



Avaya Solution & Interoperability Test Lab

Application Notes for Avaya Aura® Communication Manager 6.0, 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 6.0 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 6.0 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.

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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 alternate¹ destinations based upon SIP redirection messages from Avaya Aura® Communication Manager. The AT&T IP Transfer Connect service is typically used by enterprises that have multiple call centers that are separated geographically or otherwise not interconnected. Using SIP NCR, trunk-to-trunk routing of certain inbound calls at Avaya Aura® Communication Manager can be avoided by requesting that the AT&T network transfer the inbound caller to an alternate destination. 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 6.0 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 6.0, Avaya IP and analog 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.

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.

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 MIS/PNT does not support cRTP.

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. By default Communication Manager 6.0 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). This issue can be alleviated by enabling the ***Convert 180 to 183 for Early Media?*** parameter on the SIP trunk form (see **Section 5.2**). With this parameter enabled, Communication Manager will send a BYE upon receipt of NOTIFY/200OK.
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 call. 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.

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 provides the voice communications services for a particular enterprise site, including H.323 and Digital stations (SIP stations are not supported in the reference configuration). In this reference configuration, Communication Manager runs on an Avaya S8800 Server. This solution is extensible to other Avaya S8xxx Servers. The Avaya Media Gateway provides the resources for Communication Manager. In this reference configuration, an Avaya G450 Media Gateway is used. This solution is extensible to other Avaya Media Gateways.
- Avaya “desk” stations are represented in the reference configuration by Avaya 4610 and 9630 Series IP Telephones running H.323 software, as well as an Avaya 6211 Series Analog Telephone. An Avaya One-X® Agent, a PC based H323 softphone, was also used in the reference configuration.
- The Acme Packet Net-Net 3800³ provides SIP Session Border Controller (SBC) functionality, including address translation and SIP header manipulation between the AT&T IP Toll Free service and the enterprise internal network.
- An existing Avaya Modular Messaging system (in Multi-Site mode in the 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.

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

- Inbound calls were sent from the AT&T IP Toll Free service, through the Acme Packet SBC, to Communication Manager. Communication Manager connects the call to the appropriate phone or fax extension. The H.323 stations on the enterprise side registered directly to Communication Manager
- A PC (via analog modem) running the Ventafax application, was used to test T.38 fax.

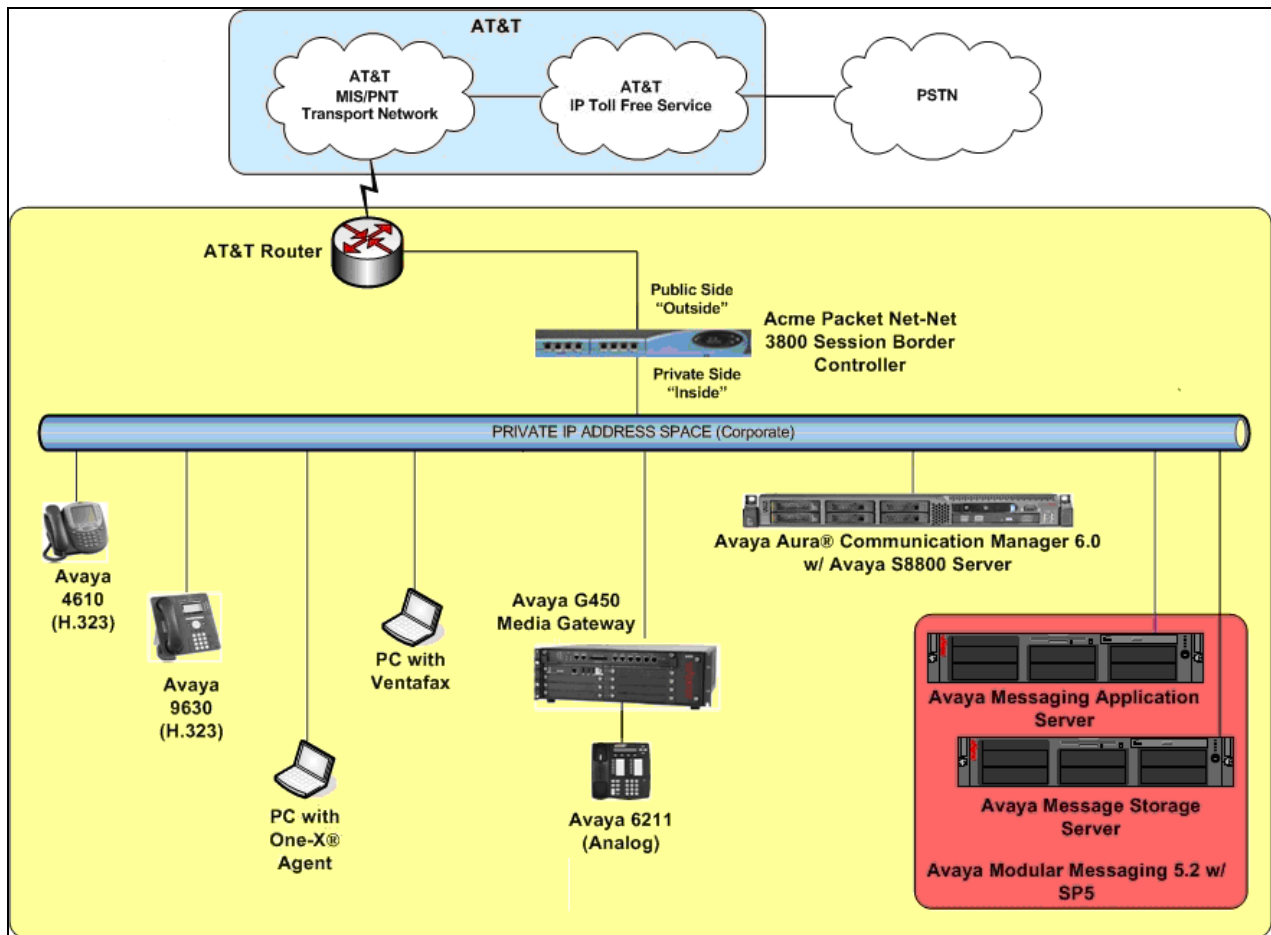


Figure 1: Reference Configuration

3.1. Illustrative Configuration Information

The specific values listed in **Table 1** below and in subsequent sections are used in the reference configuration described in these Application Notes, and are **for illustrative purposes only**. Customers must obtain and use the specific values for their own 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	
Processor Ethernet IP Address	192.168.67.202
Avaya Aura® Communication Manager extensions	40xxx = stations 44xxx = VDNs 47xxx = Agents
Avaya CPE local dial plan	4xxxx
Voice Messaging Pilot Extension	46000
Avaya Modular Messaging	
Messaging Application Server (MAS) IP Address	192.168.67.141
Messaging Server (MSS) IP Address	192.168.67.140
Modular Messaging dial plan	4xxxx
Acme Packet Net-Net SBC	
IP Address of “Outside” (Public) Interface (connected to AT&T Access Router/IP Transfer Connect 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.

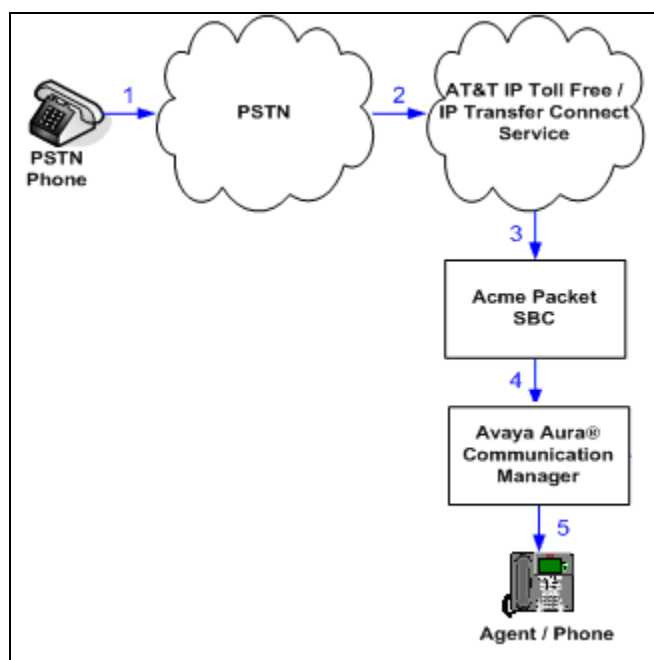


Figure 2: Inbound AT&T IP Transfer Connect Call – No Redirection

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).

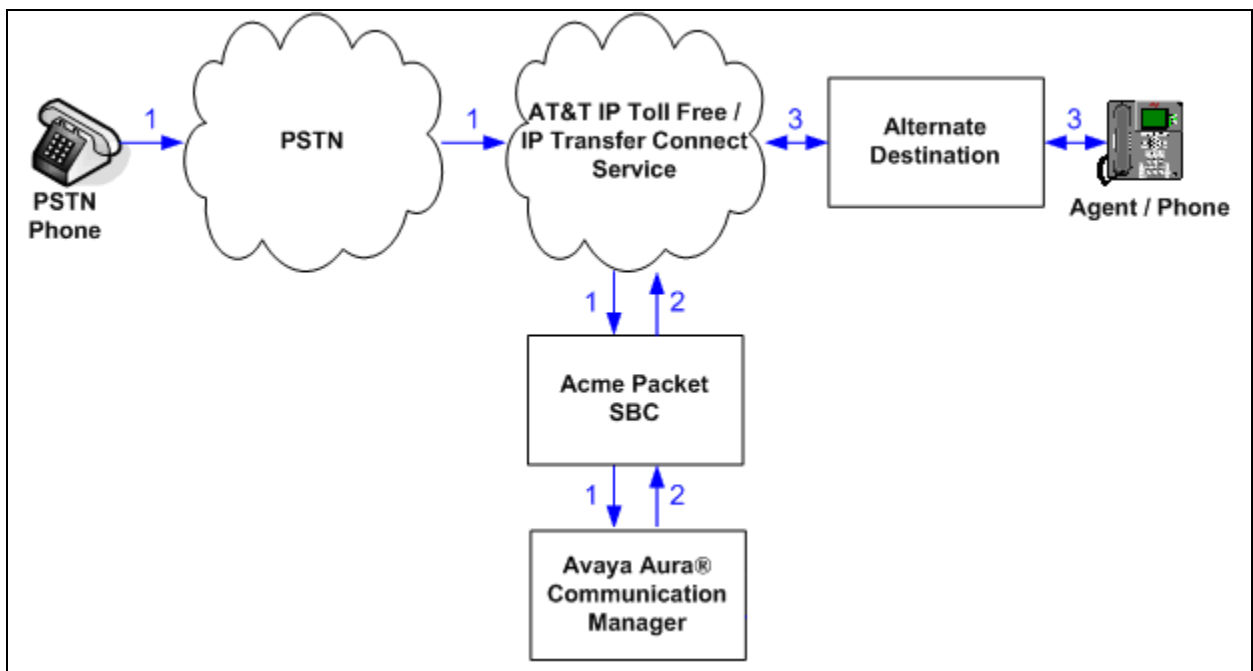


Figure 3: Inbound AT&T IP Transfer Connect Call – Pre-Answer SIP 302 Redirection

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).

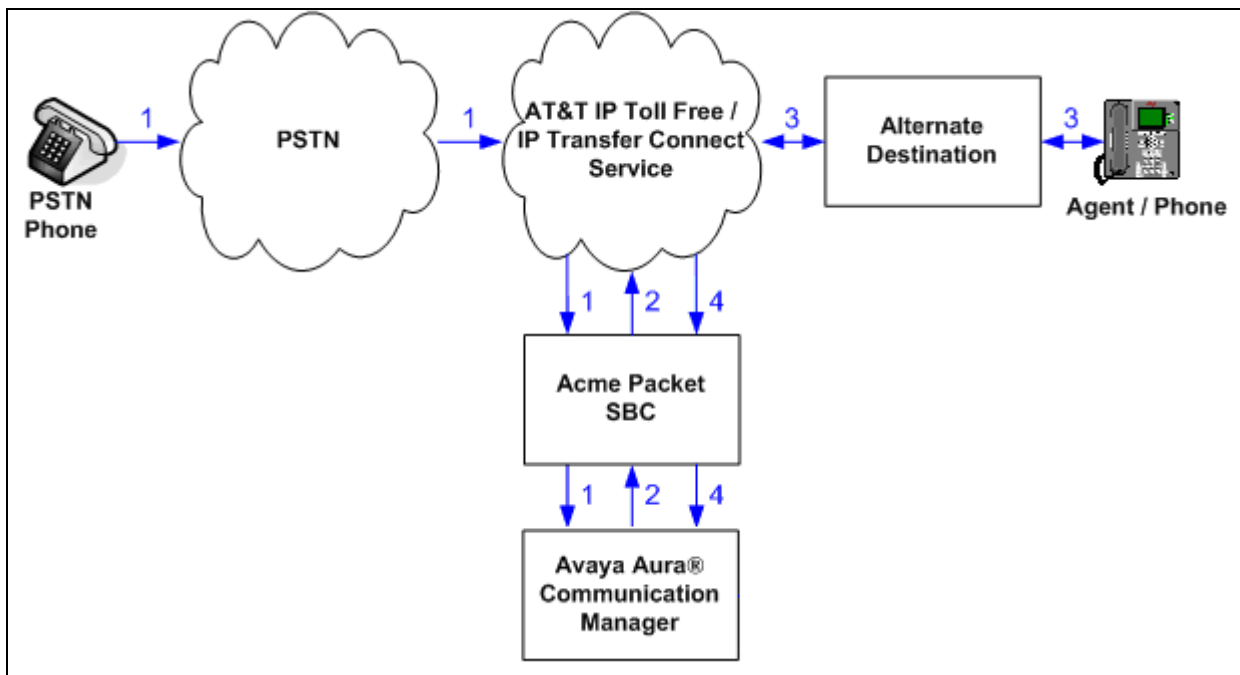


Figure 4: Inbound AT&T IP Transfer Connect Call – Post-Answer SIP REFER Redirection Successful

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.

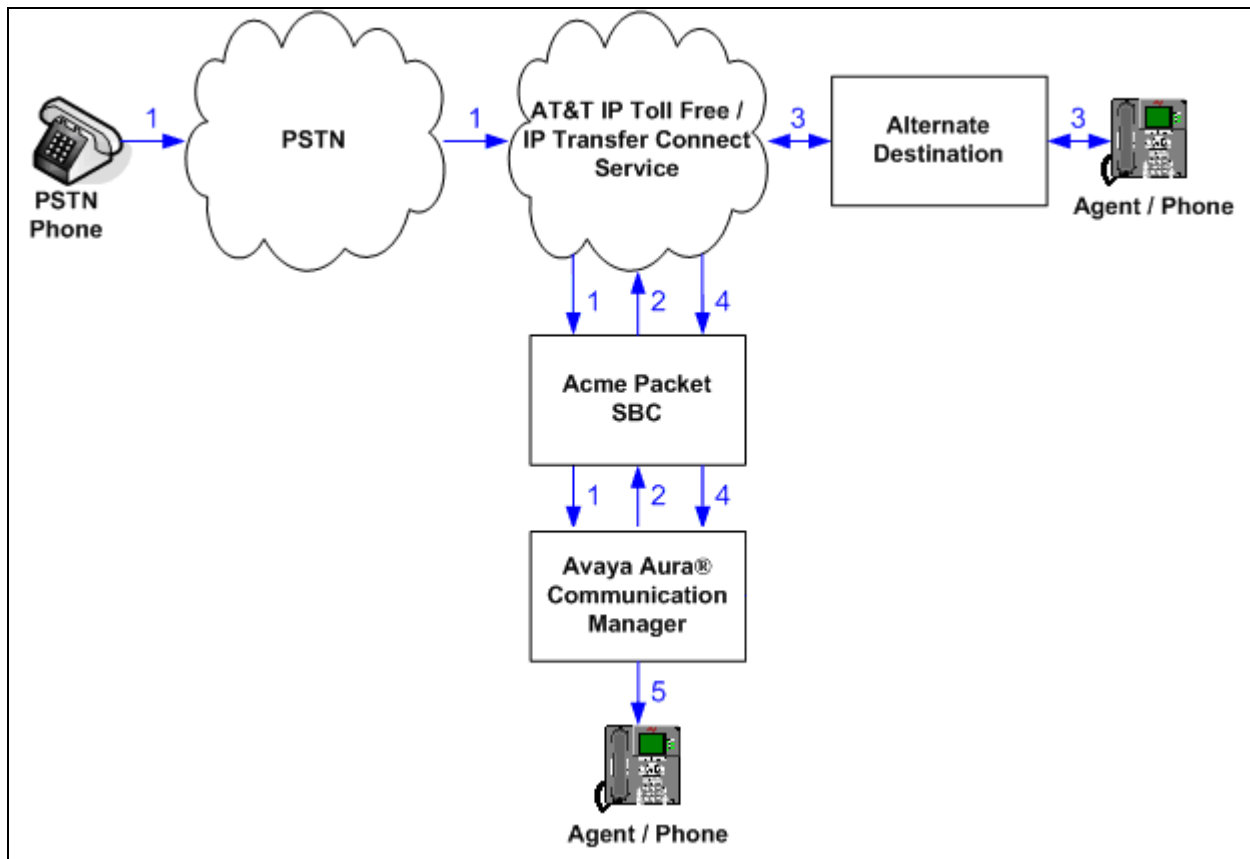


Figure 5: Inbound AT&T IP Transfer Connect Call – Post-Answer SIP REFER Redirection Unsuccessful

4. Equipment and Software Validated

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

Component	Version
Avaya S8800 Server	Avaya Aura® Communication Manager 6.0 (R016x.00.0.345.0) with SP2 (18567)
Avaya G450 Media Gateway	31.18.1
MM711 Analog	HW31 FW95
Avaya 9630 IP Telephone – H323	Avaya one-X® Deskphone Edition H.323 Version S3.110b (ha96xxua3_11.bin)
Avaya one-X® Communicator	6.0.0.26
Avaya 4610SW IP Telephone – H323	a10d01b2_9_1.bin
Avaya one-X® Agent	2.0.09184.0
Avaya Modular Messaging (MAS and MSS) on Avaya S3500 Servers	Release 5.2 – SP5 with Patch 1 (9.0.350.5019)
Fax device	Ventafax Home Version 6.3.102.288
Acme Packet Net-Net 3800	SCX6.2.0 m5p1
AT&T IP Transfer Connect Service via MIS/PNT transport service connections.	VNI 18

Table 2: Equipment and Software Versions

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

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? n
  Enhanced EC500? y           ISDN/SIP Network Call Redirection? y
Enterprise Survivable Server? n                                     ISDN-BRI Trunks? y
  Enterprise Wide Licensing? n                                       ISDN-PRI? y
  ESS Administration? y       Local Survivable Processor? n
  Extended Cvg/Fwd Admin? y   Malicious Call Trace? y
  External Device Alarm Admin? y   Media Encryption Over IP? n
Five Port Networks Max Per MCC? n   Mode Code for Centralized Voice Mail? n
  Flexible Billing? n
Forced Entry of Account Codes? y   Multifrequency Signaling? y
  Global Call Classification? y   Multimedia Call Handling (Basic)? y
  Hospitality (Basic)? y         Multimedia Call Handling (Enhanced)? y
Hospitality (G3V3 Enhancements)? y   Multimedia IP SIP Trunking? y
  IP Trunks? y

IP Attendant Consoles? y
(NOTE: You must logoff & login to effect the permission changes.)
```

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: 6.0
                                ACD? y                                Reason Codes? y
                                BCMS (Basic)? y                        Service Level Maximizer? n
                                BCMS/VuStats Service Level? y        Service Observing (Basic)? y
BSR Local Treatment for IP & ISDN? y    Service Observing (Remote/By FAC)? y
                                Business Advocate? n                Service Observing (VDNs)? y
                                Call Work Codes? y                    Timed ACW? y
                                DTMF Feedback Signals For VRU? y      Vectoring (Basic)? y
                                Dynamic Advocate? n                  Vectoring (Prompting)? y
                                Expert Agent Selection (EAS)? y       Vectoring (G3V4 Enhanced)? y
                                EAS-PHD? y                            Vectoring (3.0 Enhanced)? y
                                Forced ACD Calls? n                  Vectoring (ANI/II-Digits Routing)? y
                                Least Occupied Agent? y              Vectoring (G3V4 Advanced Routing)? y
                                Lookahead Interflow (LAI)? y         Vectoring (CINFO)? y
Multiple Call Handling (On Request)? y    Vectoring (Best Service Routing)? y
                                Multiple Call Handling (Forced)? y    Vectoring (Holidays)? y
                                PASTE (Display PBX Data on Phone)? y  Vectoring (Variables)? Y

```

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 (e.g. 2). On Page 4 of the **trunk-group** form, set **Network Call Redirection** to “y” (see item 2 in Section 2.2.1).
2. Verify **Support Request History?** is set to “n”.
3. Verify **Telephone Event Payload Type:** to 100.
4. Set **Convert 180 to 183 for Early Media?** to “y” (see item 1 in Section 2.2.1).

```

display 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
                                Convert 180 to 183 for Early Media? y
                                Always Use re-INVITE for Display Updates? n
                                Enable Q-SIP? n

```

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 44020 shown in **Figure 10**, which invokes the vector 22 shown in **Figure 11**. The vector does the following:

- Plays ringback for 3 seconds (vector step **02**).
- Assigns the data “**1234567890123456**” to ASAI UUI variable “**B**” (vector step **05**).
Note: The parameters for ASAI UUI variables “**A**” and “**B**”, and other vector variables are defined using the **change variables** command (see **Figure 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		VARIABLES FOR VECTORS				Page 1 of 39
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 44020                                     Page 1 of 3
              VECTOR DIRECTORY NUMBER
              Extension: 44020
              Name*: 302
              Destination: Vector Number           22
              Attendant Vectoring? n
              Meet-me Conferencing? n
              Allow VDN Override? n
              COR: 1
              TN*: 1
              Measured: none
              VDN of Origin Annc. Extension*:
              1st Skill*:
              2nd Skill*:
              3rd Skill*:
* Follows VDN Override Rules

```

Figure 10: Sample VDN for Pre-Answer Redirection

```

display vector 22                                     Page 1 of 6
              CALL VECTOR
              Name: 302RingUII
              Number: 22
Multimedia? n      Attendant Vectoring? n      Meet-me Conf? n      Lock? n
  Basic? y      EAS? y      G3V4 Enhanced? y      ANI/II-Digits? y      ASAI Routing? y
  Prompting? y      LAI? y      G3V4 Adv Route? y      CINFO? y      BSR? y      Holidays? y
  Variables? y      3.0 Enhanced? y
01 #      Ringing
02 wait-time      3      secs hearing ringback
03
04 #      Define UUI variable
05 set      B      = none      CATR      1234567890123456
06
07 #      Redirect
08 route-to      number ~r1012      with cov n if unconditionally
09 stop
10

```

Figure 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 44010 shown in **Figure 12**, which invokes the vector 15 shown in **Figure 13**. The vector does the following:

- Assigns the data “**1234567890123456**” to ASAI UUI variable “**A**” (vector step **02**).
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 **05**).
- Attempts to redirect the call to the number “**1012**” (vector step **08**). 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 **42009** is played to the caller.

```
display vdn 44010                                     Page 1 of 3
              VECTOR DIRECTORY NUMBER
              Extension: 44010
              Name*: REFER
              Destination: Vector Number           15
Attendant Vectoring? n
Meet-me Conferencing? n
Allow VDN Override? n
              COR: 1
              TN*: 1
              Measured: none
VDN of Origin Annc. Extension*:
              1st Skill*:
              2nd Skill*:
              3rd Skill*:
* Follows VDN Override Rules
```

Figure 12: Sample VDN for Post-Answer Redirection

```
display vector 15                                     Page 1 of 6
              CALL VECTOR
Number: 15              Name: Refer_UUI
Multimedia? n          Attendant Vectoring? n          Meet-me Conf? n          Lock? n
Basic? y              EAS? y          G3V4 Enhanced? y          ANI/II-Digits? y          ASAI Routing? y
Prompting? y          LAI? y          G3V4 Adv Route? y          CINFO? y          BSR? y          Holidays? y
Variables? y          3.0 Enhanced? y
01 #          Generate UUI
02 set          A          = none          CATR 1234567890123456
03
04 #          Play Refer announcement
05 announcement 42008
06
07 #          Refer occurs since this is post answer
08 route-to          number ~r1012          with cov n if unconditionally
09 #          If Refer fails play announcement and disconnect
10 disconnect          after announcement 42009
11
```

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 3 seconds (vector step **02**).
- Answers the call and plays the greeting announcement 42008 (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. 486 Busy), then announcement **42009** 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.

```
display vector 18                                     Page 1 of 6
                                                    CALL VECTOR
Number: 18                                           Name: Refer486toAgent
Multimedia? n      Attendant Vectoring? n      Meet-me Conf? n      Lock? n
  Basic? y      EAS? y      G3V4 Enhanced? y      ANI/II-Digits? y      ASAI Routing? y
  Prompting? y      LAI? y      G3V4 Adv Route? y      CINFO? y      BSR? y      Holidays? y
  Variables? y      3.0 Enhanced? y
01 #      Send ringing
02 wait-time      3      secs hearing ringback
03
04 #      Answer call with announcement
05 announcement 42008
06
07 #      Refer occurs since this is post answer
08 route-to      number ~r1012      with cov n if unconditionally
09
10 #      If REFER fails play announcement and go to skill 4
11 announcement 42009
12 wait-time      2      secs hearing ringback
13 announcement 42004
14 queue-to      skill 4      pri m
15 wait-time      10      secs hearing music
16 announcement 42005
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 Manager *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 40002		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 (see [1]) 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 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 16**).

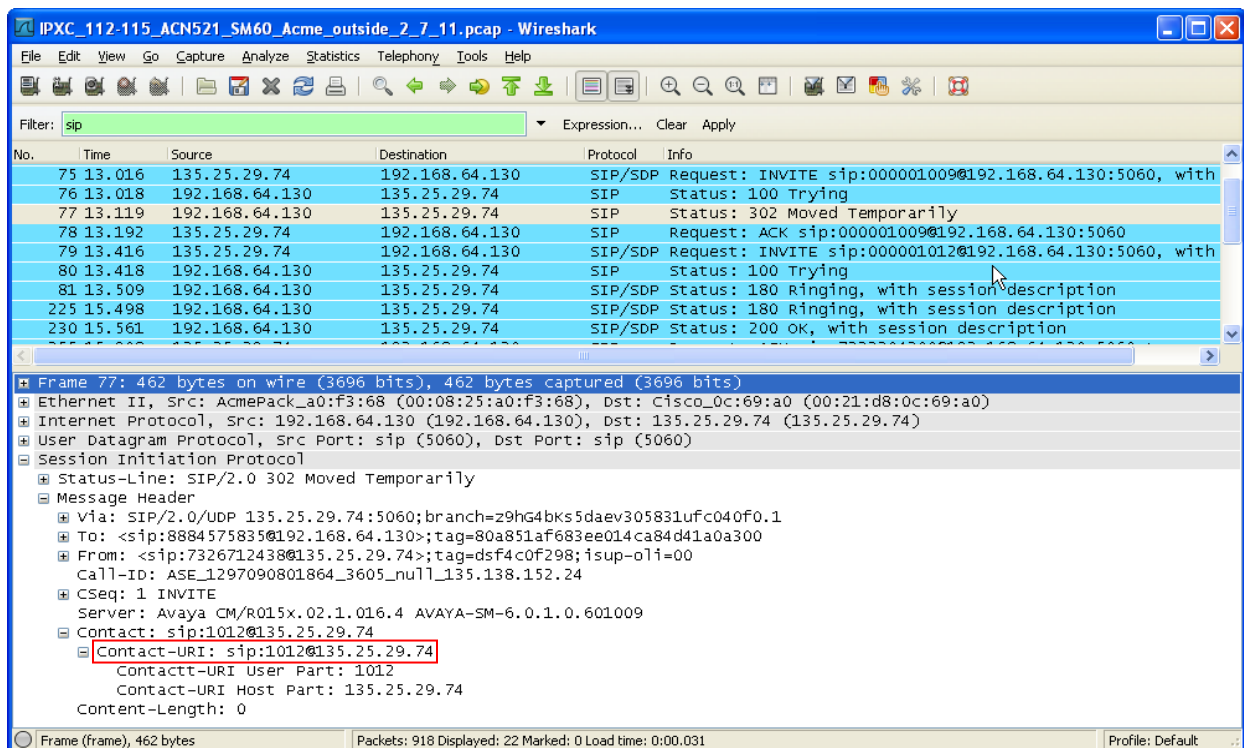


Figure 16: 302 Protocol Trace

- 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**).

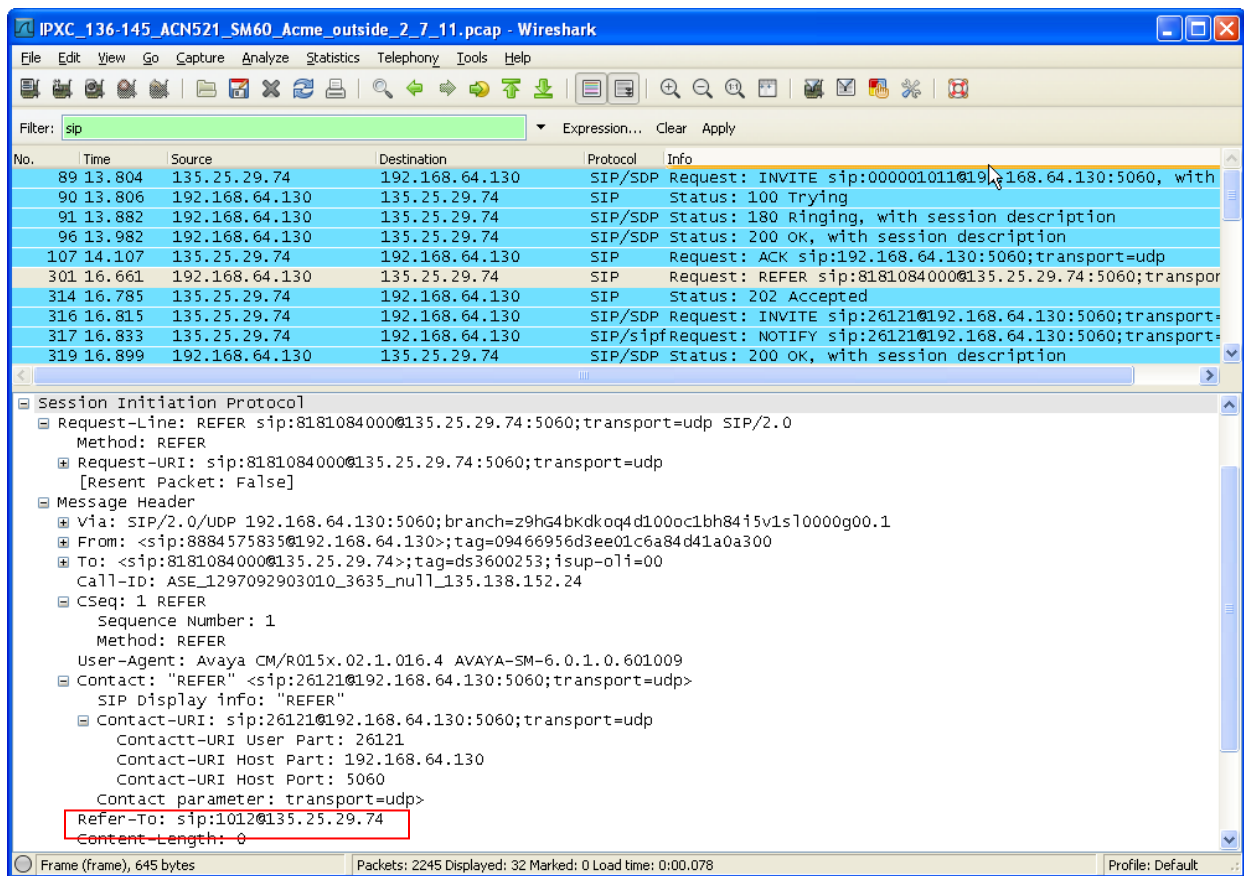


Figure 17: Refer Protocol Trace

- 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 (UII) 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 6.0 and Acme Packet Net-Net 6.2.0 with AT&T IP Toll Free Service – Issue 1.0*
- [2] *Installing and Configuring Avaya Aura® Communication Manager*, Doc ID 03-603558, Release 6.0 June, 2010
- [3] *Avaya Aura® Communication Manager Feature Description and Implementation*, Release 6.0, 555-245-205, Issue 8.0, June 2010
- [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

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