



Avaya Solution & Interoperability Test Lab

Application Notes for Avaya Aura® Communication Manager 5.2.1, Avaya Aura® Session 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), Avaya Aura® Session Manager, 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® Session Manager 6.0 is a core SIP routing and integration engine that connects disparate SIP devices and applications within an enterprise. Avaya Aura® Communication Manager 5.2.1 is a telephony application server and is the point of connection between the enterprise endpoints and Avaya Aura® Session Manager. An Acme Packet Net-Net connects Avaya Aura® Session 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 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, Avaya Aura® Session 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), Avaya Aura® Session Manager, 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 5.2.1, Avaya Aura® Session 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® System Manager 6.0, Avaya Aura® Session Manager 6.0, 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 Session Manager, 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 6**). With this feature enabled, 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 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. Refer to [1] for more information on the reference configuration.

- Session Manager provides core SIP routing and integration services that enables communications between disparate SIP-enabled entities, e.g., PBXs, SIP proxies, gateways, adjuncts, trunks, applications, etc. across the enterprise. Session Manager allows enterprises to implement centralized and policy-based routing, centralized yet flexible dial plans, consolidated trunking, and centralized access to adjuncts and applications.
- System Manager provides a common administration interface for centralized management of all Session Manager instances in an enterprise.
- 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 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 3800³ 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 Net-Net 3800, to Session Manager which routed the call 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).

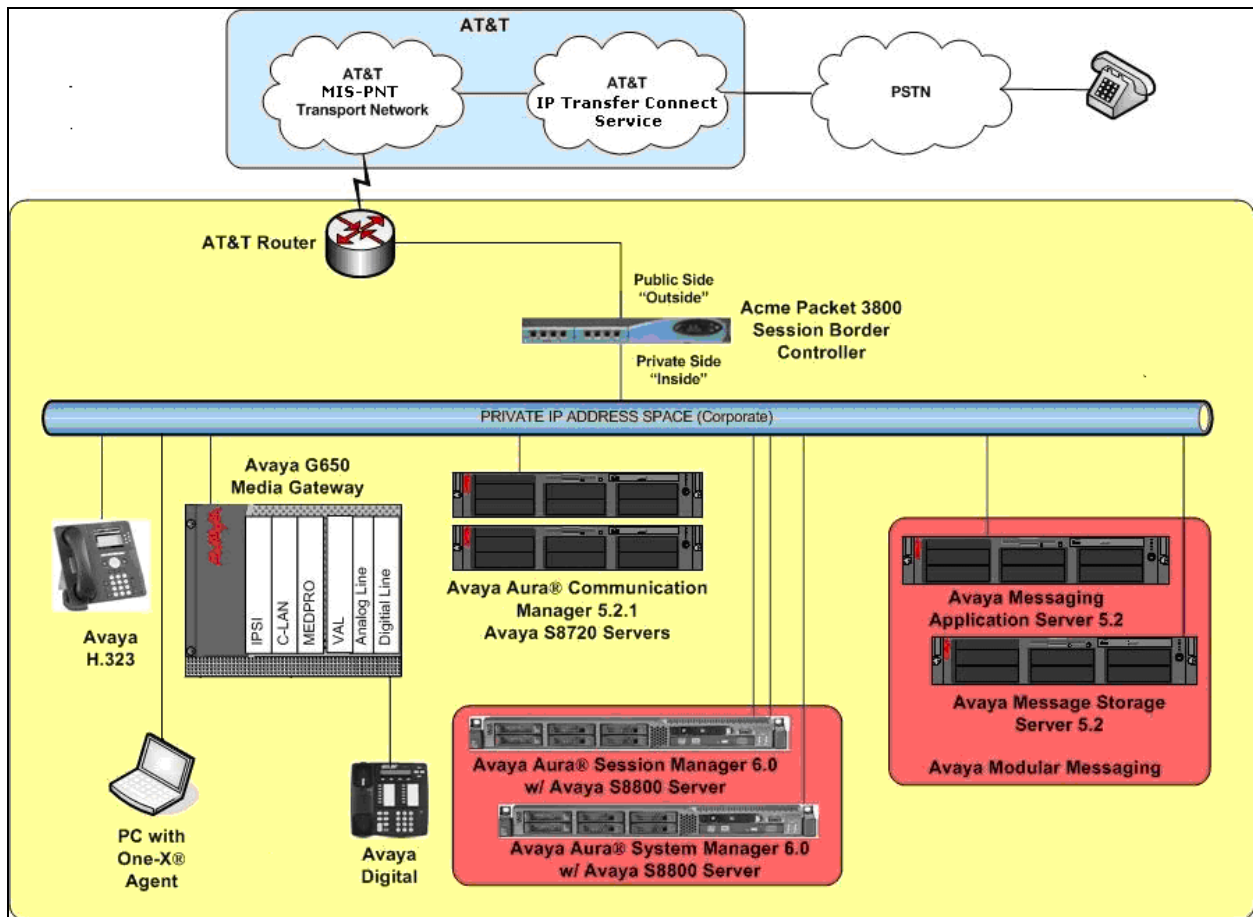


Figure 1: Reference Configuration

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® System Manager	
Management IP Address	192.168.67.207
Avaya Aura® Session Manager	
Management IP Address	192.168.67.209
SIP signaling IP Address	192.168.67.210
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® Session 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® Session Manager and Avaya Aura® Communication Manager, four general call flows are described in this section.

The first call scenario illustrated in **Figure 2** is an inbound AT&T IP Transfer Connect service call that arrives on Avaya Aura® Session Manager and is subsequently routed to Avaya Aura® Communication Manager, which in turn routes the call to a vector, agent, or phone. Note that no redirection is performed in this scenario, and thus the call flow is the same as that of an inbound AT&T IP Toll Free service call.

1. A PSTN phone originates a call to an AT&T IP Transfer Connect service number (an AT&T IP Toll Free service number that has been enabled with the AT&T IP Transfer Connect service option).
2. The PSTN routes the call to the AT&T IP Transfer Connect service network.
3. The AT&T IP Transfer Connect service routes the call to the Acme Packet SBC.
4. The Acme Packet SBC performs SIP Network Address Translation (NAT) and any necessary SIP header modifications, and routes the call to Avaya Aura® Session Manager.
5. Avaya Aura® Session Manager applies any necessary SIP header adaptations and digit conversions, and based on configured Network Routing Policies, determines where the call should be routed next. In this case, Avaya Aura® Session Manager routes the call to Avaya Aura® Communication Manager.
6. Depending on the called number, Avaya Aura® Communication Manager routes the call to a) a vector, which in turn, routes the call to an agent or phone, or b) directly to an agent or phone.

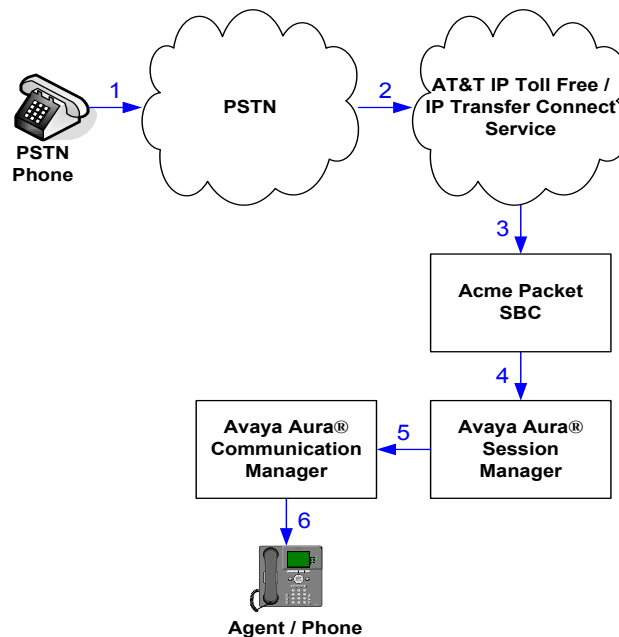


Figure 2: Inbound AT&T IP Transfer Connect Call – No Redirection

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

The second call scenario illustrated in **Figure 3** is an inbound AT&T IP Transfer Connect service call that arrives on Avaya Aura® Session Manager and is subsequently routed to Avaya Aura® Communication Manager, which in turn routes the call to a vector. The vector, without answering the call, immediately redirects the call back to the AT&T IP Transfer Connect service (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 Avaya Aura® Session Manager and then the Acme Packet SBC to the AT&T IP Transfer Connect service network. Since the SIP 302 message is a final response, the redirecting party (Avaya Aura® Communication Manager) is no longer involved in the call whether the redirection succeeds or fails, and thereby releases the trunk.
3. The AT&T IP Transfer Connect service places a call to the alternate destination and upon answer, connects the calling party to the target party (alternate destination).

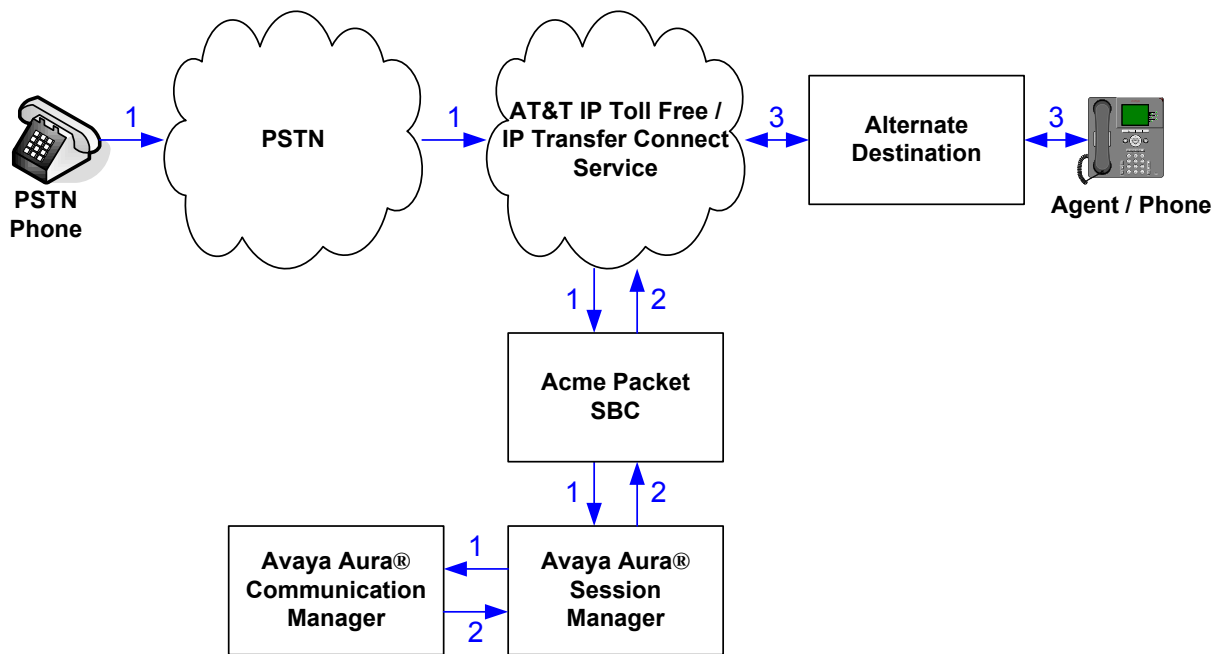


Figure 3: Inbound AT&T IP Transfer Connect Call – Pre-Answer SIP 302 Redirection

The third call scenario illustrated in **Figure 4** is an inbound AT&T IP Transfer Connect service call that arrives on Avaya Aura® Session Manager and is subsequently routed to Avaya Aura® Communication Manager, which in turn routes the call to a vector. The vector answers the call and then redirects the call back to the AT&T IP Transfer Connect service (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 Avaya Aura® Session Manager and then the Acme Packet SBC to the AT&T IP Transfer Connect service network.
3. The AT&T IP Transfer Connect service places a call to the target party (alternate destination) and upon answer, connects the calling party to the target party.
4. The AT&T IP Transfer Connect service clears the call on the redirecting/referring party (Avaya Aura® Communication Manager).

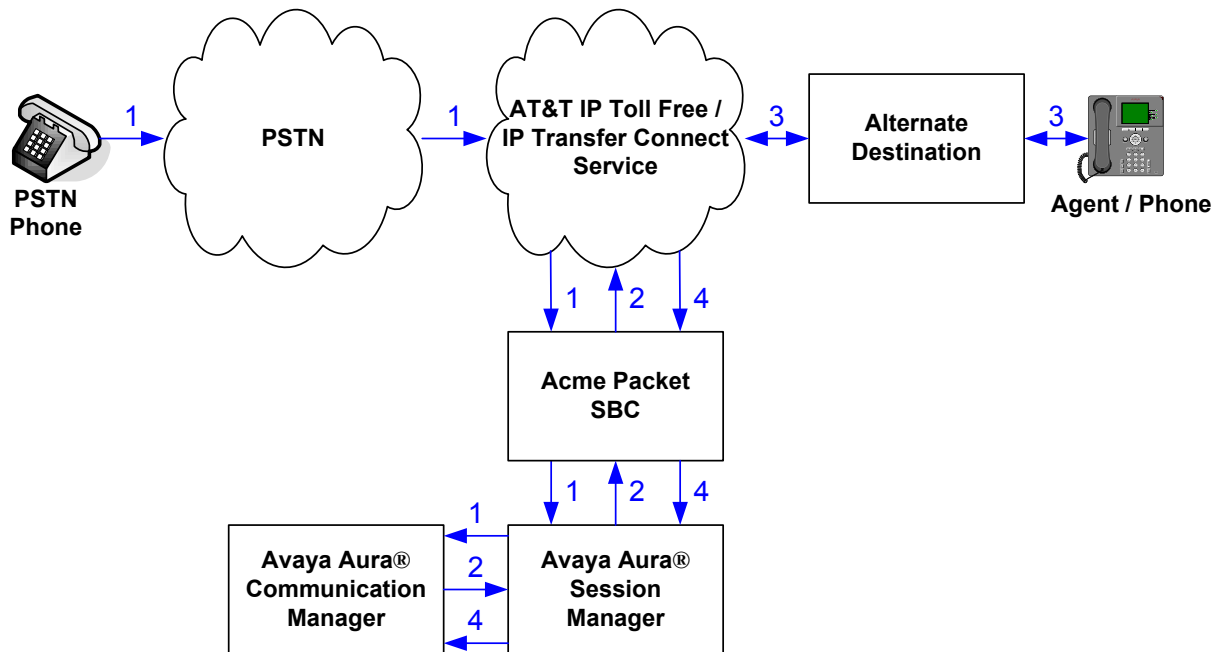


Figure 4: Inbound AT&T IP Transfer Connect Call – Post-Answer SIP REFER Redirection Successful

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

1. Step 1 is the same as the third call scenario above.
2. Step 2 is the same as the third call scenario above.
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.

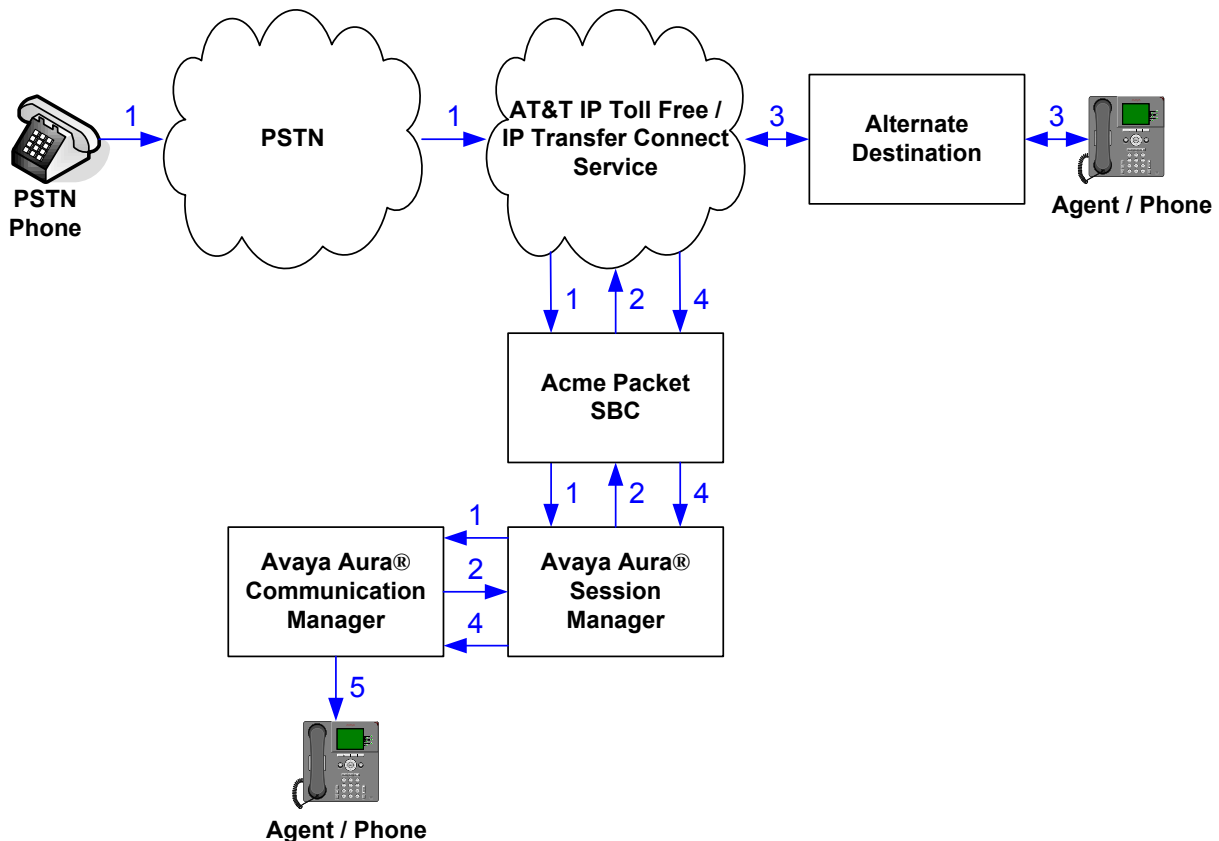


Figure 5: Inbound AT&T IP Transfer Connect Call – Post-Answer SIP REFER Redirection Unsuccessful

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® System Manager 6.0 (6.0.0.0.556-3.0.6.1)
Avaya S8800 Server	Avaya Aura® Session Manager 6.0 (6.0.0.0.600020)
Avaya 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

Note - The solution integration validated in these Application Notes should be considered valid for deployment with Avaya Aura® Communication Manager release 5.2.1 and Avaya Aura® Session Manager release 6.1. Avaya agrees to provide service and support for the integration of Avaya Aura® Communication Manager release 5.2.1 and Avaya Aura® Session Manager release 6.1 with the AT&T IP Transfer Connect service offer, in compliance with existing support agreements for Avaya Aura® Communication Manager release 5.2.1 and Avaya Aura® Session Manager 6.0, and in conformance with the integration guidelines as specified in the body of this document.

5. Avaya Aura® Session Manager

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

5.1. Dial Patterns

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

6. Avaya Aura® Communication Manager

The Avaya Aura® Communication Manager administration for interaction with the AT&T IP Toll Free service is described in [1] and are applicable for the AT&T IP Transfer Connect service as well. This section describes the additional administration steps on 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.

6.1. 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

6.2. Trunks

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

1. Enter the **change trunk-group t** command, where **t** is the number of the trunk group administered in [1] for inbound AT&T IP Toll Free service calls. On Page 4 of the **trunk-group** form, set **Network Call Redirection** to “y” (see **item 2** in **Section 2.2.1**).

```

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? y
                                Telephone Event Payload Type:

```

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.

6.3. 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 6.2** 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 [6] and [7] for further information.

6.3.1. Pre-Answer Redirection

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

- Plays ringback for 2 seconds (vector step **02**).
- Assigns the data “**1234567890123456**” to ASAI UUI variable “**A**” and “**7890123456789012**” to ASAI UUI variable “**B**” (vector steps **05** and **06**).
Note: The parameters for ASAI UUI variables “**A**” and “**B**”, and other vector variables are defined using the **change variables** command (see **Figure 9**).
- Redirects the call to the number “**1012**” (vector step **08**). Note that since this vector did not answer the call, the presence of the “~” in the “**route-to number**” instructs 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 Session Manager and the Acme Packet SBC).

```
change variables
```

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VARIABLES FOR VECTORS

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

Figure 9: Change Variables Form

```
display vdn 31009
```

Page 1 of 3

VECTOR DIRECTORY NUMBER

Extension: 31009
Name*: NCR Ringback 302 UUI
Destination: Vector Number 1009

Attendant Vectoring? n
Meet-me Conferencing? n
Allow VDN Override? n
COR: 1
TN*: 1
Measured: none

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

* Follows VDN Override Rules

Figure 10: Sample VDN for Pre-Answer Redirection


```

display vector 1009                                     Page 1 of 6
                                                    CALL VECTOR

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

```

Figure 11: Sample Vector for Pre-Answer Redirection

6.3.2. Post-Answer Redirection

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

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

```

display vdn 31010                                     Page 1 of 3
                VECTOR DIRECTORY NUMBER

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

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

* Follows VDN Override Rules

```

Figure 12: Sample VDN for Post-Answer Redirection

```

display vector 1010                                   Page 1 of 6
                CALL VECTOR

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

```

Figure 13: Sample Vector for Post-Answer Redirection

In order to display the UUI information defined in the vector 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	RC:	Grp:
2: call-appr	6: aux-work		Grp:
3:	7: auto-in		
4: send-calls Ext:	8: uui-info		
voice-mail Number: 26000			

Figure 14: UUI display button

6.4. 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.

- 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
    header-name
    action
    comparison-type
    msg-type
    methods
    match-value
    new-value
    element-rule
        name
        parameter-name
        type
        action
    modReferTo
    Refer-To
    manipulate
    case-sensitive
    any
    REFER
    modmline
    uri-host
    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

7. Verification Steps

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

7.1. 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 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 15**).

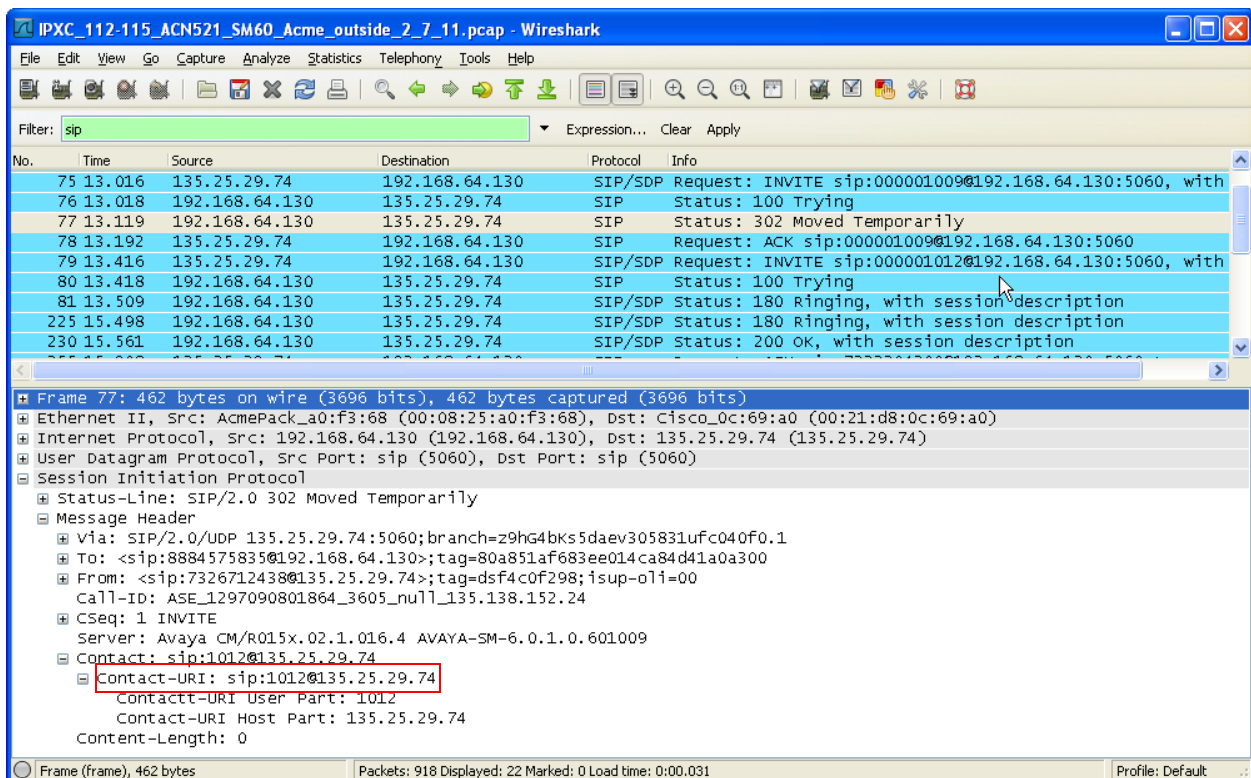


Figure 15: 302 Protocol Trace

- 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 16**).

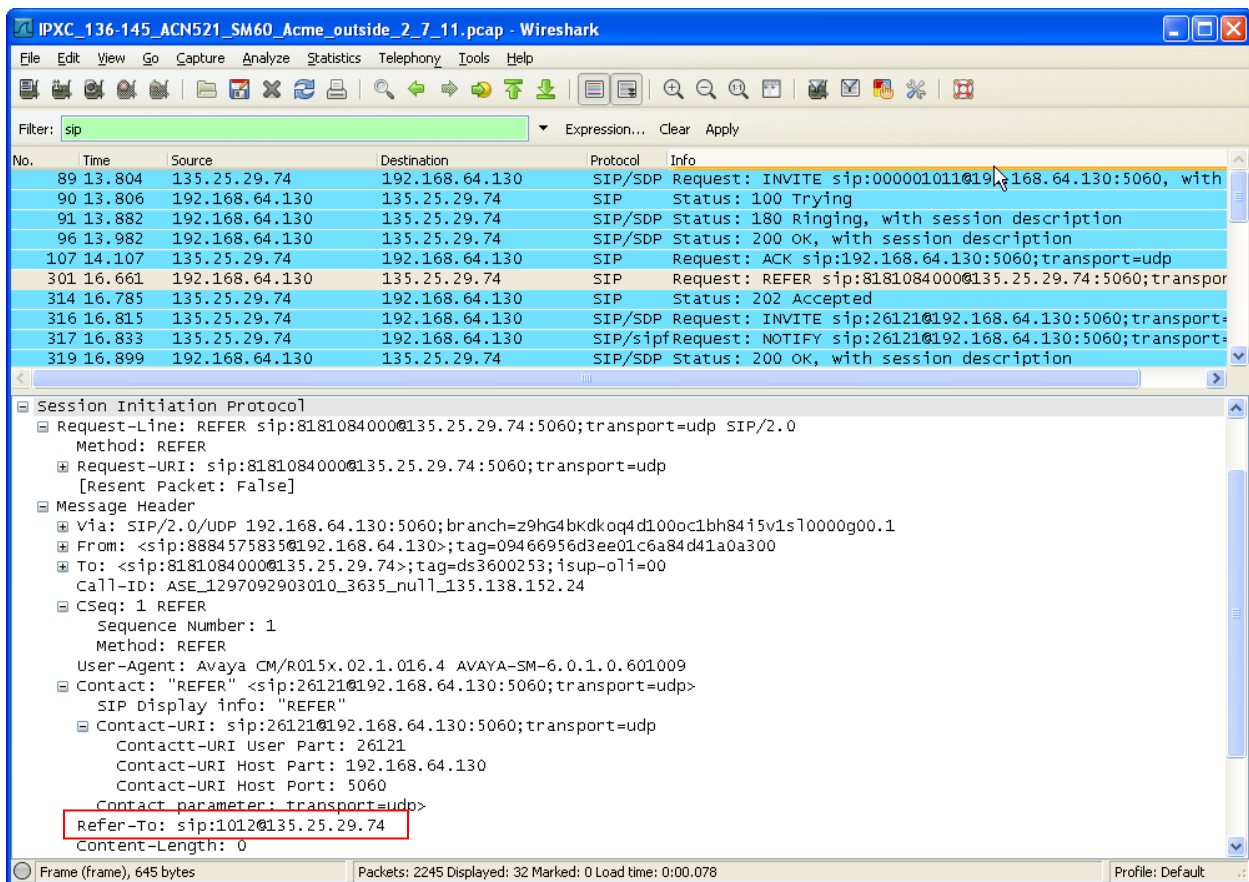


Figure 16: Refer Protocol Trace

- 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 6.3.2**).

8. Conclusion

As illustrated in these Application Notes, (and building upon the configurations described in [1]), Avaya Aura® Session Manager, 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 can work in complement with 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.

9. 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, Avaya Aura® Session 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® Session Manager*, Doc ID 03-603473 Release 6.
- [3] *Administering Avaya Aura® Session Manager*, Doc ID 03-603324, Release 6.0, June 2010
- [4] *Administering Avaya Aura® Communication Manager*, Issue 5.0, Release 5.2, May 2009, Document Number 03-300509
- [5] *Avaya Aura® Communication Manager Feature Description and Implementation*, Issue 7, Release 5.2, May 2009, Document Number 555-245-205
- [6] *Administering Avaya Aura® Call Center Features*, Release 6.0, June 2010
- [7] *Programming Call Vectors in Avaya Aura® Call Center*, 6.0, June 2010
- [8] Acme Packet Support (login required) - <http://support.acmepacket.com>

10. 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 6**). With this feature enabled, Avaya Aura® Communication Manager will also use the SIP parameter *SendOnly* to signal any hold call conditions. The *SendOnly* SIP parameter is not supported by the AT&T Flexible Reach service. Any customers that access both AT&T IP Transfer Connect and AT&T IP Flexible Reach services via the same Avaya Aura® Communication Manager environment, must use the following procedures to have the Acme Packet SBC replace the *SendOnly* parameter with the *SendRecv* parameter that the AT&T Flexible Reach service does support.

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

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

header-rule

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

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