Avaya Solution & Interoperability Test Lab

Application Notes for Avaya Aura® Communication Manager 5.2.1, and Acme Packet Net-Net 6.2.0 with AT&T IP Transfer Connect Service – Issue 1.0

Abstract

These Application Notes describe the steps for configuring Avaya Aura® Communication Manager with SIP Network Call Redirection (NCR), and the Acme Packet Net-Net (models 3800, 4250, and 4500) with the AT&T IP Transfer Connect service using **MIS/PNT** transport service connections.

Avaya Aura® Communication Manager 5.2.1 is a telephony application server and is the point of connection between the enterprise endpoints and the Acme Packet Net-Net. An Acme Packet Net-Net connects Avaya Aura® Communication Manager and the AT&T IP Transfer Connect service, and is used to not only secure the SIP trunk, but also to make adjustments to the SIP signaling for interoperability.

AT&T IP Transfer Connect is a service option available with the AT&T IP Toll Free service, and supports the rerouting of inbound Transfer Connect 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 are utilized, in conjunction with the Data Forwarding option of the AT&T IP Transfer Connect service, to transmit UUI within SIP signaling messages to the alternate destinations.

Note that these Application Notes are intended to supplement the separate document: *Applications Notes for Avaya Aura® Communication Manager 5.2.1 and Acme Packet Net-Net 6.2.0 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.

TABLE OF CONTENTS

1. Introduction

These Application Notes describe the steps for configuring Avaya Aura® Communication Manager SIP Network Call Redirection (NCR) and the Acme Packet Net-Net (models 3800, 4250, and 4500) Session Border Controller (SBC) with the AT&T IP Transfer Connect service using **MIS/PNT** transport connections. AT&T IP Transfer Connect is a service option available with the AT&T IP Toll Free service, and supports the rerouting of inbound Transfer Connect 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. 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 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 5.2.1 and Acme Packet Net-Net 6.2.0 with AT&T IP Toll Free Service – Issue 1.0*.

2. General Test Approach and Test Results

The test environment consisted of:

- A simulated enterprise with Avaya Aura® Communication Manager 5.2.1, Avaya IP and Digital phones, Acme Packet Net-Net 3800, and Avaya Modular Messaging.
- A laboratory version of the AT&T IP Toll Free service network, to which the simulated enterprise was connected via **MIS/PNT** transport.

[1] Note that this is NOT the same as the "Alternate Destination Routing (ADR)" service option available with the AT&T IP Transfer Connect service.

2.1. Interoperability Compliance Testing

The interoperability compliance testing focused on verifying inbound call flows (see **Section 3** for examples) between Communication Manager, Acme Packet Net-Net 3800, and the AT&T IP Transfer Connect service using **MIS/PNT**[2] transport.

The compliance testing was based on a test plan provided by AT&T, for the functionality required for certification as a solution supported on the AT&T network. Calls were made from the PSTN across the AT&T network. The following features were tested as part of this effort:

- SIP trunking.
- 302 based call redirection
- Refer based call redirection
- Communication Manager Network Call Redirection (NCR) functionality.
- AT&T IP Transfer Connect calls to Communication Manager stations/skills/agents, Vector Directory Numbers (VDNs), and Vectors.
- Basic telephony functions such as hold, transfer and conferencing.

2.2. Test Results

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

- Inbound AT&T IP Transfer Connect service calls to Communication Manager VDNs, vectors, agents, and phones.
- Inbound AT&T IP Transfer Connect service calls that are immediately redirected by a Communication Manager vector (pre-answer redirection) back to the AT&T IP Transfer Connect service for redirection to an alternate destination.
- Inbound AT&T IP Transfer Connect service calls that are answered by a Communication Manager vector and then redirected (post-answer redirection) back to the AT&T IP Transfer Connect service for redirection to an alternate destination.
- Redirected AT&T IP Transfer Connect service calls per above arriving on Communication Manager VDNs, vectors, agents, and phones (i.e., 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.
- Call and two-way talk path establishment between callers and Communication Manager agents/phones.

The above test objectives, with limitations as noted in **Section 2.2.1,** were verified.

2.2.1. Known Limitations

1. Communication Manager 5.2.1 issues a BYE upon receipt of a Notify/180 during Refer calls. While this did not prevent the Refers from completing successfully, this is not desired behavior (desired behavior is for Communication Manager to wait for the AT&T IP Transfer Connect service to issue a Notify/200OK before sending the BYE, or wait until the AT&T IP Transfer Connect sends the BYE). Avaya is investigating the issue.

[2] MIS/PNT does not support cRTP.

2. The 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**. With this feature enabled, Communication Manager will also use the SIP parameter *SendOnly* to signal call hold . 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 Communication Manager environment, must use the procedures described in **Addendum 1** of this document to alleviate potential issues with the AT&T IP Flexible Reach service. It describes having the Acme Packet SBC replace the *SendOnly* parameter with the *SendRecv* parameter that the AT&T Flexible Reach service does support.
3. Avaya Aura® Communication Manager vectors can specify a maximum of 96 characters for UUI.
4. Some post-answer call redirection (Refer) vectors may also include secondary routing, should the Refer-To call be denied (e.g. busy). These types of secondary routing vectors should include a wait time parameter that specifies a minimum of 1 second wait time before the secondary routing step (see **Section 5.3.3**). Adding this wait step alleviates a potential one-way audio issue.

2.3. Support

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

Avaya customers may obtain documentation and support for Avaya products by visiting http://support.avaya.com. In the United States, (866) GO-AVAYA (866-462-8292) provides access to overall sales and service support menus. Customers may also use specific numbers (provided on http://support.avaya.com) to directly access specific support and consultation services based upon their Avaya support agreements.

3. Reference Configuration

The reference configuration used in these Application Notes is shown in **Figure 1** and consists of several components. See **[1]** for more information on the reference configuration.

- Communication Manager (Access Element configuration) provides the voice communications services for a particular enterprise site, including H.323 and Digital endpoints. A Communication Manager Access Element configuration does not support SIP endpoints. In this reference configuration, 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 Communication Manager. In this reference configuration, an Avaya G650 Media Gateway is used. This solution is extensible to other Avaya Media Gateways.
- Avaya "desk" phones are represented in the reference configuration by Avaya 4610 and 9630 Series IP Telephones running H.323 software, as well as an Avaya 6400 Series Digital Telephone. An Avaya One-X® Agent, a PC based H323 softphone, was also used in the reference configuration. Note – SIP phones are not supported with the Communication Manager Access Element configuration.

- The Acme Packet Net-Net 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.
- An existing Avaya Modular Messaging system (in Multi-Site mode in this reference configuration) provides the corporate voice messaging capabilities in the reference configuration. However the provisioning of Modular Messaging is beyond the scope of this document.
- Inbound calls were sent from the AT&T IP Transfer Connect service, through the Acme Packet SBC, to Communication Manager. Communication Manager connects the call to the appropriate phone extension. The H.323 phones on the enterprise side registered directly to the Communication Manager Control LAN (C-LAN).

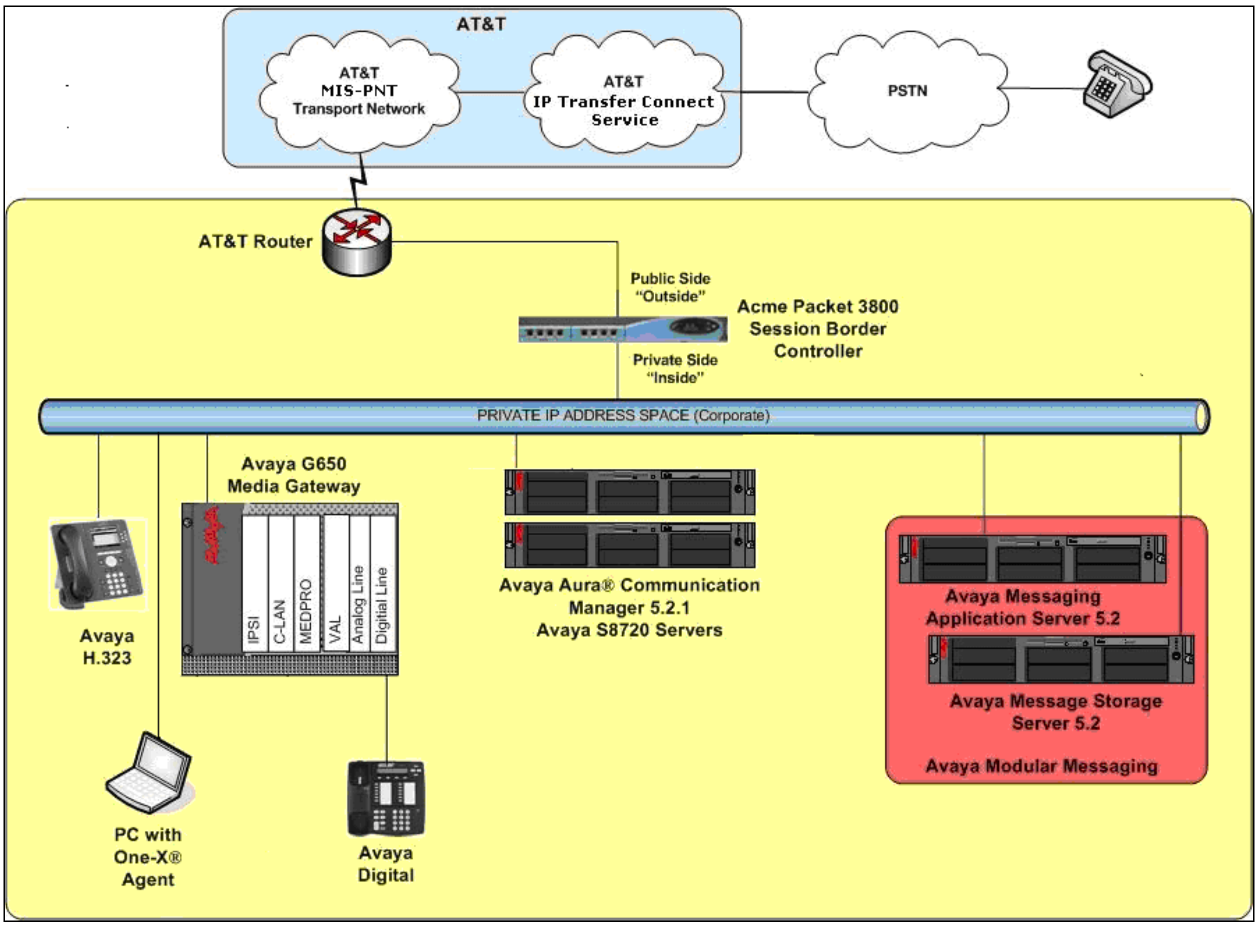

Figure 1: Reference Configuration

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

3.1. Illustrative Configuration Information

The specific values listed in **Table 1** below and in subsequent sections are used in the reference configuration described in these Application Notes, and are **for illustrative purposes only**. Customers must obtain and use the specific values for their own specific configurations.

Note - The AT&T IP Transfer Connect service Border Element IP address and DNIS digits, (destination digits specified in the SIP Request URIs sent by the AT&T Transfer Connect service), shown in this document are examples. AT&T Customer Care will provide the actual network IP addresses and DNIS digits as part of the IP Transfer Connect provisioning process.

Component	Illustrative Value in these Application Notes
Avaya Aura® Communication Manager	
C-LAN IP Address	192.168.67.14
Avaya Aura® Communication Manager extensions	26xxx
Voice Messaging Pilot Extension	26000
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	17231126xxx
Pilot Number	17231126000
Acme Packet SBC	
IP Address of "Outside" (Public) Interface (connected to AT&T Access Router/IP Toll Free Service)	192.168.64.130
IP Address of "Inside" (Private) Interface (connected to Avaya Aura® Communication Manager)	192.168.67.130
AT&T IP Toll Free Service	
Border Element IP Address	135.25.29.74
AT&T Access router interface (to Acme outside)	192.168.64.254
AT&T Access Router NAT address (Acme outside address)	135.16.170.55

Table 1: Illustrative Values Used in the Reference Configuration

3.2. Call Flows

To understand how inbound AT&T IP Transfer Connect service calls are handled by 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® 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® Communication Manager.
5. 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.

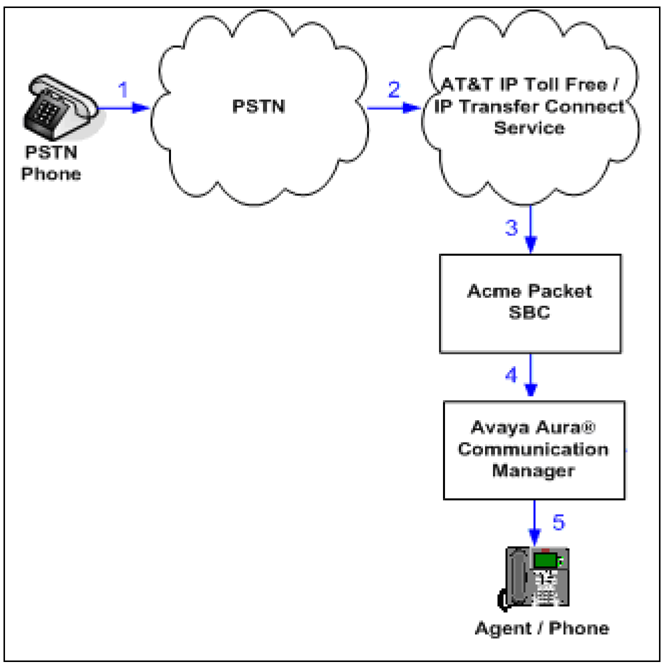

Figure 2: Inbound AT&T IP Transfer Connect Call – No Redirection

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® 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 (302 Redirection) 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 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).

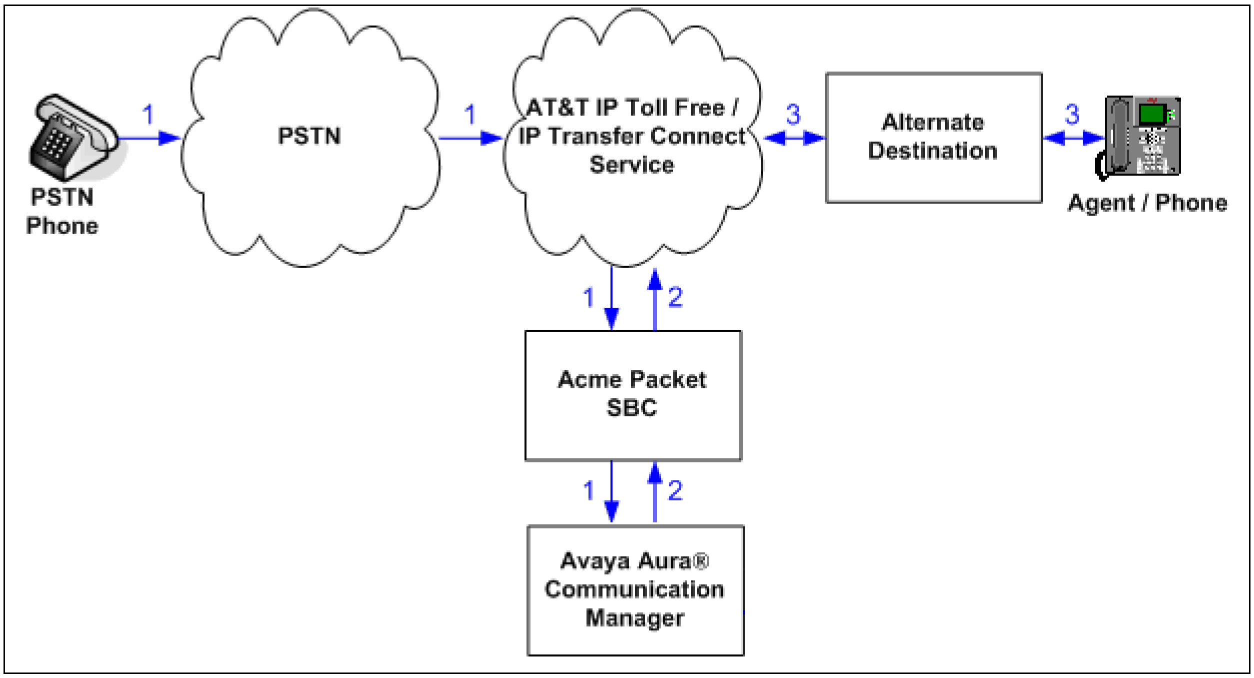

Figure 3: Inbound AT&T IP Transfer Connect Call – Pre-Answer SIP 302 Redirection

The third call scenario illustrated in **Figure 4** is an inbound AT&T IP Transfer Connect service call that arrives on Avaya Aura® 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 (Refer Redirection) 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 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).

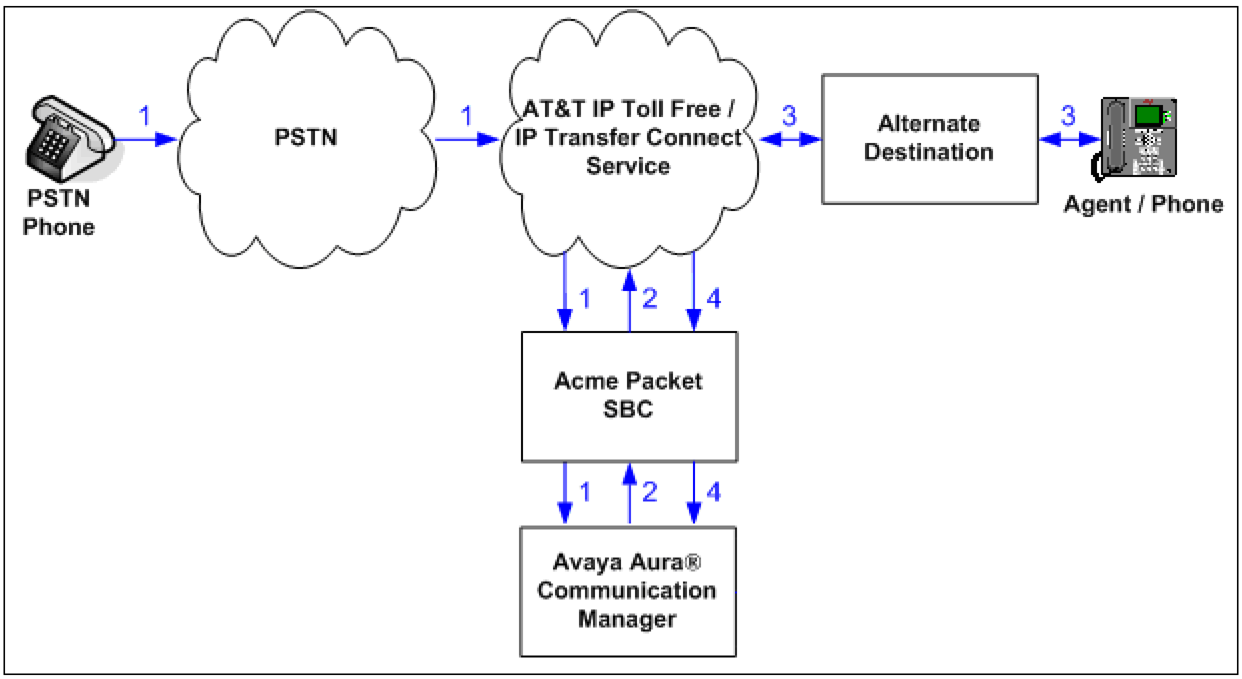

Figure 4: Inbound AT&T IP Transfer Connect Call – Post-Answer SIP REFER Redirection Successful

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.

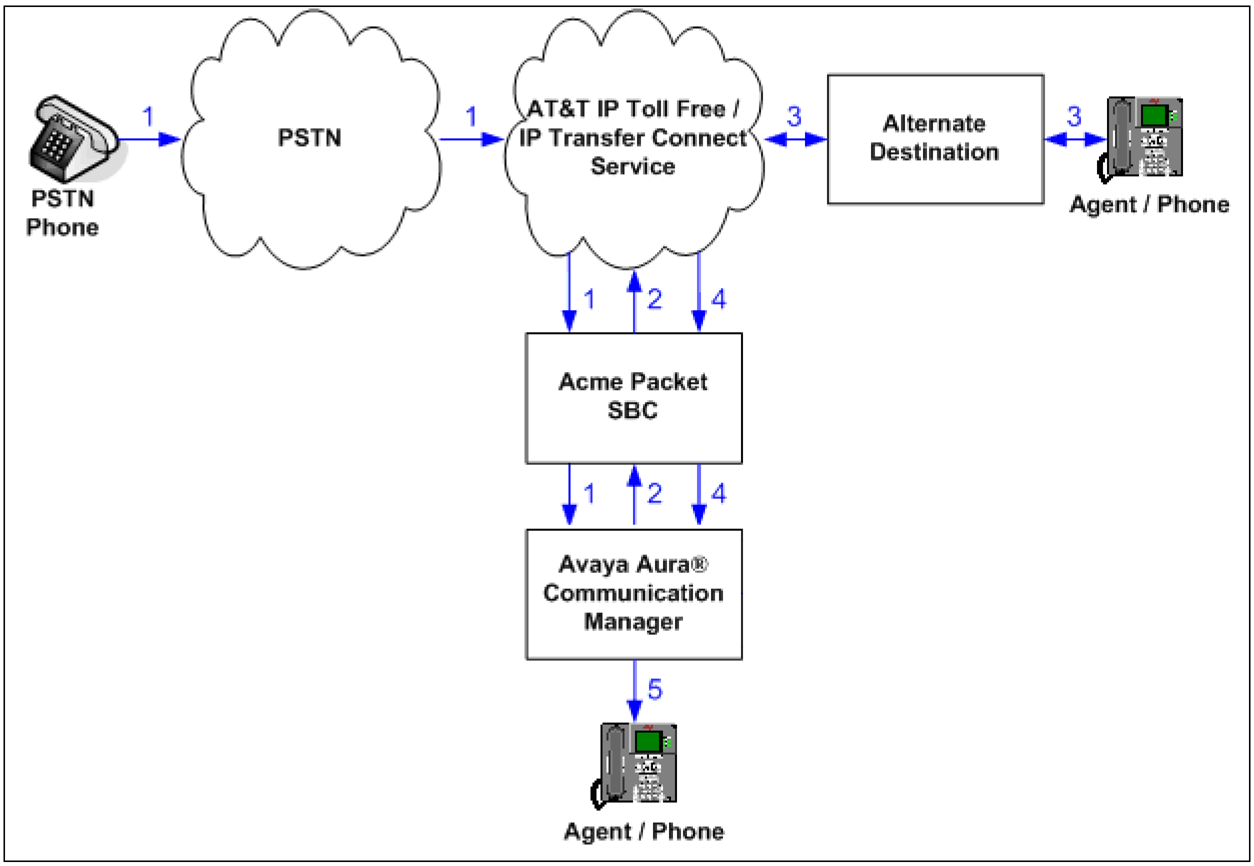

Figure 5: Inbound AT&T IP Transfer Connect Call – Post-Answer SIP REFER Redirection Unsuccessful

4. Equipment and Software Validated

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

Component	Version
Avaya S8720 Server	Avaya Aura® Communication Manager 5.2.1 (R015x.02.1.016.4) with SP6 18576
Avaya G650 Media Gateway	
TN2312BP IP Server Interface (IPSI)	HW15 FW053
TN799DP Control-LAN (C-LAN)	HW01 FW039
TN2602AP IP Media Resource 320 (MedPro)	HW02 FW058
TN2501AP VAL-ANNOUNCEMENT	HW03 FW021
TN2224CP Digital Line	HW08 FW015
Avaya 9630 IP Telephone	Avaya one-X® Deskphone Edition H.323 Release 3.110b
Avaya one-X® Agent	2.0.018.8
Avaya 6416D+ Digital Telephone	-
Avaya S3500 Servers for Avaya Modular Messaging (MAS and MSS)	Release 5.2 – SP5 with Patch 1 (9.0.350.5019)
Acme Packet Net-Net 3800	SCX6.2.0 MR5 Patch 1 (Build 783)
AT&T IP Transfer Connect Service using MIS-PNT transport service connections.	VNI 18

Table 2: Equipment and Software Versions

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 is applicable for the AT&T IP Transfer Connect service as well. This section describes the additional administration steps on Communication Manager necessary for supporting interaction with the AT&T IP Transfer Connect service. The steps are performed from the Communication Manager System Access Terminal (SAT) interface.

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

5.1. System Parameters

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

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

```
display system-parameters customer-options                    Page   4 of  11
                                OPTIONAL FEATURES
   Emergency Access to Attendant? y                           IP Stations? y
           Enable 'dadmin' Login? y
           Enhanced Conferencing? y                     ISDN Feature Plus? 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
```

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 the Acme Packet SBC 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 2** in **Section 2.2.1**).
2. Set **Support Request History?** is set to "**n**".
3. Set **Telephone Event Payload Type:** to **100**.

```
change trunk-group 2                                          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? n
                  Telephone Event Payload Type: 100
```

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

Note – See **Addendum 1** regarding the use of the **Network Call Redirection** parameter in customer environments with both AT&T IP Transfer Connect and AT&T IP Flexible Reach services.

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 Communication Manager SIP Network Call Redirection (NCR) functionality (see **Section 5.2** above). 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.

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 **[4]** and **[5]** 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 "**~r**" in the "**route-to number**" instructs Communication Manager to send a SIP 302 message with the number "**1012**" in the user part of the Contact header URI, e.g., 1012@<host/domain>, to the AT&T IP Transfer Connect service (via the Acme Packet SBC).

```
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
```

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: 302Redir_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**" (vector step **05**).

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 "**~r**" in the "**route-to number**" instructs Communication Manager to send a SIP REFER message with the number "**1012**" in the user part of the Refer-To header URI, e.g., 1012@<host/domain> to the AT&T IP Transfer Connect service (via the Acme Packet SBC).
- If the redirection fails (e.g. network denies the call), then announcement **26021** is played to the caller.

```
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
07 #    Refer to AT&T DID 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 26021
12
```

Figure 13: Sample Vector for Post-Answer Redirection

5.3.3. Post-Answer Redirection with Secondary Destination

This section provides an example of Post-Answer Redirection with a secondary destination. In this example should the Refer-To call does not complete (e.g. 486 busy), the vector can route the call to a secondary destination (e.g. an Agent station). The inbound call is routed to the vector shown in **Figure 14**. The vector does the following:

- Plays ringback for 2 seconds (vector step **02**).
- Answers the call and plays the greeting announcement 26020 (vector step **05**).
- Attempts to redirect the call to the number "**1012**" (vector step **08**).
- Vector step **09** waits 2 seconds before proceeding. **Note** – this wait step is critical in alleviating a potential one-way audio issue. See item 3 in **Section 2.2.1**.
- If the redirection fails (e.g. network denies the call), then announcement **26021** is played to the caller (vector step **11**).
- Vector step **12** waits 2 seconds, plays the Skill 4 announcement in step **13**, then the call is sent to the Skill 4 queue in step **14**.
- Vector step **15** plays music-on-hold to the caller and every 10 seconds step **16** will play a queue announcement, until a Skill 4 Agent is available.

```
change vector 1022                                              Page   1 of   6
                                  CALL VECTOR

    Number: 1022             Name: REFER486toAgent
                                             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? y   BSR? y   Holidays? n
 Variables? y   3.0 Enhanced? y
01 #    Send Ringing
02 wait-time    3   secs hearing ringback
03
04 #    Answer call immediately with announcement then NCR REFER
05 announcement 26020
06
07 #    Refer occurs since this is post answer
08 route-to     number ~r1012           with cov n if unconditionally
09 wait-time    2   secs hearing silence
10 #    If Refer fails play announcement and go to skill 4
11 announcement 26021
12 wait-time    2   secs hearing ringback
13 announcement 26014
14 queue-to     skill 4    pri h
15 wait-time    10  secs hearing music
16 announcement 26015
17
```

Figure 14: Sample Vector for Post-Answer Redirection with Secondary Destination

5.3.4. Station Provisioning to Display UUI

In order to display the UUI information defined in the **Sections 5.3.1** and **5.3.2** above, the Agent's station must have a UUI display button defined via the Communication Mange ***change station x*** form, where ***x*** is a station extension associated with the Agent.

- On page 4 of the station form add the **uui-info** feature to any available button appearance (e.g. button appearance **8**).

```
change station 26102                                          Page   4 of   5
                                     STATION
 SITE DATA
       Room:                                               Headset? n
       Jack:                                               Speaker? n
      Cable:                                              Mounting: d
      Floor:                                           Cord Length: 0
   Building:                                             Set Color:
ABBREVIATED DIALING
     List1:                     List2:                    List3:
BUTTON ASSIGNMENTS
 1: call-appr                             5: release
 2: call-appr                             6: aux-work    RC:    Grp:
 3:                                       7: auto-in            Grp:
 4: send-calls Ext:                       8: uui-info
    voice-mail Number: 26000
```

Figure 15: UUI display button

5.4. Acme Packet NET-NET 3800 Configuration

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. 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 section describes the Acme Packet header-rule provisioning. The header-rules described below were added to the existing sip-manipulation ***NAT_IP*** described in **[1]**.

- **Modify 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
```

- **Modify 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 1 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                          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
```

6. Verification Steps

The following steps may be used to verify the reference configuration in addition to the verification procedures described in **[1]**:

6.1. Redirection Verification Tests

1. Place an inbound call to an AT&T IP Transfer Connect service number enabled with Redirect features (302 redirection).

    a. Verify that an appropriate Communication Manager vector immediately redirects the call back to the AT&T IP Transfer Connect service for redirection to the alternate destination.

    i. On Communication Manager enter the command ***list trace vector x***, where *x* is an extension assigned to the associated vector. This will display the vector as it executes.

    ii. Using a SIP protocol analyzer (e.g. Wireshark), monitor the SIP traffic at the Acme Packet SBC public "outside" interface connection to the AT&T IP Toll Free service. Verify that a 302 Moved Temporarily message is sent and that it contains the alternate destination AT&T IP Transfer Connect service access number programmed in the vector, e.g. **1012** (see **Figure 16**).

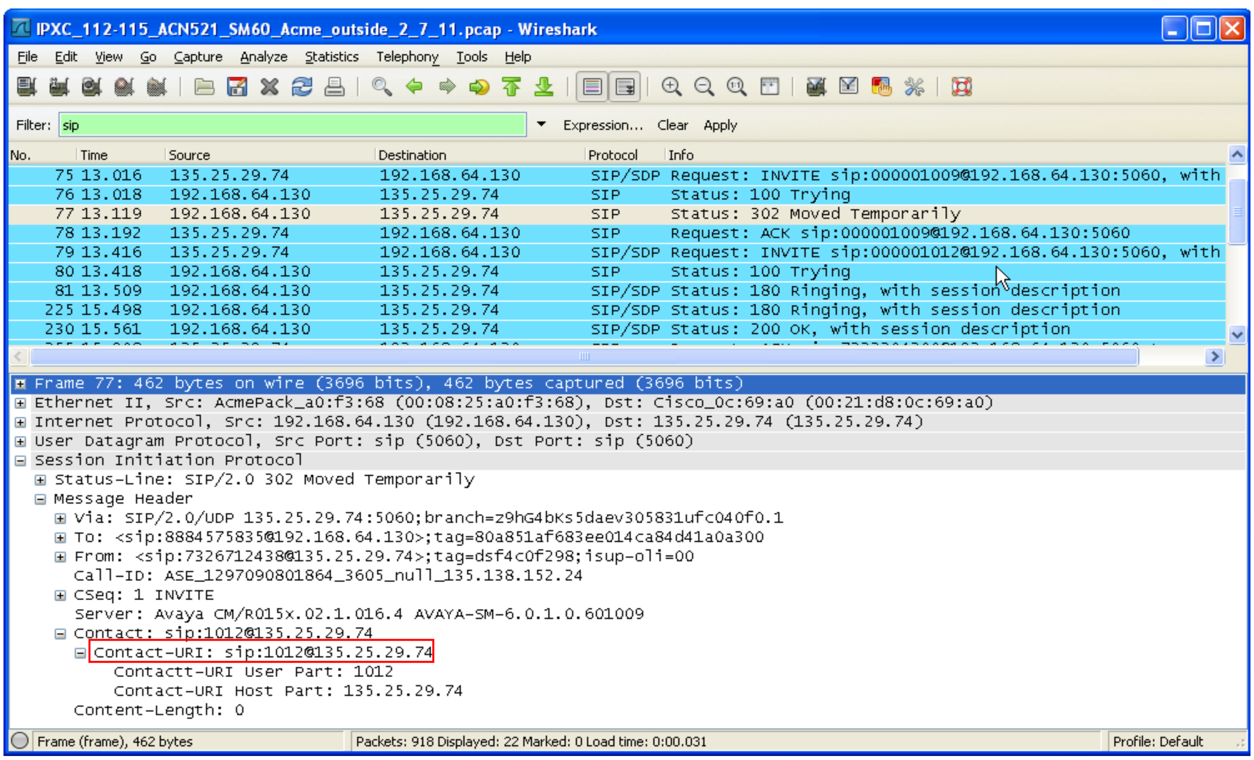

Figure 16: 302 Protocol Trace

    iii. When the redirection is complete, verify two way talk path.

2. Place an inbound call to an AT&T IP Transfer Connect service number enabled with IP Courtesy Transfer features (Refer redirection).

    a. Verify that an appropriate Communication Manager vector immediately redirects the call back to the AT&T IP Transfer Connect service for redirection to the alternate destination.

        i. On Communication Manager enter the command ***list trace vector x***, where *x* is an extension assigned to the associated vector. This will display the vector as it executes

        ii. Using a SIP protocol analyzer (e.g. Wireshark), monitor the SIP traffic at the Acme Packet SBC public "outside" interface connection to the AT&T IP Toll Free service. Verify that a Refer packet is sent and that it contains the alternate destination AT&T IP Transfer Connect service access number programmed in the vector, e.g. **1012** (see **Figure 17**).

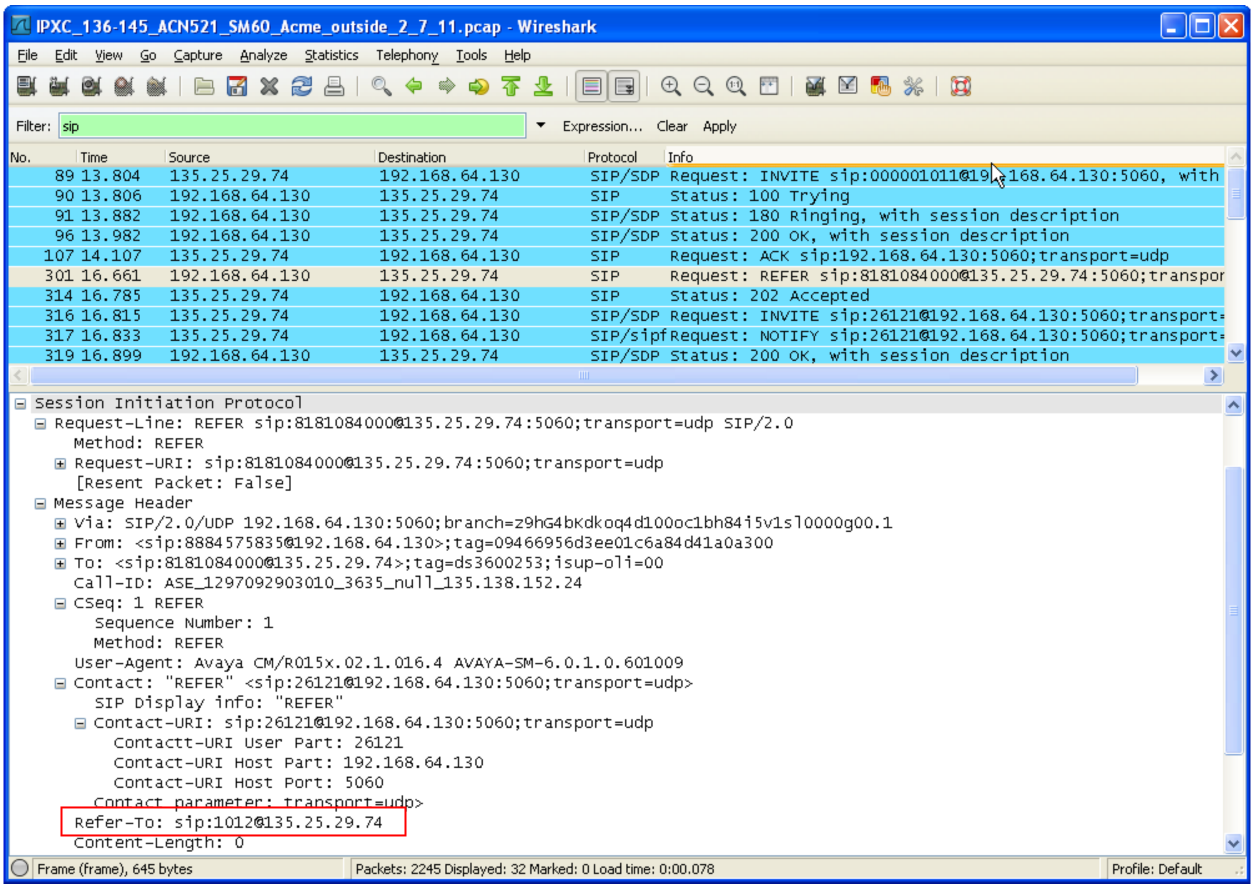

Figure 17: Refer Protocol Trace

        iii. When the redirection is complete, verify two way talk path.

3. Verify that when Communication Manager is the transfer target of redirected calls, the calls are answered with two-way talk path, and that any defined user-to-user information (UUI) is displayed on the answering station (see **Section 5.3**).

7. Conclusion

As illustrated in these Application Notes, (and building upon the configurations described in **[1]**), Avaya Aura® Communication Manager Network Call Redirection, and the Acme Packet Net-Net can be configured to interoperate successfully with the AT&T IP Transfer Connect service connection via **MIS/PNT** transport. 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 complement the AT&T IP Transfer Connect service 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.

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.

8. References

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

[1] *Applications Notes for Avaya Aura® Communication Manager 5.2.1 and Acme Packet Net-Net 6.2.0 with AT&T IP Toll Free Service – Issue 1.0*
[2] *Administering Avaya Aura® Communication Manager*, Issue 5.0, Release 5.2, May 2009, Document Number 03-300509
[3] *Avaya Aura® Communication Manager Feature Description and Implementation*, Issue 7, Release 5.2, May 2009, Document Number 555-245-205
[4] *Administering Avaya Aura® Call Center Features*, Release 6.0, June 2010
[5] *Programming Call Vectors in Avaya Aura® Call Center*, 6.0, June 2010

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

9. 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**). 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.

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.