Avaya Solution & Interoperability Test Lab

Application Notes for Avaya Aura® Communication Manager R6.2, Avaya Aura® Session Manager R6.3 and Acme Packet Net-Net 6.2.0 with AT&T IP Transfer Connect SIP Trunk Service – Issue 1.0

Abstract

These Application Notes describe the steps for configuring Avaya Aura® Communication Manager R6.2 with SIP Network Call Redirection (NCR), Avaya Aura® Session Manager R6.3, and the Acme Packet Net-Net 3800 with the AT&T IP Transfer Connect service using **AVPN** or **MIS/PNT** transport connections.

Avaya Aura® Session Manager R6.3 is a core SIP routing and integration engine that connects disparate SIP devices and applications within an enterprise. Avaya Aura® Communication Manager 6.2 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 3800, is the point of connection between Avaya Aura® Session Manager R6.3 and the AT&T IP Transfer Connect service, and is used to not only secure the SIP trunk, but also to make adjustments to the SIP signaling for interoperability.

AT&T IP Transfer Connect is a service option available with the AT&T IP Toll Free service, and supports the rerouting of inbound Transfer Connect calls to alternate destinations based upon SIP redirection messages from Avaya Aura® Communication Manager. In addition, the Avaya Aura® Communication Manager NCR and SIP User-to-User Information (UUI) features are utilized, in conjunction with the Data Forwarding option of the AT&T IP Transfer Connect service, to transmit UUI within SIP signaling messages to the alternate destinations.

Note that these Application Notes are intended to supplement the separate document: A***pplications Notes for Avaya Aura® Communication Manager R6.2, Avaya Aura® Session Manager R6.3 and Acme Packet Net-Net 6.2.0 with AT&T IP Toll Free SIP Trunk 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 at 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 3800[1] Session Border Controller (SBC) with the AT&T IP Transfer Connect service using **AVPN** or **MIS/PNT** transport connections.

Avaya Aura® Session Manager R6.3 is a core SIP routing and integration engine that connects disparate SIP devices and applications within an enterprise. Avaya Aura® Communication Manager R6.2 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 3800 is the point of connection between Avaya Aura® Session Manager R6.3 and the AT&T IP Transfer Connect service and is used to not only secure the SIP trunk, but also to make adjustments to the SIP signaling for interoperability.

AT&T IP Transfer Connect is a service option available with the AT&T IP Toll Free service, and supports the rerouting of inbound Transfer Connect calls to alternate[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. 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 R6.2, Avaya Aura® Session Manager R6.3 and Acme Packet Net-Net 6.2.0 with AT&T IP Toll Free SIP Trunk Service – Issue 1.0***.

[1] Although an Acme Net-Net 3800 was used in the reference configuration, the 4250 and 4500 platforms are also supported.
[2] 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. General Test Approach and Test Results

The test environment consisted of:

1. A simulated enterprise with System Manager, Session Manager, Communication Manager, Avaya phones, fax machines (Ventafax application), Acme Packet Net-Net, a.k.a. Acme Packet Session Border Controller (SBC), and Communication Manager Messaging .
2. A laboratory version of the AT&T IP Toll Free service, to which the simulated enterprise was connected via AVPN or MIS-PNT transport.

DevConnect Compliance Testing is conducted jointly by Avaya and DevConnect members. The jointly-defined test plan focuses on exercising APIs and/or standards-based interfaces pertinent to the interoperability of the tested products and their functionalities. DevConnect Compliance Testing is not intended to substitute full product performance or feature testing performed by DevConnect members, nor is it to be construed as an endorsement by Avaya of the suitability or completeness of a DevConnect member's solution.

2.1. Interoperability Compliance Testing

The interoperability compliance testing focused on verifying inbound call flows (see **Section 3.2** for sample call flows) between Session Manager, Communication Manager, Acme Packet SBC and the AT&T IP Transfer Connect 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:

- Inbound AT&T IP Transfer Connect service calls to Communication Manager telephones and Vector Directory Numbers (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 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 talkpath establishment between callers and Communication Manager agents/phones.

2.2. Test Results and Known Limitations

1. Communication Manager 6.2 issues a BYE upon receipt of a Notify/180 during Attended 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. 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 (**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 **Section 7** 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. Vector variables in Communication Manager can specify a maximum of 96 characters for UUI. As a result some of the boundary conditions for AT&T IP Transfer Connect service cannot be tested.

The test objectives stated in **Section 2.1** with limitations as noted in this section were verified.

2.3. Support

AT&T customers may obtain support for the AT&T IP Toll Free service by calling (888) 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:

- Session Manager provides core SIP routing and integration services that enables communication 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 provides the voice communication services for a particular enterprise site. In the reference configuration, Communication Manager 6.2 runs on an Avaya S8800 Server in a G650/Control LAN (C-LAN) configuration. This solution is extensible to other Avaya Servers.
- The Avaya Media Gateway provides the physical interfaces and resources for Communication Manager. In the reference configuration, an Avaya G650 Media Gateway is used. The G650 contains circuit packs such as the Control LAN (C-LAN) and Media Processor (MedPro). This solution is extensible to other Avaya Media Gateways.
- Avaya "desk" telephones are represented with Avaya 96x0 and 96x1 Series IP Telephones running H.323 and SIP, Avaya 6408D Series Digital Telephone, Avaya Analog phone 2500MMG and Avaya one-X® Communicator (SIP/H323), a PC based softphone.
- The Acme Packet SBC 3800[3] provides SIP Session Border Controller functionality, including address translation and SIP header manipulation between the AT&T IP Toll Free service and the enterprise internal network[4]. UDP transport protocol is used between the Acme Packet SBC and the AT&T Toll Free service.
- Communication Manager Messaging system provides the corporate voice messaging capabilities in the reference configuration. **The provisioning of Communication Manager Messaging is beyond the scope of this document**.
- Inbound calls were placed from PSTN to the Customer Premises Equipment (CPE) via the AT&T IP Transfer Connect service, through the Acme Packet SBC, Session Manager, and Communication Manager. Communication Manager terminated the calls using appropriate phone or fax stations. The H.323 phones at the CPE are registered to the Communication Manager C-LANs and the SIP phones are registered to Session Manager.

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

[4] The AT&T IP Toll Free service uses SIP over UDP to communicate with enterprise edge SIP devices (Acme Packet SBC) in this sample configuration. Session Manager may use SIP over UDP, TCP, or TLS to communicate with SIP network elements. In the reference configuration, Session Manager uses SIP over TCP to communicate with the Acme Packet SBC and Communication Manager.

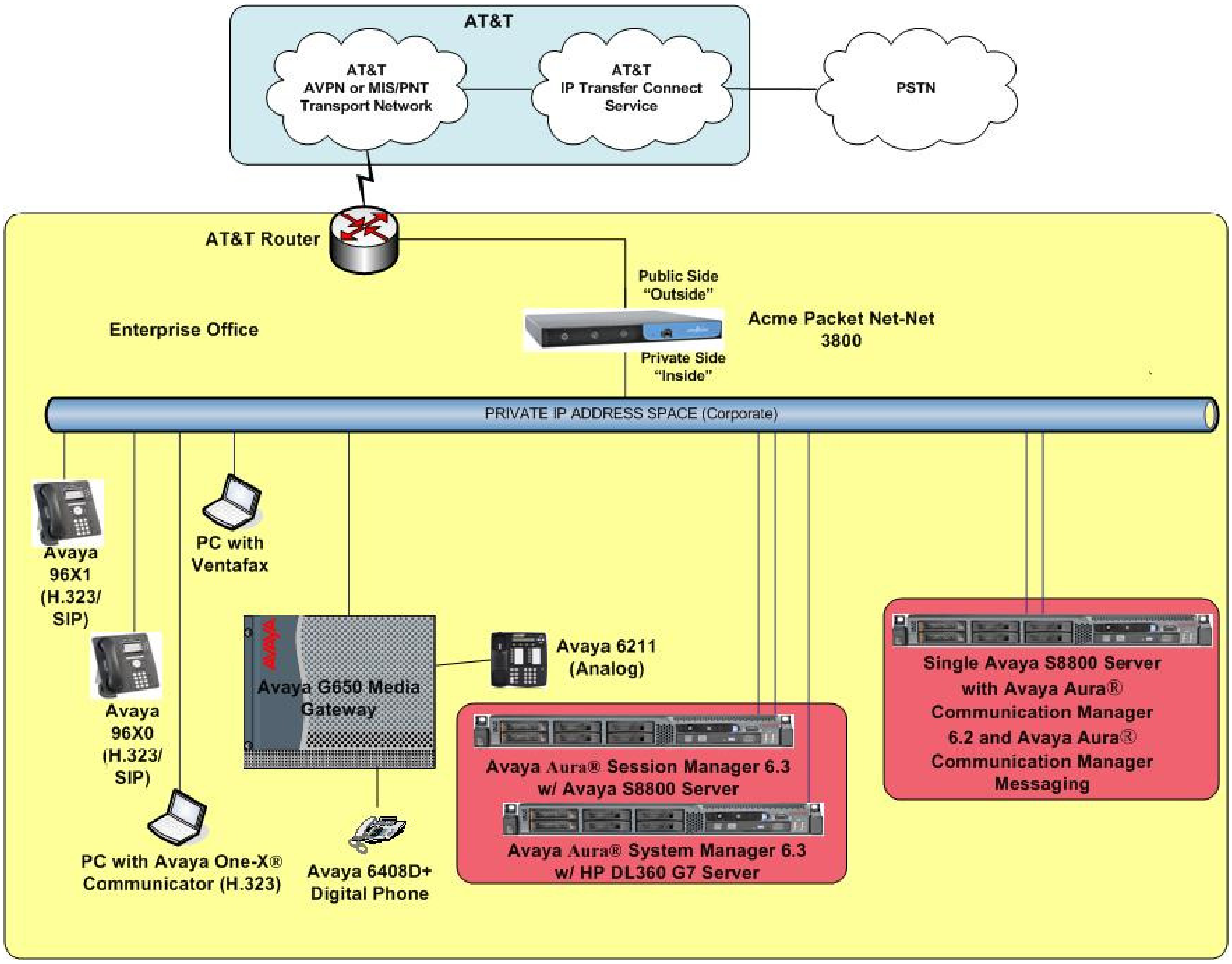
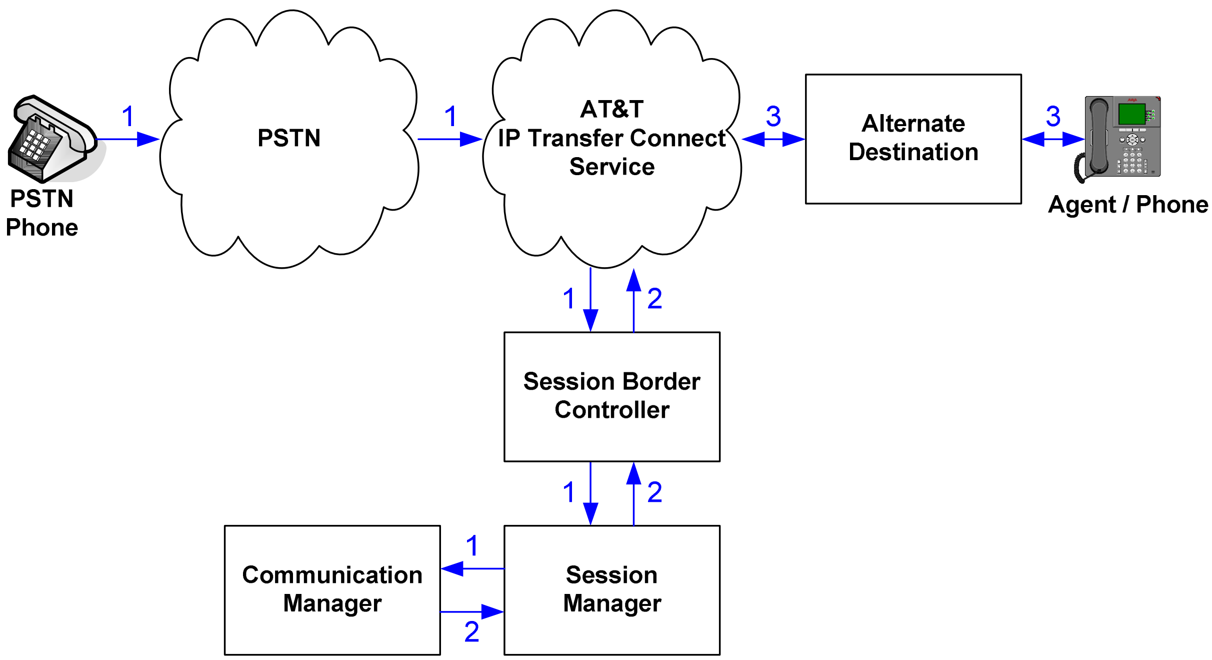


Figure 1: Reference Configuration

3.2.2. Inbound Call Redirection to Alternate Destination

The second call scenario illustrated in the figure below is an inbound AT&T IP Transfer Connect service call that arrives at 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 SBC 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).

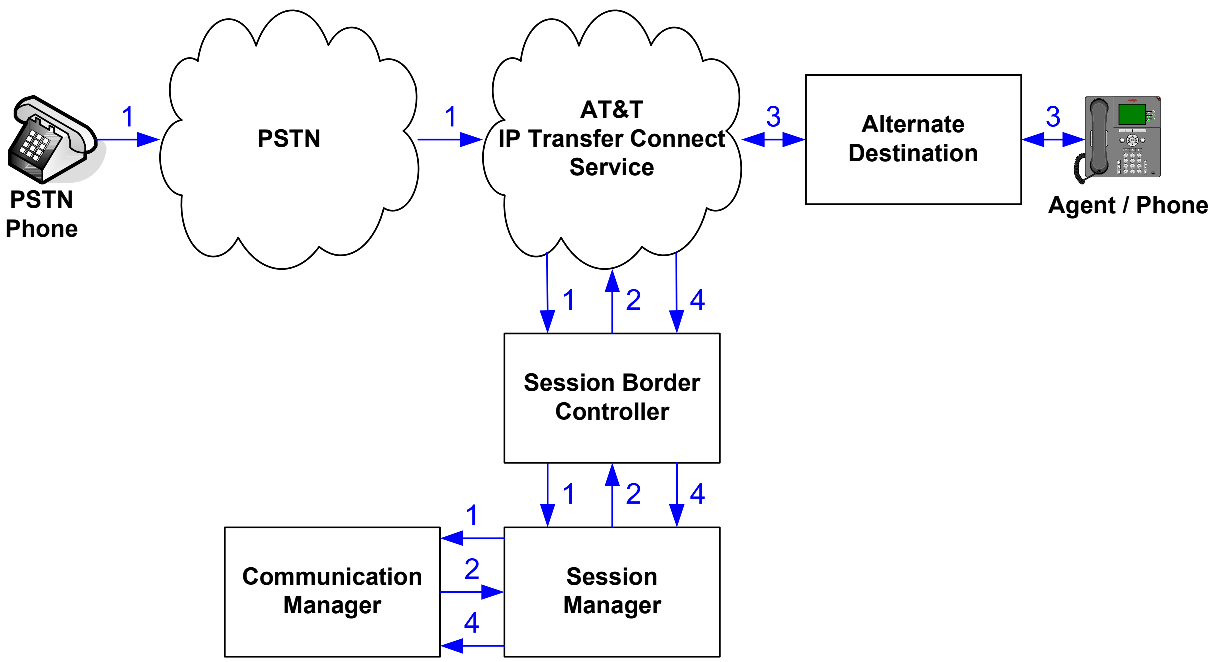


Inbound AT&T IP Transfer Connect Call – Pre-Answer SIP 302 Redirection Messaging

3.2.3. Inbound Call Referred to Alternate Destination

The third call scenario illustrated in the figure below is an inbound AT&T IP Transfer Connect service call that arrives at 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 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 (Communication Manager).



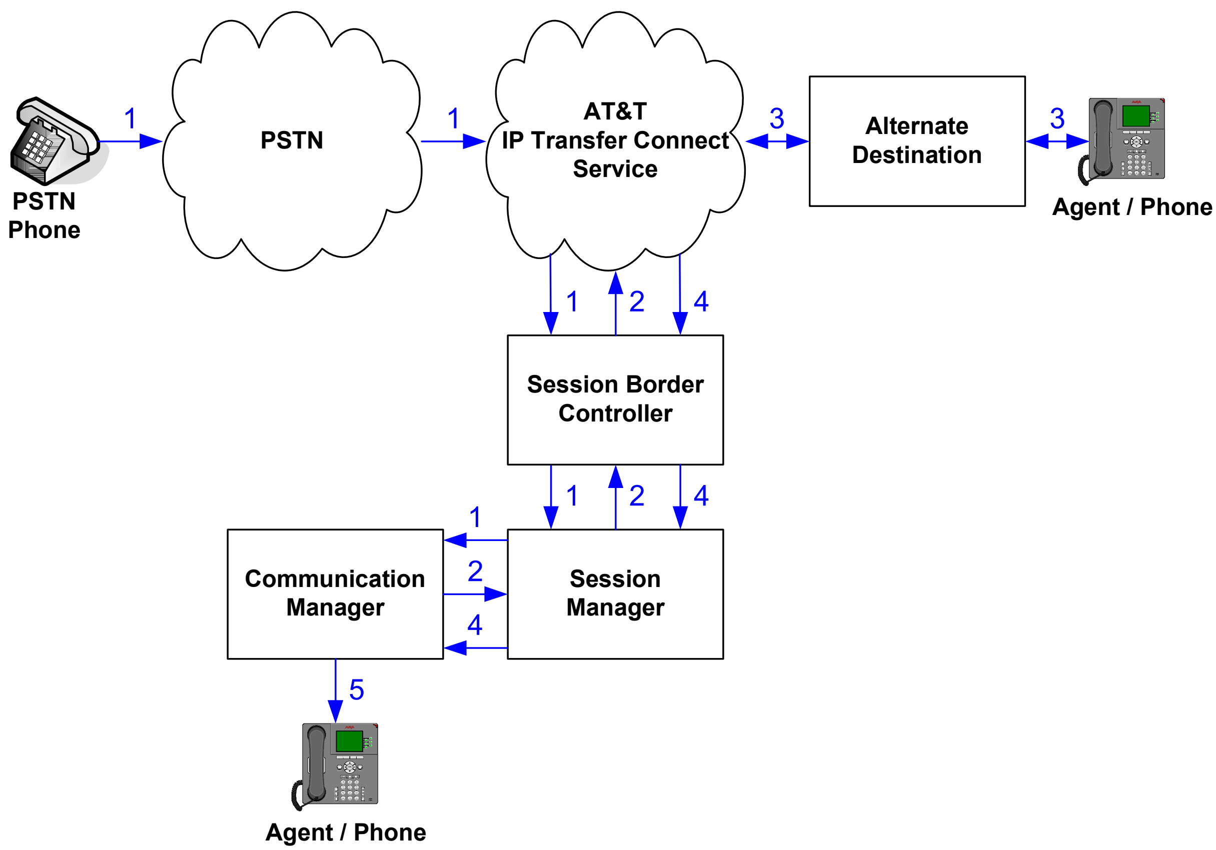
Inbound AT&T IP Transfer Connect Call – Post-Answer SIP REFER Redirection Successful

3.2.4. Unsuccessful Inbound Refer Call to Alternate Destination

The fourth call scenario illustrated in the figure below is an inbound AT&T IP Transfer Connect service call that arrives at 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. The fourth call scenario illustrated in the figure below 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, 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.



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.

Equipment/Software	Release/Version
HP DL360 G7 Server	Avaya Aura® System Manager 6.3 (6.3.0.8.923) System Platform 6.2.2.06002.0
Avaya S8800 Server	Avaya Aura® Session Manager 6.3 SP2 (6.3.0.0.630039)
Avaya S8800 Server	Avaya Aura® Communication Manager R6.2 SP5 (02.0.823.0-20396) System Platform 6.2.2.08001.0
Avaya G650 Media Gateway	
TN2312BP IP Server Interface (IPSI)	HW06 FW057
TN799DP Control-LAN (C-LAN)	HW01 FW041
TN2602AP IP Media Resource 320 (MedPro)	HW02 FW063
TN2501AP VAL-ANNOUNCEMENT	HW03 FW018
TN2224CP Digital Line	HW08 FW015
TN793B Analog Line	HW05 FW011
Avaya 9650 IP Telephone	H.323 R3.1.5
Avaya 9641G IP Telephone	H.323 R6.2.3.12
Avaya 9608 IP Telephone	SIP R6.2.1
Avaya one-X® Communicator (H323)	6.1.7.04-SP7-39506
Avaya Digital Telephone 6408D+	
Avaya Analog phone 2500MMG	-
Fax device	Ventafax Home Version 6.1.59.144
Acme Packet Net-Net 3800	SCX6.2.0 MR-6 Patch 5 (Build 916)
AT&T IP Toll Free service using AVPN/MIS-PNT transport service connection	VNI 26

Table 2: Equipment and Software Versions

5. Configure Avaya Aura® Session Manager Release 6.3

The Session Manager administration for interoperability with AT&T IP Transfer Connect service is similar to the configuration done in [**7**] for AT&T IP Toll Free service. Refer to the appropriate section in [**7**] to configure the Session Manager for AT&T IP Transfer Connect service. Additional dial patterns and modification to the existing Routing Policy may be required for proper routing of the calls from AT&T Transfer Connect service.

6. Configure Avaya Aura® Communication Manager Release 6.2

This section describes the administration steps for Communication Manager in support of the reference configuration described in these Application Notes. The steps are similar to the AT&T IP Toll Free service configuration in [**7**]. This section describes additional administration step on Communication Manager to support interoperability with AT&T IP Transfer Connect service features as listed in **Section 2.1**. The steps are performed from the Communication Manager System Access Terminal (SAT) interface. For any values not configured, defaults are used in this reference configuration. These Application Notes assume that basic Communication Manager administration has already been performed. Consult [**5**] and [**6**] for further details if necessary.

Note – In the following sections, only the parameters that are highlighted in **bold** text are specifically applicable to this Application Note. Other parameter values may or may not match based on local configurations.

6.1. System Parameters

This section reviews the Communication Manager licenses and features that are required for the reference configuration described in these Application Notes. 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 field** 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
                Flexible Billing? n
   Forced Entry of Account Codes? n                   Multifrequency Signaling? y
      Global Call Classification? n           Multimedia Call Handling (Basic)? y
             Hospitality (Basic)? y      Multimedia Call Handling (Enhanced)? y
 Hospitality (G3V3 Enhancements)? n                 Multimedia IP SIP Trunking? n
                       IP Trunks? y
           IP Attendant Consoles? n
```

System-Parameters Customer-Options Form – Page 4

1. On **Page 6** of the **System-Parameters Customer-Options** form, verify that the vectoring features shown below 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
```

System-Parameters Customer-Options Form – Page 6

6.2. SIP Trunks

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

1. Enter the **change trunk-group 2** command, where **2** is the trunk group configured in [**7**]. On **Page 4** of the **trunk-group** form, set the **Network Call Redirection** field to "**y**" (see **Item 2** in **Section 2.2**)

```
add trunk-group 2                                              Page   4 of  21
                              PROTOCOL VARIATIONS
                           Mark Users as Phone? n
                  Prepend '+' to Calling Number? n
            Send Transferring Party Information? n
                      Network Call Redirection? y
                         Send Diversion Header? n
                       Support Request History? n
                  Telephone Event Payload Type: 100

             Convert 180 to 183 For Early Media? n
      Always Use re-INIVIT for Display Updates? n
            Identity for Calling Party Display? P-Asserted-Identity
Block Sending Calling Party Location in INVITE? n
```

Trunk Group form for IP Transfer Connect service calls – Page 4

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 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 [**7**].

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 simply necessary 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 (AE Services) 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 [**5**] and [**6**] for further information.

6.3.1. Pre-Answer Redirection

This section provides an example of Pre-Answer Redirection. In this example, first the vector variables are defined for the vectors (**Figure 1**) and an inbound call from the AT&T IP Transfer Connect service is routed to a VDN shown in **Figure 2**, which invokes the vector shown in **Figure 3**. 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**). This UUI information will be returned to the AT&T IP Transfer Connect service in the SIP 302 message.
  Note: The parameters for ASAI UUI variables "**A**" and "**B**", and other vector variables are defined using the **change variables** command (see **Figure 1**).
- Redirects the call to the number "**1092**" (vector step **08**). Note that since this vector did not answer the call, the presence of the "**~r**" in the "**route-to number**" instructs Communication Manager to send a SIP 302 message with the number "**1092**" in the user part of the Contact header URI (e.g., 1092@<host/domain>) to the AT&T IP Transfer Connect service. The host/domain is populated with the Far-end Domain value administered on the signaling group defined in [**7**] on which the inbound call arrives.

```
change variables                                                 Page   1 of  39
                              VARIABLES FOR VECTORS

Var Description                     Type    Scope Length Start Assignment     VAC
A   UuiTest1                        asaiuui L     16     1
B   UuiTest2                        asaiuui L     16     17
C
```

Figure 1: Change Variables Form

```
display vdn 2016                                                 Page   1 of   3
                            VECTOR DIRECTORY NUMBER

                             Extension: 2016
                                 Name*: NCR Ringback 302 UUI
                           Destination: Vector Number        16
                   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 2: Sample VDN for Pre-Answer Redirection

```
display vector 16                                                  Page   1 of   6
                                  CALL VECTOR

    Number: 16               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 ~r1092           with cov n if unconditionally
09
10 #    Play this announcement only on redirect failure
11 announcement 33008
12
```

Figure 3: 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 AT&T IP Transfer Connect service call is routed to the VDN shown in **Figure 4**, which invokes the vector shown in **Figure 5**. 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**). This UUI information will be returned to the AT&T IP Transfer Connect service in the SIP REFER message.
  Note: The parameters for UUI variable "**A**" and other vector variables are defined using the **change variables** command (see **Figure 1**).
- Answers the call to play an announcement (vector step **08**).
- Attempts to redirect the call to the number "**1092**" (vector step **09**). Note that since this vector answered the call, the presence of the "**~r**" in the "**route-to number**" instructs Communication Manager to send a SIP REFER message with the number "**1092**" in the user part of the Refer-To header URI (e.g., 1092@<host/domain>) to the AT&T IP Transfer Connect service. The host/domain is populated with the Far-end Domain value administered on the signaling group defined in [**7**] on which the inbound call arrived.

```
display vdn 2018                                              Page   1 of   3
                            VECTOR DIRECTORY NUMBER

                             Extension: 2018
                                 Name*: NCR Ringback REFER UUI
                           Destination: Vector Number        18
                   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 4: Sample VDN for Post-Answer Redirection

```
display vector 18                                              Page   1 of   6
                                  CALL VECTOR

    Number: 18               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 33007
09 route-to     number ~r1092           with cov n if unconditionally
10 #    Play this announcement only on redirect failure
11 disconnect   after announcement 33008
12
```

Figure 5: Sample Vector for Post-Answer Redirection

6.4. Saving Translations

To save all Communication Manager provisioning changes, enter the command **save translations.**

7. Configure Acme Packet Session Border Controller

These Application Notes assume that basic Acme Packet SBC administration has already been performed. The Acme Packet SBC configuration used in the reference configuration is provided below as a reference. The notable settings are highlighted in bold and brief annotations are provided on the pertinent settings. Use **putty** or similar tool to access Acme Packet SBC for configuration. Consult with Acme Packet Support [**8**] for further details and explanations on the configuration below.

ANNOTATION: The local policies below govern the routing of SIP messages from elements on the network on which the Avaya elements, e.g., Session Manager, Communication Manager, etc., reside to the AT&T IP services. The Session Agent Groups (**SAG**) defined here, and further down, is provisioned under the session-groups **SP-PROXY** and **ENTERPRISE.** Note: Although **Enterprise** policy is not used for AT&T IP Transfer Connect service it is left in this document in case the customer is using AT&T IP Flexible Reach service.

The Acme Packet SBC configuration for interaction with the AT&T IP Toll Free service is provided in [**7**]. The additional configuration on the Acme Packet SBC necessary for supporting interaction with the AT&T IP Transfer Connect service is provided below as a reference.

1. The AT&T IP Transfer Connect service requires that SIP Refer and 302 call redirection messages contain the AT&T Border Element IP address in the Refer-to header of a Refer call and the Contact header of a 302 call. The following section describes the Acme Packet header-rule provisioning. The header-rules described below were added to the existing sip-manipulation ***NAT_IP*** described in [**7**].

   - **Modify Refer-to header**

```
header-rule
      name                          modReferTo
      header-name                   Refer-To
      action                        manipulate
      comparison-type               case-sensitive
      msg-type                      any
      methods                       REFER
      match-value
      new-value
      element-rule
      name                          modmline
      parameter-name
      type                          uri-host
      action                        find-replace-all
      match-val-type                any
      comparison-type               case-sensitive
      match-value                   customerb.com
      new-value                     $REMOTE_IP
```

   - **Modify 302 Contact header**

```
header-rule
      name                          mod302
      header-name                   Contact
      action                        manipulate
```

```
comparison-type                 case-sensitive
msg-type                        Reply
methods                         INVITE
match-value
new-value
element-rule
name                            modmline
parameter-name
type                            uri-host
action                          find-replace-all
match-val-type                  any
comparison-type                 case-sensitive
match-value                     customerb.com
new-value                       $REMOTE_IP
```

- **Modify Sendonly** (See **Item 2**, **Section 2.2** for further details)

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**

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
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
```

8. Verification Steps

The following steps may be used to verify this reference configuration:

8.1. AT&T IP Transfer Connect

1. Place an inbound call, answer the call, and verify that two-way talk path exists. Verify that the call remains stable for several minutes and disconnects properly.
2. Verify basic call functions such as hold, transfer and conference.
3. Verify the use of DTMF signaling.

8.2. Avaya Aura® Communication Manager

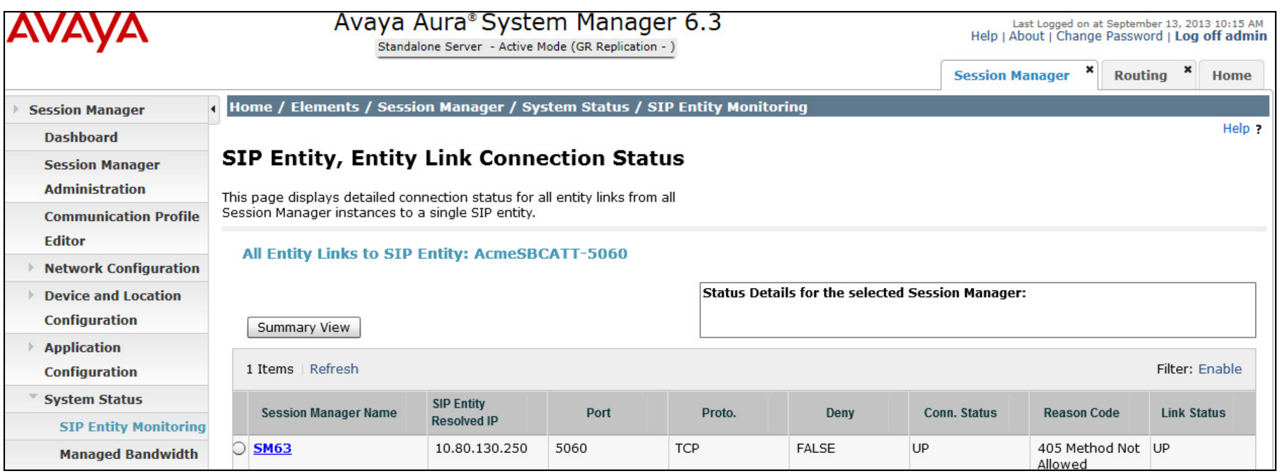
The following examples are only a few of the monitoring commands available on Communication Manager. See [**5**] and [**6**] for more information.

- From the Communication Manager SAT console connection enter the command ***list trace tac xxx***, where ***xxx*** is a trunk access code to verify that the inbound or outbound calls are using the right trunk groups. Similarly, ***list trace station***, ***list trace vdn***, and ***list trace vector***, ***status trunk*** and ***status station*** commands can be used on Communication Manager.

8.3. Avaya Aura® Session Manager

Navigate to **Home→ Elements→ Session Manager→ System Status → SIP Entity Monitoring** and click on the SIP Entity for which the status is required. Following screen shows status for the entity link between Session Manager and Acme Packet SBC.

Note: The Reason Code column indicates that Session Manager has received a **SIP 405 Method Not Allowed** response (normal for this reference configuration) to the **SIP OPTIONS** message that was generated. This response is sufficient for SIP Link Monitoring to consider the link up.



9. Conclusion

As illustrated in these Application Notes, Avaya Aura® Session Manager, Avaya Aura® Communication Manager Network Call Redirection, and Acme Packet 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 conjunction with the AT&T implementations of SIP NCR and UUI to support call redirection over SIP trunks while preserving initiating caller information. This solution provides contact center users of Avaya Aura® Communication Manager the ability to redirect inbound AT&T IP Transfer Connect service calls to alternate destinations, and deliver UUI-encoded customer information to those alternate destinations for the purposes of invoking contact center applications, e.g., triggering agent screen pop-ups with caller information, etc.

The reference 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://www.avaya.com/support unless otherwise noted.

Avaya Aura® Session Manager/System Manager

[1] Administering Avaya Aura® Session Manager, Doc ID 03-603324, Release 6.3, December 2012

[2] Installing and Configuring Avaya Aura® Session Manager, Doc ID 03-603473 Issue 2, November 2010

[3] Maintaining and Troubleshooting Avaya Aura® Session Manager, Doc ID 03-603325, Release 6.3, December 2012

[4] Administering Avaya Aura® System Manager, Release 6.3, Issue 1.0, December 2012

Avaya Aura® Communication Manager

[5] Administering Avaya Aura® Communication Manager, Issue 7.0, Release 6.2, December 2012, Document Number 03-300509

[6] Avaya Aura® Call Center 5.2 Call Vectoring and Expert Agent Selection (EAS) Reference, Release 5.2, April 2009, Document Number 07-600780

Additional References

[7] Applications Notes for Avaya Aura® Communication Manager R6.2, Avaya Aura® Session Manager R6.3 and Acme Packet Net-Net 6.2.0 with AT&T IP Toll Free SIP Trunk service – Issue 1.0

Acme Packet Support (login required):

[8] http://www.acmepacket.com/support.htm

AT&T IP Toll Free Service Descriptions:

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

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.