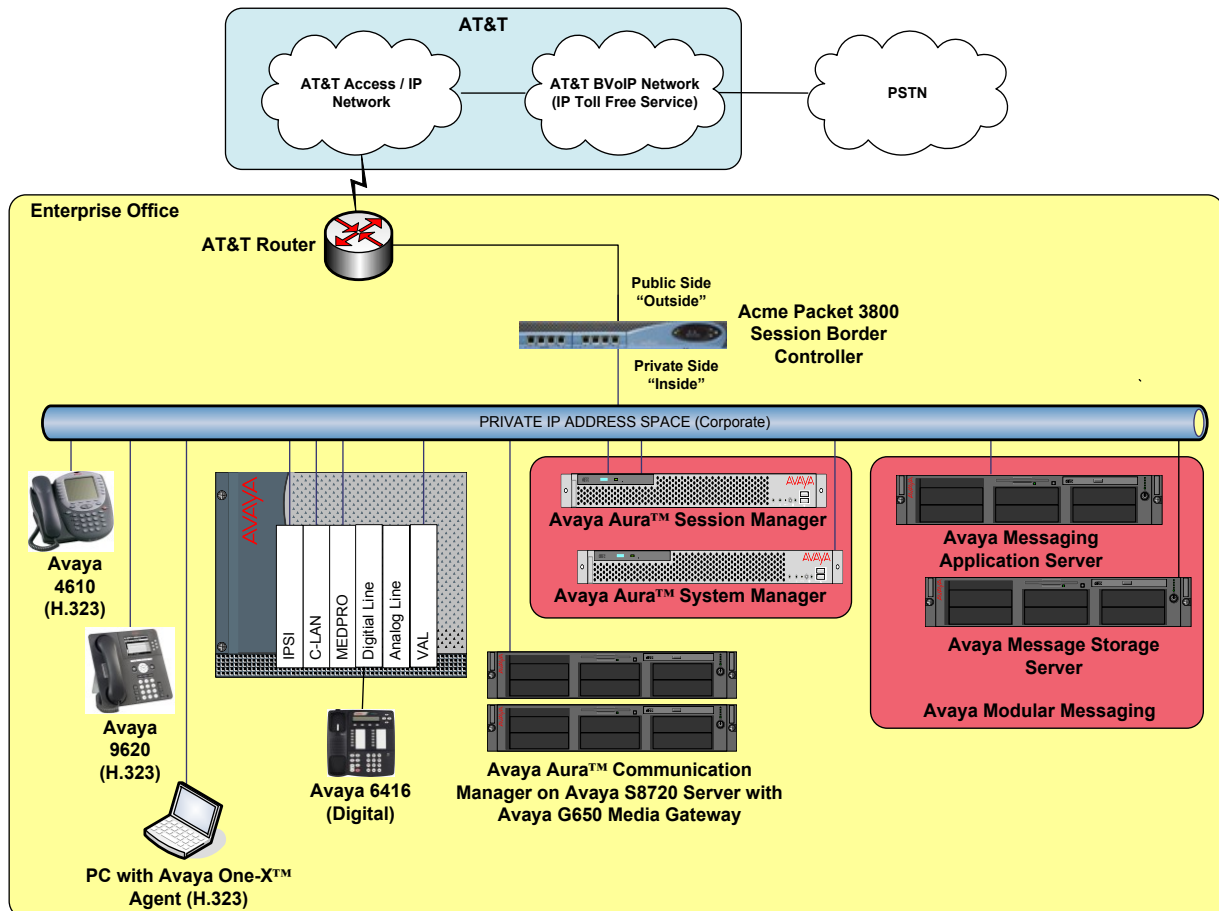

Figure 1: Reference Configuration

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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.

> **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	192.168.67.135
Avaya Aura™ Session Manager	
Management IP Address	192.168.67.136
SM100 Card IP Address	192.168.67.137
Avaya Aura™ Communication Manager	
C-LAN IP Address	192.168.67.13
Vector Directory Number (VDN) Extensions	1010 - 1020
Skill (Hunt Group) Extensions	1002, 1003, 1004
Agent Extensions	26666, 26667, 26668
VDN Extensions	26120, 26121
Phone Extensions	26102, 26103, 26104
Voice Messaging Pilot Extension	26000
Avaya Modular Messaging	
Messaging Application Server (MAS) IP Address	192.168.67.141
Message Storage Server (MSS) IP Address	192.168.67.140
Pilot Number	17231126000
Acme Packet SBC	
IP Address of Public Interface (connected to AT&T IP Toll Free Service)	192.168.64.130
IP Address of Private Interface (connected to Avaya CPE)	192.168.67.130
AT&T IP Transfer Connect Service	
Border Element IP Address	135.25.29.74
Digits Passed in SIP Request-URI	00000108x

Table 1: Illustrative Values Used in the Reference Configuration

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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.
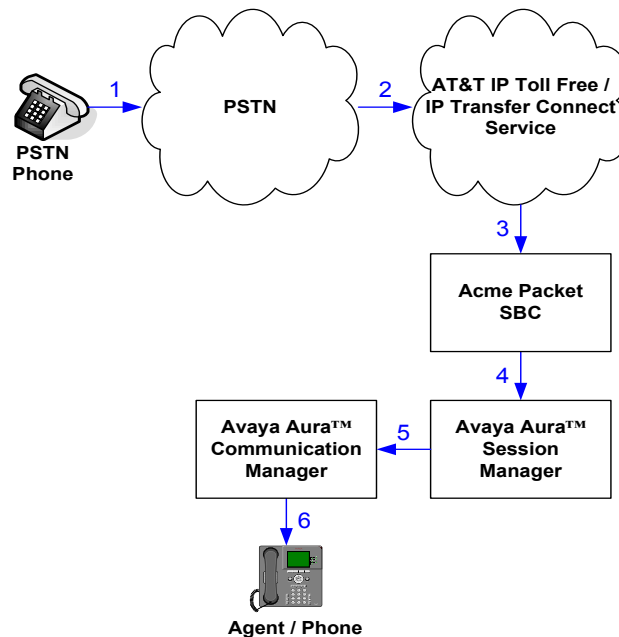

Figure 2: Inbound AT&T IP Transfer Connect Call – No Redirection

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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).
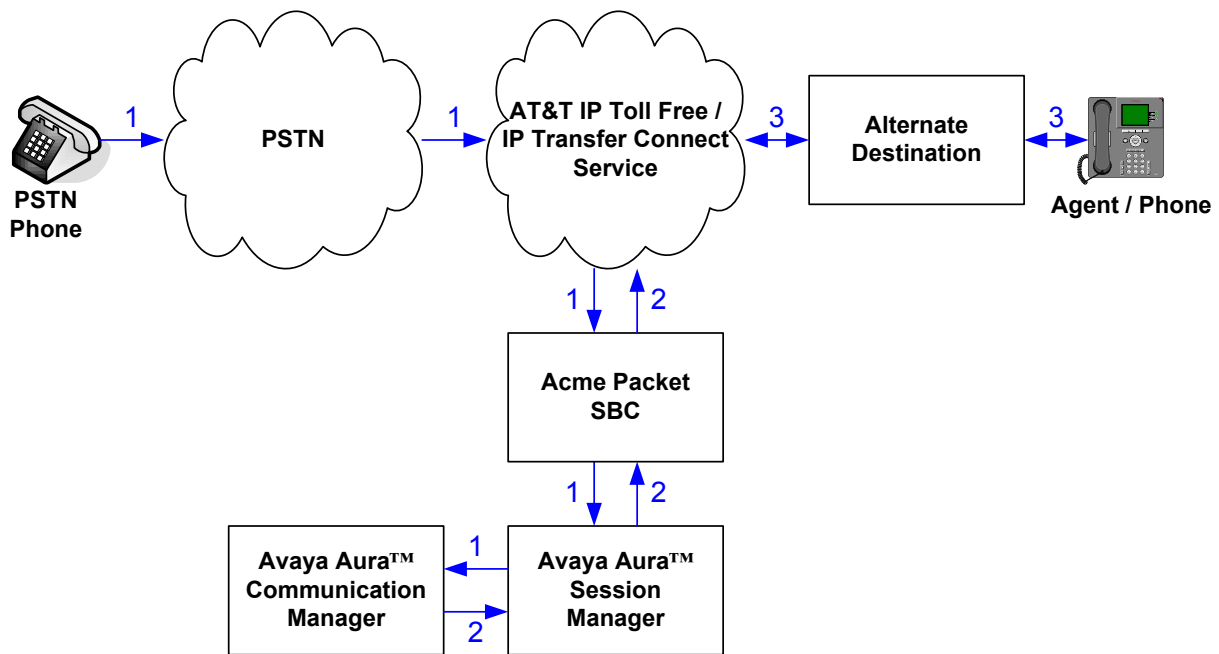

Figure 3: Inbound AT&T IP Transfer Connect Call – Pre-Answer SIP 302 Redirection

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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).

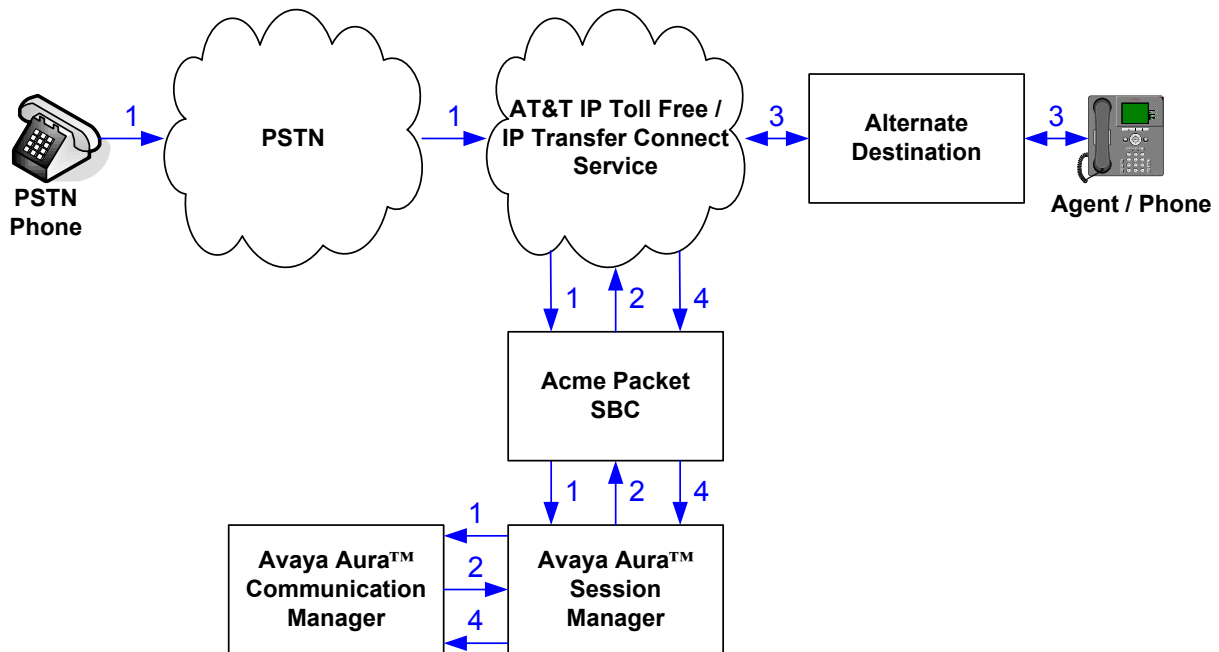
Figure 4: Inbound AT&T IP Transfer Connect Call – Post-Answer SIP REFER Redirection Successful

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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.

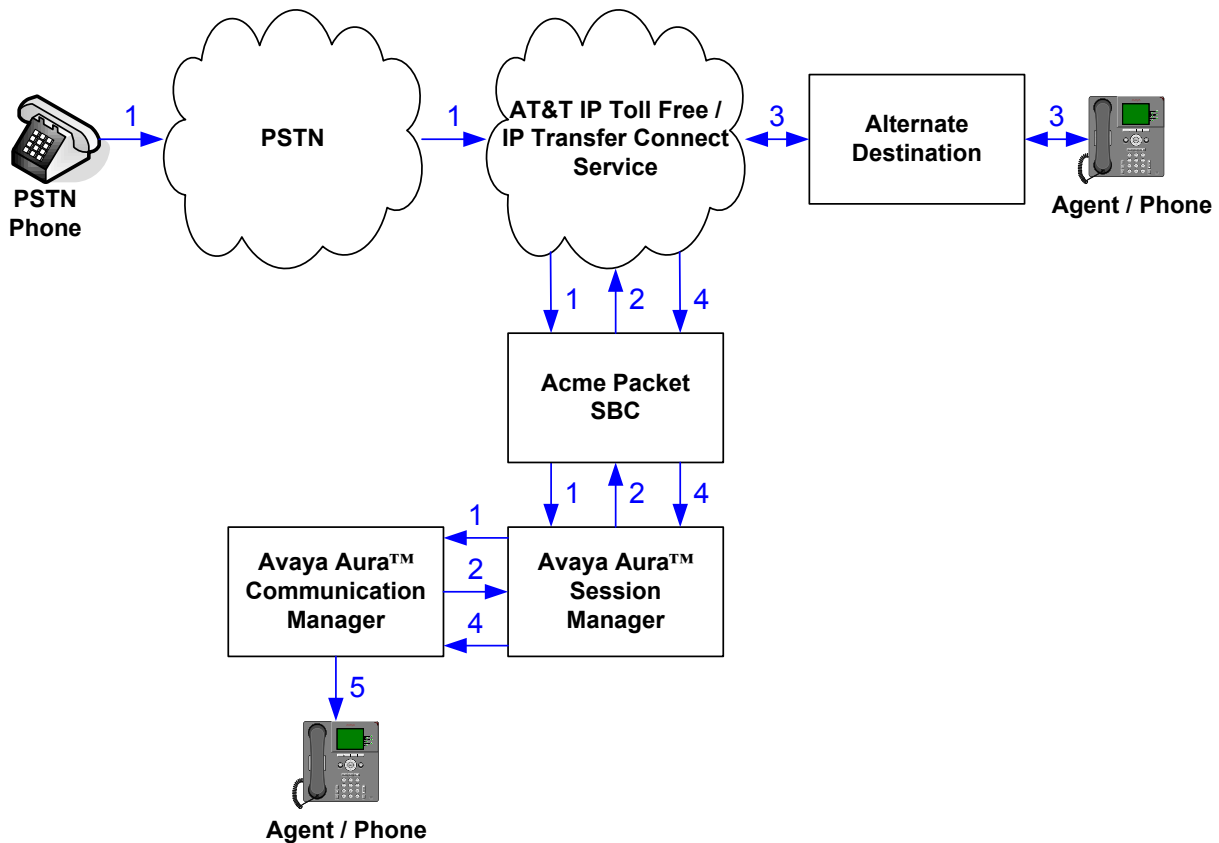**Figure 5: Inbound AT&T IP Transfer Connect Call – Post-Answer SIP REFER Redirection Unsuccessful**

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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 Session Manager 5.2 – 5.2.1.1.521012
Avaya S8510 Server	Avaya System Manager 5.2 – 5.2.1.0.521001
Avaya S8720 Server	Avaya Communication Manager 5.2.1 - R015x.02.1.016.4 with Service Pack 1 - 02.1.016.4-17959
Avaya G650 Media Gateway	-
TN2312BP IP Server Interface (IPSI)	HW15 FW050
TN799DP Control-LAN (C-LAN)	HW01 FW037
TN2602AP IP Media Resource 320 (MedPro)	HW02 FW054
TN2501AP VAL- Announcement	HW03 FW021
TN2224CP Digital Line	HW08 FW015
TN793CP Analog Line	HW05 FW010
Avaya 9630 IP Telephone	ha96xxua3_10.bin
Avaya 4610SW IP Telephone	a10d01b2_9_1.bin
Avaya 6416D+ Digital Telephone	-
Avaya S3500 Server	Avaya Modular Messaging 5.2 with Service Pack 1- 520101
Acme Packet Net-Net Session Director 3800	SCX610m3p1 (requiring sip-manipulation) or SCX 610m4p4 (see Section 6)
AT&T IP Transfer Connect Service	VNI 16

Table 2: Equipment and Software Versions

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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.

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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.

> **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 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

2. 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 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 signaling-group x** command, where **x** 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 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>, to the AT&T IP Transfer Connect service (via Avaya Aura™ Session Manager and the Acme Packet Net-Net Session Director). [2]

---

[2] 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.

```
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 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> to the AT&T IP Transfer Connect service (via Avaya Aura™ Session Manager and the Acme Packet Net-Net Session Director).[3]

---

[3] 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.

```
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.

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 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 [9] for further details of these parameters.

> **Note** – Acme later provided a new load, *SCX610m4p4* that includes a fix for this issue, making the *sip-manipulation* shown below, unnecessary.

```
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
     last-modified-by          admin@console
     last-modified-date        2010-03-30 17:09:04
```

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 with limitations as noted in **Section 1.3** were verified.

8. Verification Steps

8.1. Call Verification Tests

The following steps may be used to verify the configuration:

1. Place an inbound call, answer the call, and verify that two-way talk path exists. Verify that the call remains stable for several minutes and disconnect properly.
2. Place an inbound call to an agent or phone, but do not answer the call. Verify that the call covers to voicemail.
3. Verify the call routing administration on Session Manager. In the left pane of the 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.
   a. **Figure 15** shows a sample routing test for an inbound call with the **Called Party URI** information of the Acme Packet SBC "private" interface (192.168.67.130) with the called AT&T IP Toll Free DNIS number 000001041. The **Calling Party Address** contains the Acme Packet SBC "public" interface (192.168.64.130) with the **Calling Party URI** 7326712438. Click on "**Execute**".

Figure 15: Call Routing Test Page

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b. 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 Session Manager in **Section 4**. In the example the final call routing is displayed under the **Routing Decisions** heading, with all the routing logic steps listed below under the **Routing Decision Process** heading.

Routing Decisions

Route < sip:26000@customera.com > to SIP Entity Main_Site_Clan1 (192.168.67.13). Terminating Location is Main.

Routing Decision Process

NRP Sip Entities: Replacing Session Manager FQDN/IP address < 192.168.67.137 > with < customera.com > in request URI.

Checking NRP to determine if this is a call to an emergency number.

Originating Location is Main. Using digits < 000001041 > and host < customera.com > for routing.

NRP Dial Patterns: No matches for digits < 000001041 > and domain < customera.com >.

NRP Dial Patterns: No matches for digits < 000001041 > and domain < null >.

NRP Dial Patterns: No matches found for Main. Trying again using NRP Dial Patterns that specify -ALL- NRP Locations.

NRP Dial Patterns: No matches for digits < 000001041 > and domain < customera.com >.

NRP Dial Patterns: No matches for digits < 000001041 > and domain < null >.

NRP Dial Patterns: No matches found. Proceeding to the next phase.

NRP Regular Expressions: Full URI did not match any Regular Expression. Trying only < 000001041@customera.com >.

NRP Regular Expressions: No matches found. Proceeding to the next phase.

No Outbound Proxy found for Session Manager 1

NRP Adaptations: AT&T Adaptation applied.

NRP Adaptations: Request URI set to sip:26000@customera.com

NRP Adaptations: P-Asserted-Identity set to sip:7326712438@192.168.64.130

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Figure 16: Call Routing Test Page – Test Results – Page 1

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Routing Decision Process

NRP Sip Entities: Originating SIP Entity is Acme.

Originating Location is Main. Using digits < 26000 > and host < customera.com > for routing.

NRP Dial Patterns: No matches for digits < 26000 > and domain < customera.com >.

NRP Dial Patterns: No matches for digits < 26000 > and domain < null >.

NRP Dial Patterns: No matches found for Main. Trying again using NRP Dial Patterns that specify -ALL- NRP Locations.

NRP Dial Patterns: No matches for digits < 26000 > and domain < customera.com >.

NRP Dial Patterns: Found a Dial Pattern match for pattern < 26000 > Min/Max length 5/5 and domain < null >.

NRP Routing Policies: Ranked destination NRP Sip Entities: Main_Site_Clan1.

NRP Routing Policies: Removing disabled routes.

NRP Routing Policies: Ranked destination NRP Sip Entities: Main_Site_Clan1.

Adapting and proxying for SIP Entity Main_Site_Clan1.

NRP Entity Links: Found direct link to destination. Link uses TCP to port 5060.

NRP Adaptations: CLAN applied.

NRP Adaptations: P-Asserted-Identity set to sip:7326712438@192.168.64.130

Route < sip:26000@customera.com > to SIP Entity Main_Site_Clan1 (192.168.67.13). Terminating Location is Main.

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Figure 17: Call Routing Test Page – Test Results – Page 2

4. 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.
5. 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.
6. 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.

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.

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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.

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 5.2, Avaya Aura™ Communication Manager 5.2.1, and Acme Packet Net-Net Session Director 6.1.0 with AT&T IP Toll Free Service* – Issue 1.0
https://devconnect.avaya.com/

[2] *Avaya Aura™ Session Manager Overview*, Issue 2, Release 5.2, November 2009, Document Number 03-603323

[3] *Administering Avaya Aura™ Session Manager*, Issue 2, Release 5.2, November 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

[8] *Modular Messaging with Avaya Message Storage Server (MSS), Messaging Application Server (MAS) Administration Guide, November* 2009

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

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

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