Avaya Solution & Interoperability Test Lab

Application Notes for Avaya Aura™ SIP Enablement Services, Avaya Aura™ Communication Manager SIP Network Call Redirection with AT&T IP Transfer Connect Service – Issue 1.0

Abstract

These Application Notes describe the steps for configuring Avaya Aura™ SIP Enablement Services and Avaya Aura™ Communication Manager SIP Network Call Redirection, with the AT&T IP Transfer Connect service using MIS-PNT transport service connections.

The AT&T IP Transfer Connect service is a service option available with the AT&T IP Toll Free service, and supports the rerouting of inbound toll free calls to alternate destinations based upon SIP redirection messages from Avaya Aura™ Communication Manager. In addition, 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. Note that these Application Notes are intended to supplement separate Application Notes covering Avaya Aura™ SIP Enablement Services, Avaya Aura™ Communication Manager interoperability with the AT&T IP Toll Free service (https://devconnect.avaya.com/public/download/dyn/CMSES521_IPTF.pdf).

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™ SIP Enablement Services (SES), and Avaya Aura™ Communication Manager SIP Network Call Redirection (NCR) with the AT&T IP Transfer Connect service using MIS-PNT transport service connections.

The AT&T IP transfer Connect service is a service option available with the AT&T IP Toll Free service, and supports the rerouting of inbound toll free calls to alternate[1] 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.

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™ SIP Enablement Services 5.2.1, Avaya Aura™ Communication Manager 5.2.1, interoperability with the AT&T IP Toll Free service [11].

[1] 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 IP Transfer Connect service to SES and Communication Manager.

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. The "Connect with Avaya" section provides the worldwide support directory. 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. In an attended Refer call scenario, the initial call leg is supposed to be maintained until the refer-to call leg has been established. During REFER testing, it was found that Communication Manager sends a BYE upon receipt of a NOTIFY/Ringing message. The call completes if the refer-to party answers the call; however if the refer-to party does not answer the call, Communication Manager cannot resume control of the original call leg since it sent a BYE. This issue is being investigated by the Avaya product team and will be addressed in a future release.
2. Depending on how accurate CDR (Call Detail Records) need to be, reports may be off by the elapsed time between the 180 Ringing sipfrag NOTIFY/Ringing and BYE messages from the AT&T network.
3. An AT&T IP Transfer Connect circuit can support up to 4 C-LAN cards and one SES. If additional C-LANS and SES servers must be supported, an AT&T IP Transfer Connect/Avaya certified Session Border Controller (SBC) will be required.

Note: Communication Manager R5.2.1 will send a SIP UPDATE message for any display information changes.

2. Reference Configuration

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

- Communication Manager provides the enterprise voice communications services. In this sample configuration, Communication Manager runs on an Avaya S8720 Server. This solution is extensible to other Avaya S8xxx Servers as well as Communication Manager supported on the Midsize Business Template.
- The Avaya Media Gateway provides the physical interfaces and resources for enterprise voice communications. In this sample configuration, an Avaya G650 Media Gateway is used. This solution is extensible to other Avaya Media Gateways.
- SES runs on an Avaya S8500 server. SES serves as a SIP proxy between Communication Manager and the AT&T IP Toll Free service with Transfer Connect option. SES also provides registrar services to SIP phones at the enterprise.
- Avaya "office" phones include Avaya H.323, Analog and Digital phones, and Avaya one-X™ Agent, which is a Softphone that runs on a Desktop (not shown). Avaya SIP phones were only used for call delivery to a phone extension and not to an agent.
- Avaya Modular Messaging provides the corporate voice messaging capabilities for enterprise users. The provisioning of Modular Messaging is beyond the scope of this document.
- IP Network Address Translation (NAT) devices, firewalls, Application Layer Gateway (ALG) devices, and Session Border Controller (SBC) devices that may exist between the enterprise site and the AT&T IP Transfer Connect service are not explicitly shown. Those devices generally must be SIP-aware and configured properly for SIP trunking to function properly. When configured correctly, those devices are transparent to the Avaya communications infrastructure.
- Enterprise networks of sufficient size or complexity may use separate servers for SES Home and Edge roles. This configuration is functionally equivalent to the SES combined Home/Edge server configuration illustrated in these Application Notes.

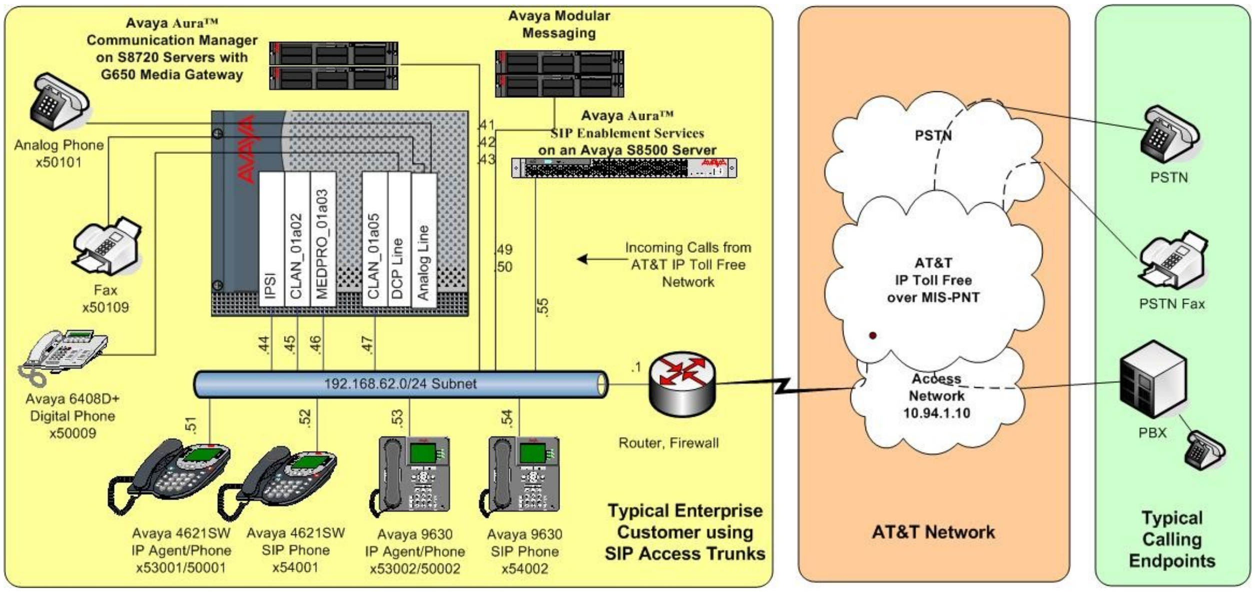

Figure 1: Reference SIP Trunking Configuration

2.1. Illustrative Configuration Information

The specific values listed in **Table 1** below and in subsequent sections are used in the sample 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 as part of the IP Transfer Connect provisioning process.

Component	Illustrative Value in these Application Notes
Avaya Aura™ SIP Enablement Services	
IP Address	192.168.62.55
Avaya Aura™ Communication Manager	
C-LAN IP Address	192.168.62.45
VDN	2015-2025
Skill (Hunt Group)	11, 12, 13
Agent Extensions	53001, 53002, 53003, 53004, 53005
Hunt Group Extensions	53011 (11), 53012 (12), 53013 (13)
Phone Extensions	50001, 50002, 50003, 50009, 50101, 54001, 54002
Voice Messaging Pilot Extension	55000
Avaya Modular Messaging	
Messaging Application Server (MAS) IP Address	192.168.62.49
Exchange Server IP Address	192.168.62.50
Pilot Number	17323255000
AT&T IP Transfer Connect Service	
Border Element IP Address	10.94.1.10
Digits Passed in SIP Request-URI (RRN)	000001009 (SIP Redirect) 000001010 (SIP Refer Attended) 000001011 (SIP Refer Unattended) 000001012, 000001013 (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 SES and Communication Manager, the following call flows are described in this section.

The call scenario illustrated in the figure below is an inbound call from a PSTN phone via a SIP trunk from the AT&T IP Transfer Connect service to SES and Communication Manager.

1. A PSTN phone or fax originates a call to an AT&T 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 SES.
4. SES routes the call to Communication Manager.
5. Depending on the RRN, Communication Manager routes the call
    - To a vector, which in turn, routes the call to an agent
    - Directly to an agent or a phone extension.

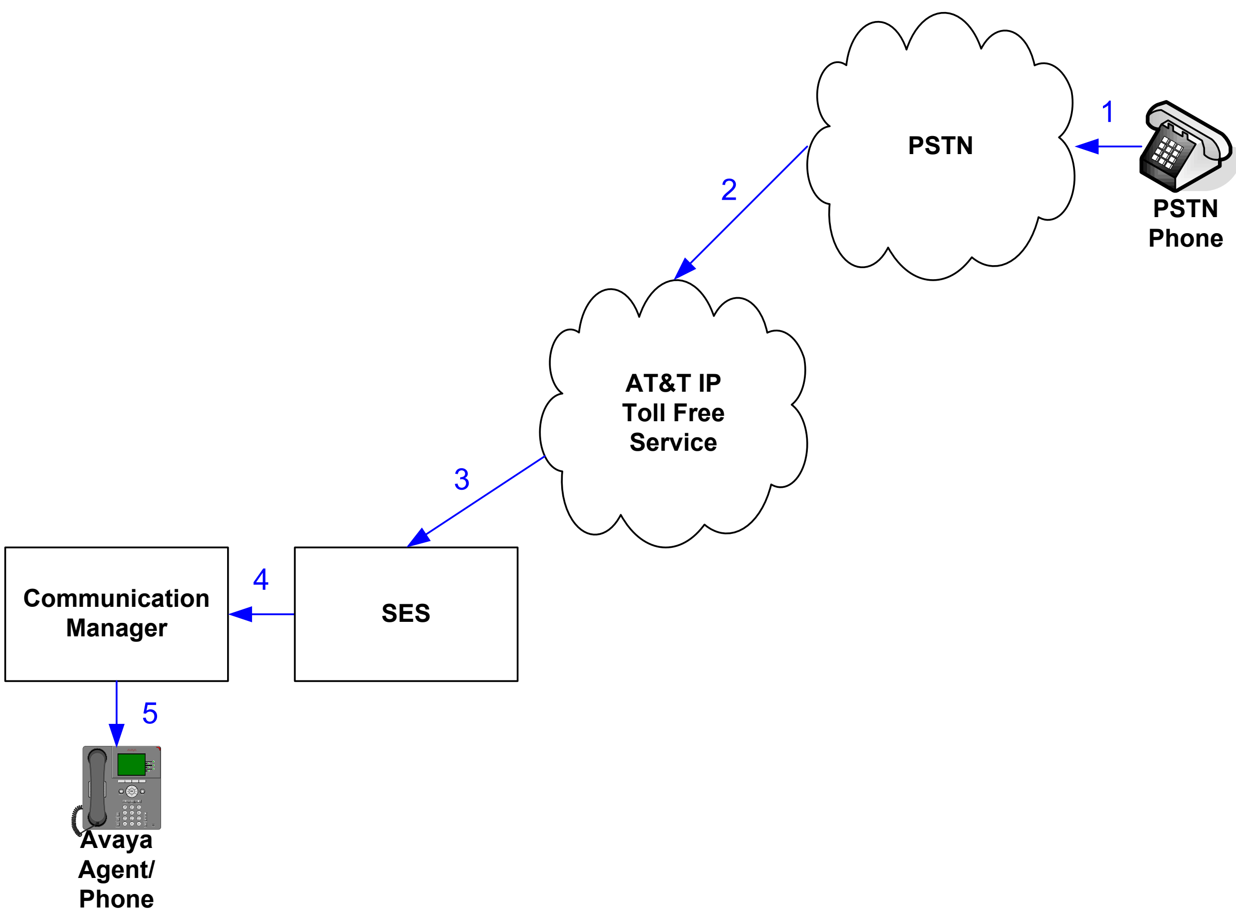

Figure 2: AT&T IP Transfer Connect Service Call – No Redirection

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

The next call scenario illustrated in the figure below is an inbound call from a PSTN phone via a SIP trunk from the AT&T IP Transfer Connect service to SES and 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 it to an alternate destination.

1. Same as first four steps from call scenario in **Figure 2**.
2. Communication Manager routes the call to a vector, which redirects the call by sending a SIP 302 message to AT&T IP Transfer Connect service network via the SIP trunk on which the inbound call arrived. Since SIP 302 message is a final response from the Communication Manager, it releases the trunk whether redirection succeeds or not.
3. The AT&T IP Transfer Connect service routes the call to an alternate destination and upon answer, connects the calling party to the target party (alternate destination).

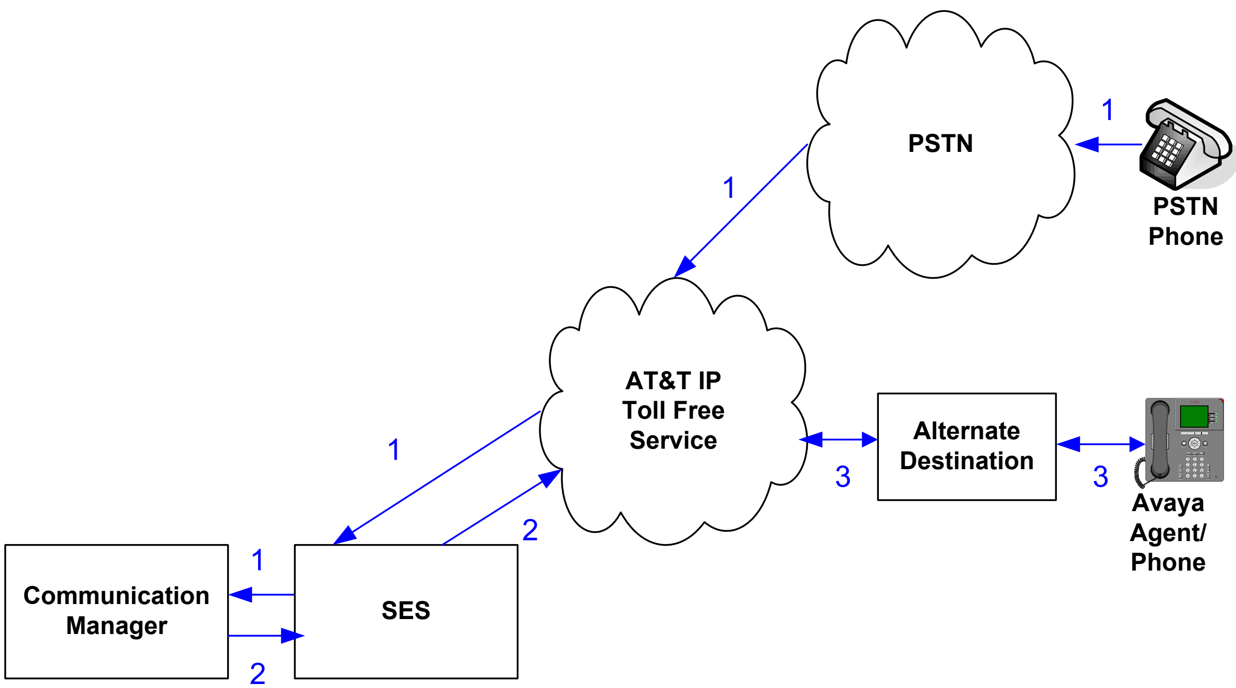

Figure 3: AT&T IP Transfer Connect Service Call – Pre-Answer SIP 302 Redirection

The third call scenario illustrated in the figure below is an inbound call from a PSTN phone via a SIP trunk from the AT&T IP Transfer Connect service to SES and 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 it to an alternate destination.

1. Same as first four steps from call scenario in **Figure 2**.
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 to the AT&T IP Transfer Connect service network via the SIP trunk on which the inbound call arrived.
3. The AT&T IP Transfer Connect service places the call to the target party at the 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).

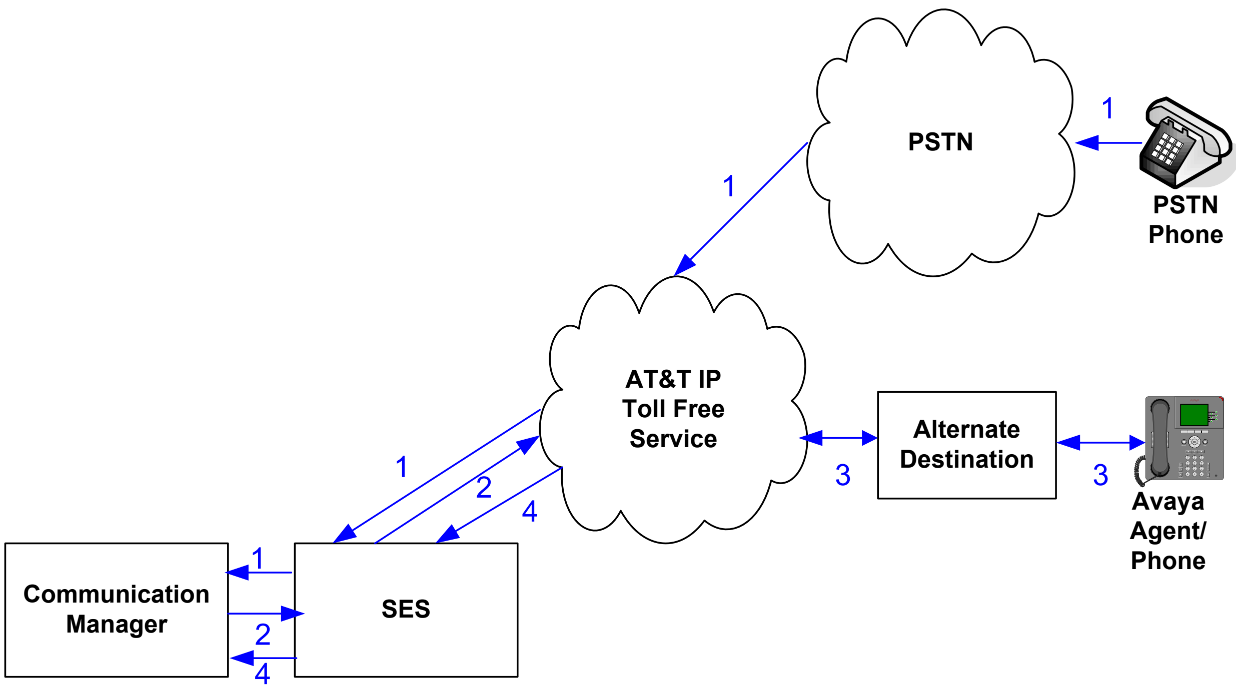

Figure 4: AT&T Transfer Connect service call – Post-Answer SIP REFER Redirection Successful

The last call scenario illustrated in the figure below is similar to the previous call except the redirection is not successful as the alternate destination is busy or otherwise unavailable. As a result, the alternate destination routing is cancelled and Communication Manager routes the call to an agent/phone.

1. Same as first four steps from call scenario in **Figure 2**.
2. Same as call scenario in **Figure 4**.
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.

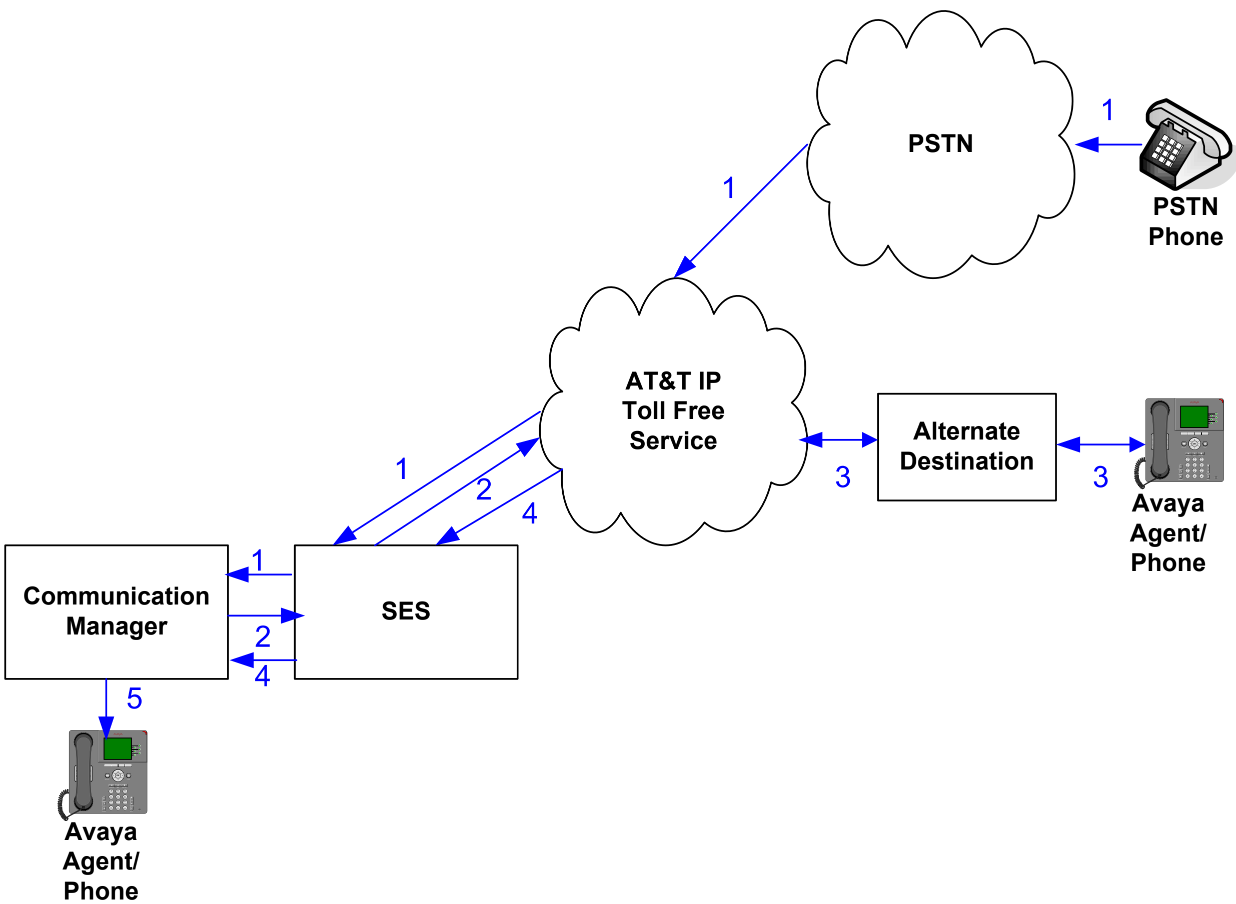

Figure 5: AT&T Transfer Connect service call – Post-Answer SIP REFER Redirection Unsuccessful

3. Equipment and Software Validated

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

Component	Version
Avaya S8710 Server	Avaya Aura™ Communication Manager 5.2.1, Service Pack 2 (R015x.02.1.016.4-18111)
Avaya G650 Media Gateway	
TN2312BP IP Server Interface (IPSI)	HW03 FW050
TN799DP Control-LAN (C-LAN)	HW00 FW037
TN2602AP IP Media Resource 320 (MedPro)	HW02 FW054
TN2501AP VAL-Announcement	HW02 FW018
TN2224B Digital Line	HW03
TN793B Analog Line	HW05
Avaya S8500B Server	Avaya Aura™ SIP Enablement Services 5.2.1, Service Pack 1b (SES05.2.1.016.4-SP1b)
Avaya 9650 IP Telephone	Avaya one-X™ Deskphone Edition H.323 Release S3.0
Avaya 9630 IP Telephone	Avaya one-X™ Deskphone Edition SIP Release 2.5
Avaya 4610SW H323 Telephone	2.8
Avaya 4621SW SIP Telephone	2.2.2
Avaya 6408D+ Digital Telephone	-
Avaya 6211 Analog Telephone	-
Avaya one-X Agent	2.0
AT&T IP Transfer Connect service over MIS-PNT	VNI 16

Table 2: Equipment and Software Versions

4. Avaya Aura™ Communication Manager

Communication Manager administration for interoperability with the AT&T IP Toll Free service is described in **[11]**. This section describes the additional administration steps on Communication Manager necessary for supporting interoperability with the AT&T IP Transfer Connect service. The steps are performed from the Communication Manager System Access Terminal (SAT) interface.

Note – In the following sections, only the **highlighted** parameters are applicable to these Application Notes. Other parameters shown should be considered informational.

4.1. System Parameters

This section reviews the additional Communication Manager licenses and features that are required for supporting the interoperability with the AT&T IP Transfer Connect service. For required licenses that are not enabled in the steps that follow, contact an authorized Avaya account representative to obtain the licenses.

- Enter the **display system-parameters customer-options** command. On **Page 4** of the **system-parameters customer-options** form, verify that the bolded fields in the following screenshots are set to "**y**".

```
display system-parameters customer-options                    Page   4 of  10
                                OPTIONAL FEATURES

   Emergency Access to Attendant? y                            IP Stations? y
           Enable 'dadmin' Login? y
           Enhanced Conferencing? y                      ISDN Feature Plus? n
                  Enhanced EC500? y     ISDN/SIP Network Call Redirection? y
    Enterprise Survivable Server? n                       ISDN-BRI Trunks? y
       Enterprise Wide Licensing? n                              ISDN-PRI? y
              ESS Administration? y              Local Survivable Processor? n
          Extended Cvg/Fwd Admin? y                    Malicious Call Trace? 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)? n
             Hospitality (Basic)? y     Multimedia Call Handling (Enhanced)? n
 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

- On **Page 6** of the **system-parameters customer-options** form, verify that the bolded fields in the following screenshots are set to "**y**".

```
display system-parameters customer-options                      Page   6 of  10
                         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

4.2. Trunks

This section describes the steps for modifying the Signaling Group and SIP trunk to SES to support the interoperability with the AT&T IP Transfer Connect service.

- Enter the **change signaling-group s** command, where **s** is the number of the signaling group administered in [11] for inbound AT&T IP Toll Free service calls. The **Far-end Domain** was left blank in this Reference configuration. The SIP 302 message will have the IP address of the SES in the **Contact** header. The **Refer-To** header in SIP REFER message will have the domain name configured in SES.

Note: Set the **Far-end Domain** field to the IP Address of the AT&T Border Element if the SIP 302 message **Contact** header and SIP REFER message **Refer-T**o header need to reflect the AT&T Border Element address.

```
change signaling-group 5                                     Page   1 of   1

 Group Number: 5              Group Type: sip
                        Transport Method: tls


   Near-end Node Name: CLAN-1A02            Far-end Node Name: SES
 Near-end Listen Port: 5061               Far-end Listen Port: 5061
                                       Far-end Network Region: 3
       Far-end Domain:

                                          Bypass If IP Threshold Exceeded? n

         DTMF over IP: rtp-payload         Direct IP-IP Audio Connections? y
                                                     IP Audio Hairpinning? n
         Enable Layer 3 Test? n
 Session Establishment Timer(min): 3           Alternate Route Timer(sec): 6
```

Figure 8: Signaling-Group Form for AT&T IP Transfer Connect Calls

- Enter the **change trunk-group t** command, where **t** is the number of the trunk group administered in [11] for inbound AT&T IP Toll Free service calls. On **Page 4** of the **trunk-group** form, set **Network Call Redirection** to "**y**".

```
change trunk-group 5                                        Page   4 of  21
                              PROTOCOL VARIATIONS

                      Mark Users as Phone? n
            Prepend '+' to Calling Number? n
      Send Transferring Party Information? n
                 Network Call Redirection? y
                    Send Diversion Header? n
                  Support Request History? y
             Telephone Event Payload Type:
```

Figure 9: Trunk-Group Form for Inbound AT&T IP Toll Free / IP Transfer Connect Calls – Page 4

4.3. 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 **[7]** and **[8]** for further information.

4.3.1. 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 11**, which invokes the vector shown in **Figure 12**. 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 10**).
- Redirects the call to the number "**1012**" (vector step **08**). Note that since this vector did not answer the call, the presence of the "~" in the "**route-to number**" instructs Communication Manager to send a SIP 302 message with the number "**1012**" in the user part of the Contact header URI (e.g., 1012@<host/domain>) to the AT&T IP Transfer Connect service. The host/domain is populated with the Far-end Domain value administered in the signaling group on which the inbound call arrived; see **Section 4.2**.

```
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 10: Change Variables Form

```
display vdn 31009                                             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 11: 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 ~r1012           with cov n if unconditionally
09
10 #    Play this announcement only on redirect failure
11 announcement 33008
12
```

Figure 12: Sample Vector for Pre-Answer Redirection

4.3.2. 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 13**, which invokes the vector shown in **Figure 14**. 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 10**).
- Answers the call to play an announcement (vector step **08**).
- Attempts to redirect the call to the number "**1012**" (vector step **09**). Note that since this vector answered the call, the presence of the "~" in the "**route-to number**" instructs Communication Manager to send a SIP REFER message with the number "**1012**" in the user part of the Refer-To header URI (e.g., 1012@<host/domain>) to the AT&T IP Transfer Connect service. The host/domain is populated with the Far-end Domain value administered in the signaling group on which the inbound call arrived; see **Section 4.2**.

```
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 13: 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 ~r1012           with cov n if unconditionally
10 #    Play this announcement only on redirect failure
11 disconnect   after announcement 33008
12
```

Figure 14: Sample Vector for Post-Answer Redirection

5. Configure Avaya Aura™ SIP Enablement Services

SES administration for interoperability with the AT&T IP Toll Free / IP Transfer Connect service is described in **[11]**. Additional Host maps may need to be defined to handle the digits sent by the AT&T IP Transfer Connect service.

6. General Test Approach and Test Results

The test environment consisted of:

- A simulated enterprise with SES, Communication Manager, and Avaya phones.
- A laboratory version of the AT&T IP Transfer Connect service, to which the simulated enterprise was connected.

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

- Inbound AT&T IP Transfer Connect service calls to Communication Manager VDNs, agents, and phones.
- Inbound AT&T IP Transfer Connect service calls that are immediately redirected by a Communication Manager vector (pre-answer redirection) back to the AT&T IP Transfer Connect service for redirection to an alternate destination.
- Inbound AT&T IP Transfer Connect service calls that are answered by a Communication Manager vector and then redirected (post-answer redirection) back to the AT&T IP Transfer Connect service for redirection to 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 at the alternate destination.
- Call and two-way talkpath establishment between callers and Communication Manager agents/phones.

The test objectives of **Section 6** with limitations as noted in **Section 1.3** were verified.

7. Verification Steps

7.1. Verification Tests

The following steps may be used to verify the configuration:

- Place an inbound call, and verify that two-way talkpath exists, and that the calls remain stable for several minutes and disconnect properly.
- Place an inbound call to an agent or a phone and verify that the call goes to coverage if it is not answered.
- 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.
- 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.
- Verify that when Communication Manager is the transfer target of redirected calls, the calls are answered with two-way talk path. Verify that the calls remain stable for several minutes and disconnect properly.

7.2. Troubleshooting Tools

The Communication Manager **list trace vector**, **list trace vdn**, **list trace tac**, and/or **status trunk *trunk-group-no*** commands are helpful diagnostic tools to verify correct operation and to troubleshoot problems. MST (Message Sequence Trace) diagnostic traces (performed by Avaya Support) can be helpful in understanding the specific interoperability issues.

The **traceSES** function within the SES may be used to capture SIP traces between SES and the AT&T IP Transfer Connect service. In addition, if port monitoring is available, a SIP protocol analyzer such as Wireshark (a.k.a. Ethereal) can be used to capture SIP traces at the various interfaces. SIP traces can be instrumental in understanding SIP protocol issues resulting from configuration problems. Note that the SIP messaging between Communication Manager and SES uses TLS encryption and cannot be viewed using Wireshark.

8. Conclusion

As illustrated in these Application Notes, Avaya Aura™ SIP Enablement Services, and Avaya Aura™ Communication Manager Network Call Redirection 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.).

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

9. References

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

[1] *Administering Avaya Aura™ Communication Manager*, Issue 5.0, Release 5.2, May 2009, Document Number 03-300509

[2] *Avaya Aura™ Communication Manager Feature Description and Implementation*, Issue 7, Release 5.2, May 2009, Document Number 555-245-205

[3] *SIP Support in Avaya Aura™ Communication Manager Running on the Avaya S8xxx Servers*, Issue 9, May 2009, Document Number 555-245-206

[4] *Installing, Administering, Maintaining, and Troubleshooting Avaya Aura™ SIP Enablement Services*, Issue 7.0, May 2009, Document Number 03-600768

[5] *Avaya one-X™ Deskphone Edition for 9600 Series SIP IP Telephones Administrator Guide Release 2.5*, Issue 5, November 2009, Document Number 16-601944

[6] *Avaya Aura™ SIP Enablement Services (SES) Implementation Guide*, Issue 6, May 2009, Document Number 16-300140

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

[8] *Avaya Aura™ Call Center 5.2 Automatic Call Distribution Reference*, Release 5.2, April 2009, Document Number 07-602568

[9] *Modular Messaging Multi-Site Guide Release 5.2*, November 2009

[10] *Modular Messaging for Microsoft Exchange Release 5.2 Installation and Upgrades*, Issue 1.0, November 2009

[11] *Application Notes for Avaya Aura™ Communication Manager and Avaya Aura™ SIP Enablement Services SIP Trunking with AT&T IP Toll Free Service – Issue 1.0* - https://devconnect.avaya.com/public/download/dyn/CMSES521_IPTF.pdf

AT&T IP Toll Free Service Description:

[12] *AT&T IP Toll Free Service Description* 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.