



## Avaya Solution & Interoperability Test Lab

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# Application Notes for Avaya Aura® Communication Manager 6.2, Avaya Aura® Session Manager 6.3, and Avaya Session Border Controller for Enterprise 4.0.5 with AT&T IP Transfer Connect Service – Issue 1.0

### Abstract

These Application Notes describe the steps for configuring Avaya Aura® Communication Manager 6.2 with SIP Network Call Redirection (NCR), Avaya Aura® Session Manager 6.3, and the Avaya Session Border Controller for Enterprise 4.0.5 over the AT&T IP Transfer Connect service using AT&T **AVPN** or **MIS/PNT** transport connections.

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

Note that these Application Notes are intended to supplement the separate document: *Applications Notes for Avaya Aura® Communication Manager 6.2, Avaya Aura® Session Manager 6.3 and Avaya Session Border Controller for Enterprise 4.0.5 with AT&T IP Toll Free SIP Trunk Service*.

AT&T is a member of the Avaya DevConnect Service Provider program. Information in these Application Notes has been obtained through compliance testing and additional technical discussions. Testing was conducted via the DevConnect Program by the Avaya Solution and Interoperability Test Lab.

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# 1. Introduction

These Application Notes describe the steps for configuring Avaya Aura® Communication Manager 6.2 with SIP Network Call Redirection (NCR), Avaya Aura® Session Manager 6.3, and the Avaya Session Border Controller for Enterprise 4.0.5, (referred to in the remainder of this document as *Avaya SBCE*), with the AT&T IP Transfer Connect service using AT&T **AVPN** or **MIS/PNT** transport connections<sup>1</sup>. 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<sup>2</sup> destinations based upon SIP redirection messages from Avaya Aura® Communication Manager.

The AT&T IP Transfer Connect service is typically used by enterprises that have multiple call centers that are separated geographically or otherwise not interconnected. Using SIP NCR, trunk-to-trunk routing of certain inbound calls at Avaya Aura® Communication Manager can be avoided by requesting that the AT&T network transfer the inbound caller to an alternate destination.

In addition, the Avaya Aura® Communication Manager SIP User-to-User Information (UII) feature can be utilized with the SIP NCR feature to transmit UII 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 UII 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 the separate document: *Applications Notes for Avaya Aura® Communication Manager 6.2, Avaya Aura® Session Manager 6.3 and Avaya Session Border Controller for Enterprise 4.0.5 with AT&T IP Toll Free SIP Trunk Service 1.0*. This document is listed in **Section 10** as reference document [1]. It is recommended that this AT&T IP Toll Free service document should be available as a reference during provisioning of the AT&T IP Transfer Connect service.

## 2. General Test Approach and Test Results

The test environment consisted of:

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

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<sup>1</sup> MIS/PNT transport does not support compressed RTP (cRTP), however AVPN transport does support cRTP

<sup>2</sup> Note that this is NOT the same as the “Alternate Destination Routing (ADR)” service option available with the AT&T IP Toll Free service.

## 2.1. Interoperability Compliance Testing

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

The compliance testing was based on a test plan provided by AT&T, for the functionality required for certification as a solution supported on the AT&T network. Calls were made from the PSTN across the AT&T network. The following features were tested as part of this effort:

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

**Note** – Documents used to provision the reference configuration are listed in **Section 10**. Specific references to these documents are indicated in the following sections by the notation [x], where x is the document reference number.

## 2.2. Test Results

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

- Inbound AT&T IP Transfer Connect service calls to Communication Manager VDNs, agents, and phones.
- Inbound AT&T IP Transfer Connect service calls that are immediately redirected by a Communication Manager vector (pre-answer redirection) back to the AT&T IP Transfer Connect service for redirection to an alternate destination.
- Inbound AT&T IP Transfer Connect service calls that are answered by a Communication Manager vector and then redirected (post-answer redirection) back to the AT&T IP Transfer Connect service for redirection to an alternate destination.
- Redirected AT&T IP Transfer Connect service calls per above arriving on Communication Manager VDNs, agents, and phones (e.g., Communication Manager as the target party for the redirected calls).
- Recovery from unsuccessful post-answer redirection attempts per above due to busy or error conditions on the alternate destination.
- Call and two-way talk path establishment between callers and Communication Manager agents/phones.

The above test objectives with limitations as noted in **Section 2.2.1** were verified.

### 2.2.1. Known Limitations

1. **Avaya SIP endpoints may generate three Bandwidth headers; b=TIAS:64000, b=CT:64, and b=AS:64, causing AT&T network issues.** Certain Avaya SIP endpoints (e.g., 9620, 9630, 9601, and 9608 models) may generate various Bandwidth headers depending on the call flow. It has been observed that sending these Bandwidth headers may cause issues with

AT&T services. Therefore an Avaya SBCE Signaling Manipulation Rule is used to remove these headers (see document [1], **Section 2.2.1, Item 4**).

- Note - It was found that when all three Bandwidth headers are sent, the Avaya SBCE will only pass one of the headers to AT&T and block the other two.
  - The Avaya SBCE support team has been notified and an MR submitted.
- 2. **302 redirections and 180 Ringing** - The IP Transfer Connect service specifies that 18x responses should not be used in conjunction with 302 redirection calls. Therefore ring back should not be specified in Communication Manager 302 redirection vectors (see **Section 6.3.1**).
- 3. **Communication Manager 18x responses and Refer** - Communication Manager 6.2 SIP trunk form may be configured to send either a 180 Ringing message (default), or a 183 Session Progress message (see **Section 6.2**). The message type selected alters Communication Managers behavior upon receipt of a Notify from AT&T during Refer calls. If 180 Ringing is selected, then Communication Manager will issue a BYE upon receipt of the Notify/Ringing message from AT&T. If 183 Session Progress is selected, then Communication Manager will issue a BYE upon receipt of the Notify/200OK from AT&T. In both cases the expected behavior was for the Avaya CPE to wait for AT&T IP Transfer Connect service to issue the BYE (in some cases a network BYE may arrive before Communication Manager issues one). However no issues were encountered during testing due to either behavior.
- 4. **Customers subscribing to AT&T IP Transfer Connect and IP Flexible Reach Enhanced Features services** - Communication Managers Network Call Redirection (NCR) feature is required to enable Refer and 302 call redirection with the AT&T IP Transfer Connect Service (see **Section 6.2**). With this feature enabled, Communication Manager will also use the SIP parameter *SendOnly* to signal any hold call conditions (as well as the *RecvOnly* response as required). The *SendOnly* SIP parameter is not currently supported by the AT&T IP Flexible Reach - Enhanced Features service (see document [2]). Therefore any customers subscribing to both AT&T IP Transfer Connect and AT&T IP Flexible Reach - Enhanced Features services via the same Communication Manager environment, should follow the procedures described in **Addendum 1** of this document.

## 2.3. Support

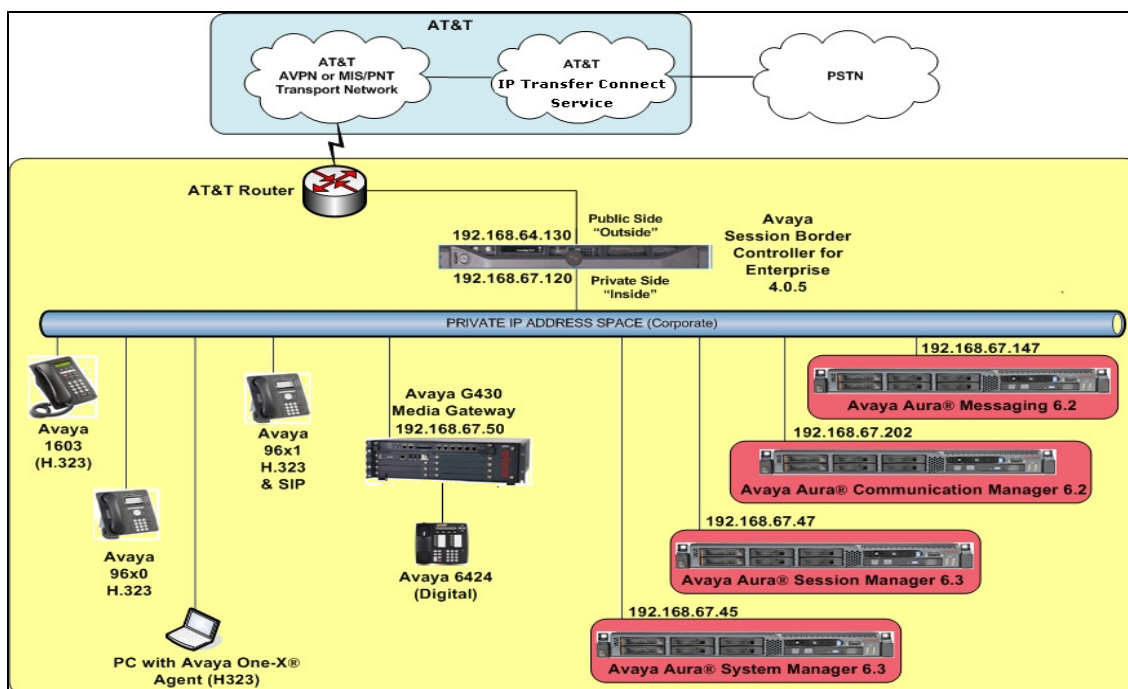
AT&T customers may obtain support for the AT&T IP Transfer Connect service by calling (800) 325-5555.

Avaya customers may obtain documentation and support for Avaya products by visiting <http://support.avaya.com>. In the United States, (866) GO-AVAYA (866-462-8292) provides access to overall sales and service support menus. Customers may also use specific numbers (provided on <http://support.avaya.com>) to directly access specific support and consultation services based upon their Avaya support agreements.

### 3. Reference Configuration

The reference configuration used in these Application Notes is shown in **Figure 1** and consists of several components:

- Session Manager 6.3 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. Avaya SIP endpoints register to Session Manager.
- System Manager 6.3 provides a common administration interface for centralized management of all Session Manager instances in an enterprise.
- Communication Manager 6.2 provides the voice communication services for a particular enterprise site. Avaya H.323 endpoints register to Communication Manager.
- The Avaya Media Gateway provides the physical interfaces and resources for Communication Manager. In the reference configuration, an Avaya G430 Media Gateway is used. This solution is extensible to other Avaya Media Gateways.
- Avaya desk telephones are represented with Avaya 1603(H.323), 960x Series IP Telephones (running H.323 firmware), and 96x1 Series IP Telephones (running H.323 or SIP firmware), Avaya 6424 Digital Telephones, as well as Avaya one-X® Agent soft phone ( H323).
- The Avaya SBCE 4.0.5 provides SIP Session Border Controller (SBC) functionality, including address translation and SIP header manipulation between the AT&T IP Transfer Connect service and the enterprise internal network.
- The AT&T IP Transfer Connect service uses SIP over UDP to communicate with enterprise edge SIP devices, e.g., the Avaya SBCE in this sample configuration. Session Manager may use SIP over UDP, TCP, or TLS to communicate with SIP network elements, e.g., the Avaya SBCE and Communication Manager. In the reference configuration, Session Manager uses SIP over TCP to communicate with the Avaya SBCE, and SIP over TCP and TLS to communicate with Communication Manager. UDP transport protocol is used between the Avaya SBCE and the AT&T IP Transfer Connect service.
- Avaya Messaging was used in the reference configuration to provide voice messaging capabilities. The provisioning of Avaya Messaging is beyond the scope of this document.
- Inbound calls were placed from PSTN via the AT&T IP Transfer Connect service, through the Avaya SBCE to Session Manager, which routed the call to Communication Manager. Communication Manager terminated the call to the appropriate agent/phone or fax extension.



**Figure 1: Reference configuration**

### 3.1. Illustrative Configuration Information

The specific values listed in **Table 1** below and in subsequent sections are used in the reference configuration described in these Application Notes, and are **for illustrative purposes only**. Customers must obtain and use the specific values for their own specific configurations.

**Note** - The AT&T IP Transfer Connect service Border Element IP address and DNIS digits, (destination digits specified in the SIP Request URIs sent by the AT&T Transfer Connect service) are shown in this document as examples. AT&T Customer Care will provide the actual IP addresses and DNIS digits as part of the IP Transfer Connect provisioning process.

Component	Illustrative Value in these Application Notes
<b>Avaya Aura® System Manager</b>	
Management IP Address	192.168.67.46
<b>Avaya Aura® Session Manager</b>	
Management IP Address	192.168.67.45
Network IP Address	192.168.67.47
<b>Avaya Aura® Communication Manager</b>	
IP Address	192.168.67.44
Avaya Aura® Communication Manager extensions	19xxx
Voice Messaging Pilot Extension	36000
<b>Avaya Session Border Controller for Enterprise (Avaya SBCE)</b>	
IP Address of Outside (Public) Interface (to	192.168.64.130

<b>Component</b>	<b>Illustrative Value in these Application Notes</b>
AT&T IP Transfer Connect Service)	
IP Address of Inside (Private) Interface (connected to Avaya Aura® Session Manager)	192.168.67.120
<b>Avaya Aura® Messaging</b>	
IP Address	192.168.67.147
Messaging Mailboxes	19xxx
<b>AT&amp;T IP Transfer Connect Service</b>	
Border Element IP Address	135.25.29.74

**Table 1: Illustrative Values Used in these Application Notes**

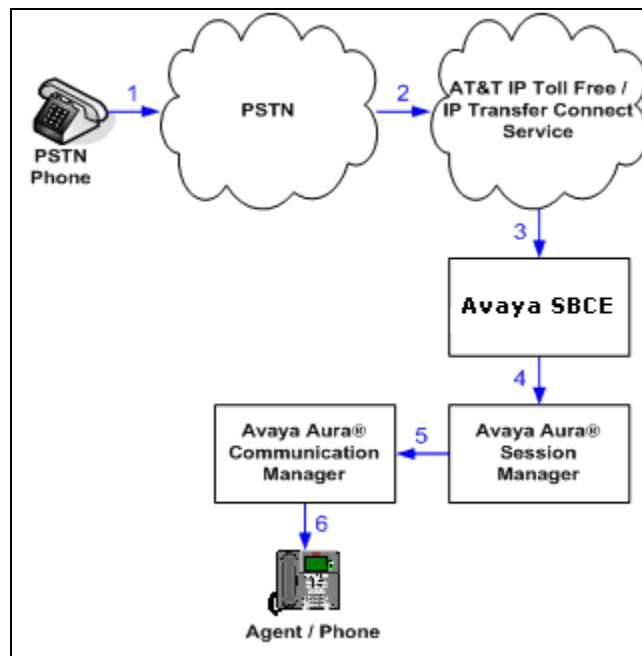


### 3.2. Call Flows

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

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

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

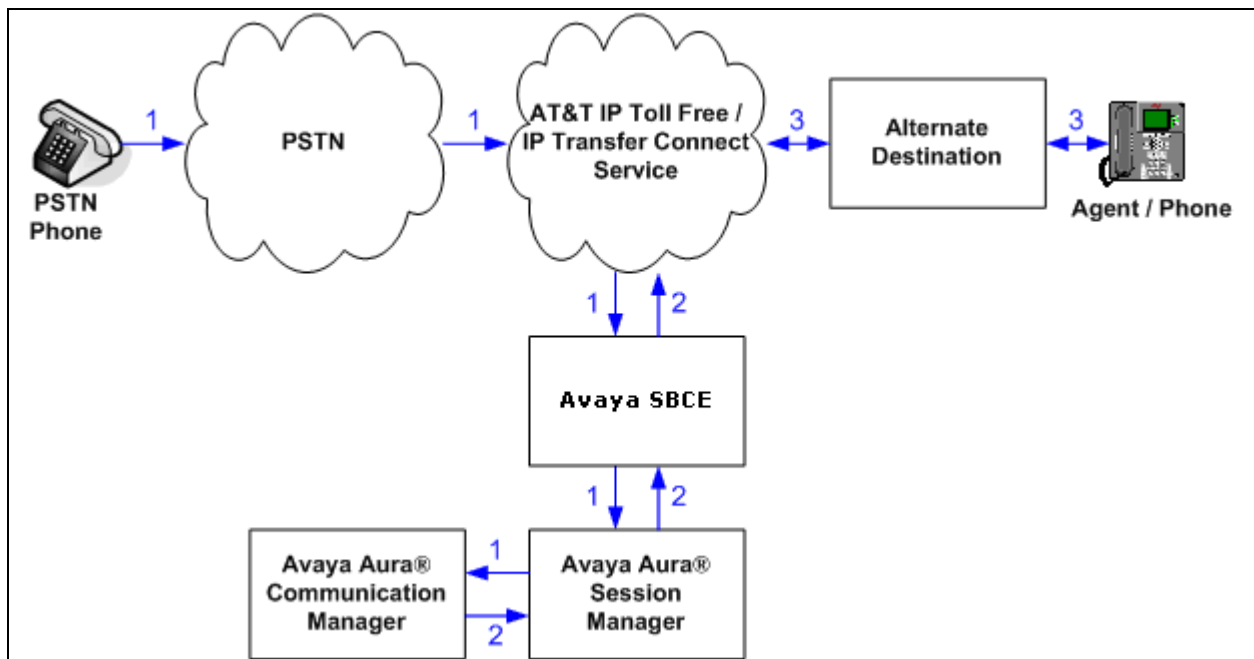


**Figure 2: Inbound AT&T IP Transfer Connect Call – No Redirection**

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

The second call scenario illustrated in **Figure 3** is an inbound AT&T IP Transfer Connect service call that arrives at the Avaya SBCE, sent to Session Manager, 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.

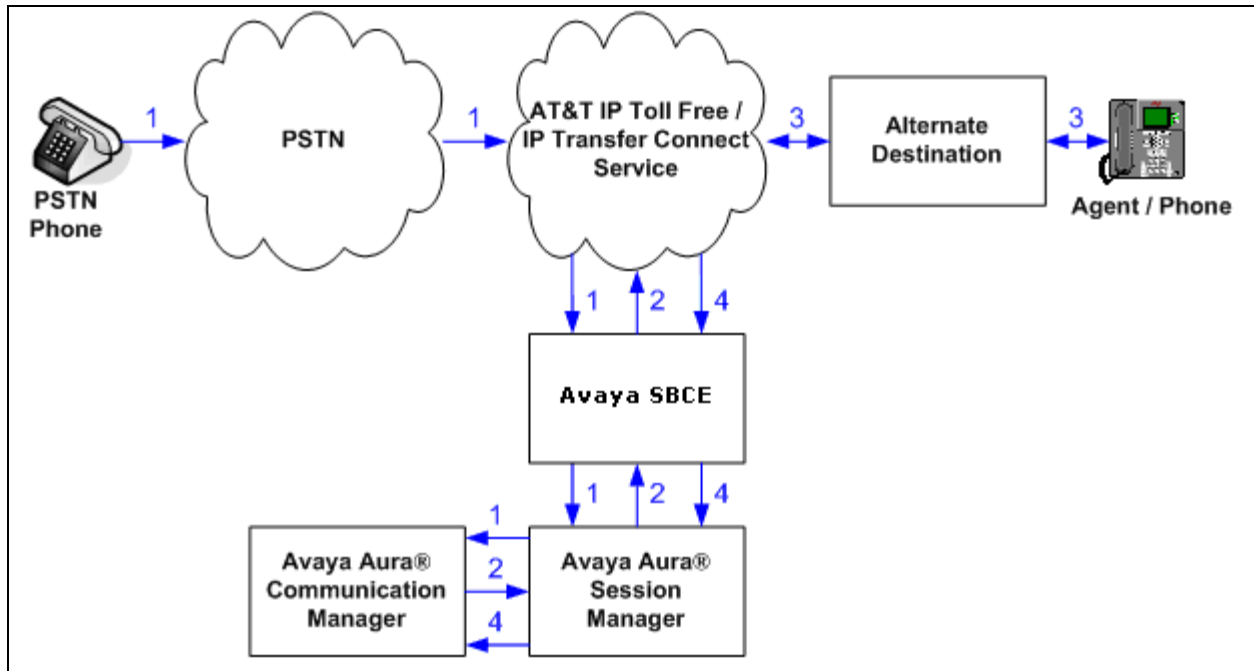
1. Same as the first five steps from the first call scenario.
2. Communication Manager routes the call to a vector, which redirects the call by sending a SIP 302 message back out on the SIP trunk on which the inbound call arrived. The SIP 302 message is routed back through Session Manager and then the Avaya SBCE sends the call to the AT&T IP Transfer Connect service network. Since the SIP 302 message is a final response, the redirecting party (Communication Manager) is no longer involved in the call whether the redirection succeeds or fails, and thereby releases the trunk.
3. The AT&T IP Transfer Connect service places a call to the alternate destination and upon answer, connects the calling party to the target party (alternate destination).



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

The third call scenario illustrated in **Figure 4** is an inbound AT&T IP Transfer Connect service call that arrives on the Avaya SBCE, then is transferred to 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 Avaya SBCE sends the call 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).

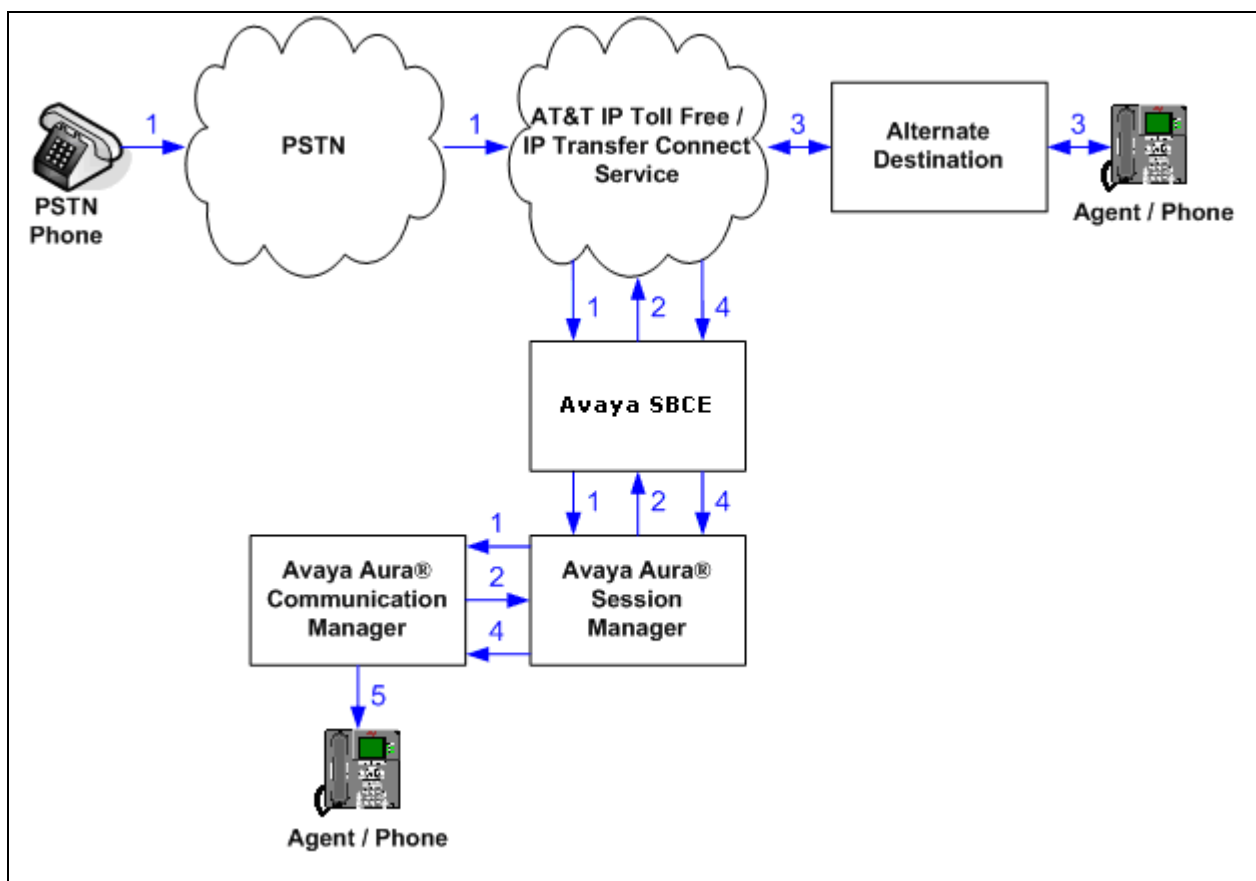


**Figure 4: Inbound AT&T IP Transfer Connect Call – Post-Answer SIP REFER Redirection**

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

1. Same as the third call scenario.
2. Same as the third call scenario.
3. The AT&T IP Transfer Connect service places a call to the target party (alternate destination), but the target party is busy or otherwise unavailable.
4. The AT&T IP Transfer Connect service notifies the redirecting/referring party (Communication Manager) of the error condition.
5. Communication Manager routes the call to a local agent or phone.

**Note:** This “error handling” scenario occurs only with AT&T IP Transfer Connect service lines enabled with the Attended IP Courtesy Transfer feature.



**Figure 5: Inbound AT&T IP Transfer Connect Call - Post-Answer SIP REFER Redirection. Refer is unsuccessful, Communication Manager re-routes to Agent.**

## 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 Proliant DL360 G7 server <ul style="list-style-type: none"> <li>System Platform</li> <li>Avaya Aura® System Manager</li> </ul>	<ul style="list-style-type: none"> <li>6.2.1.0.9 with patch 6.2.2.08001</li> <li>6.3 (6.3.1.9), with SP1 (r1212)</li> </ul>
IBM 8800 server <ul style="list-style-type: none"> <li>Avaya Aura® Session Manager</li> </ul>	<ul style="list-style-type: none"> <li>6.3.0.0.630039 with SP1(6.3.1.0.631004)</li> </ul>
IBM 8800 server <ul style="list-style-type: none"> <li>System Platform</li> <li>Avaya Aura® Communication Manager</li> </ul>	<ul style="list-style-type: none"> <li>6.2.1.0.9 with patch 6.2.2.08001</li> <li>6.2 SP6 (02.0.823.0-20558)</li> </ul>
Dell R610 <ul style="list-style-type: none"> <li>System Platform</li> <li>Avaya Aura® Messaging</li> </ul>	<ul style="list-style-type: none"> <li>6.2.1.0.9</li> <li>6.2 SP2 (6.2.02.0.823.0-109_0202) with patch (m62109cd_pt+a.rpm)</li> </ul>
Avaya G430 Media Gateway	32.26.0
MM711 Analog card	HW31 FW094
Dell R310 <ul style="list-style-type: none"> <li>Avaya Session Border Controller for Enterprise</li> </ul>	<ul style="list-style-type: none"> <li>4.0.5 Q19</li> </ul>
Avaya 96x0 IP Telephone	H.323 Version S3.2
Avaya 96x1 IP Telephone	H.323 Version S6.2313 SIP Version 6.2.1.26
Avaya 9601 IP Telephone	SIP version 6.1.5.12
Avaya one-X® Agent	2.5.1072.11103
Avaya 1603 IP Telephone	H323 (ha1603ua1_3200.bin)
Avaya 6424 Digital telephone	-
Windows PC/ Ventafax Home Version (Fax device)	6.1.59.144

**Table 2: Equipment and Software Versions**

## 5. Avaya Aura® Session Manager

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

### 5.1. Dial Patterns

Provision additional AT&T IP Transfer Connect service DNIS numbers (digits delivered in the Request URIs of inbound Invites), so they can be converted to Communication Manager VDN, Agent, or station extensions, according to the procedures described in **Section 5.8** of document [1].

## 6. Avaya Aura® Communication Manager

Communication Managers administration for interaction with the AT&T IP Toll Free service is described in document [1] and is applicable for the AT&T IP Transfer Connect service as well. This section describes the additional administration steps on Communication Manager necessary for supporting interaction with the AT&T IP Transfer Connect service. The steps are performed from the Communication Manager System Access Terminal (SAT) interface.

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

### 6.1. System Parameters

This section reviews the additional Communication Manager licenses and features that are required for supporting the interaction with the AT&T IP Transfer Connect service.

**NOTE - For any required features that cannot be enabled in the steps that follow, contact an authorized Avaya account representative to obtain the necessary licenses.**

1. Enter the **display system-parameters customer-options** command. On **Page 4** of the **system-parameters customer-options** form, verify that the **ISDN/SIP Network Call Redirection?** feature is set to “y”.

## OPTIONAL FEATURES

Emergency Access to Attendant? y	<b>IP Stations? y</b>
Enable 'dadmin' Login? y	
Enhanced Conferencing? y	ISDN Feature Plus? n
Enhanced EC500? y	<b>ISDN/SIP Network Call Redirection? y</b>
Enterprise Survivable Server? n	ISDN-BRI Trunks? y
Enterprise Wide Licensing? n	ISDN-PRI? y
ESS Administration? y	Local Survivable Processor? n
Extended Cvg/Fwd Admin? y	Malicious Call Trace? y
External Device Alarm Admin? y	Media Encryption Over IP? n
Five Port Networks Max Per MCC? n	Mode Code for Centralized Voice Mail? n
Flexible Billing? n	
Forced Entry of Account Codes? y	Multifrequency Signaling? y
Global Call Classification? y	Multimedia Call Handling (Basic)? y
Hospitality (Basic)? y	Multimedia Call Handling (Enhanced)? y
Hospitality (G3V3 Enhancements)? y	Multimedia IP SIP Trunking? y
IP Trunks? y	
IP Attendant Consoles? y	

(NOTE: You must logoff & login to effect the permission changes.)

2. On Page 6 of the **system-parameters customer-options** form, verify that the **ACD**, **EAS** and **Vectoring** features are set to “y”.

## CALL CENTER OPTIONAL FEATURES

Call Center Release: 6.0

<b>ACD? y</b>	Reason Codes? y
BCMS (Basic)? y	Service Level Maximizer? n
BCMS/VuStats Service Level? y	Service Observing (Basic)? y
BSR Local Treatment for IP & ISDN? y	Service Observing (Remote/By FAC)? y
Business Advocate? n	Service Observing (VDNs)? y
Call Work Codes? y	Timed ACW? y
DTMF Feedback Signals For VRU? y	<b>Vectoring (Basic)? y</b>
Dynamic Advocate? n	<b>Vectoring (Prompting)? y</b>
<b>Expert Agent Selection (EAS)? y</b>	<b>Vectoring (G3V4 Enhanced)? y</b>
EAS-PHD? y	<b>Vectoring (3.0 Enhanced)? y</b>
Forced ACD Calls? n	<b>Vectoring (ANI/II-Digits Routing)? y</b>
Least Occupied Agent? y	<b>Vectoring (G3V4 Advanced Routing)? y</b>
Lookahead Interflow (LAI)? y	<b>Vectoring (CINFO)? y</b>
Multiple Call Handling (On Request)? y	<b>Vectoring (Best Service Routing)? y</b>
Multiple Call Handling (Forced)? y	<b>Vectoring (Holidays)? y</b>
PASTE (Display PBX Data on Phone)? y	<b>Vectoring (Variables)? y</b>

(NOTE: You must logoff & login to effect the permission changes.)

## 6.2. Trunks

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

1. Enter the **change trunk-group x** command, where **x** is the number of the trunk group administered in document [1] for inbound AT&T IP Toll Free service calls (e.g., trunk 2). On **Page 4** of the **trunk-group** form, set **Network Call Redirection** to **y**.
2. Note whether the setting for **Convert 180 to 183 for Early Media?** is **n** (default) or **y**. The value defined will alter the Refer NOTIFY response behavior (see **Section 2.2.1, item 2**).

<b>change trunk-group 2</b>	<b>Page 4 of 21</b>
PROTOCOL VARIATIONS	
Mark Users as Phone? n	
Prepend '+' to Calling Number? n	
Send Transferring Party Information? n	
<b>Network Call Redirection? y</b>	
Send Diversion Header? n	
Support Request History? y	
Telephone Event Payload Type: 100	
<b>Convert 180 to 183 for Early Media? n</b>	
Always Use re-INVITE for Display Updates? n	
Identity for Calling Party Display: P-Asserted-Identity	
Block Sending Calling Party Location in INVITE? n	
Enable Q-SIP? n	

## 6.3. Inbound Call Routing

This section describes the steps for routing inbound AT&T IP Transfer Connect service calls to reach Vector Directory Numbers (VDNs) with corresponding programmable vectors. These vectors contain steps that invoke the Communication Manager SIP Network Call Redirection (NCR) functionality.

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

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

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

These Application Notes provide rudimentary vector definitions to demonstrate and test the SIP NCR and UII functionalities. In general, call centers will use vector functionality that is more complex and tailored to their individual needs. Call centers may also use customer hosts running applications used in conjunction with Avaya Aura® Application Enablement Services (AES) to



define call routing and provide associated UUI. The definition and documentation of those complex applications and associated vectors are beyond the scope of these Application Notes. Consult documents [7] and [8] for further information.

### 6.3.1. Pre-Answer Redirection - 302

This section provides an example of Pre-Answer Redirection. In this example, the inbound call is routed to the VDN shown in **Figure 7**, which invokes the vector shown in **Figure 8**. Note that the vector does not specify ring back (see **Section 2.2.1, item 1**). The vector does the following:

1. Assigns the data “**1234567890123456**” to ASAI UUI variable “**A**” (vector step **02**).  
**Note:** The parameters for ASAI UUI variables “**A**” and “**B**”, and other vector variables are defined using the **change variables** command (see **Figure 6**).
2. Redirects the call to the number “**1012**” (vector step **05**). Note that since this vector did not answer the call, the presence of the “~” in the “**route-to number**” line instructs Communication Manager to send a SIP 302 message with the number “**1012**” in the user part of the Contact header URI, (e.g., 1012@<host/domain>), to the AT&T IP Transfer Connect service (via Session Manager and the Avaya SBCE).

change variables						Page 1 of 39
VARIABLES FOR VECTORS						
Var	Description	Type	Scope	Length	Start	Assignment VAC
A	UuiTest1	asaiuui	L	16	1	
B						
C						

**Figure 6: Change Variables Form**

display vdn 19020		Page 1 of 3
VECTOR DIRECTORY NUMBER		
Extension: 19020		
Name*: 302		
Destination: Vector Number	22	
Attendant Vectoring? n		
Meet-me Conferencing? n		
Allow VDN Override? n		
COR: 1		
TN*: 1		
Measured: none		
VDN of Origin Annc. Extension*:		
1st Skill*:		
2nd Skill*:		
3rd Skill*:		
* Follows VDN Override Rules		

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

<b>display vector 22</b>	Page 1 of 6
CALL VECTOR	
Number: 22	Name: 302NoRingUII
Multimedia? n	Attendant Vectoring? n      Meet-me Conf? n      Lock? n
Basic? y	EAS? y    G3V4 Enhanced? y    ANI/II-Digits? y    ASAI Routing? y
Prompting? y	LAI? y    G3V4 Adv Route? y    CINFO? y    BSR? y    Holidays? y
Variables? y	3.0 Enhanced? y
01	
02	
03	
04 #	Define UII variable
05 set	B            = none    CATR    1234567890123456
06	
07 #	Redirect
08 route-to	number ~r1012                      with cov n if unconditionally
09 stop	
10	

**Figure 8: Sample Vector for Pre-Answer Redirection (302)**

### 6.3.2. Post-Answer Redirection - Refer

This section provides an example of Post-Answer Redirection. In this example, the inbound call is routed to the VDN shown in **Figure 9**, which invokes the vector shown in **Figure 10**. The vector does the following:

1. Assigns the data “**1234567890123456**” to ASAI UII variable “**A**” (vector steps **02**).  
**Note:** The parameters for UII variable “**A**” and other vector variables are defined using the **change variables** command (see **Figure 6**).
2. Answers the call to play an announcement (vector step **05**).
3. Attempts to redirect the call to the number “**1012**” (vector step **08**). Note that since this vector answered the call, the presence of the “~” in the “**route-to number**” line instructs Communication Manager to send a SIP REFER message with the number “**1012**” in the user part of the Refer-To header URI, (e.g., 1012@<host/domain>) to the AT&T IP Transfer Connect service (via Session Manager and the Avaya SBCE).

<b>display vdn 19010</b>	Page 1 of 3
VECTOR DIRECTORY NUMBER	
Extension: 19010	
Name*: REFER	
Destination: Vector Number	15
Attendant Vectoring? n	
Meet-me Conferencing? n	
Allow VDN Override? n	
COR: 1	
TN*: 1	
Measured: none	
VDN of Origin Annc. Extension*:	
1st Skill*:	
* Follows VDN Override Rules	

**Figure 9: Sample VDN for Post-Answer Redirection**

<b>display vector 15</b>	Page 1 of 6
CALL VECTOR	

```

Number: 15                      Name: Refer_UUI
Multimedia? n      Attendant Vectoring? n      Meet-me Conf? n      Lock? n
Basic? y      EAS? y      G3V4 Enhanced? y      ANI/II-Digits? y      ASAI Routing? y
Prompting? y      LAI? y      G3V4 Adv Route? y      CINFO? y      BSR? y      Holidays? y
Variables? y      3.0 Enhanced? y
01 #      Generate UUI
02 set      B      = none      CATR      1234567890123456
03
04 #      Play Refer announcement
05 announcement 42008
06
07 #      Refer occurs since this is post answer
08 route-to      number ~r1012      with cov n if unconditionally
09 #      If Refer fails play announcement and disconnect
10 disconnect      after announcement 42009
11
12

```

**Figure 10: Sample Vector for Post-Answer Redirection (Refer)**

## 7. Avaya Session Border Controller for Enterprise

The Avaya SBCE configuration for interaction with the AT&T IP Toll Free service provided in document [1] should also be followed for interoperability with the AT&T IP Transfer Connect service.

## 8. Verification Steps

### 8.1. Call Verification Tests

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

1. Place an inbound call to an AT&T IP Transfer Connect service line enabled with Redirect features. Verify that an appropriate Communication Manager vector immediately redirects the call back to the AT&T IP Transfer Connect service for redirection to an alternate destination using 302. Verify two-way talk path and transmission of UUI information as appropriate.
2. Place an inbound call to an AT&T IP Transfer Connect service line enabled with Refer features. Verify that an appropriate Communication Manager vector answers the call and then redirects the call back to the AT&T IP Transfer Connect service for redirection to an alternate destination using Refer. Verify two-way talk path and transmission of UUI information as appropriate.
3. Verify that when Communication Manager is the transfer target of redirected calls, the calls are answered with two-way talk path. Verify that the calls remain stable for several minutes and disconnect properly.

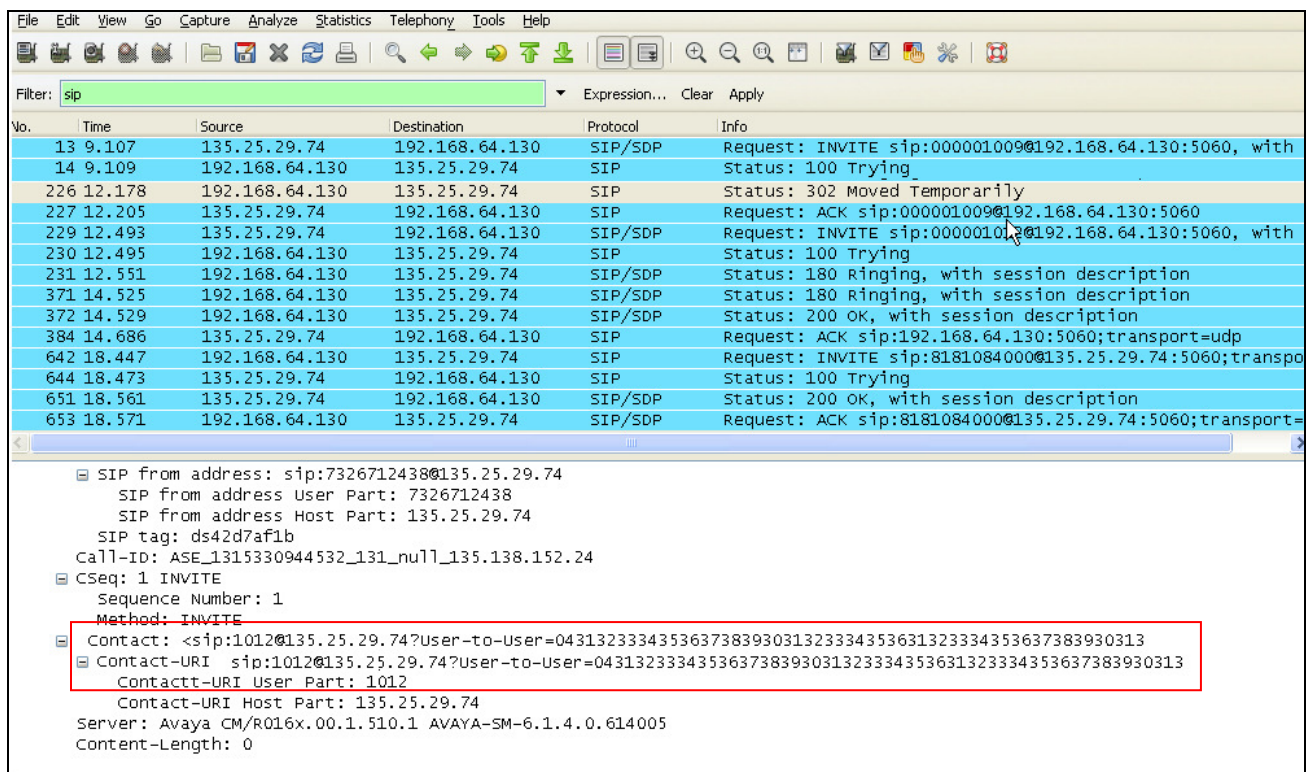
## 8.2. Protocol Traces

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

### 8.2.1. 302 Redirection

The following is an example of a 302 redirection call filtering on the SIP protocol. Note the following:

- The Contact header contains the new called number (1012) as defined in vector 22 (see **Section 6.3.1**).
- The User-to-User Information (UII) that is defined in vector 22.
- That 180 Ringing is not sent prior to the 302 Moved Temporarily (see **Section 6.3.1**).



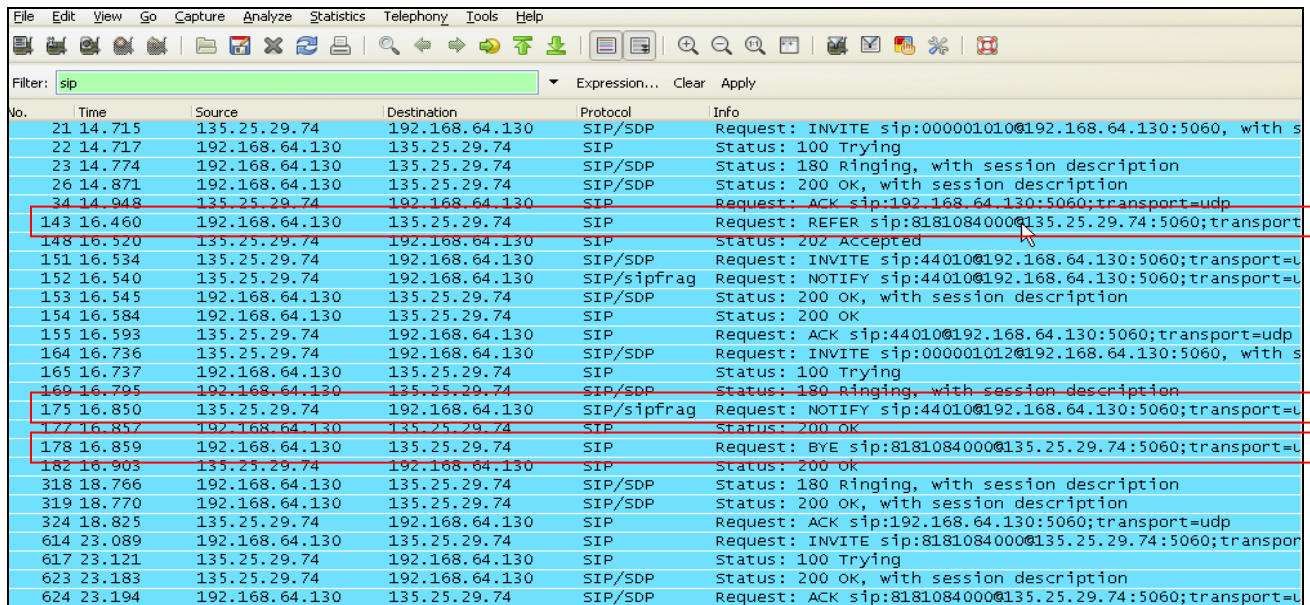
No.	Time	Source	Destination	Protocol	Info
13	9.107	135.25.29.74	192.168.64.130	SIP/SDP	Request: INVITE sip:000001009@192.168.64.130:5060, with
14	9.109	192.168.64.130	135.25.29.74	SIP	Status: 100 Trying
226	12.178	192.168.64.130	135.25.29.74	SIP	Status: 302 Moved Temporarily
227	12.205	135.25.29.74	192.168.64.130	SIP	Request: ACK sip:000001009@192.168.64.130:5060
229	12.493	135.25.29.74	192.168.64.130	SIP/SDP	Request: INVITE sip:000001012@192.168.64.130:5060, with
230	12.495	192.168.64.130	135.25.29.74	SIP	Status: 100 Trying
231	12.551	192.168.64.130	135.25.29.74	SIP/SDP	Status: 180 Ringing, with session description
371	14.525	192.168.64.130	135.25.29.74	SIP/SDP	Status: 180 Ringing, with session description
372	14.529	192.168.64.130	135.25.29.74	SIP/SDP	Status: 200 OK, with session description
384	14.686	135.25.29.74	192.168.64.130	SIP	Request: ACK sip:192.168.64.130:5060;transport=udp
642	18.447	192.168.64.130	135.25.29.74	SIP	Request: INVITE sip:8181084000@135.25.29.74:5060;transport=
644	18.473	135.25.29.74	192.168.64.130	SIP	Status: 100 Trying
651	18.561	135.25.29.74	192.168.64.130	SIP/SDP	Status: 200 OK, with session description
653	18.571	192.168.64.130	135.25.29.74	SIP/SDP	Request: ACK sip:8181084000@135.25.29.74:5060;transport=

[-] SIP from address: sip:7326712438@135.25.29.74 SIP from address User Part: 7326712438 SIP from address Host Part: 135.25.29.74 SIP tag: ds42d7af1b Call-ID: ASE_1315330944532_131_nu11_135.138.152.24 [-] CSeq: 1 INVITE Sequence Number: 1 Method: INVITE [-] Contact: <sip:1012@135.25.29.74?User-to-User=043132333435363738393031323334353631323334353637383930313 [-] Contact-URI sip:1012@135.25.29.74?User-to-User=043132333435363738393031323334353631323334353637383930313 Contact-URI User Part: 1012 Contact-URI Host Part: 135.25.29.74 Server: Avaya CM/R016x.00.1.510.1 AVAYA-SM-6.1.4.0.614005 Content-Length: 0
--

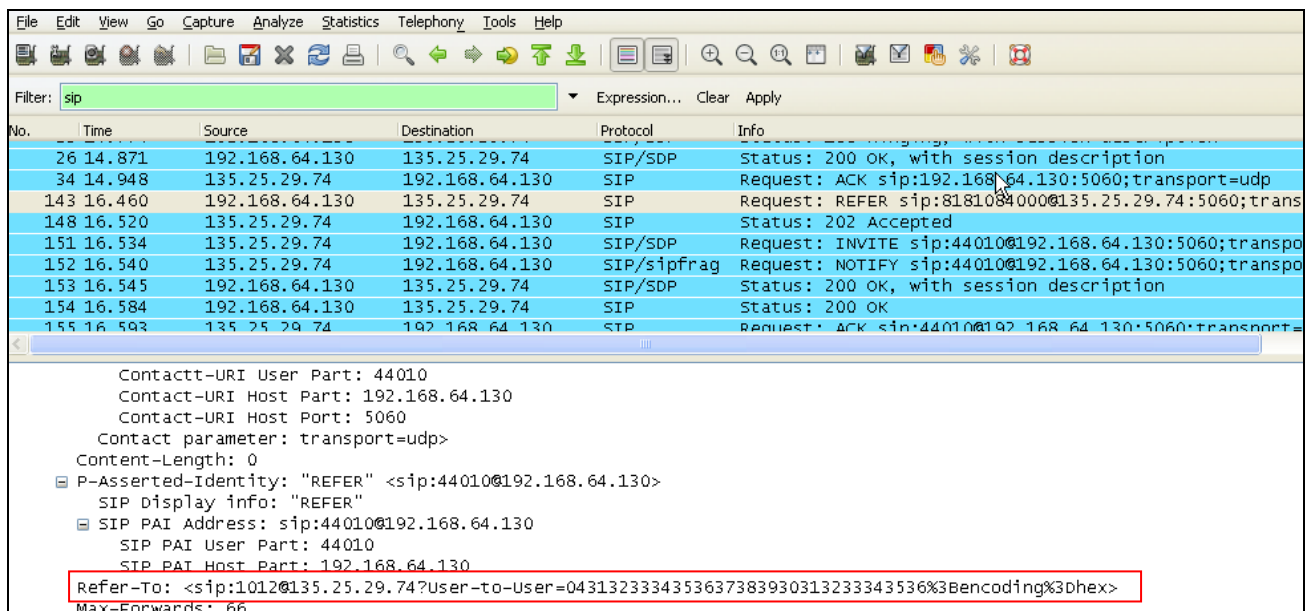
## 8.2.2. Refer with 180

The following is an example of a Refer (frame 143) redirection call filtering on the SIP protocol. Note that Communication Manager is sending 180 Ringing. As described in **Section 2.2.1, item 2**, this causes Communication Manager to send a BYE (frame 178) upon receipt of the Notify/180 Ringing sent by AT&T in frame 175.



No.	Time	Source	Destination	Protocol	Info
21	14.715	135.25.29.74	192.168.64.130	SIP/SDP	Request: INVITE sip:000001010@192.168.64.130:5060, with s
22	14.717	192.168.64.130	135.25.29.74	SIP	Status: 100 Trying
23	14.774	192.168.64.130	135.25.29.74	SIP/SDP	Status: 180 Ringing, with session description
26	14.871	192.168.64.130	135.25.29.74	SIP/SDP	Status: 200 OK, with session description
34	14.948	135.25.29.74	192.168.64.130	SIP	Request: ACK sip:192.168.64.130:5060;transport=udp
143	16.460	192.168.64.130	135.25.29.74	SIP	Request: REFER sip:8181084000@135.25.29.74:5060;transport=
148	16.520	135.25.29.74	192.168.64.130	SIP	Status: 202 Accepted
151	16.534	135.25.29.74	192.168.64.130	SIP/SDP	Request: INVITE sip:44010@192.168.64.130:5060;transport=u
152	16.540	135.25.29.74	192.168.64.130	SIP/sipfrag	Request: NOTIFY sip:44010@192.168.64.130:5060;transport=u
153	16.545	192.168.64.130	135.25.29.74	SIP/SDP	Status: 200 OK, with session description
154	16.584	192.168.64.130	135.25.29.74	SIP	Status: 200 OK
155	16.593	135.25.29.74	192.168.64.130	SIP	Request: ACK sip:44010@192.168.64.130:5060;transport=udp
164	16.736	135.25.29.74	192.168.64.130	SIP/SDP	Request: INVITE sip:000001012@192.168.64.130:5060, with s
165	16.737	192.168.64.130	135.25.29.74	SIP	Status: 100 Trying
169	16.795	192.168.64.130	135.25.29.74	SIP/SDP	Status: 180 Ringing, with session description
175	16.850	135.25.29.74	192.168.64.130	SIP/sipfrag	Request: NOTIFY sip:44010@192.168.64.130:5060;transport=u
177	16.857	192.168.64.130	135.25.29.74	SIP	Status: 200 OK
178	16.859	192.168.64.130	135.25.29.74	SIP	Request: BYE sip:8181084000@135.25.29.74:5060;transport=u
182	16.903	135.25.29.74	192.168.64.130	SIP	Status: 200 OK
318	18.766	192.168.64.130	135.25.29.74	SIP/SDP	Status: 180 Ringing, with session description
319	18.770	192.168.64.130	135.25.29.74	SIP/SDP	Status: 200 OK, with session description
324	18.825	135.25.29.74	192.168.64.130	SIP	Request: ACK sip:192.168.64.130:5060;transport=udp
614	23.089	192.168.64.130	135.25.29.74	SIP	Request: INVITE sip:8181084000@135.25.29.74:5060;transpor
617	23.121	135.25.29.74	192.168.64.130	SIP	Status: 100 Trying
623	23.183	135.25.29.74	192.168.64.130	SIP/SDP	Status: 200 OK, with session description
624	23.194	192.168.64.130	135.25.29.74	SIP/SDP	Request: ACK sip:8181084000@135.25.29.74:5060;transport=u

This screen shows the Refer in frame 143 in detail. The Refer-To header specifies the new called number (1012) as defined in vector 15 shown in **Section 6.3**. Also note the UII information also defined on vector 15.



No.	Time	Source	Destination	Protocol	Info
26	14.871	192.168.64.130	135.25.29.74	SIP/SDP	Status: 200 OK, with session description
34	14.948	135.25.29.74	192.168.64.130	SIP	Request: ACK sip:192.168.64.130:5060;transport=udp
143	16.460	192.168.64.130	135.25.29.74	SIP	Request: REFER sip:8181084000@135.25.29.74:5060;trans
148	16.520	135.25.29.74	192.168.64.130	SIP	Status: 202 Accepted
151	16.534	135.25.29.74	192.168.64.130	SIP/SDP	Request: INVITE sip:44010@192.168.64.130:5060;transpo
152	16.540	135.25.29.74	192.168.64.130	SIP/sipfrag	Request: NOTIFY sip:44010@192.168.64.130:5060;transpo
153	16.545	192.168.64.130	135.25.29.74	SIP/SDP	Status: 200 OK, with session description
154	16.584	192.168.64.130	135.25.29.74	SIP	Status: 200 OK
155	16.593	135.25.29.74	192.168.64.130	SIP	Request: ACK sip:44010@192.168.64.130:5060;transport=

Contact-URI User Part: 44010
Contact-URI Host Part: 192.168.64.130
Contact-URI Host Port: 5060
Contact parameter: transport=udp>
Content-Length: 0
P-Asserted-Identity: "REFER" <sip:44010@192.168.64.130>
SIP Display info: "REFER"
SIP PAI Address: sip:44010@192.168.64.130
SIP PAI User Part: 44010
SIP PAI Host Part: 192.168.64.130
Refer-To: <sip:1012@135.25.29.74?User-to=User=0431323334353637383930313233343536%3Bencoding%3Dhex>
Max-Forwards: 66

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

Filter: sip

No.	Time	Source	Destination	Protocol	Info
153	16.543	192.168.64.130	135.25.29.74	SIP/SDP	Status: 200 OK, with session description
154	16.584	192.168.64.130	135.25.29.74	SIP	Status: 200 OK
155	16.593	135.25.29.74	192.168.64.130	SIP	Request: ACK sip:44010@192.168.64.130:5060;transport=
164	16.736	135.25.29.74	192.168.64.130	SIP/SDP	Request: INVITE sip:000001012@192.168.64.130:5060, wi
165	16.737	192.168.64.130	135.25.29.74	SIP	Status: 100 Trying
169	16.795	192.168.64.130	135.25.29.74	SIP/SDP	Status: 180 Ringing, with session description
175	16.850	135.25.29.74	192.168.64.130	SIP/sipfrag	Request: NOTIFY sip:44010@192.168.64.130:5060;transport=
177	16.857	192.168.64.130	135.25.29.74	SIP	Status: 200 OK
178	16.859	192.168.64.130	135.25.29.74	SIP	Request: BYE sip:8181084000@135.25.29.74:5060;transport=

Contact: <sip:8181084000@135.25.29.74:5060;transport=udp>  
 Contact-URI: sip:8181084000@135.25.29.74:5060;transport=udp  
 Contact-URI User Part: 8181084000  
 Contact-URI Host Part: 135.25.29.74  
 Contact-URI Host Port: 5060  
 Contact parameter: transport=udp>  
 Event: refer;id=1  
 Subscription-State: active;expires=119  
 Content-Type: message/sipfrag;version=2.0  
 Message Body  
 sipfrag  
 SIP/2.0 180 Ringing

### 8.2.3. Refer with 183

The following is an example of a Refer (frame 134) redirection call filtering on the SIP protocol. Note that Communication Manager is sending 183 Session Progress. As described in **Section 2.2.1, item 2**, this causes Communication Manager to send a BYE (frame 323) upon receipt of the Notify/183 Session Progress sent by AT&T in frame 316.

Filter: sip

No.	Time	Source	Destination	Protocol	Info
11	8.186	135.25.29.74	192.168.64.130	SIP/SDP	Request: INVITE sip:000001010@192.168.64.130:5060, with s
12	8.188	192.168.64.130	135.25.29.74	SIP	Status: 100 Trying
13	8.204	192.168.64.130	135.25.29.74	SIP/SDP	Status: 183 Session Progress, with session description
18	8.302	192.168.64.130	135.25.29.74	SIP/SDP	Status: 200 OK, with session description
26	8.382	135.25.29.74	192.168.64.130	SIP	Request: ACK sip:192.168.64.130:5060;transport=udp
134	9.890	192.168.64.130	135.25.29.74	SIP	Request: REFER sip:8181084000@135.25.29.74:5060;transport=
139	9.948	135.25.29.74	192.168.64.130	SIP	Status: 202 Accepted
141	9.963	135.25.29.74	192.168.64.130	SIP/SDP	Request: INVITE sip:44010@192.168.64.130:5060;transport=
142	9.970	135.25.29.74	192.168.64.130	SIP/sipfrag	Request: NOTIFY sip:44010@192.168.64.130:5060;transport=
143	9.975	192.168.64.130	135.25.29.74	SIP/SDP	Status: 200 OK, with session description
145	10.013	192.168.64.130	135.25.29.74	SIP	Status: 200 OK
146	10.028	135.25.29.74	192.168.64.130	SIP	Request: ACK sip:44010@192.168.64.130:5060;transport=
154	10.169	135.25.29.74	192.168.64.130	SIP/SDP	Request: INVITE sip:000001012@192.168.64.130:5060, with s
155	10.171	192.168.64.130	135.25.29.74	SIP	Status: 100 Trying
158	10.217	192.168.64.130	135.25.29.74	SIP/SDP	Status: 183 Session Progress, with session description
164	10.271	135.25.29.74	192.168.64.130	SIP/sipfrag	Request: NOTIFY sip:44010@192.168.64.130:5060;transport=
165	10.277	192.168.64.130	135.25.29.74	SIP	Status: 200 OK
311	12.191	192.168.64.130	135.25.29.74	SIP/SDP	Status: 183 Session Progress, with session description
313	12.201	192.168.64.130	135.25.29.74	SIP/SDP	Status: 200 OK, with session description
316	12.245	135.25.29.74	192.168.64.130	SIP/sipfrag	Request: NOTIFY sip:44010@192.168.64.130:5060;transport=
318	12.251	135.25.29.74	192.168.64.130	SIP/sipfrag	Request: NOTIFY sip:44010@192.168.64.130:5060;transport=
319	12.252	192.168.64.130	135.25.29.74	SIP	Status: 200 OK
320	12.255	135.25.29.74	192.168.64.130	SIP	Request: ACK sip:192.168.64.130:5060;transport=udp
322	12.258	192.168.64.130	135.25.29.74	SIP	Status: 200 OK
323	12.259	192.168.64.130	135.25.29.74	SIP	Request: BYE sip:8181084000@135.25.29.74:5060;transport=
326	12.299	135.25.29.74	192.168.64.130	SIP	Status: 200 OK
622	16.641	192.168.64.130	135.25.29.74	SIP	Request: INVITE sip:8181084000@135.25.29.74:5060;transport=
625	16.671	135.25.29.74	192.168.64.130	SIP	Status: 100 Trying
630	16.733	135.25.29.74	192.168.64.130	SIP/SDP	Status: 200 OK, with session description
631	16.744	192.168.64.130	135.25.29.74	SIP/SDP	Request: ACK sip:8181084000@135.25.29.74:5060;transport=



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

No.	Time	Source	Destination	Protocol	Info
105	10.277	192.168.64.130	135.25.29.74	SIP	Status: 200 OK
311	12.191	192.168.64.130	135.25.29.74	SIP/SDP	Status: 183 Session Progress, with session description
313	12.201	192.168.64.130	135.25.29.74	SIP/SDP	Status: 200 OK, with session description
316	12.245	135.25.29.74	192.168.64.130	SIP/sipfrag	Request: NOTIFY sip:44010@192.168.64.130:5060;transport=udp
318	12.251	135.25.29.74	192.168.64.130	SIP/sipfrag	Request: NOTIFY sip:44010@192.168.64.130:5060;transport=udp
319	12.252	192.168.64.130	135.25.29.74	SIP	Status: 200 OK
320	12.255	135.25.29.74	192.168.64.130	SIP	Request: ACK sip:192.168.64.130:5060;transport=udp
322	12.258	192.168.64.130	135.25.29.74	SIP	Status: 200 OK
323	12.259	192.168.64.130	135.25.29.74	SIP	Request: BYE sip:8181084000@135.25.29.74:5060;transport=udp
326	12.299	135.25.29.74	192.168.64.130	SIP	Status: 200 OK
622	16.641	192.168.64.130	135.25.29.74	SIP	Request: INVITE sip:8181084000@135.25.29.74:5060;transport=udp
625	16.671	135.25.29.74	192.168.64.130	SIP	Status: 100 Trying
630	16.733	135.25.29.74	192.168.64.130	SIP/SDP	Status: 200 OK, with session description
631	16.744	192.168.64.130	135.25.29.74	SIP/SDP	Request: ACK sip:8181084000@135.25.29.74:5060;transport=udp

Call-ID: ASE_1317554708878_17554708878_135.25.29.74
<div> <div>CSeq: 5 NOTIFY</div> <div>Sequence Number: 5</div> <div>Method: NOTIFY</div> <div>Content-Length: 28</div> <div>Contact: &lt;sip:8181084000@135.25.29.74:5060;transport=udp&gt;</div> <div>Contact-URI: sip:8181084000@135.25.29.74:5060;transport=udp</div> <div>Contact-URI User Part: 8181084000</div> <div>Contact-URI Host Part: 135.25.29.74</div> <div>Contact-URI Host Port: 5060</div> <div>Contact parameter: transport=udp</div> <div>Event: refer;id=1</div> <div>Subscription-State: active;expires=117</div> <div>Content-Type: message/sipfrag;version=2.0</div> <div> <div>Message Body</div> <div> <div>Sipfrag</div> <div>SIP/2.0 183 Session Progress</div> </div> </div> </div>

## 9. Conclusion

As illustrated in these Application Notes, Avaya Aura® Session Manager 6.3, Avaya Aura® Communication Manager 6.2, and the Avaya Session Border Controller for Enterprise 4.0.5 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 (UII) features can work in complement with the AT&T implementations of SIP NCR and UII 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 (using Refer and 302 redirection), and deliver UII-encoded customer information to those alternate destinations for the purposes of invoking contact center applications, e.g., triggering agent screen pop-ups with caller information, etc. Both intra-site and IP Transfer Connect call scenarios were tested.

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

## 10. References

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

1. *Applications Notes for Avaya Aura® Communication Manager 6.2, Avaya Aura® Session Manager 6.3 and Avaya Session Border Controller for Enterprise 4.0.5 with AT&T IP Toll Free SIP Trunk Service 1.0*
2. *Application Notes for Avaya Aura® Communication Manager 6.2, Avaya Aura® Session Manager 6.3, and Avaya Session Border Controller for Enterprise 4.0.5, with AT&T IP Flexible Reach - Enhanced Features – Issue 1.0*

### **Avaya Aura® Session Manager/System Manager**

3. *Administering Avaya Aura® Session Manager*, Release 6.3, December, 2012
4. *Implementing Avaya Aura® Session Manager*, Release 6.3, March, 2013
5. *Implementing Avaya Aura® System Manager*, Release 6.3, Issue 1, December, 2012
6. *Administering Avaya Aura® System Manager*, Release 6.3, Issue 1.0, December, 2012

### **Avaya Aura® Communication Manager**

7. *Administering Avaya Aura® Communication Manager*, Release 6.2 03-300509, Issue 7, December, 2012
8. *Programming Call Vectors in Avaya Aura® Call Center*, 6.0, June 2010

### **Avaya Aura® Messaging**

9. *Administering Avaya Aura® Messaging*, Release 6.2, Issue 2.1, February, 2013

### **Avaya Session Border Controller for Enterprise**

Product documentation for UC-Sec version 4.0.5, can be obtained from Sipera using the link at <http://www.sipera.com>

10. *E-SBC 1U Installation Guide*, Release 4.0.5, Part Number: 101-5225-405v1.00, Release Date: November 2011
11. *E-SBC Administration Guide*, Release 4.0.5, Part Number: 010-5424-405v1.00, Release Date: November 2011



## **11. Addendum 1 – Additional Considerations for Customers that Subscribe to AT&T IP Transfer Connect and AT&T IP Flexible Reach - Enhanced Features**

As described in **Section 2.2.1, Item 3**, Communication Managers NCR feature is required to enable Refer and 302 call redirection with the AT&T IP Transfer Connect Service. With this feature enabled, Communication Manager will also use the SIP parameter *SendOnly* to signal any hold call conditions.

The AT&T IP Flexible Reach - Enhanced Features service also supports the use of Refer for specific call scenarios, and therefore NCR is enabled for that service as well. However the *SendOnly* SIP parameter is not currently supported by the AT&T IP Flexible Reach - Enhanced Features service. For this, and other reasons, the AT&T IP Flexible Reach - Enhanced Features service Application Notes [2], defines the use of a separate NCR enabled SIP trunk specifically for Refer based features (a SIP trunk with NCR disabled is defined for all other AT&T IP Flexible Reach traffic).

Customers that subscribe to both the AT&T IP Transfer Connect and AT&T IP Flexible Reach - Enhanced Features services via the same Communication Manager environment, should use the NCR enabled trunk(s) for both the AT&T IP Flexible Reach - Enhanced Features that support Refer, as well as for the AT&T IP Transfer Connect features that support Refer and 302 call redirection.

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