Avaya Solution & Interoperability Test Lab

Applications Notes for Avaya Aura™ Communication Manager 6.0, Avaya Aura™ Session Manager 6.0 and Avaya Aura™ Session Border Controller with AT&T IP Transfer Connect SIP Trunk Service – Issue 1.1

Abstract

These Application Notes describe the steps for configuring Avaya Aura™ Session Manager, Avaya Aura™ Communication Manager with SIP Network Call Redirection, and the Avaya Aura™ Session Border Controller with the AT&T IP Transfer Connect service using **MIS/PNT** transport connection.

The AT&T IP Transfer Connect service is a service option available with the AT&T IP Transfer Connect 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, Avaya Aura™ Communication Manager Network Call Redirection and SIP User-to-User Information features can be utilized together, in conjunction with the Data Forwarding option of the AT&T IP Transfer Connect service, to transmit User-to-User Information within SIP signaling messages to the alternate destinations. Avaya Aura™ Session Manager 6.0 is a core SIP routing and integration engine that connects disparate SIP devices and applications within an enterprise.

Note: These Application Notes are intended to supplement separate Application Notes covering Avaya Aura™ Session Manager 6.0, Avaya Aura™ Communication Manager 6.0, using Avaya Aura™ Session Border Controller interoperability with the AT&T IP Toll Free service [11]

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™ Session Manager, Avaya Aura™ Communication Manager with SIP Network Call Redirection (NCR), and the Avaya Aura™ Session Border Controller (AA-SBC) with the AT&T IP Transfer Connect service using **MIS/PNT** transport connections[1].

The AT&T IP Transfer Connect service is a service option available with the AT&T IP Transfer Connect 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. 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 toll free 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: These Application Notes are intended to supplement separate Application Notes covering Avaya Aura™ Session Manager 6.0, Avaya Aura™ Communication Manager 6.0, using Avaya Aura™ Session Border Controller interoperability with the AT&T IP Toll Free service [11].

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[1] MIS/PNT does not support compressed RTP (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.

1.1. Interoperability Compliance Testing

The interoperability compliance testing focused on verifying inbound call flows (see **Section 2.2**) from AT&T network to Session Manager and Communication Manager with subsequent redirection to an alternate destination via AT&T IP Transfer Connect service.

The compliance testing was based on a test plan provided by AT&T, for 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
- SIP Redirect
- SIP Refer (with Attended and Unattended Transfers)

1.2. Support

AT&T customers may obtain support for the AT&T IP Transfer Connect 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.

1.3. Known Limitations

1. Communication Manager 6.0 issues a BYE upon receipt of a Notify/200OK during attended Refer calls (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 due to this behavior.
2. Shuffling must be disabled on the Communication Manager "local" SIP trunk due to codec negotiation issues with Avaya SIP telephones (see [12] for more information).
3. Communication Manager 6.0 inserts a leading plus sign to calling number headers 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 headers containing calling numbers. To avoid the introduction of this ("+") sign as a leading digit, SIP trunks were configured with "private" numbering format [11].
4. Communication Manager Network Call Redirection (NCR) feature is required to enable Refer and 302 call redirection with the AT&T IP Transfer Connect Service (see **Section 5.1**). 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 separate trunks for supporting these features.
5. In case of Multiple Refers, Communication Manager sends a BYE and AT&T IP Transfer Connect service responds with a 200 OK. Session Border Controller tears down the call after receiving the 200 OK in response to the BYE sent by Communication Manager. A response to Notify (Ringing) is lost in this transaction. AT&T repeats the Notify (ringing) but the AA-SBC responds with a 404 not found as the call does not exists anymore in the AA-SBC. The call works fine in spite of this signaling issue. This issue is still being investigated.
6. Communication Manager supports a maximum of 96 Octets for UUI information transmission.

2. Reference Configuration

The reference configuration used in these Application Notes is shown in the figure below and consists of several components:

- Session Manager provides core SIP routing and integration services that enables communications between disparate SIP-enabled entities, e.g., PBXs, SIP proxies, gateways, adjuncts, trunks, applications, etc. across the enterprise. Session Manager allows enterprises to implement centralized and policy-based routing, centralized yet flexible dial plans, consolidated trunking, and centralized access to adjuncts and applications.
- System Manager provides a common administration interface for centralized management of all Session Manager instances in an enterprise.
- Communication Manager provides the voice communications services for a particular enterprise site. In this reference configuration, Communication Manager runs on an Avaya S8800 Server. This solution is extensible to other Avaya S8xxx Servers.
- The Avaya Media Gateway provides the physical interfaces and resources for Communication Manager. In the reference configuration, an Avaya G650 Media Gateway is used. This solution is extensible to other Avaya Media Gateways.
- Avaya “desk” phones are represented with Avaya 4600 and 9600 Series IP Telephones running H.323 software, 9600 Series IP Telephones running SIP software, Avaya 6211 series Analog Telephones, and Avaya one-X® Agent, a PC based Softphone.
- AA-SBC provides SIP header manipulation between the AT&T IP Transfer Connect service and the enterprise internal network. UDP transport protocol is used between the AA-SBC 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 and its provisioning is beyond the scope of this document.
- Inbound calls from PSTN were sent from AT&T IP Transfer Connect service, through the AA-SBC to the Session Manager which routed the call to Communication Manager. Communication Manager terminated the call to the appropriate agent/phone or fax extension. The H.323 phones on the enterprise side registered to the Communication Manager C-LAN. The SIP phones on the enterprise side registered to the Session Manager.

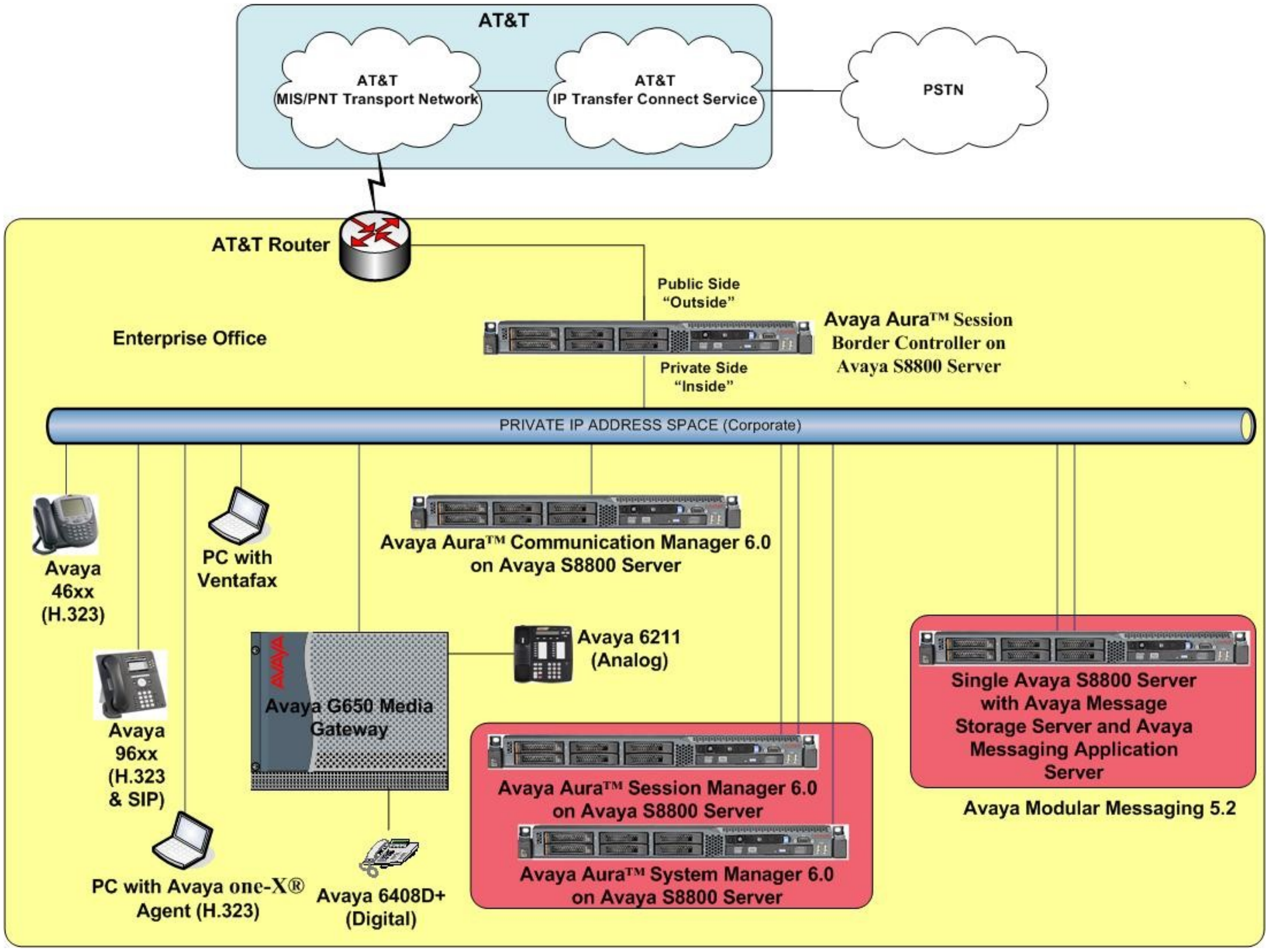

Figure 1: Reference configuration

2.1. Illustrative Configuration Information

The specific values listed in the table 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 Redirect Routing Number (RRN) digits (Destination digits specified in the SIP Request URIs sent by AT&T Transfer Connect service), shown in this document are examples. AT&T Customer Care will provide the actual IP addresses and RRN digits as part of the IP Transfer Connect provisioning process.

Component	Illustrative Value in these Application Notes
Avaya Aura™ System Manager	
Management IP Address	10.80.130.21
Avaya Aura™ Session Manager	
Management IP Address	10.80.120.27
Network IP Address	10.80.120.28
Avaya Aura™ Communication Manager	
CLAN IP Address	10.80.111.31
VDN	6665315-6665327
Skill/Hunt Group	11, 12, 13
Agent Extensions	6665611-6665615
Hunt Group Extensions	6665611(11), 6665612(12), 6665613(13)
Phone Extensions	66650xx (H323), 66654xx (SIP) 66651xx(Analog), 66652xx(Digital)
Avaya Modular Messaging	
Messaging Application Server (MAS) IP Address	10.80.100.30
Messaging Server (MSS) IP Address	10.80.100.29
Avaya Aura™ Session Border Controller	
IP Address of "Outside" (Public) Interface (connected to AT&T Border Element)	192.168.62.55 (active)
IP Address of "Inside" (Private) Interface (connected to Avaya Aura™ Session Manager)	10.80.130.12 (active)
AT&T IP Transfer Service	
Border Element IP Address	135.242.225.210
Digits Passed in SIP Request-URI (RRN)	000001089 (SIP Redirect) 000001090 (SIP Refer Attended) 000001091 (SIP Refer Unattended) 000001092, 000001093 (Speed Dial Codes used for Redirection/Refer)

Table 1: Illustrative Values Used in these Application Notes

2.2. Call Flows

To understand how inbound AT&T IP Transfer Connect service calls are handled by Session Manager and Communication Manager, following general call flows are described in this section.

The first call scenario illustrated in the figure below 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.

1. A PSTN phone originates a call to an AT&T IP Transfer Connect service number.
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 AA-SBC.
4. The AA-SBC performs 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 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 vector, which in turn, routes the call to an agent
   - Directly to an agent or a phone/fax extension.

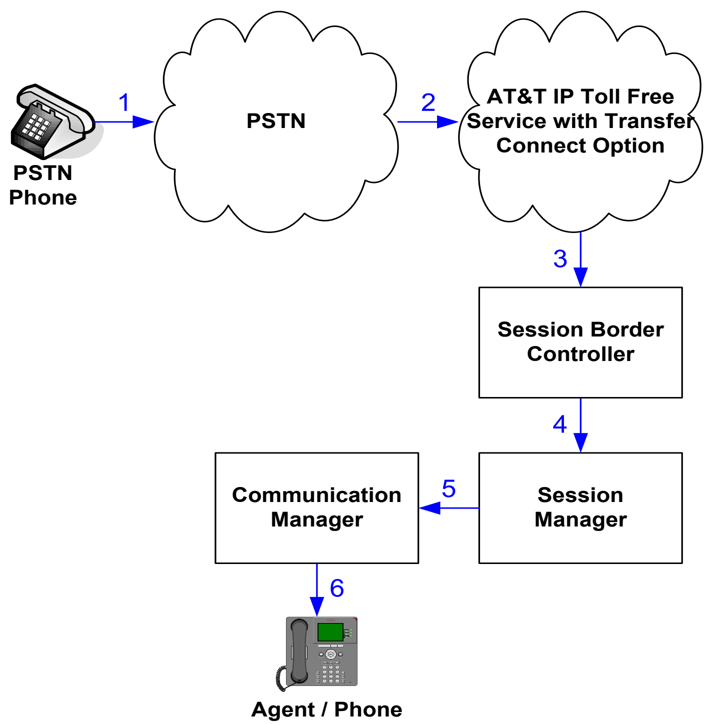

Figure 2: Inbound AT&T IP Transfer Connect Call to VDN/Agent/Phone

Note: **In the call scenarios that follows, 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 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 AA-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).

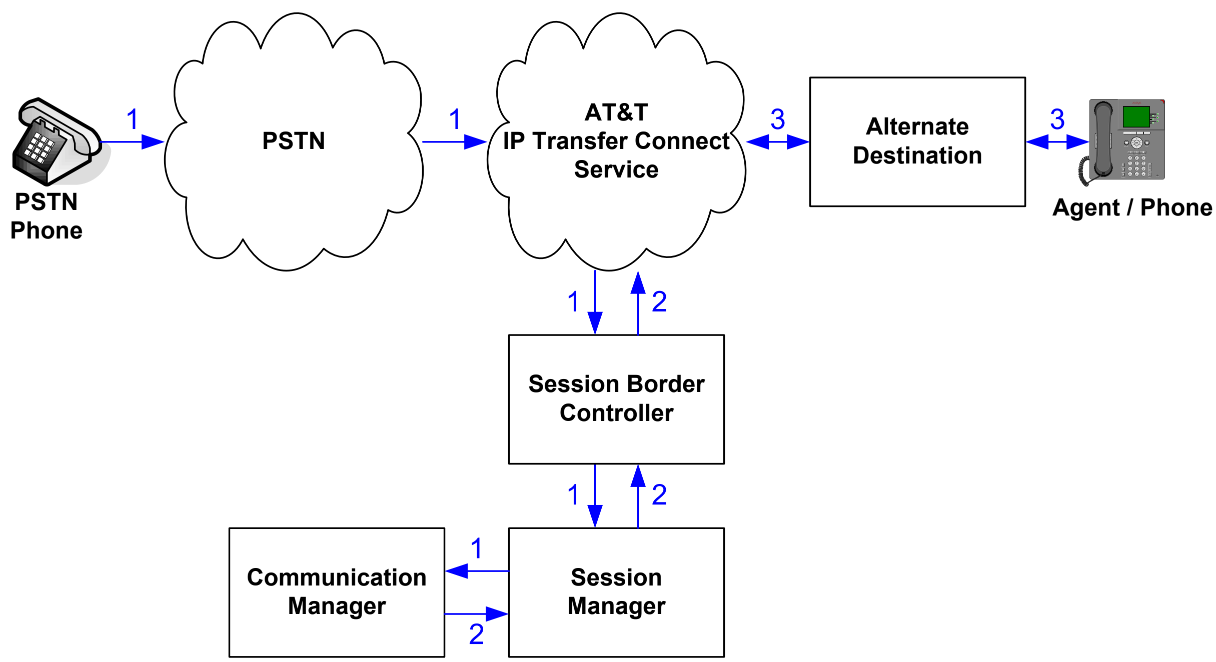

Figure 3: Inbound AT&T IP Transfer Connect Call – Pre-Answer SIP 302 Redirection Messaging

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 AA-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).

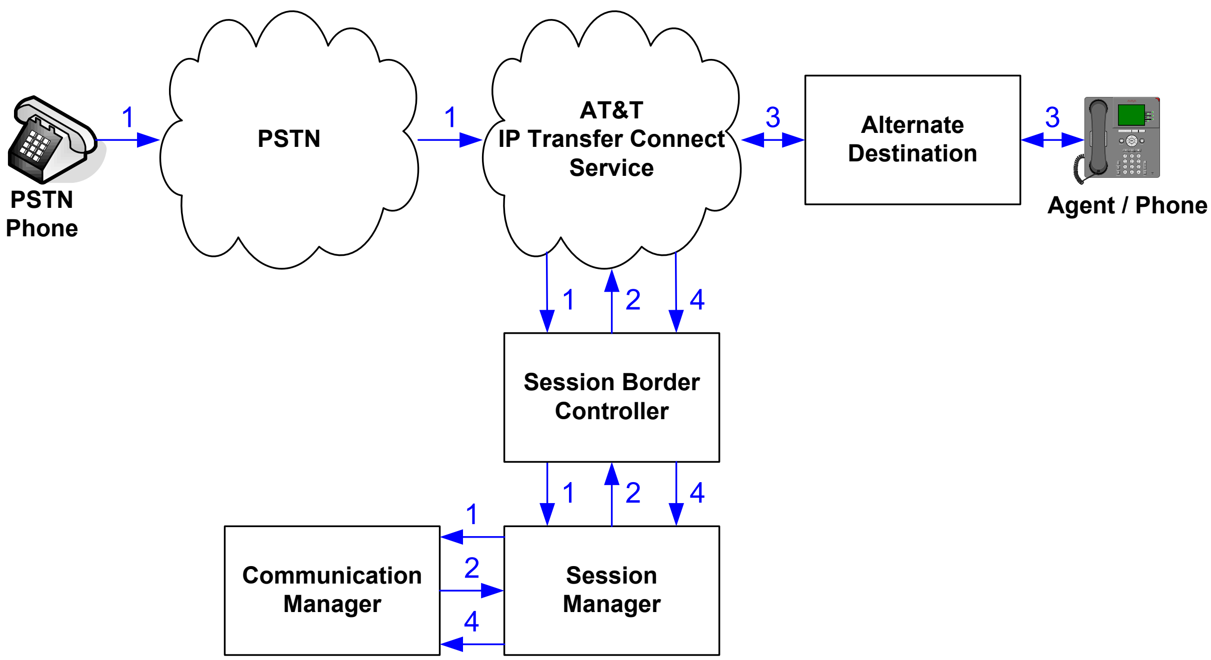

Figure 4: Inbound AT&T IP Transfer Connect Call – Post-Answer SIP REFER Redirection Successful

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.

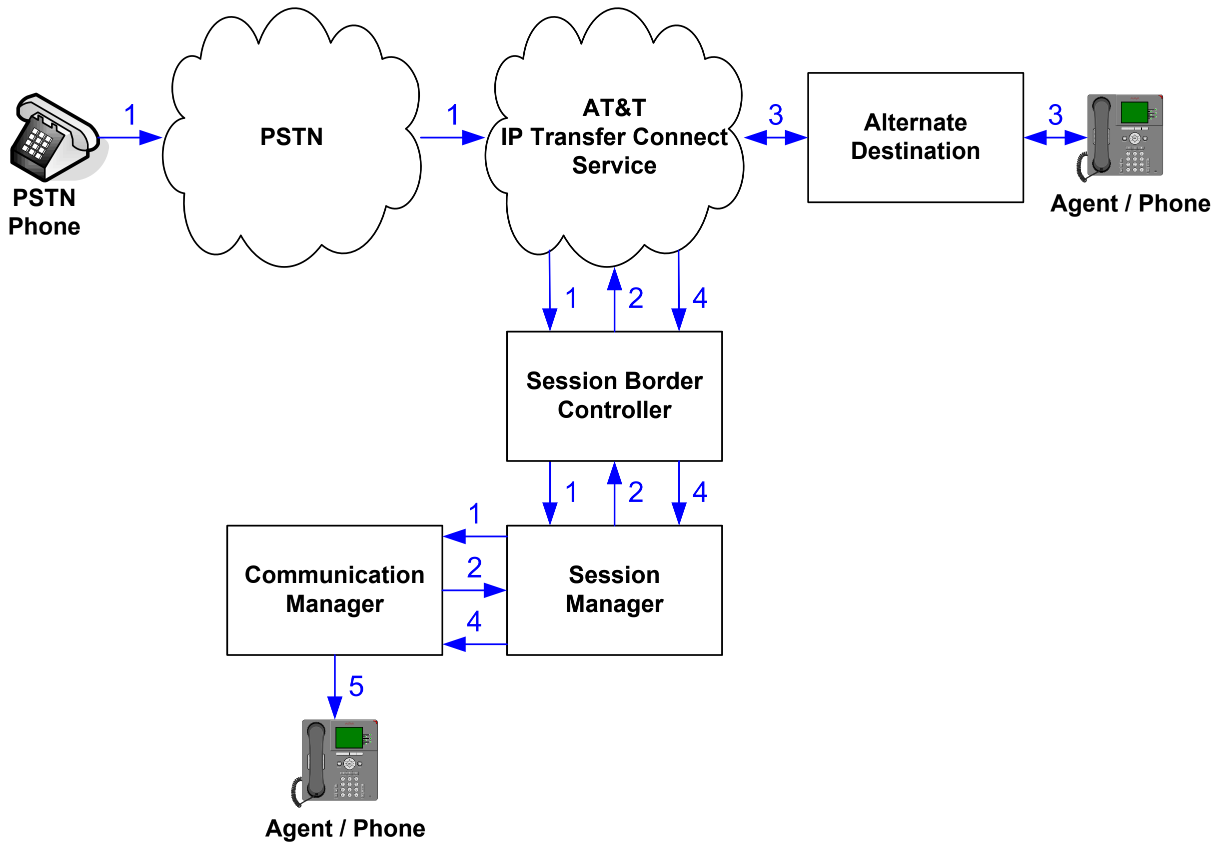

Figure 5: Inbound AT&T IP Transfer Connect Call – Post-Answer SIP REFER Redirection Unsuccessful

3. Equipment and Software Validated

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

Component	Version
Avaya S8800 Server	Avaya Aura™ System Manager 6.0 (6.0.0.0.556-3.0.6.1)
Avaya S8800 Server	Avaya Aura™ Session Manager 6.0 (6.0.0.0.600020)
Avaya S8800 Server	Avaya Aura™ Communication Manager 6.0 (R016x.00.0.345.0) with SP1
Avaya S8800 Server	Avaya Aura™ Session Border Controller 6.0 (R6.0.0.3.4), Product Version 36M2, Build Version 3.6.0, Build 46752 on VSP-6.0.1.0.5
Avaya G650 Media Gateway	
TN2312BP IP Server Interface (IPSI)	HW15 FW050
TN799DP Control-LAN (C-LAN)	HW01 FW037
TN2602AP IP Media Resource 320 (MedPro)	HW02 FW057
TN2501AP VAL-ANNOUNCEMENT	HW02 FW018
TN2224CP Digital Line	HW08 FW015
TN793CP Analog Line	HW04 FW010
Avaya 9630 IP Telephone	Avaya one-X® Deskphone Edition H.323 Version S3.1
Avaya 9620C IP Telephone	Avaya one-X® Deskphone Edition SIP Version 2.6.0 (sip96xx_2_6_0_0.bin)
Avaya one-X® Agent	2.0 with SP3
Avaya 4625SW IP Telephone	a25d01a2_8.bin
Avaya 6211 Analog phone	-
Avaya S8800 Single Server	Avaya Modular Messaging 5.2
Fax device	Ventafax Home Version 6.2
AT&T IP Transfer Connect Service using MIS/PNT transport service connection	VNI 18

Table 2: Equipment and Software Versions

Note - The solution integration validated in these Application Notes should be considered valid for deployment with Avaya Aura® Communication Manager release 6.0.1 and Avaya Aura® Session Manager release 6.1. Avaya agrees to provide service and support for the integration of Avaya Aura® Communication Manager release 6.0.1 and Avaya Aura® Session Manager release 6.1 with the AT&T IP Transfer Connect service offer, in compliance with existing support agreements for

Avaya Aura® Communication Manager release 6.0 and Avaya Aura® Session Manager 6.0, and in conformance with the integration guidelines as specified in the body of this document.

4. Avaya Aura™ Session Manager

The Session Manager administration for interoperability with AT&T IP Transfer Connect service is similar to the configuration done in [11] for AT&T IP Toll Free service. Refer to the appropriate section in [11] 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.

5. Avaya Aura™ Communication Manager

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 [11]. This section describes additional administration step on Communication Manager to support interoperability with AT&T IP Transfer Connect service. 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 [3] and [4] for further details if necessary.

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

5.1. 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
```

Figure 6: System-Parameters Customer-Options Form – Page 4

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

Figure 7: System-Parameters Customer-Options Form – Page 6

5.2. 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 20** command, where 20 is the trunk group configured in [11]. On **Page 4** of the **trunk-group** form, set the **Network Call Redirection** field to "**y**" (see **Item 5** in **Section 1.3**)

```
change trunk-group 20                                          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? y
 Always Use re-INVITE for Display Updates? n
                             Enable Q-SIP? n
```

Figure 8: Trunk Group form for IP Transfer Connect service calls – Page 4

5.3. Inbound Call Routing

This section describes the steps for routing inbound AT&T IP Transfer Connect service calls to reach Vector Directory Numbers 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 [**11**].

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 information provided by AT&T from **Section 2.1**.

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 (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 [**5**] and [**6**] for further information.

5.3.1. Pre-Answer Redirection

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

- Plays ringback for 2 seconds (vector step **02**).
- Assigns the data "**1234567890123456**" to ASAI UUI variable "**A**" and "**7890123456789012**" to ASAI UUI variable "**B**" (vector steps **05** and **06**).
  Note: The parameters for ASAI UUI variables "**A**" and "**B**", and other vector variables are defined using the **change variables** command (see **Figure 9)**.
- Redirects the call to the number "**1092**" (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 "**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 in the signaling group [11] on which the inbound call arrived.

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

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

Figure 9: Change Variables Form

```
display vdn 6665316                                            Page   1 of   3
                            VECTOR DIRECTORY NUMBER

                             Extension: 6665316
                                 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 10: 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 11: Sample Vector for Pre-Answer Redirection

5.3.2. Post-Answer Redirection

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

- Plays ringback for 2 seconds (vector step **02**).
- Assigns the data "**1234567890123456**" to ASAI UUI variable "**A**" and "**7890123456789012**" to ASAI variable "**B**" (vector steps **05** and **06**).
  Note: The parameters for UUI variable "**A**" and other vector variables are defined using the **change variables** command (see **Figure 9)**.
- Answers the call to play an announcement (vector step **08**).
- Attempts to redirect the call to the number "**1092**" (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 "**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 in the signaling group on which the inbound call arrived.

```
display vdn 6665318                                           Page   1 of   3
                            VECTOR DIRECTORY NUMBER

                             Extension: 6665318
                                 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 12: 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 13: Sample Vector for Post-Answer Redirection

6. Avaya Aura™ Session Border Controller Configuration

This section lists additional configuration of the AA-SBC to support AT&T IP Toll Free service. Please refer to [**11**] for other AA-SBC installation and configuration details.

6.1. Refer-To Header in REFER Message

This section presents a sample configuration that will cause the AA-SBC to modify the host portion of the Refer-To header in a REFER message, while preserving the user portion (containing the Refer-To destination telephone number) and any User-User Information. In this example, the host portion was changed such that AT&T would receive the AT&T IP Transfer Connect service IP Address and port as the host portion. On the production circuit used to verify these Application Notes, this header manipulation may not be required.

In the left side menu, navigate to **vsp → session-config-pool → entry ToPBX → header-settings**. On the right panel, select **Add reg-ex-header** as shown below.

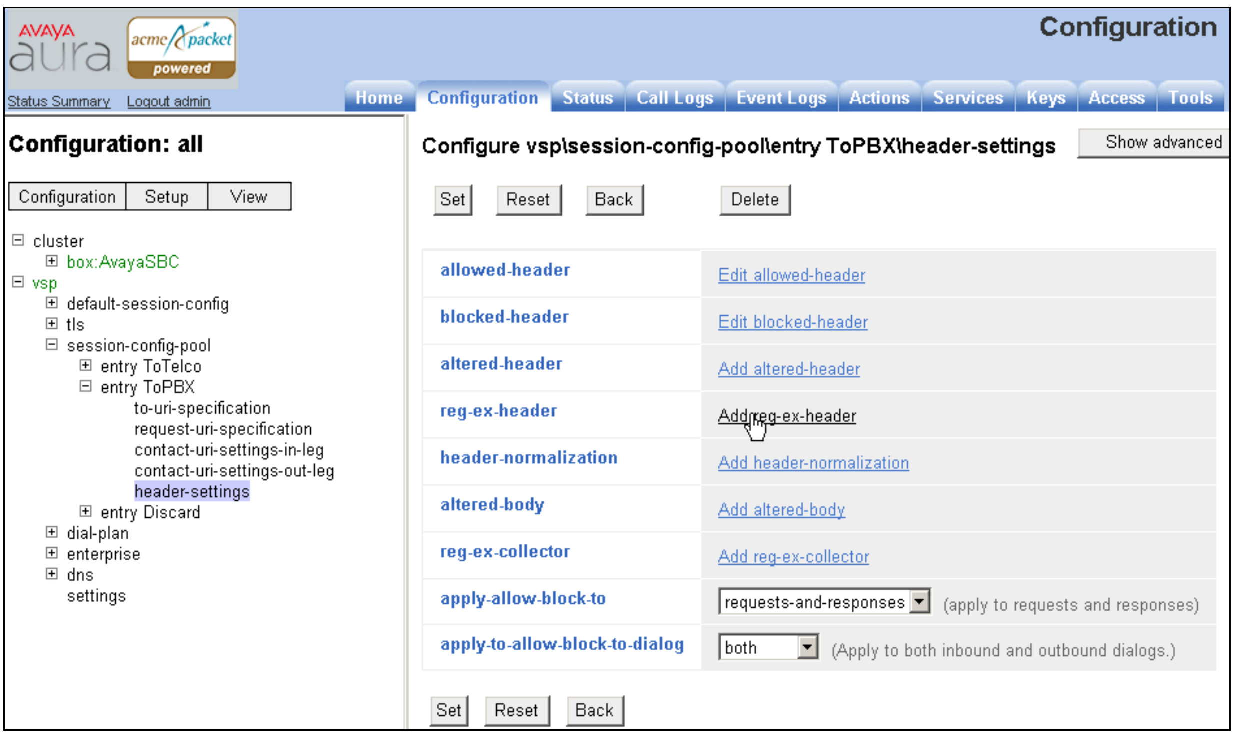

Figure 14: Configuration - Header Settings screen

In the resultant screen, enter any number in the **number** field and enter “**Refer-To**” in the **destination** field and click **Create**.

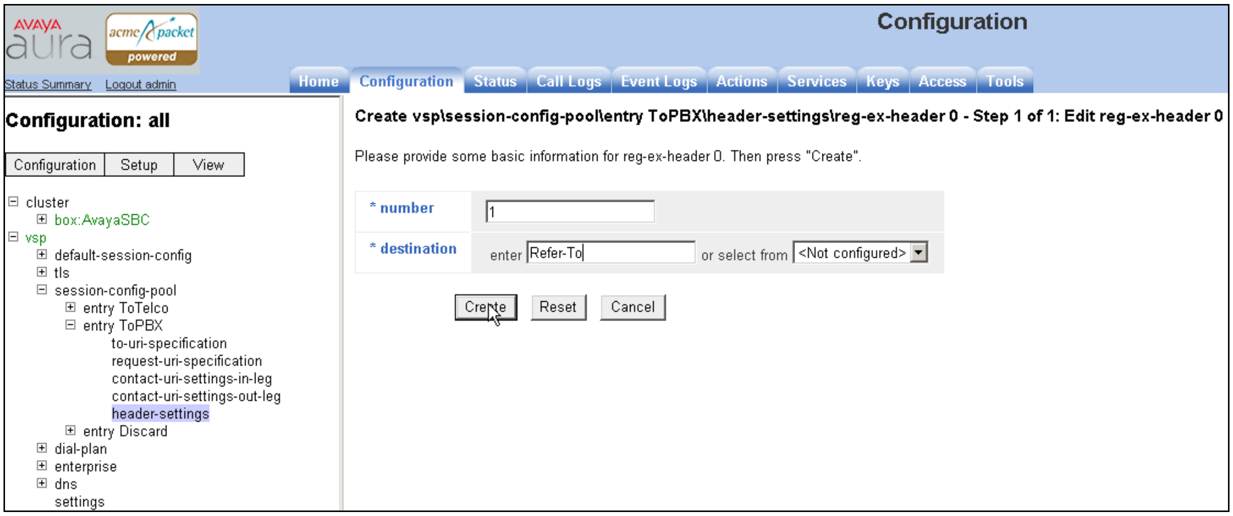

Figure 15: Configuration - Create Regular Expression Header screen

In the resultant screen, select “REFER” for **apply-to-methods** and “**both**” for **type** field in **apply-to-responses** section. Select the **Configure** link to the right of **create**.

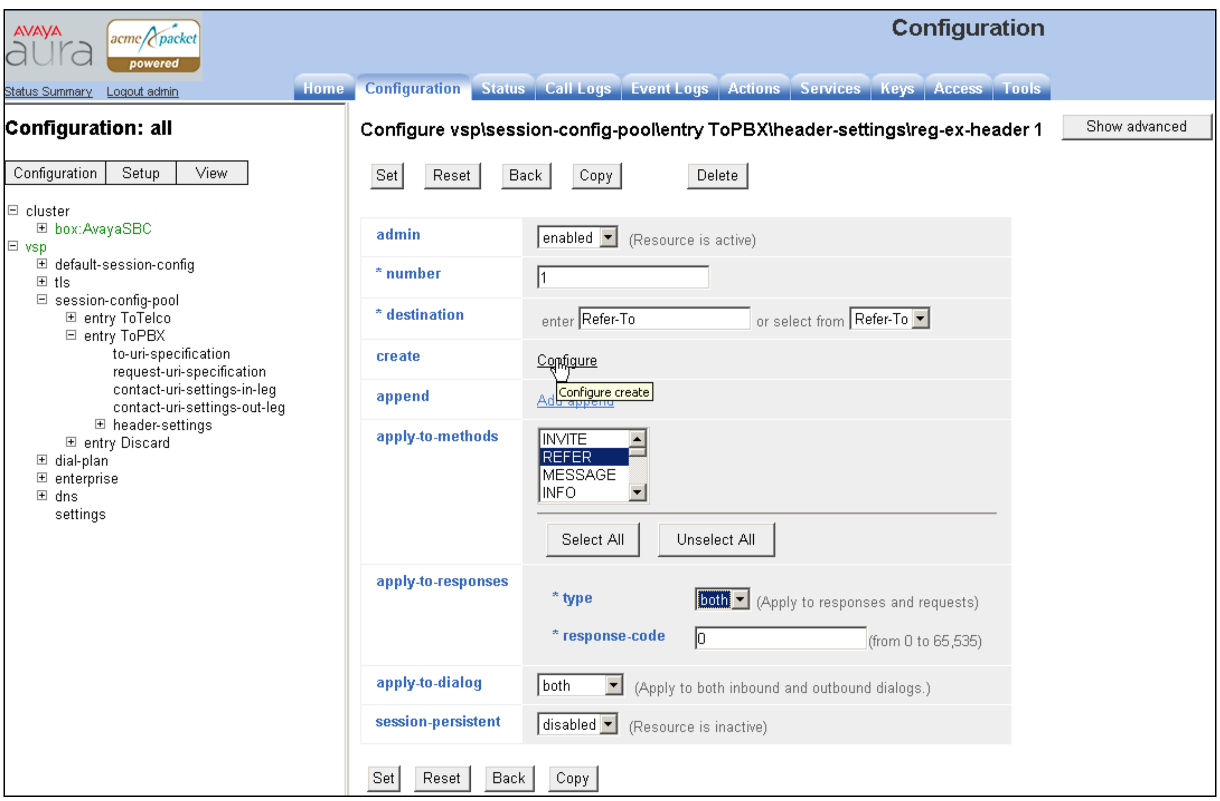

Figure 16: Configuration – Regular Expression Header 1 screen

The following screen is presented. In the **source** area, select "**Refer-To**" from the drop-down list or type "**Refer-To**" in the **select from** field.

In the **expression** field, enter a regular expression to match. In the sample configuration, "**<sip:(.*)@avaya\.com(.*)>**" was entered. In this expression, the first (.*) will match and store any user part of the Refer-To header. The second instance of (.*) matches and stores any UUI if present. The domain "avaya.com" is what the AA-SBC would otherwise put in the Refer-To header host part.

In the **replacement** field, "**<sip:\1@\r:\R\2>**" was entered in the sample configuration. The variable "\1" is the stored user part from the original Refer-To header containing the Refer-To number, corresponding to the first instance of "(.*)" from the **expression**. The variable "\2" is any stored UUI from the original Refer-To header, corresponding to the second instance of "(.*)" from the **expression**. The "\r" inserts the "remote IP Address" corresponding to the AT&T IP Transfer Connect service IP Address. This is followed by a colon and "\R" corresponding to the AT&T IP Transfer Connect SIP signaling port, which is 5060 in this case.

After completing the **source**, **expression** and **replacement** fields as appropriate, click **Create**.

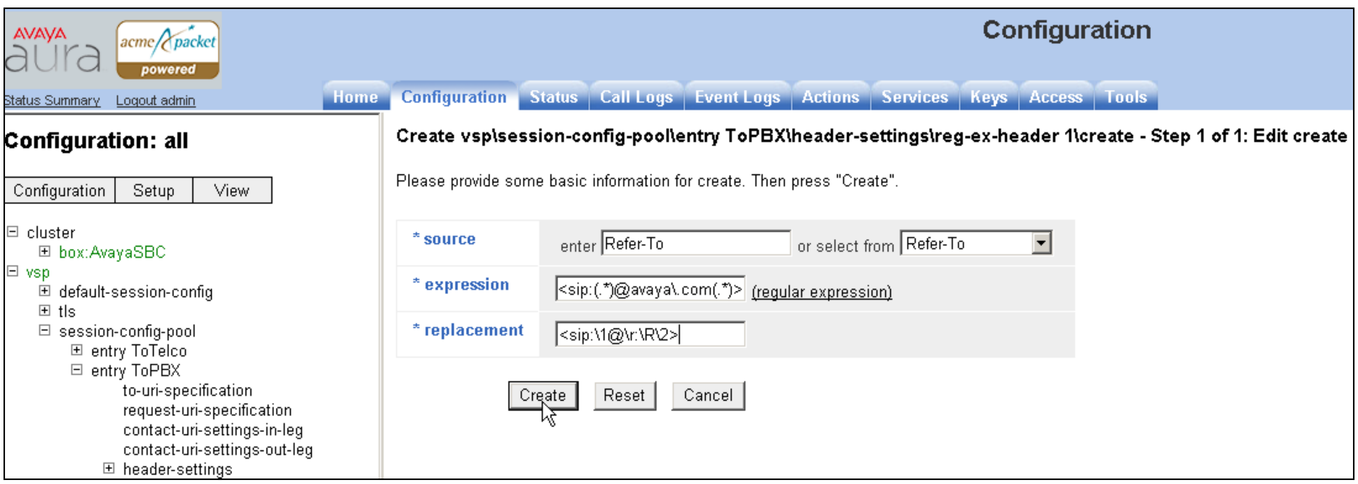

Figure 17: Configuration – Create Regular Expression Header screen

The following screen shows the completed rule. Click the **Set** button. Proceed to save and activate the configuration as described in [**11**].

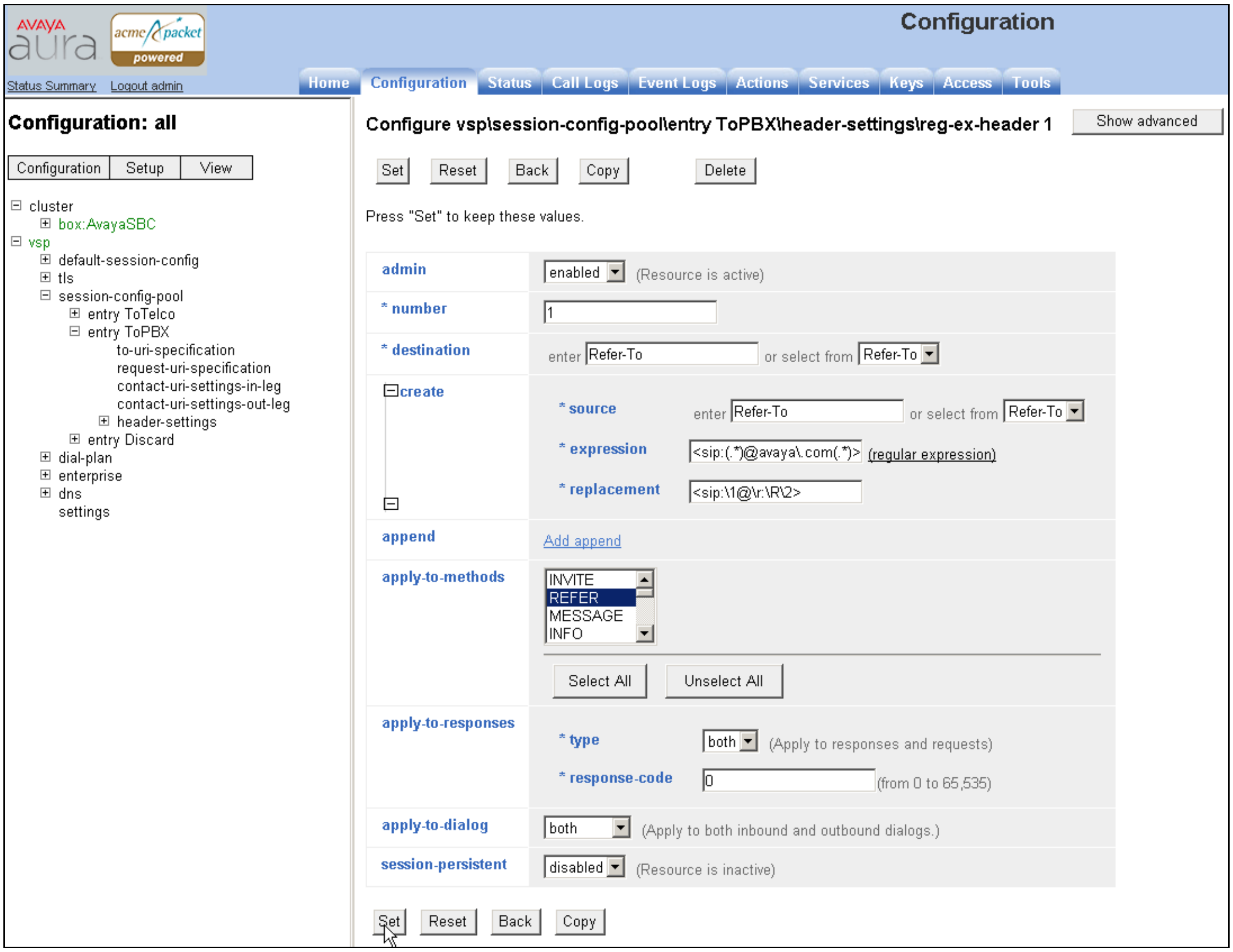

Figure 18: Configuration – Regular Expression Header (Final) screen

6.2. Avaya Aura™ Session Border Controller Element Manager Configuration

The notable settings are highlighted in bold on the pertinent settings done during installation and further configuration.

```
cat cxc.cfg
#
#  Copyright (c) 2004-2010  Acme Packet Inc.
#  All Rights Reserved.
#
#  File: /cxc/cxc.cfg
#
config cluster
 config box 1
  set hostname AvayaSBC
  set timezone America/Denver
  set name AvayaSBC
  set identifier 00:ca:fe:07:98:42
  config interface eth0
   config ip inside
    set ip-address static 10.80.130.12/24
    config ssh
    return
    config snmp
     set trap-target 10.80.130.11 162
     set trap-filter generic
     set trap-filter dos
     set trap-filter sip
     set trap-filter system
    return
    config web
    return
    config web-service
     set protocol https 8443
     set authentication certificate "vsp\tls\certificate ws-cert"
    return
    config sip
     set udp-port 5060 "" "" any 0
     set tcp-port 5060 "" "" any 0
     set tls-port 5061 "" "" any 0
    return
```

```
  config icmp
  return
  config media-ports
  return
  config routing
   config route Default
    set gateway 10.80.130.1
   return
   config route Static0
    set destination network 192.11.13.4/30
    set gateway 10.80.130.10
   return
   config route Static1
    set admin disabled
   return
   config route Static2
    set admin disabled
   return
   config route Static3
    set admin disabled
   return
   config route Static4
    set admin disabled
   return
   config route Static5
    set admin disabled
   return
   config route Static6
    set admin disabled
   return
   config route Static7
    set admin disabled
   return
   config route internal-sip-media
    set destination host 10.80.120.28
    set gateway 10.80.130.1
   return
  return
 return
return
config interface eth2
 config ip outside
  set ip-address static 205.168.62.55/25
  config sip
   set udp-port 5060 "" "" any 0
```

```
     set tcp-port 5060 "" "" any 0
     set tls-port 5061 "" "" any 0
    return
    config icmp
    return
    config media-ports
    return
    config routing
     config route Default
      set admin disabled
     return
     config route external-sip-media
      set destination network 207.242.225.0/24
      set gateway 205.168.62.1
     return
    return
   return
  return
  config cli
   set prompt AvayaSBC
  return
  config os

  return
 return
return
config services
 config event-log
  config file access
   set filter access info
  return
  config file system
   set filter general info
   set filter system info
  return
  config file errorlog
   set filter all error
  return
  config file db
   set filter db debug
   set filter dosDatabase info
  return
  config file management
   set filter management info
  return
```

```
  config file peer
   set filter sipSvr info
  return
  config file cac
   set filter sipCAC warning
  return
  config file dos
   set filter dos alert
   set filter dosSip alert
   set filter dosTransport alert
   set filter dosUrl alert
  return
  config file krnlsys
   set filter krnlsys debug
  return
  config file acct
   set filter acct debug
  return
 return
return
config master-services
 config accounting
 return
 config database
  set media enabled
 return
return
config vsp
 set admin enabled
 config default-session-config
  config media
   set anchor enabled
   set rtp-stats enabled
  return
  config sip-directive
   set directive allow
  return
  config log-alert
   set apply-to-methods-for-filtered-logs
  return
  config header-settings
   set blocked-header P-Site
  return
  config third-party-call-control
  return
```

```
return
config tls
 config certificate ws-cert
  set certificate-file /cxc/certs/ws.cert
 return
return
config session-config-pool
 config entry ToTelco
  config to-uri-specification
   set host next-hop
  return
  config from-uri-specification
   set host local-ip
  return
  config request-uri-specification
   set host next-hop
  return
  config p-asserted-identity-uri-specification
   set host local-ip
  return
  config contact-uri-settings-in-leg
   set add-maddr disabled
   set use-incoming-contact enabled
  return
  config contact-uri-settings-out-leg
   set add-maddr disabled
   set use-incoming-contact enabled
  return
  config header-settings
   config reg-ex-header 1
    set destination Refer-To
    set create Refer-To "<sip:(.*)@avaya\.com(.*)>" "<sip:\1@\r:\R\2>"
    set apply-to-methods REFER
    set apply-to-responses both 0
   return
 return
 config entry ToPBX
  config to-uri-specification
   set host next-hop-domain
  return
  config request-uri-specification
   set host next-hop-domain
  return
  config contact-uri-settings-in-leg
   set add-maddr disabled
```

```
   set use-incoming-contact enabled
  return
  config contact-uri-settings-out-leg
   set add-maddr disabled
   set use-incoming-contact enabled
  return
 return
 config entry Discard
  config sip-directive
  return
 return
return
config dial-plan
 config route Default
  set priority 500
  set location-match-preferred exclusive
  set session-config vsp\session-config-pool\entry Discard
 return
 config source-route FromTelco
  set peer server "vsp\enterprise\servers\sip-gateway PBX"
  set source-match server "vsp\enterprise\servers\sip-gateway Telco"
 return
 config source-route FromPBX
  set peer server "vsp\enterprise\servers\sip-gateway Telco"
  set source-match server "vsp\enterprise\servers\sip-gateway PBX"
 return
return
config enterprise
 config servers
  config sip-gateway PBX
   set peer-identity ""
   set domain avaya.com
   set outbound-session-config-pool-entry vsp\session-config-pool\entry ToPBX
   config server-pool
    config server PBX1
     set host 10.80.120.28
     set transport TCP
    return
   return
  return
  config sip-gateway Telco
   set peer-identity ""
   set outbound-session-config-pool-entry vsp\session-config-pool\entry ToTelco
   config server-pool
    config server Telco1
```

```
      set host 207.242.225.210
     return
    return
   return
  return
 return
 config dns
  config resolver
   config server 135.9.1.2
   return
  return
 return
 config settings
  set stack-socket-threads-max 2
 return
return
config external-services
return
config preferences
 config gui-preferences
 return
return
config access
 config permissions superuser
  set cli advanced
 return
 config permissions read-only
  set config view
  set actions disabled
 return
 config users
  config user admin
   set password 0x002bdd5d9fea2fefeb97b0115854a47db2c8b27a2fe0187e0274977f4b
   set permissions access\permissions superuser
  return
  config user cust
   set password 0x004803cd9fae4ee1b2462598359d6c5e179008f9083caa7b30b9b19b43
   set permissions access\permissions read-only
  return
 return
return
config features
return
```

7. 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 Avaya Aura™ Session Border Controller.
- A laboratory version of the AT&T IP Transfer Connect service, to which the simulated enterprise was connected via MIS/PNT transport.

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

- Inbound AT&T IP Transfer Connect service calls to Communication Manager telephones and 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.

The above test objectives with limitations as noted in **Section 1.3** were verified.

8. Verification Steps

The call verification steps and troubleshooting tools described for the AT&T Toll Free service described in [**11**], 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.
2. Place an inbound call to an AT&T IP Transfer Connect service line enabled with IP Courtesy Transfer 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.
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 disconnects properly.

9. Conclusion

As illustrated in these Application Notes, Avaya Aura™ Session Manager, Avaya Aura™ Communication Manager Network Call Redirection, and Avaya Aura™ 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] *Installing and Configuring Avaya Aura™ Session Manager,* Doc ID 03-603473 Release 6.
[2] *Administering Avaya Aura™ Session Manager*, Doc ID 03-603324, Release 6.0, June 2010
[3] *Installing and Configuring Avaya Aura™ Communication Manager*, Doc ID 03-603558, Release 6.0 June, 2010
[4] *Avaya Aura™ Communication Manager Feature Description and Implementation,* Release 6.0, 555-245-205, Issue 8.0, June 2010
[5] *Administering Avaya Aura™ Call Center Features*, Release 6.0, June 2010
[6] *Programming Call Vectors in Avaya Aura™ Call Center*, 6.0, June 2010
[7] *Modular Messaging Multi-Site Guide Release 5.1*, June 2009
[8] *Modular Messaging for Microsoft Exchange Release 5.1 Installation and Upgrades*, June 2009
[9] *Modular Messaging for the Avaya Message Storage Server (MSS) Configuration Release 5.1 Installation and Upgrades*, June 2009
[10] *Modular Messaging for IBM Lotus Domino 5.1 Installation and Upgrades*, June 2009
[11] *Application Notes for Avaya Aura™ Communication Manager 6.0, Avaya Aura™ Session Manager 6.0 and Avaya Aura™ Session Border Controller AT&T IP Transfer Connect service Issue 1.0 –* *https://devconnect.avaya.com/*

AT&T IP Transfer Connect Service Descriptions:
[12] *AT&T IP Transfer Connect*
http://www.business.att.com/enterprise/Service/business-voip-enterprise/network-based-voip-enterprise/ip-flexible-reach-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.