Avaya Solution & Interoperability Test Lab

Application Notes for Avaya Aura® Communication Manager 6.0.1, Avaya Aura® Session Manager 6.1, 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), Avaya Aura® Session Manager, and the Acme Packet Net-Net Session Border Controller (models 3800, and 4500) with the AT&T IP Transfer Connect service using **AVPN** or **MIS/PNT** transport connections. The AT&T IP Transfer Connect service is a service option available with the AT&T IP Toll Free service, and supports the rerouting of inbound toll free calls to alternate destinations based upon SIP redirection messages from Avaya Aura® Communication Manager. In addition, the Avaya Aura® Communication Manager NCR and SIP User-to-User Information (UUI) features can be utilized together, in conjunction with the Data Forwarding option of the AT&T IP Transfer Connect service, to transmit UUI within SIP signaling messages to the alternate destinations. Avaya Aura® Session Manager is a core SIP routing and integration engine that connects disparate SIP devices and applications within an enterprise. Note that these Application Notes are intended to supplement the separate document: *Applications Notes for Avaya Aura® Communication Manager 6.0.1, Avaya Aura® Session Manager 6.1 and Acme Packet Net-Net 6.2.0 with AT&T IP Toll Free SIP Trunk Service*.

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, and 4500) Session Border Controller (SBC) with the AT&T IP Transfer Connect service using **AVPN** or **MIS/PNT** transport connections[1]. The AT&T IP Transfer Connect service is a service option available with the AT&T IP Toll Free service, and supports the rerouting of inbound toll free calls to alternate[2] 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.

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 6.0.1, Avaya Aura® Session Manager 6.1 and Acme Packet Net-Net 6.2.0 with AT&T IP Toll Free SIP Trunk Service*.

2. General Test Approach and Test Results

The test environment consisted of:

- A simulated enterprise with Avaya Aura® System Manager, Avaya Aura® Session Manager, Avaya Aura® Communication Manager, Avaya phones, and an Acme Packet Net-Net Session Border Controller.
- A laboratory version of the AT&T IP Transfer Connect service, to which the simulated enterprise was connected.

2.1. Interoperability Compliance Testing

The interoperability compliance testing focused on verifying inbound call flows (see **Section 3.2** for descriptions) to Session Manager and subsequent routing to Communication Manager, and subsequent redirection messages to AT&T for rerouting to alternate destinations.

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[1] MIS/PNT transport does not support compressed RTP (cRTP), however AVPN transport does support cRTP

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

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.
- Call redirection functionality utilizing 302 and Refer SIP call processing.
- Communication Manager features such as hold, resume, and transfer.

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, 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, 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 on the alternate destination.
- 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. The Communication Manager 6.0.1 SIP trunk form may be configured to send either a 180 (default) or 183 session progress message (see **Section 6.2**). The session progress message selected alters the Communication Manager behavior upon receipt of a Notify form AT&T during Refer calls. If 180 is selected, then Communication Manager will issue a BYE upon receipt of the Notify/Ringing message from AT&T. If 183 is selected, then Communication Manager will issue a BYE upon receipt of the Notify/200OK from AT&T. In both cases the expected behavior was for the Avaya CPE to wait for AT&T IP Transfer Connect service to issue the BYE. However no issues were encountered during testing due to either behavior.
2. Communication Manager 6.0 inserts a leading plus sign to destination headers containing number strings by default (e.g. Update, From, PAI, Contact). The AT&T IP Transfer Connect service does not support the use of digit strings with a leading plus sign ("+") in the headers. See **Section 7.1** that describes utilizing the Acme Packet Net-Net Session Border Controller, to remove the plus signs inserted by Avaya Aura® Communication Manager.
3. 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 6.2**).

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 that describes having the Acme Packet Net-Net Session Border Controller replace the *SendOnly* parameter with the *SendRecv* parameter that the AT&T Flexible Reach service does support.

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:

- Avaya Aura® Session Manager provides core SIP routing and integration services that enables communications between disparate SIP-enabled entities, e.g., PBXs, SIP proxies, gateways, adjuncts, trunks, applications, etc. across the enterprise.
  Session Manager allows enterprises to implement centralized and policy-based routing, centralized yet flexible dial plans, consolidated trunking, and centralized access to adjuncts and applications.
- Avaya Aura® System Manager provides a common administration interface for centralized management of all Session Manager instances in an enterprise.
- Avaya Aura® Communication Manager provides the voice communications services for a particular enterprise site. In the reference configuration, Communication Manager runs on an Avaya S8800 Server in a Processor Ethernet (Procr) configuration. This solution is extensible to other Avaya S8xxx Servers.
- The Avaya Media Gateway provides the physical interfaces and resources for Communication Manager. In the reference configuration, an Avaya G450 Media Gateway is used. This solution is extensible to other Avaya Media Gateways.
- Avaya "desk" phones are represented with Avaya A175, 46x0, 96x0, and 96x1 Series IP Telephones running H.323 or SIP software, Avaya 6211 Series Analog Telephones, as well as Avaya one-X® Communicator and Avaya one-X® Agent, PC based softphones. Note that agent phones are H.323.

- The Acme Packet Net-Net 3800[3] Session Border Controller provides Back to Back User Agent (B2BUA) functionality, including address translation and SIP header manipulation between the AT&T IP Transfer Connect service and the enterprise internal network. UDP transport protocol is used between the Acme Packet Net-Net Session Border Controller and the AT&T IP Transfer Connect service.
- An existing Avaya Modular Messaging system (in Multi-Site mode in this reference configuration) provides the corporate voice messaging capabilities in the reference configuration. The provisioning of Modular Messaging is beyond the scope of this document.
- Inbound calls were placed from PSTN via the AT&T IP Transfer Connect service, through the Acme Packet Net-Net Session Border controller to the Session Manager which routed the call to Communication Manager. Communication Manager terminated the call to the appropriate agent/phone. The H.323 phones on the enterprise side registered to the Communication Manager Processor Ethernet interface (Procr, see [1]) . The SIP phones on the enterprise side registered to the Session Manager.

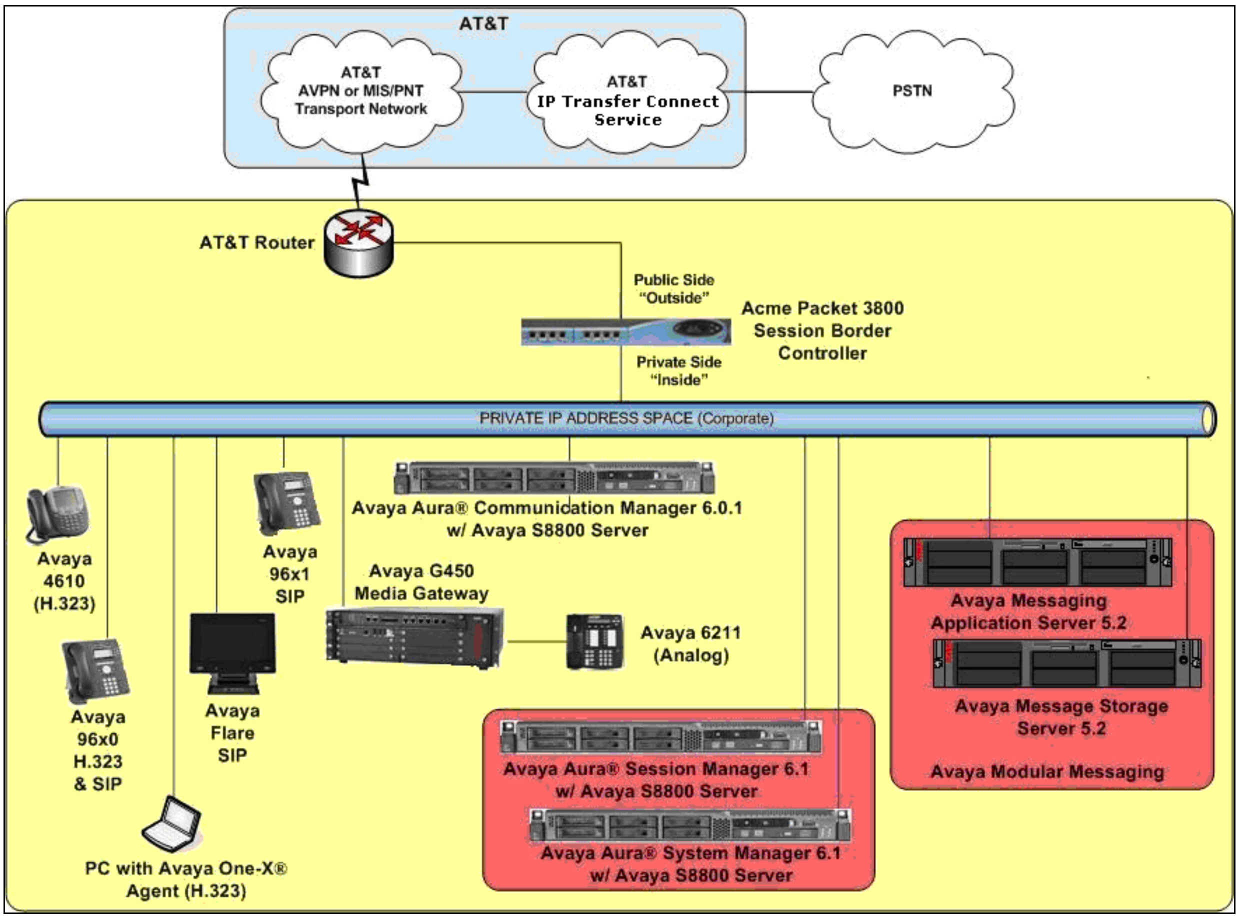


Figure 1: Reference Configuration

[3] Although an Acme Packet Net-Net SD 3800 was used in the reference configuration, the 4500 platform is 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 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
Network IP Address	192.168.67.210
Avaya Aura® Communication Manager	
Procr IP Address	192.168.67.202
Avaya Aura® Communication Manager extensions	40xxx = H323 and Analog 41xxx = SIP
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	1723114xxxx
Acme Packet Net-Net Session Border Controller	
IP Address of "Outside" (Public) Interface (connected to AT&T Access Router/IP Toll Free Service)	192.168.64.130 (active)
IP Address of "Inside" (Private) Interface (connected to Avaya Aura® Session Manager)	192.168.67.130 (active)
AT&T IP Transfer Connect 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 Session Manager and 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 Session Manager and is subsequently routed to 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 Net-Net Session Border Controller.
4. The Acme Packet Net-Net Session Border Controller performs SIP Network Address Translation (NAT) and any necessary SIP header modifications, and routes the call to Session Manager.
5. 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, Session Manager routes the call to Communication Manager.
6. Depending on the called number, 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.

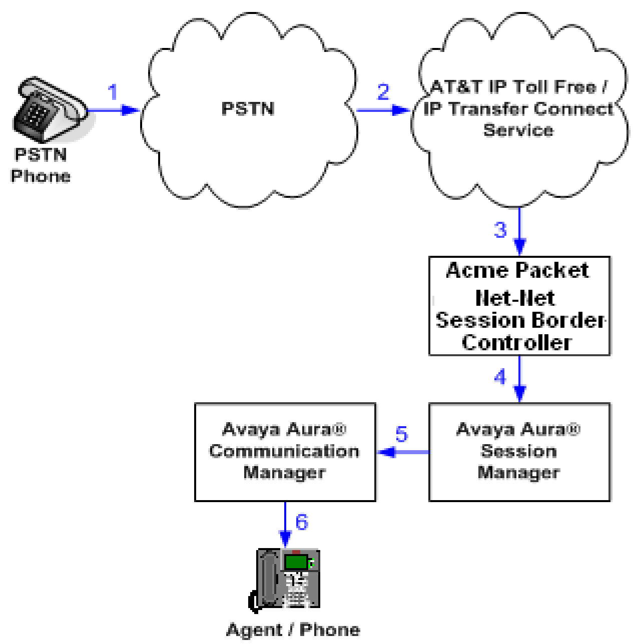


Figure 2: Inbound AT&T IP Transfer Connect Call – No Redirection

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 Session Manager and is subsequently routed to Communication Manager, which in turn routes the call to a vector. The vector, without answering the call, immediately redirects the call back to the AT&T IP Transfer Connect service for routing to an alternate destination.

1. Same as the first five steps from the first call scenario.
2. 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 Session Manager and then the Acme Packet Net-Net Session Border Controller to the AT&T IP Transfer Connect service network. Since the SIP 302 message is a final response, the redirecting party (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).

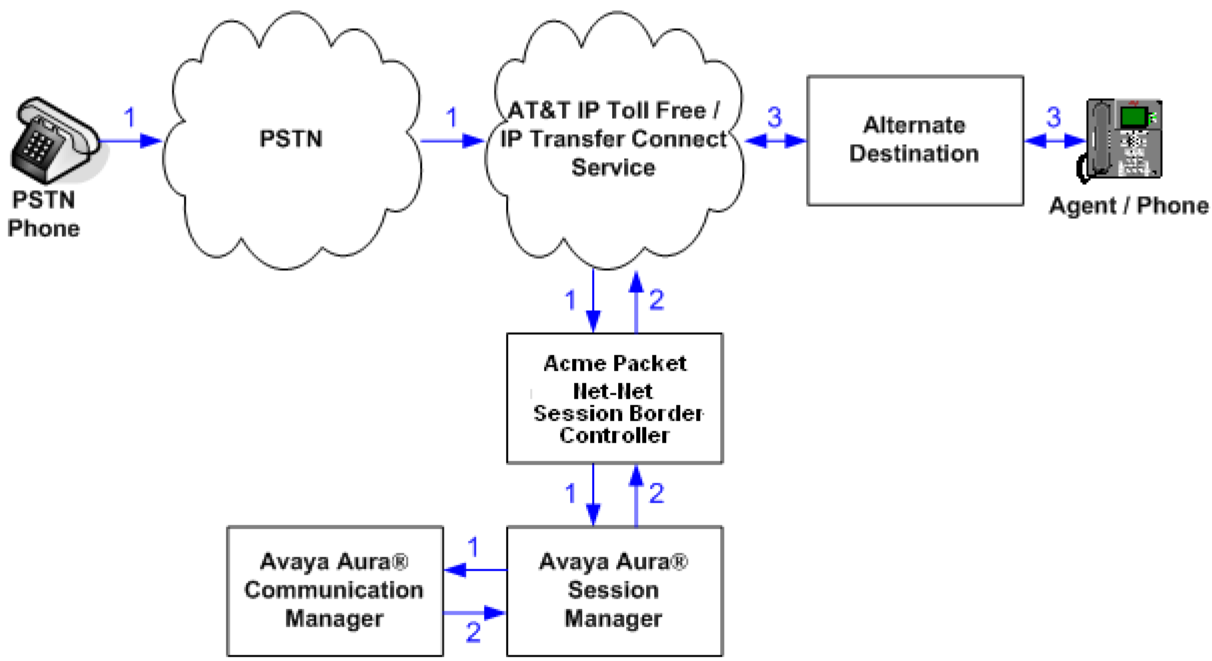


Figure 3: Inbound AT&T IP Transfer Connect Call – Pre-Answer SIP 302 Redirection

The third call scenario illustrated in **Figure 4** is an inbound AT&T IP Transfer Connect service call that arrives on Session Manager and is subsequently routed to Communication Manager, which in turn routes the call to a vector. The vector answers the call and then redirects the call back to the AT&T IP Transfer Connect service for routing to an alternate destination.

1. Same as the first five steps from the first call scenario.
2. 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 Session Manager and then the Acme Packet Net-Net Session Border Controller 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 (Communication Manager).

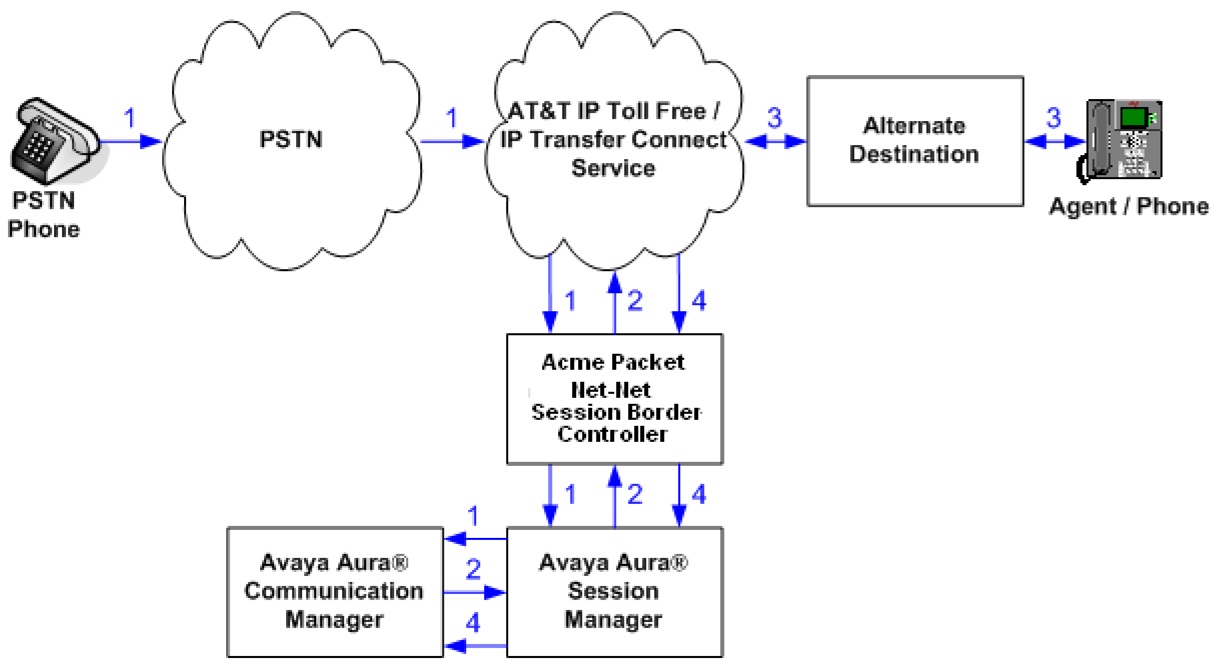


Figure 4: Inbound AT&T IP Transfer Connect Call – Post-Answer SIP REFER Redirection Successful

The fourth call scenario illustrated in **Figure** 5 is similar to the fourth call scenario, except that the redirection is unsuccessful due to the alternate destination being busy or otherwise unavailable. As a result, 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 (Communication Manager) of the error condition.
5. 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.

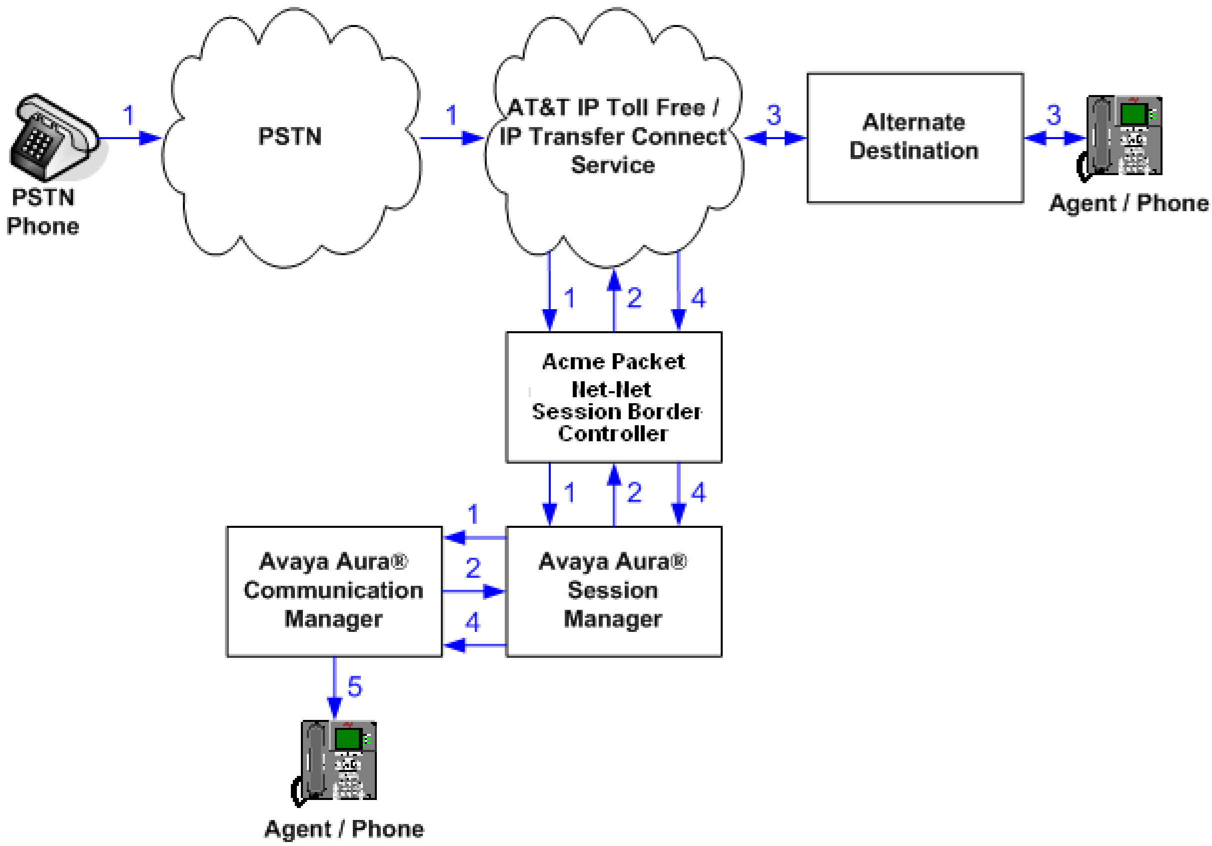


Figure 5: Inbound AT&T IP Transfer Connect Call – Post-Answer SIP REFER Redirection Unsuccessful

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 6.1 SP4 (6.1.0.0.7345-6.1.5.112 update 6.1.8.1.1455) System Platform 6.0.3.1.3
Avaya S8800 Server	Avaya Aura® Session Manager 6.1 (6.1.4.0.614005)
Avaya S8800 Server	Avaya Aura® Communication Manager 6.0.1 SP3 (00.1.510.1-19009) System Platform 6.0.3.1.3
Avaya G450 Media Gateway	31.19.2
MM711 Analog card	HW31 FW094
Avaya 9630 IP Telephone	H.323 Version S3.110b (ha96xxua3_11.bin) SIP Version 2.6.4 (SIP96xx_2_6_4_0.bin)
Avaya 9621 IP Telephone	SIP Version 6.0.1 (S96x1_SALBR6_0_1_V452)
Avaya A175 Desktop Video Device (SIP telephone function)	SIP Version 1.0.3 (SIP_A175_1_0_3_000011)
Avaya one-X® Agent	2.5.00467.09
Avaya 4610SW IP Telephone	H323 Version 2.9.1 (a10d01b2_9_1.bin)
Avaya 6211 Analog phone	-
Acme Packet Net-Net 3800	SCX6.2.0m6p3
AT&T IP Transfer Connect Service using AVPN/MIS-PNT transport service connection	VNI 20 & VNI 21

Table 2: Equipment and Software Versions

5. Avaya Aura® Session Manager

The Avaya Aura® Session Manager administration for interaction with the AT&T IP Toll Free / IP Transfer Connect service is described in [1]. This section describes the additional administration steps on 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 licenses that are not enabled in the steps that follow, contact an authorized Avaya account representative to obtain the licenses.

1. Enter the **display system-parameters customer-options** command. On Page 4 of the **system-parameters customer-options** form, verify that the **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.)
```

2. On Page 6 of the **system-parameters customer-options** form, verify that the vectoring features 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
        (NOTE: You must logoff & login to effect the permission changes.)
```

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 x** command, where **x** 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 3** in **Section 2.2.1** and **Addendum 1**).
2. Note whether the setting for **Convert 180 to 183 for Early Media?** is "**n**" (default) or "**y**" (based on customer configuration). The value defined will alter the Refer NOTIFY response behavior (see **item 1** in **Section 2.1.1)**.

```
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: 100

       Convert 180 to 183 for Early Media? n
 Always Use re-INVITE for Display Updates? n
       Identity for Calling Party Display: P-Asserted-Identity
                            Enable Q-SIP? n
```

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. The routing of inbound AT&T IP Toll Free service calls that do not invoke the SIP NCR functionality is addressed in [1].

Two different inbound call routing scenarios are described in these Application Notes:

- Pre-Answer Redirection - An inbound AT&T IP Transfer Connect service call that invokes SIP NCR (using a SIP 302 message) prior to the call being answered.
- Post-Answer Redirection - An inbound AT&T IP Transfer Connect service call that invokes SIP NCR (using a SIP REFER message) after the call has been answered by a vector.

The following inbound call treatment information is defined using the inbound number information provided by AT&T in **Section 3.1**.

These Application Notes provide rudimentary vector definitions to demonstrate and test the SIP NCR and UUI functionalities. In general, call centers will use vector functionality that is more complex and tailored to their individual needs. Call centers may also use customer hosts running applications used in conjunction with Avaya Aura® 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 7**, which invokes the vector shown in **Figure 8**. 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 ASAI UUI variables "**A**" and "**B**", and other vector variables are defined using the **change variables** command (see **Figure 6**).
- 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 Net-Net 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
C
D
E
F
G
H
I
J
K
L
M
N
O
P
Q
R
```

Figure 6: 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 7: Sample VDN for Pre-Answer Redirection

```
display vector 22                                               Page   1 of   6
                                   CALL VECTOR

    Number: 22               Name: 302RingUUI
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    2   secs hearing ringback
03
04 #    Define UUI variable
05 set          A      = none   CATR  1234567890123456
06
07 #    Redirect
08 route-to     number ~r1012           with cov n if unconditionally
09 stop
10
11
12
```

Figure 8: 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 9**, which invokes the vector shown in **Figure 10**. The vector does the following:

- Plays ringback for 2 seconds (vector step **02**).
- Assigns the data "**1234567890123456**" to ASAI UUI variable "**A**" (vector steps **05**).
  Note: The parameters for UUI variable "**A**" and other vector variables are defined using the **change variables** command (see **Figure 6**).
- 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 Net-Net 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 9: 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 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 10: Sample Vector for Post-Answer Redirection

7. Acme Packet Net-Net Session Border Controller

The Acme Packet Net-Net SBC configuration for interaction with the AT&T IP Toll Free service is provided in [1]. The additional configuration on the Acme Packet Net-Net SBC necessary for supporting interaction with the AT&T IP Transfer Connect service is provided below as a reference. The following sections describe the Acme Packet Net-Net SBC header-rule provisioning. The header-rules described below were added to the existing sip-manipulation ***NAT_IP*** described in [1].

7.1. Removing Plus (+) from Number Strings

Communication Manager will insert a plus (+) into number strings. These leading plus signs may cause issue to the AT&T IP Toll Free service. The Acme Packet Net-Net SBC may be used to remove the plus signs before they are passed to AT&T. In the reference configuration the following headers were monitored to remove any plus signs – PAI, Contact, From, and Update.

- **PAI Header**

```
header-rule
      name                               modPAIPlus
      header-name                        P-Asserted-Identity
      action                             manipulate
      comparison-type                    pattern-rule
      msg-type                           any
      methods                            INVITE
      match-value
      new-value
      element-rule
            name                               modVal
            parameter-name
            type                               uri-user
            action                             find-replace-all
            match-val-type                     any
            comparison-type                    case-sensitive
            match-value                        \+(.*)
            new-value                          $modPAI.$modVal.$1
```

- **Contact Header**

```
header-rule
      name                               modContactPlus
      header-name                        Contact
      action                             manipulate
      comparison-type                    pattern-rule
      msg-type                           any
      methods                            INVITE
      match-value
      new-value
      element-rule
            name                               modVal
            parameter-name
            type                               uri-user
            action                             find-replace-all
            match-val-type                     any
```

```
        comparison-type          case-sensitive
        match-value              \+(.*)
        new-value                $modContact.$modVal.$1
```

- **From Header**

```
header-rule
    name                     modFromPlus
    header-name              From
    action                   manipulate
    comparison-type          pattern-rule
    msg-type                 any
    methods                  INVITE
    match-value
    new-value
    element-rule
        name                     modVal
        parameter-name
        type                     uri-user
        action                   find-replace-all
        match-val-type           any
        comparison-type          case-sensitive
        match-value              \+(.*)
        new-value                $modFrom.$modVal.$1
```

- **Update Header**

```
header-rule
    name                     modUpdatePlus
    header-name              Update
    action                   manipulate
    comparison-type          pattern-rule
    msg-type                 any
    methods
    match-value
    new-value
    element-rule
        name                     modVal
        parameter-name
        type                     uri-user
        action                   find-replace-all
        match-val-type           any
        comparison-type          case-sensitive
        match-value              \+(.*)
        new-value                $modUpdate.$modVal.$1
```

7.2. Modifying 302 Contact Header and Refer-To Header

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.

- **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
```

- **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
```

8. Verification Steps

8.1. Call Verification Tests

The call verification steps and troubleshooting tools described for the AT&T Toll Free service described in [1], apply to the AT&T IP Transfer Connect service as well.

1. Place an inbound call to an AT&T IP Transfer Connect service line enabled with Redirect features. Verify that an appropriate Communication Manager vector immediately redirects the call back to the AT&T IP Transfer Connect service for redirection to an alternate

destination using 302. Verify two-way talk path and transmission of UUI information as appropriate.

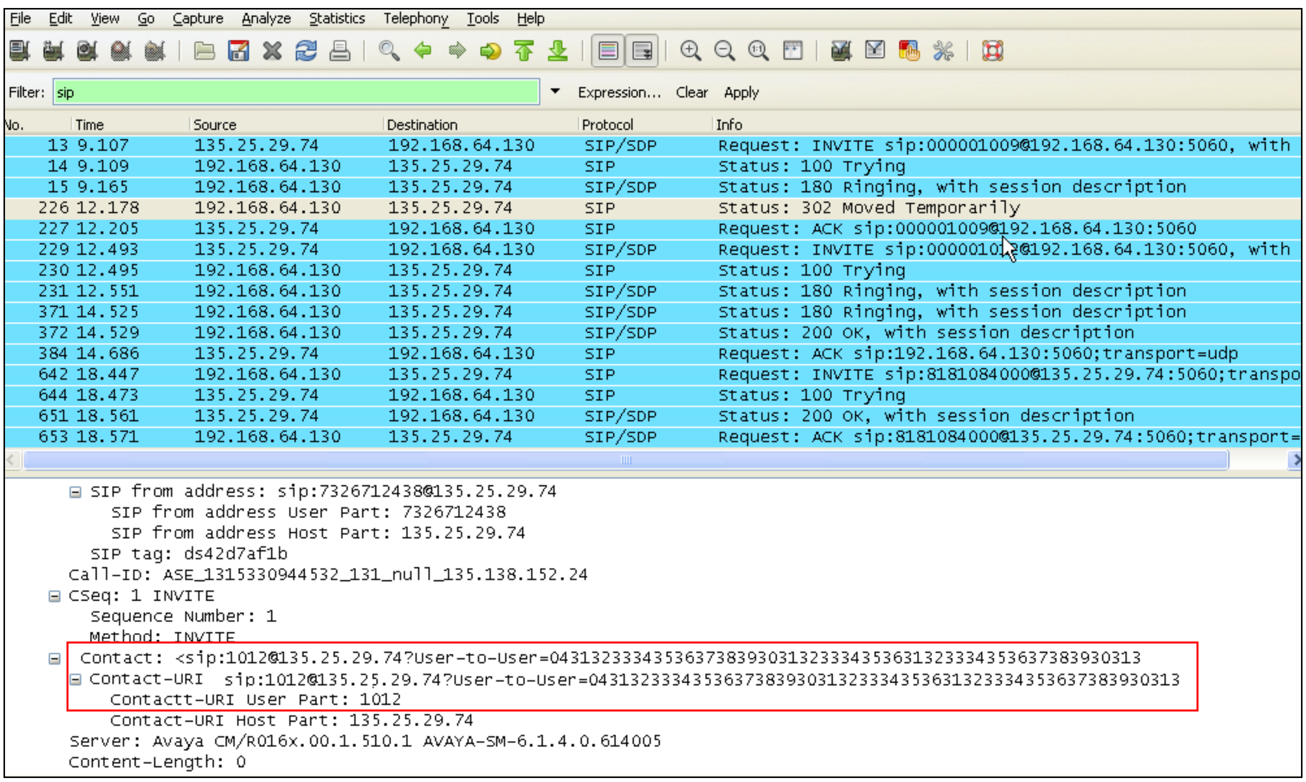
2. Place an inbound call to an AT&T IP Transfer Connect service line enabled with Refer features. Verify that an appropriate Communication Manager vector answers the call and then redirects the call back to the AT&T IP Transfer Connect service for redirection to an alternate destination using Refer. Verify two-way talk path and transmission of UUI information as appropriate.
3. Verify that when Communication Manager is the transfer target of redirected calls, the calls are answered with two-way talk path. Verify that the calls remain stable for several minutes and disconnect properly.

8.2. Protocol Traces

Using a SIP protocol analyzer (e.g. Wireshark), monitor the SIP traffic at the Acme Packet Net-Net SBC public “outside” interface connection to the AT&T IP Transfer Connect service.

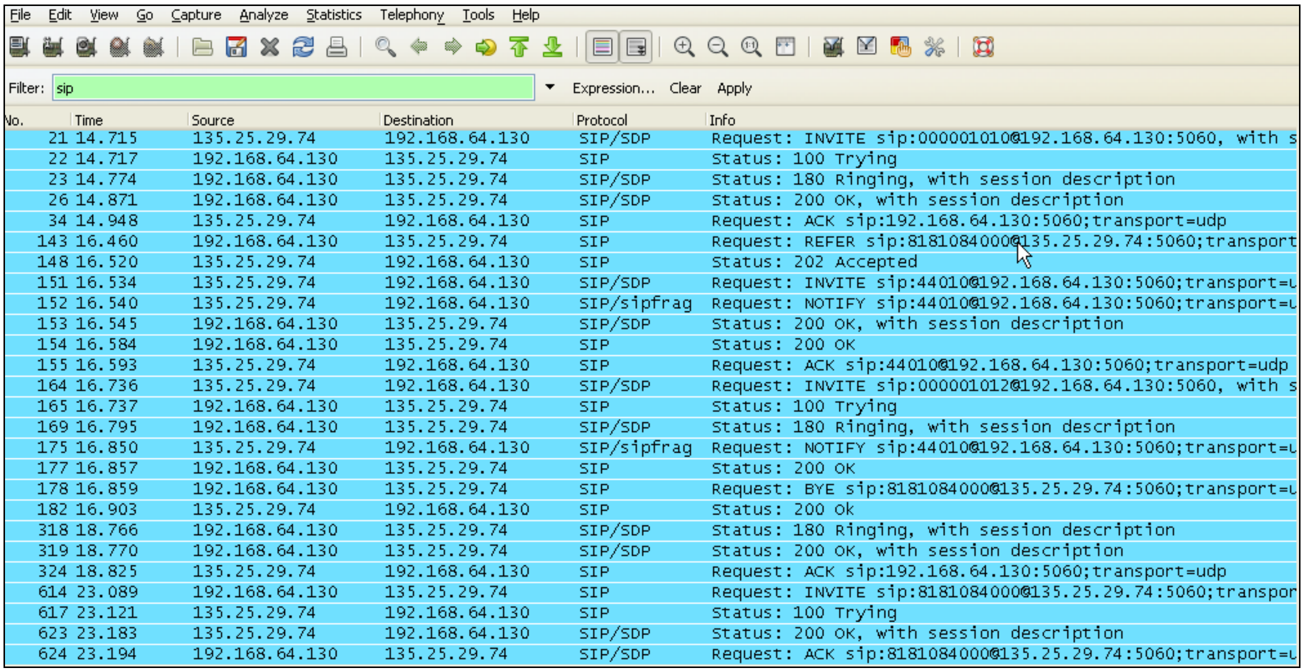
8.2.1. 302

The following is an example of a 302 redirection call filtering on the SIP protocol. Note that the Contact header contains the new called number (1012) as defined in **Section 6.3**. Also note the UUI information.

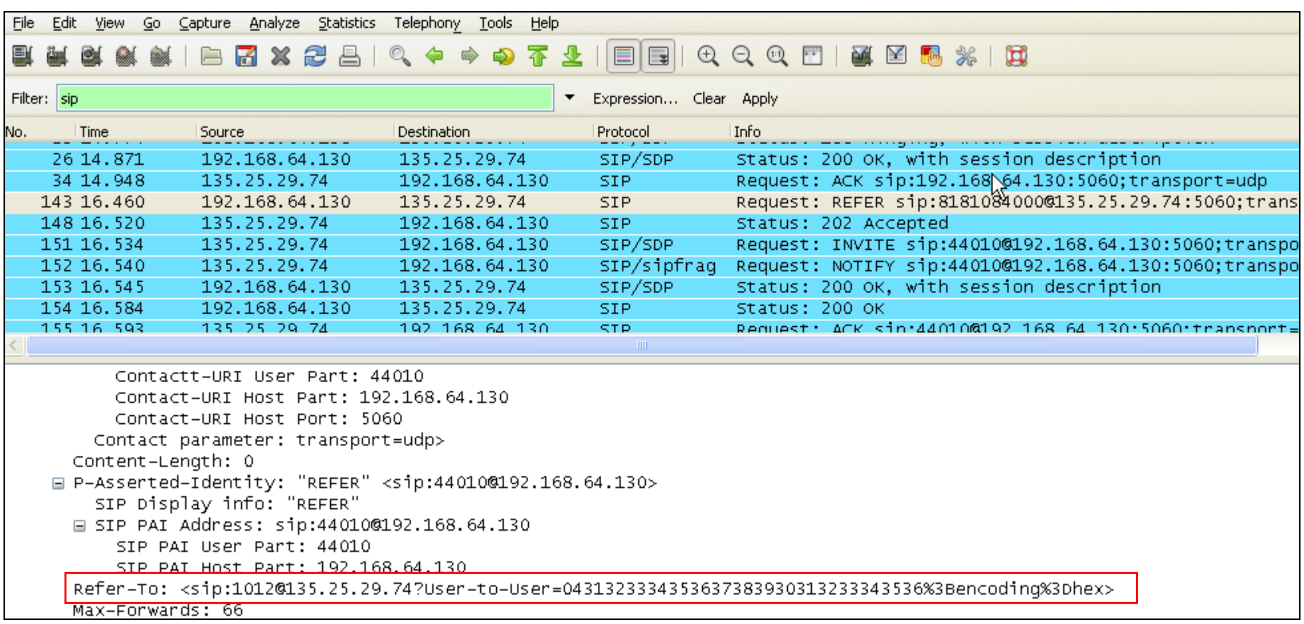


8.2.2. Refer with 180

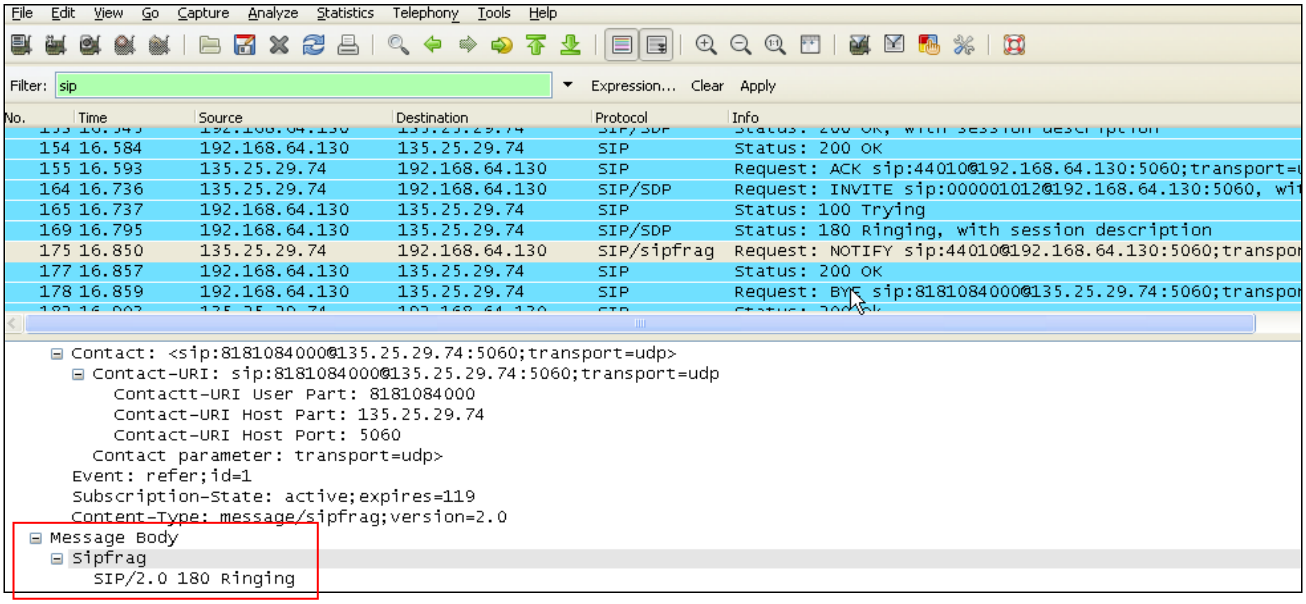
The following is an example of a Refer (frame 143) redirection call filtering on the SIP protocol. Note that Communication Manager is sending 180 Ringing. As described in **Section 2.2.1**, this causes Communication Manager to send a BYE (frame 178) upon receipt of the Notify/180 Ringing sent by AT&T in frame 175.



This screen shows the Refer in frame 143 in detail. The Refer-To header specifies the new called number (1012) as defined in **Section 6.3**. Also note the UUI information.

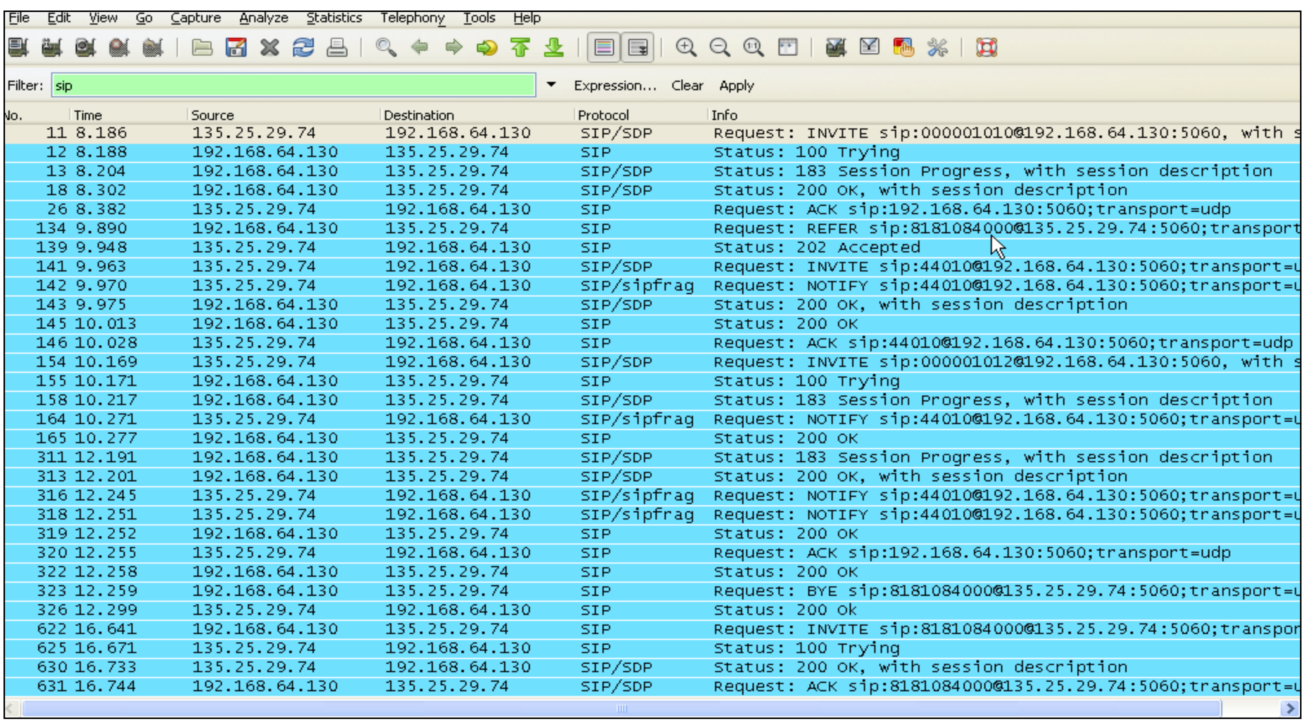


This screen shows the Notify/180 Ringing sent by AT&T in frame 175. Note that this Notify is in response to the 180 Ringing sent in frame 169.

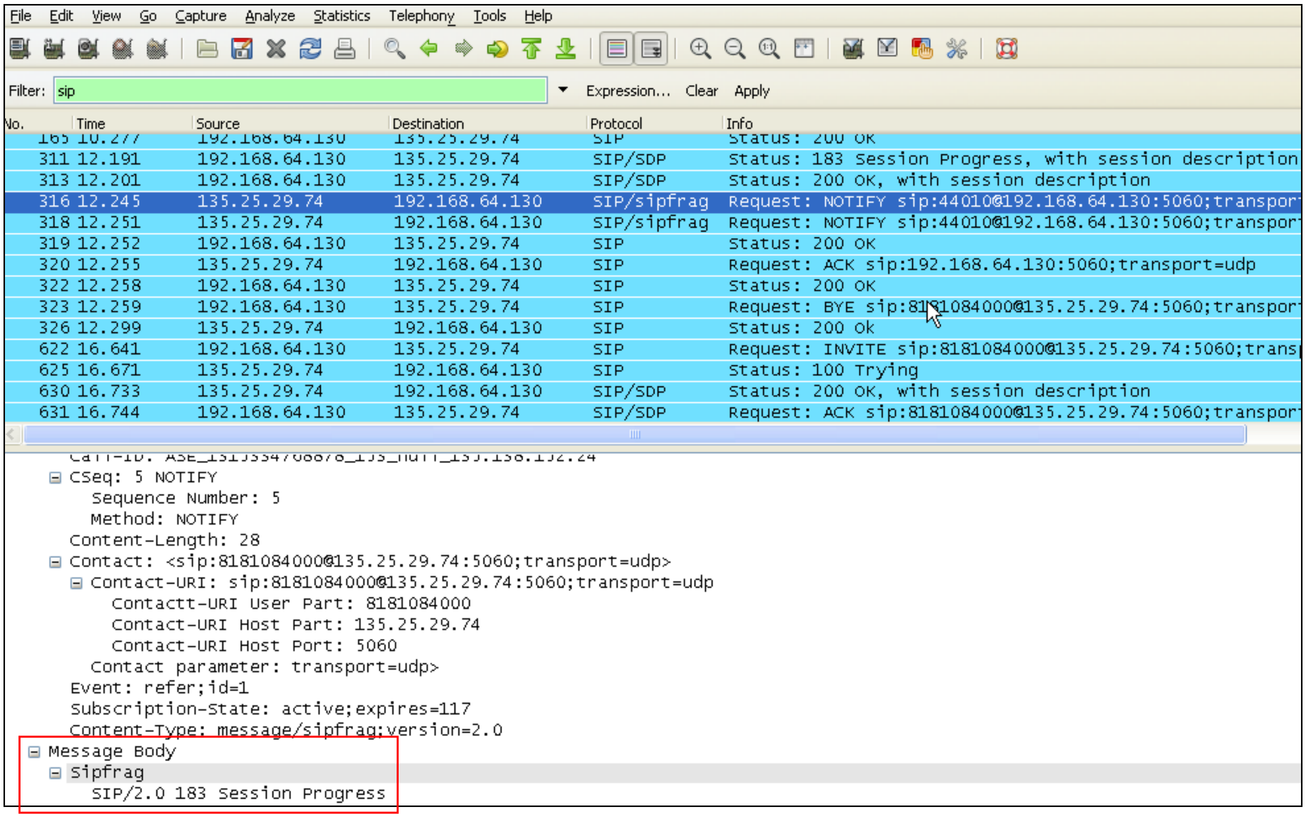


8.2.3. Refer with 183

The following is an example of a Refer (frame 134) redirection call filtering on the SIP protocol. Note that Communication Manager is sending 183 Session Progress. As described in **Section 2.2.1**, this causes Communication Manager to send a BYE (frame 323) upon receipt of the Notify/183 Session Progress sent by AT&T in frame 316.



This screen shows the Notify/183 Session Progress sent by AT&T in frame 316. Note that this Notify is in response to the 183 Session Progress sent in frame 311.



9. Conclusion

As illustrated in these Application Notes, Avaya Aura® Session Manager, Avaya Aura® Communication Manager, and the Acme Packet Net-Net Session Border Controller can be configured to interoperate successfully with the AT&T IP Transfer Connect service. In addition, these Application Notes further demonstrate that the Avaya Aura® Communication Manager SIP Network Call Redirection (NCR) and User-to-User Information (UUI) features can work in complement with the AT&T implementations of SIP NCR and UUI to support call redirection over SIP trunks while preserving initiating caller information. This solution provides contact center users of Avaya Aura® Communication Manager the ability to redirect inbound AT&T IP Transfer Connect service calls to alternate destinations, and deliver UUI-encoded customer information to those alternate destinations for the purposes of invoking contact center applications, e.g., triggering agent screen pop-ups with caller information, etc. Both intra-site and IP Transfer Connect call scenarios were tested.

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

10. References

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

[1] Applications Notes for Avaya Aura® Communication Manager 6.0.1, Avaya Aura® Session Manager 6.1 and Acme Packet Net-Net 6.2.0 with AT&T IP Toll Free SIP Trunk Service, Version 1.0
[2] Administering Avaya Aura™ Session Manager, Doc ID 03-603324, Issue 4, Feb 2011
[3] Installing and Configuring Avaya Aura™ Session Manager, Doc ID 03-603473 Issue 2, November 2010
[4] Maintaining and Troubleshooting Avaya Aura™ Session Manager, Doc ID 03-603325, Issue 3.1, March 2011
[5] Administering Avaya Aura™ System Manager, Document Number 03-603324, June 2010
[6] Administering Avaya Aura™ Communication Manager, Release 6.003-300509, Issue 6.0, June 2010Administering Avaya Aura® Call Center Features, Release 6.0, June 2010
[7] Programming Call Vectors in Avaya Aura® Call Center, 6.0, June 2010
[8] Modular Messaging Multi-Site Guide Release 5.1, June 2009
[9] Modular Messaging Messaging Application Server (MAS) Administration Guide, July 2011

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

AT&T IP Transfer Connect service support:

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

11. 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 6.2**). 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 following procedures to have the Acme Packet Net-Net 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.