Avaya Solution & Interoperability Test Lab

Application Notes for Configuring Avaya Communication Manager SIP Network Call Redirection with AT&T IP Transfer Connect Service – Issue 1.1

Abstract

These Application Notes describe the steps for configuring the Avaya SIP Network Call Redirection (NCR) capability with the AT&T IP Transfer Connect service. The Avaya SIP NCR capability is a feature of an Avaya SIP trunking solution using Avaya Communication Manager (R5.0) and Avaya SIP Enablement Services (R5.0). The AT&T IP Transfer Connect service is a service option with the AT&T IP Toll Free service family. It supports the rerouting of inbound toll free calls to predefined destinations based upon redirection commands from Avaya Communication Manager. The Avaya SIP NCR capability can also utilize the SIP User-to-User Information (UUI) capability to transmit UUI between locations within SIP signaling messages. This capability is used in conjunction with the AT&T Information Forwarding option with the IP Transfer Connect 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 between the Avaya Solution and Interoperability Test Lab and the AT&T Virtual Interoperability Test Lab.

1. Introduction

These Application Notes describe the steps for configuring the Avaya SIP Network Call Redirection (NCR) capability with the AT&T IP Transfer Connect service. The Avaya SIP NCR capability is a feature of an Avaya SIP trunking solution using Avaya Communication Manager (R5.0) and Avaya SIP Enablement Services (R5.0). The AT&T IP Transfer Connect service is a service option of the AT&T IP Toll Free service family. IP Transfer Connect supports the rerouting of inbound toll free calls to predefined destinations based upon redirection commands from Avaya Communication Manager. This service is typically used within enterprise networks that have multiple geographically separated call centers. Using SIP NCR, trunk-to-trunk routing of certain calls at Avaya Communication Manager can be avoided by requesting the AT&T network to efficiently transfer an active call to an alternate predefined destination.

Note: AT&T IP Transfer Connect service does not support rerouting of inbound toll free calls to international destinations.

The Avaya SIP NCR capability can also utilize the SIP User-to-User Information (UUI) capability to transmit UUI between locations within SIP signaling messages. This capability is used in conjunction with the AT&T Information Forwarding option with IP Transfer Connect service to transmit a limited amount of call related data (typically less than 100 bytes) 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 query of database or the best service routing data exchanged between Avaya Communication Manager systems.

SIP (Session Initiation Protocol) is a standards-based communications approach designed to provide a common framework to support multimedia communication. SIP manages the establishment and termination of connections and the transfer of related information such as the desired codec, calling party identity, etc. Within these Application Notes, SIP is used as the signaling protocol between the Avaya components and the network service offered by AT&T. RFC 3261 [13], RFC 3515 [15] and a User-to-User information working draft [16] are the primary external specifications governing the SIP NCR capability.

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 between the Avaya Solution and Interoperability Test Lab and the AT&T Virtual Interoperability Test Lab.

1.1. Typical Enterprise Customer Location

Figure 1 illustrates a typical enterprise customer location using an Avaya Communication Manager based solution to support SIP trunking with AT&T. This typical configuration includes:

- Avaya Communication Manager providing the communication services for this customer location. Associated with Avaya Communication Manager is:

- An Avaya S8500 Server serving as the host processor for the Avaya Communication Manager software.
- An Avaya G650 Media Gateway supporting various VoIP resources and port cards.

- Avaya SIP Enablement Services (SES) operating on an Avaya S8500 server. Avaya SES serves as the SIP proxy between the AT&T services and one or more Avaya Communication Manager systems.
- Various Avaya telephones and other endpoints.
- IP routing and data network infrastructure to support IP connectivity between the enterprise location and the AT&T services.

SIP Network Call Redirection with AT&T IP Transfer Connect Service

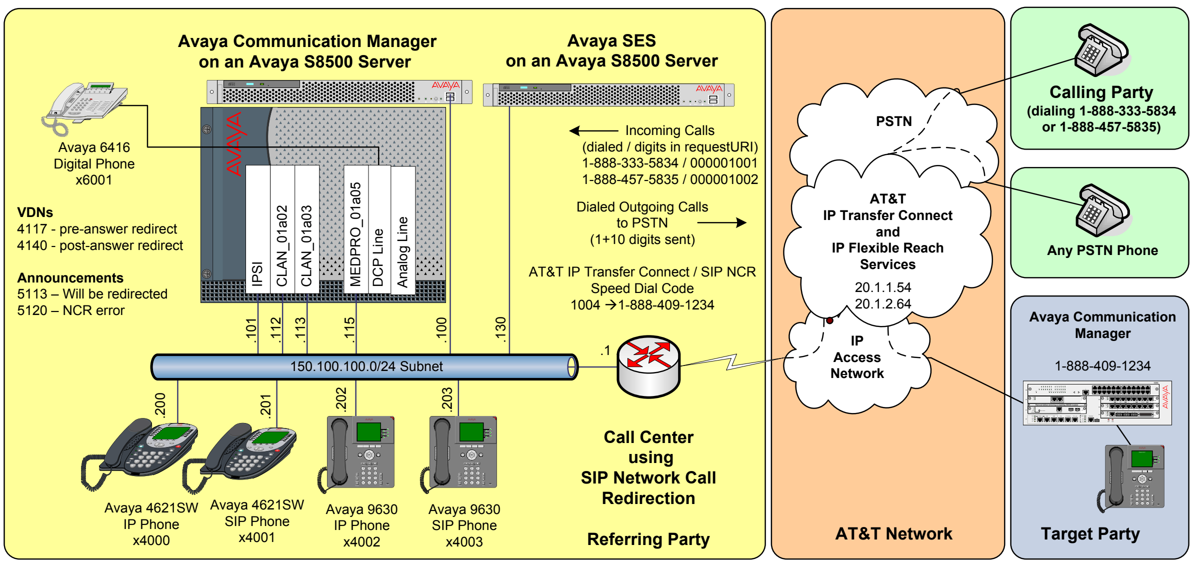

Figure 1 – SIP Network Call Redirection Configuration

1.2. AT&T Services Configuration Information

These Application Notes provide **an illustrative example** of how the Avaya SIP NCR capability is configured with the AT&T IP Transfer Connect service. These Application Notes assumes that the configuration for the AT&T IP Toll Free and IP Flexible Reach services are already in place (see Reference [1]).

The specific values provided below are illustrative only and must not be used for customer configurations. *Each customer must obtain the specific values for their configuration from AT&T during service provisioning of their AT&T IP Transfer Connect service.*

AT&T Provisioning Information	Illustrative Values in these Application Notes
AT&T Border Element IP Address(es)	20.1.1.54 20.1.2.64
G.729B, G.711MU Codecs Support	Yes
RFC 2833 (DTMF Event) Supported	Yes
Via Header Routing	Yes
IP Transfer Connect with Redirect Feature: PSTN Number Dialed	888 -333-5834
IP Transfer Connect with Redirect Feature: Digits Passed in SIP Request URI for 888-333-5834	000001001
IP Transfer Connect with Redirect Feature: Incoming Toll-Free Digits Passed in SIP To Header for 888 -333-5834	4154011001
IP Transfer Connect with Redirect Feature: Data Forwarding Option Enabled for 888-333-5834	Yes
IP Transfer Connect with Attended Courtesy Transfer Feature: PSTN Number Dialed	888-457-5835
IP Transfer Connect with Attended Courtesy Transfer Feature: Digits Passed in SIP Request URI for 800-333-5835	000001002
IP Transfer Connect with Attended Courtesy Transfer Feature: Incoming Toll-Free Digits Passed in SIP To Header for 800-333-5835	4154011002
IP Transfer Connect with Attended Courtesy Transfer Feature: Data Forwarding Option Enabled for 888-333-5835	Yes
IP Transfer Connect Speed Dial Code corresponding to destination 1-888-409-1234	1004

1.3. Call Flows

To understand how Avaya Communication Manager redirects calls, five call flows are described in this section. Although not shown in the figures that follow in this section, note that the Avaya SES serves as the SIP proxy between Avaya Communication Manager and the AT&T services. The Avaya SES is omitted from those figures simply for the purpose of focusing on the redirection capabilities of Avaya Communication Manager.

The first call scenario illustrated in **Figure 2** is a PSTN call to the enterprise site terminating on a telephone supported by Avaya Communication Manager. The inbound call arrives on Avaya Communication Manager on a trunk group enabled with SIP NCR, but no redirection is performed in this scenario.

1. A user on the PSTN dials a toll free number assigned by the AT&T IP Toll Free service to the customer location. The PSTN routes the call to the AT&T service. Based on that number, the AT&T service offers the call to Avaya Communication Manager using SIP signaling messages sent over the IP access facility. In this scenario, the called number received from the AT&T service corresponds to an extension assigned to an Avaya telephone, so Avaya Communication Manager terminates the call to the Avaya telephone.

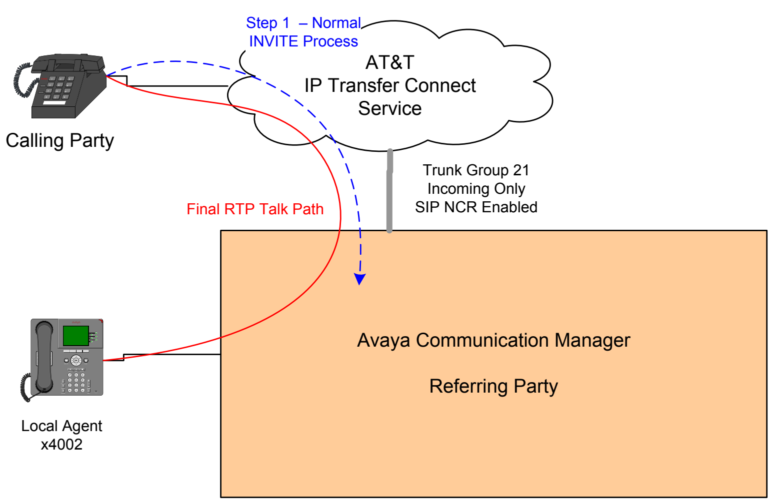

Figure 2 – SIP NCR Call Flow – Basic Incoming Call with No Redirection

The second call scenario illustrated in **Figure 3** is a PSTN call to the enterprise site that is immediately redirected by Avaya Communication Manager back to the AT&T service for routing to another destination (in **Figure 3**, the target party is another Avaya Communication Manager site).

1. As in the first call scenario, a user on the PSTN dials a toll free number assigned by the AT&T IP Toll Free service to the customer location, and the inbound call arrives on Avaya Communication Manager on a trunk group enabled with SIP NCR. In this scenario, the called number received from the AT&T service corresponds to an extension assigned to a Vector Directory Number (VDN), so Avaya Communication Manager routes the call to the VDN.

2. The vector associated with the VDN immediately redirects the call by sending a 302 Response back on the SIP NCR enabled trunk group. The 302 Response contains the speed dial code referencing the target party to which the call should be redirected. Since the 302 Response is a final response, Avaya Communication Manager drops off the call, thereby releasing the trunk.

3. Based on that number, the AT&T service offers the call to the target party and upon answer, the call is established between the calling party and the target party.

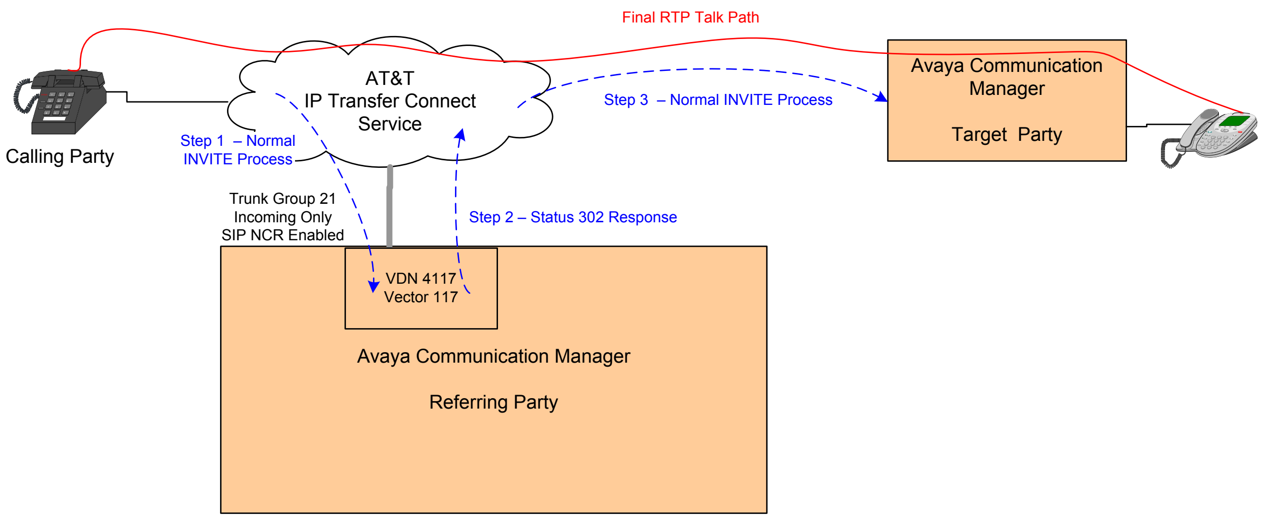

Figure 3 – SIP NCR Call Flow – Pre-Answer SIP 302 Redirection

The third call scenario illustrated in **Figure 4** is a PSTN call to the enterprise site that is initially answered by an Avaya Communication Manager announcement, and then redirected back to the AT&T service for routing to another destination (in **Figure 4**, the target party is another Avaya Communication Manager site).

1. As in the first call scenario, a user on the PSTN dial a toll free number assigned by the AT&T IP Toll Free service to the customer location, and the inbound call arrives on Avaya Communication Manager on a trunk group enabled with SIP NCR. In this scenario, the called number received from the AT&T service corresponds to an extension assigned to a Vector Directory Number (VDN), so Avaya Communication Manager routes the call to the VDN.

2. The vector associated with the VDN answers the call and plays an announcement. Following the announcement, the vector redirects the call by sending a REFER message back on the SIP NCR enabled trunk group. The REFER message contains the speed dial code referencing the target party to which the call should be redirected.

3. Based on that number, the AT&T service offers the call to the target party and upon answer, the call is established between the calling party and the target party.

4. The AT&T service also clears the call on the referring Avaya Communication Manager.

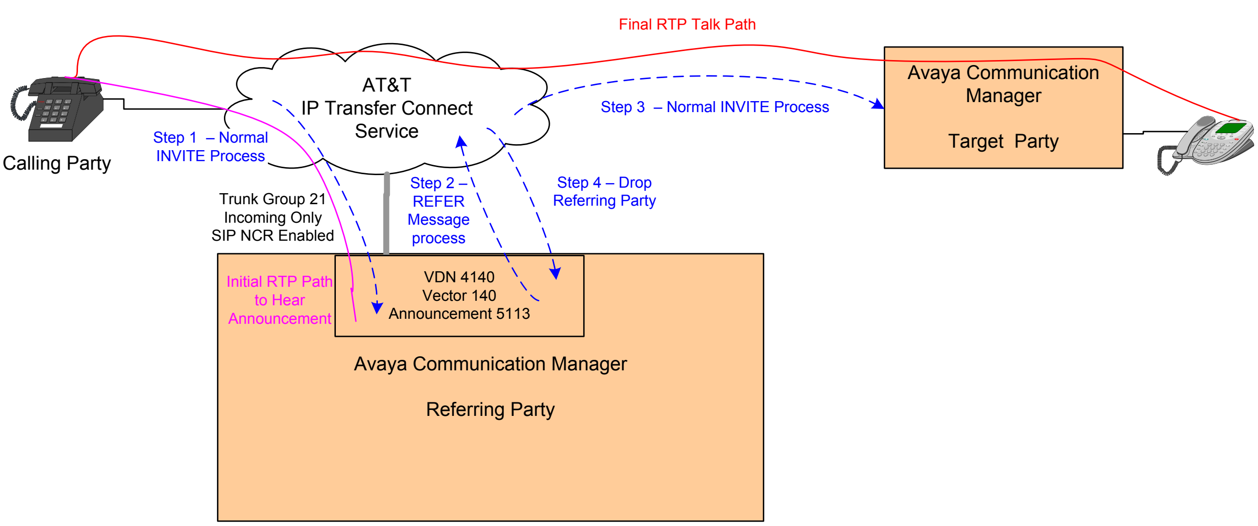

Figure 4 – SIP NCR Call Flow – Post-Answer SIP REFER Redirection

The fourth call scenario illustrated in **Figure 5** is similar to the third call scenario, except that the target party of the redirection is busy or otherwise unavailable.

1. Same as the third call scenario.
2. Same as the third call scenario.
3. The AT&T service offers the call to the target party, but the target party is busy or otherwise unavailable.
4. The AT&T service reports the error condition back to the referring party, and the vector routes the call to a local agent.

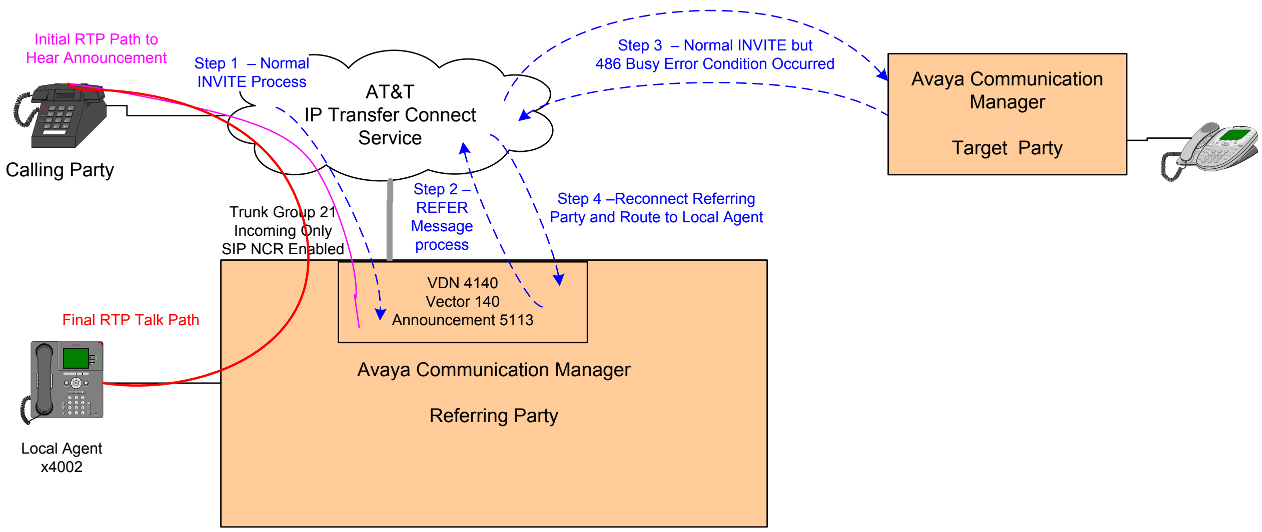

Figure 5 – SIP NCR Call Flow – Post-Answer SIP REFER Redirection with Target Party Busy – Call Answered by 2nd Choice Local Extension

The fifth call scenario illustrated in **Figure 6** is a PSTN call to the enterprise site that is terminated on a telephone supported by Avaya Communication Manager, and then transferred by the telephone to another PSTN destination.

1. Same as the first call scenario.

2. The local agent initiates the transfer, which puts the call on hold, and places a second call by dialing the number of the target PSTN destination. Auto Route Selection (ARS) routing in Avaya Communication Manager routes the second call out another trunk group to the AT&T IP Flexible Reach service. This outbound trunk group is NOT enabled with SIP NCR.

3. The AT&T service routes the call to the target PSTN destination, and the call is established between the local agent and the target PSTN destination.

4. The local agent completes the transfer and drops off the call, and talkpath is established between the calling party and the target PSTN destination. The media path between the two parties still goes through Avaya Communication Manager (trunks on trunk groups 21 and 11 below are occupied).

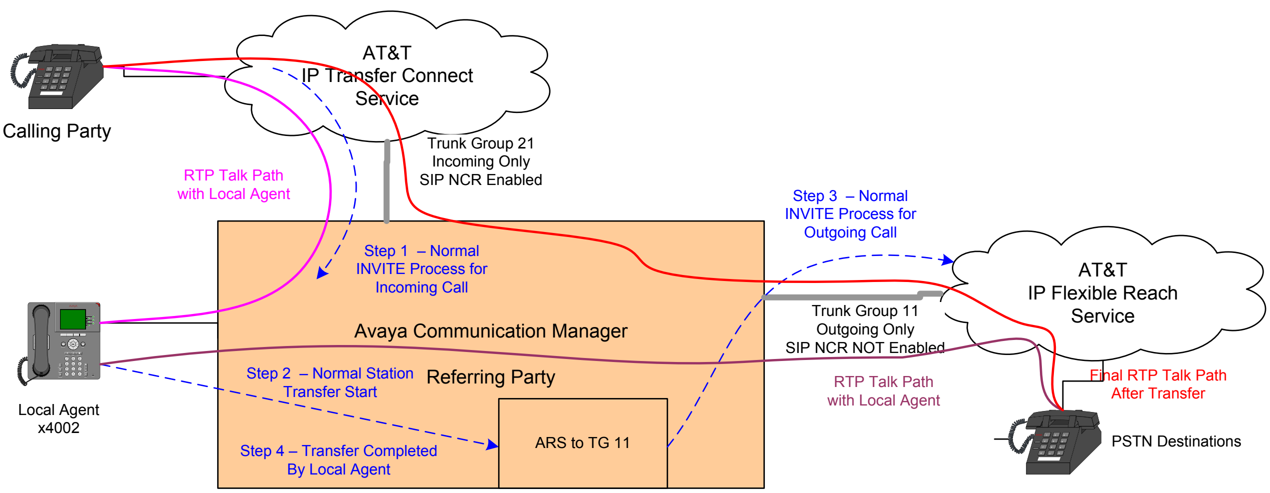

Figure 6 – SIP NCR Call Flow – Local Agent Transfers an Incoming NCR Trunk Call to a PSTN Destination.

2. Equipment and Software Validated

The following equipment and software was used during the DevConnect compliance testing with the AT&T services. This compliance testing is extensible to all other Avaya S8xxx series servers and Avaya Media Gateway platforms running the same version of Avaya Communication Manager and Avaya SIP Enablement Services.

Component	Version
Avaya	
Avaya S8500B Server	Avaya Communication Manager 5.0 (R015x.00.0.825.4 with update 15943)
Avaya G650 Media Gateway	
TN2312BP IP Server Interface (IPSI) TN799DP Control-LAN (C-LAN)	HW03 FW042 HW01 FW026
TN2602AP IP Media Processor (Medpro)	HW02 FW033
TN2224CP Digital Line	HW08 FW015
TN793B Analog Line	000005
Avaya 9630 one-X™ Deskphone Edition Telephone	Release S2.0
Avaya 4621SW IP (H.323) Telephone	Release 2.8.3
Avaya 9630 one-X™ Deskphone SIP Telephone	Release 2.0.4
Avaya 4621 SIP Telephone	Release 2.2.2
Avaya 6416D+M Digital Telephone	n/a
Avaya S8500B Server	Avaya SIP Enablement Services 5.0 (SES05.0-00.0.825.31)
AT&T	
IP Toll Free Service with IP Transfer Connect Service Option	Network Version VNI 13

Table 1 – Equipment and Version

NOTE: The solution integration validated in these Application Notes (Customer Configuration Guide) should be considered valid for deployment with Avaya Aura™ Communication Manager release 5.2 and Avaya Aura™ SIP Enablement Services release 5.2. Avaya agrees to provide service and support for the integration of Avaya Aura™ Communication Manager release 5.2 and Avaya Aura™ SIP Enablement Services release 5.2 with the AT&T IP Toll Free Service with IP Transfer Connect Service Option in compliance with existing support agreements for Avaya Communication Manager release 5.1.2 and Avaya SIP Enablement Services release 5.1.2 and in conformance with the integration guidelines as specified in the body of this document.

3. Configure Avaya Communication Manager

These Application Notes assume that Avaya Communication Manager and the Avaya SES are previously installed and configured according the Avaya document titled "Application Notes for Configuring SIP Trunking between AT&T IP Flexible Reach and IP Toll Free Services with Avaya Communication Manager / SIP Enablement Services SIP Trunking – Issue 1.0". Reference [1] provides the link to locate this document on the DevConnect section of the Avaya.com web site.

3.1. SIP Trunk Configuration

3.1.1. Verify System Capacity and Required Features

The Avaya Communication Manager license controls the customer options. Contact an authorized Avaya sales representative for assistance if insufficient capacity exists or a required feature is not enabled.

Verify that there is sufficient remaining Avaya Communication Manager SIP trunk capacity available for the SIP Trunks with the AT&T IP Transfer Connect service, taking into consideration other applications that may require Avaya Communication Manager SIP trunk resources. This is done by displaying Page 2 of the **System-Parameters Customer-Options** form. The number of SIP trunks available to add to new or existing trunk groups is the difference between the **Maximum Administered SIP Trunks** and the **USED** value.

```
display system-parameters customer-options                      Page   2 of  10
                                OPTIONAL FEATURES

IP PORT CAPACITIES                                              USED
                     Maximum Administered H.323 Trunks: 0     0
           Maximum Concurrently Registered IP Stations: 5     2
             Maximum Administered Remote Office Trunks: 0     0
Maximum Concurrently Registered Remote Office Stations: 0     0
              Maximum Concurrently Registered IP eCons: 0     0
  Max Concur Registered Unauthenticated H.323 Stations: 0     0
                  Maximum Video Capable H.323 Stations: 0     0
                   Maximum Video Capable IP Softphones: 0     0
                       Maximum Administered SIP Trunks: 100   50

   Maximum Number of DS1 Boards with Echo Cancellation: 0     0
                             Maximum TN2501 VAL Boards: 10    1
                     Maximum Media Gateway VAL Sources: 0     0
           Maximum TN2602 Boards with 80 VoIP Channels: 128   2
          Maximum TN2602 Boards with 320 VoIP Channels: 128   0
   Maximum Number of Expanded Meet-me Conference Ports: 0     0

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

Figure 7: System-Parameters Customer-Options Form – Page 2

Verify that the Automatic Route Selection (**ARS**) feature is enabled on Page 3 of the **System-Parameters Customer-Options** form.

```
display system-parameters customer-options                      Page   3 of  10
                                OPTIONAL FEATURES

    Abbreviated Dialing Enhanced List? n          Audible Message Waiting? n
        Access Security Gateway (ASG)? n              Authorization Codes? n
        Analog Trunk Incoming Call ID? n                       CAS Branch? n
 A/D Grp/Sys List Dialing Start at 01? n                         CAS Main? n
Answer Supervision by Call Classifier? n                Change COR by FAC? n
                                 ARS? y  Computer Telephony Adjunct Links? n
                ARS/AAR Partitioning? y   Cvg Of Calls Redirected Off-net? n
          ARS/AAR Dialing without FAC? n                      DCS (Basic)? n
          ASAI Link Core Capabilities? n                DCS Call Coverage? n
          ASAI Link Plus Capabilities? n               DCS with Rerouting? n
       Async. Transfer Mode (ATM) PNC? n
  Async. Transfer Mode (ATM) Trunking? n    Digital Loss Plan Modification? n
              ATM WAN Spare Processor? n                          DS1 MSP? n
                                ATMS? n             DS1 Echo Cancellation? n
                  Attendant Vectoring? n




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

Figure 8: System-Parameters Customer-Options Form – Page 3

Verify that the **ISDN/SIP Network Call Redirection** and **ISDN-PRI** features are enabled on Page 4 of the **System-Parameters Customer-Options** form.

```
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? y
                  Enhanced EC500? y        ISDN/SIP Network Call Redirection? y
    Enterprise Survivable Server? n                          ISDN-BRI Trunks? n
       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
        (NOTE: You must logoff & login to effect the permission changes.)
```

Figure 9: System-Parameters Customer-Options Form – Page 4

Verify that the **ACD** and **Vectoring** features shown are enabled on Page 6 of the **System-Parameters Customer-Options** form.

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

Figure 10: System-Parameters Customer-Options Form – Page 6

3.1.2. Determine Node Names

Use the "change node-names ip" command to view (or assign) the node names to be used in this configuration.

- "**ses**" and "**150.100.100.130**" are the **Name** and **IP Address**, respectively, of the Avaya SIP Enablement Services combined home-edge (or home) server interface where Avaya Communication Manager SIP trunk messages are sent.
- "**clan_01a03**" and "**150.100.100.113**" are the **Name** and **IP Address**, respectively, of the TN799DP C-LAN interface used for the SIP signaling group.

```
change node-names ip                                            Page   1 of   2
                                 IP NODE NAMES
    Name              IP Address

clan_01a03          150.100.100.113
default             0.0.0.0
medpro_01a05        150.100.100.115
procr               150.100.100.100
ses                 150.100.100.130
val_01a08           150.100.100.118
```

Figure 11: IP Node Names

3.1.3. Define IP Codec Set for AT&T SIP Trunk Calls

This configuration uses IP codec set 2 to assign G.729B, G729A and G.711mu codecs (in that priority) for voice calls traversing the SIP trunks to the AT&T services. T.38 will be used for Group 3 fax calls to PSTN connected fax machines via the AT&T services.

Using the "change ip-codec-set 2" command, enter "**G.729B**", "**G.729A**" and "**G.711MU**" as the **Audio Codec** values on Page 1 of the form. Retain the defaults for the remaining fields. On Page 2 of the form, enter "**t.38-standard**" for **FAX** and "**off**" for the **Modem** and **TTD/TTY** fields.

```
change ip-codec-set 2                                        Page   1 of   2

                          IP Codec Set

    Codec Set: 2

    Audio        Silence      Frames   Packet
    Codec        Suppression  Per Pkt  Size(ms)
 1: G.729B            n          2        20
 2: G.729A            n          2        20
 3: G.711MU           n          2        20
```

Figure 12: IP Codec Set 2 – Audio Codec Settings

```
change ip-codec-set 2                                        Page   2 of   2

                          IP Codec Set

                              Allow Direct-IP Multimedia? n


                   Mode                 Redundancy
    FAX            t.38-standard         0
    Modem          off                   0
    TDD/TTY        off                   0
    Clear-channel  n                     0
```

Figure 13: IP Codec Set 2 – Fax, Modem, and TTD/TTY Mode Settings

3.1.4. Verify Near End IP Network Region

These Application Notes use IP network region 1 (the normal default) for the G650 Media Gateway, the IP telephones and the C-LAN (in slot 1a03) used for IP telephone registration and the SIP signaling groups to the AT&T services. Verify this using the "display cabinet 1" command.

```
display cabinet 1
                                    CABINET
 CABINET DESCRIPTION
                  Cabinet: 1
           Cabinet Layout: G650-rack-mount-stack
             Cabinet Type: expansion-portnetwork

                 Location: 1         IP Network Region: 1

 Rack: row6          Room: sitl         Floor:             Building:

 CARRIER DESCRIPTION
   Carrier       Carrier Type        Number

      E        not-used              PN  01
      D        not-used              PN  01
      C        not-used              PN  01
      B        not-used              PN  01
      A        G650-port             PN  01
```

Figure 14: Display Cabinet Settings

3.1.5. Verify the C-LAN IP Network Region Assignment

In these Application Notes, the C-LAN was previously installed as part of the initial Avaya Communication Manager basic installation (using the procedures as described in [4]) and assigned the Node Name shown in **Figure 11**. Using the "display ip-interface 01a03" command (where 01 is the cabinet, a is carrier, and 03 is the slot of the respective C-LAN), verify the C-LAN is assigned to **Network Region 1**.

```
display ip-interface 01a03                                       Page   1 of   1
                                  IP INTERFACES

                  Type: C-LAN
                  Slot: 01A03
           Code/Suffix: TN799  D
             Node Name: clan_01a03
            IP Address: 150.100.100.113
           Subnet Mask: 255.255.255.0                            Link: 13
       Gateway Address: 150.100.100.1
  Enable Ethernet Port? y                     Allow H.323 Endpoints? y
        Network Region: 1                      Allow H.248 Gateways? y
                  VLAN: n                        Gatekeeper Priority: 5

 Target socket load and Warning level: 400
       Receive Buffer TCP Window Size: 8320
                                ETHERNET OPTIONS
                  Auto? n
                 Speed: 100Mbps
                Duplex: Full
```

Figure 15: IP Interface of C-LAN 1a03 used for SIP Signaling Group 3

3.1.6. Define IP Network Region

IP network regions set various IP network properties for SIP trunk groups and other IP elements (such as IP telephones, media processor cards, etc.) assigned to the region. In these Application Notes,

- IP network region 1 defines the properties for the main Avaya Communication Manager site previously configured during installation. These properties apply to items such as the IP telephones, IP Media Processors, etc., that are part of the local configuration.
- IP network region 11 is assigned to the AT&T services to allow codec preferences different from network region 1 to be used.
- IP network regions 1 and 11 are defined to be directly connected with 512 Kbps of bandwidth.
- IP codec-set 2 (defined in Section 3.1.3) will be used for calls between IP network regions 1 and 11.

Using the "change ip-network-region 1" command, enter on Page 1:

- **Name:** a descriptive string such as "Avaya CM Main Location".
- **Authoritative Domain:** the SIP domain of the Avaya SES (in this case "customer-sipdomain.com").
- **Codec Set:** the value "1" corresponding to the ip-codec-set (defined during initial configuration, not shown) for local calls between telephones on Avaya Communication Manager.
- **Intra-region IP-IP Direct Audio**: the value "yes" (the default).
- **Inter-region IP-IP Direct Audio:** the value "yes" (the default).

The IP-IP Direct Audio settings ensure the most efficient use of TN2602AP Media Processor resources. Defaults for the remaining values are used.

```
change ip-network-region 1                                      Page   1 of  19
                               IP NETWORK REGION
  Region: 1
Location: 1      Authoritative Domain: customer-sipdomain.com
    Name: Avaya CM Main Location
MEDIA PARAMETERS                Intra-region IP-IP Direct Audio: yes
      Codec Set: 1              Inter-region IP-IP Direct Audio: yes
   UDP Port Min: 2048                      IP Audio Hairpinning? n
   UDP Port Max: 3329
DIFFSERV/TOS PARAMETERS                  RTCP Reporting Enabled? y
 Call Control PHB Value: 46      RTCP MONITOR SERVER PARAMETERS
        Audio PHB Value: 46       Use Default Server Parameters? y
        Video PHB Value: 26
802.1P/Q PARAMETERS
 Call Control 802.1p Priority: 6
        Audio 802.1p Priority: 6
        Video 802.1p Priority: 5      AUDIO RESOURCE RESERVATION PARAMETERS
H.323 IP ENDPOINTS                                      RSVP Enabled? n
  H.323 Link Bounce Recovery? y
 Idle Traffic Interval (sec): 20
   Keep-Alive Interval (sec): 5
            Keep-Alive Count: 5
```

Figure 16: IP Network Region 1 – Page 1

Page 3 of the IP network region form is used to define the codec set and connectivity characteristics between IP network regions. Configure the "**src rgn 1 dst rgn 11**" row as follows (some rows in the figure below are deleted for clarity):

- **codec set**: enter "**2**", to use the codec choices defined in Section 3.1.3.
- **direct WAN**: enter "**y**" to indicate that regions 1 and 11 are directly connected.
- **WAN-BW-limits**: enter "**Kbits**" and "**512**"to indicate that the WAN access between IP network region 1 and 11 is 512 Kbps.

```
change ip-network-region 1                                      Page   3 of  19

                   Inter Network Region Connection Management

 src dst codec direct   WAN-BW-limits   Video                           Dyn
 rgn rgn  set   WAN   Units    Total Norm  Prio Shr Intervening-regions  CAC IGAR
 1   1    1
 1   2
 1   3
 1   4
 1   5
 1   6
 1   7
 1   8
 1   9
 1   10
 1   11   2     y    Kbits     512     0     0  y                             n
```

Figure 17: IP Network Region 1 – Page 3

Configure IP Network Region 11, using the "change ip-network-region 11" command. Enter:

- **Name:** a descriptive string such as "SIP Trks ATT"
- **Authoritative Domain:** leave blank (since no element in this region establishes a secure TLS link with the Avaya SES).
- **Codec Set:** the value "2" corresponding to the ip-codec-set defined in Section 3.1.3 for calls to the AT&T services.
- **Intra-region IP-IP Direct Audio:** the value "yes" (the default).
- **Inter-region IP-IP Direct Audio:** the value "yes" (the default).

```
change ip-network-region 11                                     Page   1 of  19
                               IP NETWORK REGION
  Region: 11
Location:          Authoritative Domain:
    Name: SIP Trks ATT
MEDIA PARAMETERS                Intra-region IP-IP Direct Audio: yes
      Codec Set: 2              Inter-region IP-IP Direct Audio: yes
   UDP Port Min: 20000                    IP Audio Hairpinning? n
   UDP Port Max: 20999
DIFFSERV/TOS PARAMETERS                 RTCP Reporting Enabled? y
 Call Control PHB Value: 46      RTCP MONITOR SERVER PARAMETERS
        Audio PHB Value: 46       Use Default Server Parameters? y
        Video PHB Value: 26
802.1P/Q PARAMETERS
 Call Control 802.1p Priority: 6
        Audio 802.1p Priority: 6
        Video 802.1p Priority: 5      AUDIO RESOURCE RESERVATION PARAMETERS
H.323 IP ENDPOINTS                                       RSVP Enabled? n
  H.323 Link Bounce Recovery? y
 Idle Traffic Interval (sec): 20
   Keep-Alive Interval (sec): 5
            Keep-Alive Count: 5
```

Figure 18: IP Network Region 11 – Page 1

Verify that Page 3 of the "change ip-network-region 11" command appears as shown below. The codec set and inter-region connectivity characteristics for the "**src rgn 11 dst rgn 1**" row were established during the configuration of IP network region 1.

```
change ip-network-region 11                                     Page   3 of  19

                   Inter Network Region Connection Management

 src dst codec direct   WAN-BW-limits   Video                           Dyn
 rgn rgn  set    WAN  Units    Total Norm  Prio Shr Intervening-regions  CAC IGAR
 11  1    2      y    Kbits     512     0     0  y                             n
```

Figure 19: IP Network Region 11 – Page 3

3.1.7. Define SIP Trunk Groups

An incoming only SIP trunk group (number 21) is defined for all incoming calls with the AT&T IP Transfer Connect, IP Toll Free and/or IP Flexible Reach DID services. A separate outbound only SIP trunk group (number 11) is defined to support outbound calls to the AT&T IP Flexible Reach service.

3.1.7.1 Establish the SIP Signaling Group

Using the "add signaling-group 11" command, configure signaling group 11 as follows:

- **Group Type**: set to "sip".
- **Transport Method**: automatically set to "tls". The Transport Layer Security (TLS) transport protocol is used between Avaya Communication Manager and the Avaya SES. Note this is not the transport protocol used to communicate between the Avaya SES and the AT&T services.
- **Near-end Node Name**: set to the C-LAN node name (defined in Section 3.1.2) used for the respective signaling group. In these Application Notes, "clan_01a03" is used for signaling group 11.

- **Far-end Node Name**: set to the Avaya combined home-edge (or home) SES. In these Application Notes, the node name "ses" is used as defined in Section 3.1.2.
- **Near-end Listen Port**: set to "5061", the default port for SIP signaling using tls transport.
- **Far-end Listen Port**: set to "5061".
- **Far-end Network Region**: set to "11", the network region defined for AT&T services as defined in Section 3.1.6.
- **Far-end Domain**: set to the Border Element IP address provided by AT&T as their node that will send and receive SIP messages. In these Application Notes, the IP address "20.1.1.54" will be used.
- **Direct IP-IP Audio Connections**: set to "y", indicating that the RTP paths should be optimized to reduce the use of media processing resources when possible.
- **DTMF over IP:** set to "rtp-payload". This value enables Avaya Communication Manager to send DTMF transmissions using RFC 2833 [14].

The default values for the other fields are used. The resulting form for signaling group 11 is shown below.

```
add signaling-group 11                                          Page   1 of   1
                               SIGNALING GROUP

Group Number: 11             Group Type: sip
                       Transport Method: tls








  Near-end Node Name: clan_01a03           Far-end Node Name: ses
Near-end Listen Port: 5061               Far-end Listen Port: 5061
                                      Far-end Network Region: 11
      Far-end Domain: 20.1.1.54

                                            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
```

Figure 20: Signaling Group 11

3.1.7.2 Establish an Incoming SIP Trunk Group

Using the "add trunk-group 21" command, configure trunk group 21 as follows.

On Page 1 of the Trunk Group form:

- **Group Type**: set to "sip".
- **Group Name**: enter a descriptive string such as "AT&T IB SIP TRKS".
- **TAC**: enter a trunk access code such as "#021".
- **Direction**: set to "incoming". This trunk group will not be used for outbound calls.
- **Service Type**: set to "public-ntwrk" for trunks to the PSTN.

- **Signaling Group**: set to "11" as defined within Section 3.1.7.1.
- **Number of Members**: set to the maximum number of simultaneous calls permitted for each trunk group. Within these Application Notes, "10" was used.

The default values are used on the remaining pages of the trunk-group form. The resulting form for trunk group 21 is shown below.

```
add trunk-group 21                                            Page   1 of  21
                                TRUNK GROUP

Group Number: 21                  Group Type: sip           CDR Reports: y
  Group Name: AT&T IB SIP TRKS           COR: 1        TN: 1        TAC: #021
   Direction: incoming       Outgoing Display? n
 Dial Access? n                                  Night Service:
Queue Length: 0
Service Type: public-ntwrk          Auth Code? n

                                                       Signaling Group: 11
                                                     Number of Members: 10
```

Figure 21: SIP Trunk Group 21 – Page 1

On Page 4 of the Trunk Group form, set **Network Call Redirection:** set to "y".

```
add trunk-group 21                                            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
             Telephone Event Payload Type: 127
```

Figure 22: SIP Trunk Group 21 – Page 4

The second inbound SIP trunk group and signaling group for incoming AT&T IP Transfer Connect service is defined in a similar manner but using the second AT&T Border Element IP address as the Far-end Domain value in the signaling group form. This configuration is not shown in these Application Notes. This is necessary to allow incoming calls to be received from either AT&T Border Element.

3.1.7.3 Establish an Outbound SIP Trunk Group

A separate trunk group 11 dedicated to outbound calls is created. Using the "add trunk-group 11" command, configure trunk group 11 as follows.

On Page 1 of the Trunk Group form:

- **Group Type**: set to "sip".
- **Group Name**: enter a descriptive string such as "AT&T OB SIP TRKS".
- **Direction**: set to "outgoing". This trunk group will not be used for inbound calls.
- **TAC**: enter a trunk access code such as "#011".
- **Service Type**: set to "public-ntwrk" for trunks to the PSTN.

- **Signaling Group**: set to “11” as defined within Section 3.1.7.1.
- **Number of Members**: set to the maximum number of simultaneous calls permitted for each trunk group. Within these Application Notes, “10” was used.

The default values are used on the remaining pages of the trunk-group form. The resulting form for trunk group 11 is shown below.

```
add trunk-group 11                                               Page   1 of  21
                                TRUNK GROUP

Group Number: 11                   Group Type: sip           CDR Reports: y
  Group Name: AT&T OB  SIP TRKS           COR: 1        TN: 1        TAC: #011
   Direction: outgoing       Outgoing Display? n
 Dial Access? n                               Night Service:
Queue Length: 0
Service Type: public-ntwrk          Auth Code? n

                                                       Signaling Group: 11
                                                     Number of Members: 10
```

Figure 23: SIP Trunk Group 11 – Page 1

On Page 4 of the Trunk Group form, set **Network Call Redirection:** set to “n”.

```
add trunk-group 11                                               Page   4 of  21

                              PROTOCOL VARIATIONS

                      Mark Users as Phone? n
            Prepend '+' to Calling Number? n
      Send Transferring Party Information? n
                 Network Call Redirection? n
             Telephone Event Payload Type: 127
```

Figure 24: SIP Trunk Group 11 – Page 4

3.1.8. Configure Calling Party Number Information

This step is not necessary for the incoming AT&T IP Transfer Connect service numbers.

The configuration required for outbound IP Flexible Reach calling (using trunk group 11) is covered in Reference [1].

3.1.9. Configure Call Routing

3.1.9.1 Outbound Calls

This step is not necessary for the incoming AT&T IP Transfer Connect service.

The ARS configuration required for outbound IP Flexible Reach calling (via trunk group 11) is covered in Reference [1]. The ARS Route Patterns must ensure that outbound calling does not attempt to use the inbound-only trunk group 21.

3.1.9.2 Incoming Calls

This section configures the routing of incoming AT&T IP Transfer Connect service calls to reach Vector Directory Numbers (VDNs) with corresponding programmable vectors. These vectors contain steps that invoke the Avaya SIP Network Call Redirection functionality.

The routing of incoming calls that do not invoke SIP NCR functionality is addressed within Reference [1].

Two different incoming call routing scenarios are described within these Application Notes:

- An incoming AT&T IP Transfer Connect call that invokes SIP NCR (using a 302 status response) prior to the call being answered.
- An incoming AT&T IP Transfer Connect call that invokes SIP NCR (using the SIP REFER message) following the call being answered. Note in this case, the answer is due to a vector.

The following incoming call treatment information is defined using the information provided by AT&T from **Section 1.2**.

AT&T Provided Information		Avaya Communication Manager Assignments	
Dialed PSTN Number	**Digits Received (within SIP INVITE message)**	**Vector Directory Number**	**Vector**
1-888-457-5834	000001001	4117	117
1-888-457-5835	000001002	4140	140

Table 2 - Incoming Number Assignments

The following AT&T speed dial codes are sent to AT&T in the SIP redirection messaging that routes calls to the alternate destination specified.

Note: The AT&T IP Transfer Connect service does not support redirection to PSTN numbers or any other numbers other than speed dial codes provided by AT&T.

AT&T Provided Information		
Dialed PSTN Number (not used)	**Speed Dial Code**	**Call Information Delivery (e.g., UUI) Supported**
1-888-904-4321	1004	Yes

Table 3 - Redirection Assignments

Begin the incoming call routing administration using the "change inc-call-handling-trmt trunk-group n" command (where "n" is the SIP trunk group number "21") to administer the incoming number routing. This administration must be done for each incoming SIP NCR trunk group.

- **Called Len**: enter the total number of incoming digits received (e.g., "9").
- **Called Number**: enter the specific digit pattern to be matched. (e.g., digits received in the SIP INVITE message).

- **Del**: enter the number of leading digits that should be deleted (e.g., “9”)
- **Insert**: enter the specific Vector Directory Number for processing the incoming call.

The completed inc-call-handling-trmt form for trunk group 21 is shown below.

```
change inc-call-handling-trmt trunk-group 21                 Page   1 of  30
                       INCOMING CALL HANDLING TREATMENT
 Service/      Called    Called      Del Insert
 Feature        Len      Number
 public-ntwrk     9  000001001       9   4117
 public-ntwrk     9  000001002       9   4140
```

Similar incoming call handling treatment should be repeated for the second incoming SIP trunk group configured to use the second AT&T Border Element IP address.

3.1.10. Vector Directory Number and Vector Definition

These Application Notes provide rudimentary vector definitions simply necessary to demonstrate and test the SIP NCR routing and UUI functionality. In general, call centers will use vector functionality that is more complicated and tailored to their individual needs. Call centers may also use customer hosts running applications used in conjunction with the Avaya Application Enablement Services to define call routing and provide associated UUI. The definition and documentation of those complex applications and associated vectors are beyond the scope of these Application Notes. Reference [2] on call center vectors should be consulted for further information.

3.1.10.1 Pre-Answer Redirection with Audible Ringback

In this case, after dialing 1-888-457-5834 a caller will hear 2 seconds of audible ringing and then the call will be automatically routed to the final destination at 1-888-904-4321. The caller will be unaware of this routing step.

From the perspective of Avaya Communication Manager, an incoming AT&T IP Transfer Connect call arrives with the incoming SIP Request URI digits 000001001 (corresponding to the 1-888-457-5834 dialed number). The Request URI digits are mapped to the vector directory number 4117 that invokes vector number 117 for call processing rules. The vector route-to step instructs the call to be redirected to the AT&T speed dial code 1004 (that corresponds to the 1-888-904-4321 final destination).

Also included in this case is a test of the User-to-User Information (UUI) feature that uses the Data Forwarding feature of the AT&T IP Transfer Connect service. Here a string (e.g., “123456790123456”) will be passed as a hex-encoded string within the User-to-User attribute of the Contact header within the SIP 302 status response. This UUI hex-encoded string will be delivered to the final destination within a User-to-User header of the SIP INVITE message.

Begin this configuration using the “add vdn 4117” command to configure vector directory number 4117 as follows:

- **Name**: enter a descriptive string such as “NCR Redirect wRingbackUui”.

- **Vector Number**: enter the number of the vector (.e.g, "117") that contains the call processing instructions

```
add vdn 4117                                                    Page   1 of   3
                            VECTOR DIRECTORY NUMBER
                             Extension: 4117
                                 Name*: NCR Redirect wRingbackUui
                         Vector Number: 117
                  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
```

The default values are used on the remaining pages of the vector directory number form.

Create a UUI variable for this test case using the "change variables" command and entering the following for **Var A**:

- **Description**: enter a descriptive string such as "UuiTest1".
- **Type**: enter "asaiuui".
- **Length**: enter "16".
- **Start**: enter "1".

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

Configure vector 117 using the "change vector 117" command. Note that this illustrative test case contains comments and blank steps to improve readability and understanding.

- **Name**: enter a short descriptive string such as "NcrRedirect_wUui".
- Steps **01, 04, 07, 10**: enter the comment symbol ("#""), a tab character, and a comment description such as "NCR Redirection with ringback and uui forwarding".
- Step **02:** enter the vector command "wait-time", tab, number of seconds (e.g., "2"), tab, unit of time (e.g., "secs"), tab, and type of treatment (e.g., "ringback").
- Step **05**: enter the vector command "set", tab, the **Var** defined in the change variable form above (e.g., "A"), tab, the variable reference "none", tab, the concatenate operation "CATR", tab, and a 16 digit UUI numeric value (e.g., "1234567890123456").
- Step **08:** enter the vector command "route-to", tab, "number", tab, and "~r" followed by the speed dial code (e.g., "1004") corresponding to the target destination.
- Step **11**: enter the vector command "announcement", tab, and the extension number (e.g., "5120") of a recorded announcement that will be heard only in the event of failure of the route-to command in step 07.

```
change vector 117                                               Page   1 of   6
```

```
                                   CALL VECTOR
    Number: 117              Name: NcrRedir_wUui
Multimedia? n                                 Meet-me Conf? n            Lock? n
     Basic? y   EAS? y   G3V4 Enhanced? y   ANI/II-Digits? n   ASAI Routing? n
 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 a UUI variable to send with the redirection
05 set          A      = none   CATR  1234567890123456
06
07 #    Immediate redirect to AT&T speed dial number
08 route-to     number ~r1004           with cov n if unconditionally
09
10 #    Play this announcement only on redirect failure
11 announcement 5120
12
```

3.1.10.2 Post Answer Redirection

In this case, after dialing 1-888-457-5835 a caller will hear a recorded announcement (e.g., "Welcome! You will soon be directed to the target destination") and then the call will be automatically routed to the final destination at 1-888-904-4321. Should an error occur (such the number being busy) during the routing to 1-888-904-4321, the caller will be routed to extension 4002 on the local Avaya Communication Manager.

From the perspective of Avaya Communication Manager, an incoming AT&T IP Transfer Connect call arrives with the incoming SIP Request URI digits 000001002 (corresponding to the 1-888-457-5835 dialed number). The Request URI digits are mapped to the vector directory number 4140 that invokes vector number 140 for call processing rules. The vector route-to step instructs the call to be redirected to the AT&T speed dial code 1004 (that corresponds to the 1-888-904-4321 final destination).

Also included in this case is a test of the User-to-User Information (UUI) feature and a demonstration of routing to a local extension should the redirection fail.

Begin this configuration using the "add vdn 4140" command and administer vector directory number 4140 as shown below.

```
change vdn 4140                                                 Page   1 of   3
                            VECTOR DIRECTORY NUMBER
                             Extension: 4140
                                 Name*: NcrReferwUuiError2LocalPhn
                         Vector Number: 140
                  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
```

Configure vector 140 using the "change vector 140" command as shown.

```
change vector 140                                               Page   1 of   6
                                  CALL VECTOR
    Number: 140              Name: ReferUuiErr2Phn
Multimedia? n                                Meet-me Conf? n           Lock? n
     Basic? y   EAS? y   G3V4 Enhanced? y   ANI/II-Digits? n   ASAI Routing? n
 Prompting? y   LAI? n  G3V4 Adv Route? y   CINFO? n   BSR? y   Holidays? n
 Variables? y   3.0 Enhanced? y
01 #    Refer with answer by local phone following error from target
02 announcement 5113
03
04 #    add uui and refer to target party
05 set          A      = none   CATR  1234567890123456
06 route-to     number ~r1004           with cov n if unconditionally
07
08 #    On error, send to local extension
09 route-to     number 4002             with cov n if unconditionally
10
11 #    Play this announcement on failure to reach local extension
12 announcement 5120
```

Note the following:

- Step **02** directs the incoming call to be answered and plays an announcement heard by the calling party.
- Step **05** uses the variable definition defined within **Section 3.1.10.1**.
- Step **06** instructs Avaya Communication Manager to redirect the call back to the network, using the speed dial code 1004 as the target destination.
- Step **09** instructs Avaya Communication Manager to route the call to the local extension 4002 if an error condition occurs during the routing to the target destination (e.g., “1004”).
- Step **12** occurs only if both steps **06** and **09** fail.

3.1.11. Save Avaya Communication Manager Changes

Use the “save translation” command to make the changes permanent.

4. Configure Avaya SIP Enablement Services

The Avaya SES is configured exactly as described within Section 4 of Reference [1] and is not repeated within these Application Notes.

SES **Media Server Address Maps** must be defined for the incoming AT&T IP Transfer Connect numbers as defined within **Table 2** within **Section 3.1.9.2** in these Application Notes.

5. Interoperability Compliance Testing

This section describes the interoperability compliance testing used to verify SIP Network Call Redirection interoperability with the AT&T IP Transfer Connect service. This section covers the general test approach and the test results.

5.1. General Test Approach

A simulated enterprise site consisting of an Avaya SIP telephony solution supporting SIP trunking was connected to a laboratory version of the AT&T IP Transfer Connect and IP Flexible Reach services via simulated managed access facilities. The enterprise site was configured as if using the generally available services provided by AT&T.

The Avaya Communication Manager incoming trunks were enabled for SIP Network Call Redirection functionality.

The AT&T IP Transfer Connect service was enabled to support redirection, attended courtesy transfer and UUI data forwarding, and UUI call information delivery options.

The following features and functionality were covered during the SIP trunking interoperability compliance testing:

- Incoming AT&T IP Transfer Connect service calls (from PSTN telephones) routed to local Avaya Communication Manager telephones and call center splits.
- Incoming AT&T IP Transfer Connect service calls to Avaya Communication Manager vector directory numbers executing the following SIP NCR functionality:
  - Pre-answer redirection (using the SIP 302 status response) using vectors.
  - Post-answer redirection (using REFER without a Replaces header) using vectors. (This is also known as the "one-channel transfer scenario".)
  - Error conditions during post-answer redirection with vector processing.
  - UUI Information forwarding in the above cases.
- Incoming AT&T IP Transfer Connect service calls answered by a telephone with subsequent transfers and/or conference (via a separate outbound SIP trunk group). In this scenario AT&T is not sent a REFER with a Replaces header (a.k.a. two-channel transfer scenario, not supported by the AT&T services).
- Direct IP-to-IP media (a.k.a "shuffling") with IP and SIP telephones.

5.2. Test Results

Interoperability testing of the sample configuration (as described in **Section 1.1**) was completed with successful results. No known problems exist with this configuration.

Note this configuration requires update 15943 for Avaya Communication Manger release 5.0 (load 825.4) as noted. Subsequent releases of Avaya Communication Manager are expected to incorporate this update within the primary release. This update includes the corrections contained in update 15653 used during compliance testing performed in Reference [1].

6. Verification Steps

6.1. Verification Tests

This section provides steps that may be performed to verify the operation of the SIP trunking configuration described in the Application Notes.

- Incoming Call to an Extension – Verify that calls placed from a PSTN telephone to an AT&T IP Transfer Connect number assigned are properly routed via the incoming SIP trunk group (with the SIP NCR option enabled) to the expected telephone, hunt group, ACD split, etc. Verify the talk-path exists in both directions, that calls remain stable for several minutes and disconnect properly.

- Incoming Call to an VDN with pre-answer redirection – Verify that calls placed from a PSTN telephone to an AT&T IP Transfer Connect number is routed properly to the pre-answer redirection VDN (e.g., 4117). The redirection should occur and the call should be routed to the target destination. Verify that no trunk-to-trunk connection exists following the redirection.

- Incoming Call to an VDN with post-answer redirection – Verify that calls placed from a PSTN telephone to an AT&T IP Transfer Connect number is routed properly to the post-answer redirection VDN (e.g., 4140). The caller should hear an announcement and then redirection should occur and the call should be routed to the target destination. Verify that no trunk-to-trunk connection exists following the redirection. Verify that the expected error-handling occurs (e.g. call routed to extension 4002) if the target destination is busy.

- Verify that UUI information is forwarded to the target destination. The Uui-Info feature on an Avaya Communication Manager telephone answering the call is one means to verify this operation.

- Direct IP-IP Connections – This applies if the incoming AT&T IP Transfer Connect calls are answered locally and IP and/or SIP telephones with the Direct IP-IP option is enabled. Verify that stable calls are using Direct IP-IP talk paths using the "status station" or "status trunk-group" commands. When Direct IP-IP is used, the Audio Connection field will indicate "ip-direct" instead of "ip-tdm".

- Verify that incoming IP Transfer Connect calls answered by an Avaya Communication Manager extension can be transferred to another destination via an outbound SIP trunk using the AT&T IP Flexible Reach service.

6.2. Troubleshooting Tools

The Avaya Communication Manager "list trace station", "list trace tac", "list trace vdn", list trace vector", "status station" and/or "status trunk-group" 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 SIP interoperability issues.

The "Trace Logger" function within the Avaya SES Administration Web Interface may be used to capture SIP traces between Avaya SES and the AT&T services. These traces can be instrumental in understanding SIP protocol issues resulting from configuration problems.

If port monitoring is available, a SIP protocol analyzer such as WireShark (a.k.a., Ethereal) to monitor the SIP messaging at the various interfaces is a very powerful tool for troubleshooting. Note that SIP messaging between Avaya Communication Manager and Avaya SES uses TLS encryption and cannot be viewed using WireShark.

7. Support

AT&T customers can get support for the AT&T IP Transfer Connect service and IP Toll Free Service by calling 1-800-325-5555. Support for the AT&T IP Flexible Reach Service should be directed to 1-877-288-8362.

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, 1-866-GO-AVAYA (1-866-462-8292) provides access to overall sales and service support menu. Customers may also use specific numbers (provided on support.avaya.com) to directly access specific support and consultation services based upon their Avaya support agreements.

8. Conclusion

These Application Notes describe the steps for configuring SIP trunking using the Avaya Communication Manager SIP Network Call Redirection feature with the AT&T IP Transfer Connect service. These Application Notes are intended to be used in conjunction with Reference [1] that provide the basis for general purpose SIP trunking with the AT&T IP Flexible Reach and IP Toll Free services.

The configuration shown in these Application Notes is representative of a typical 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] *Application Notes for Configuring SIP Trunking between AT&T IP Flexible Reach and IP Toll Free Services with Avaya Communication Manager / SIP Enablement Services SIP Trunking – Issue 1.0*, Available from the Avaya DevConnect web site at https://devconnect.avaya.com/public/download/dyn/AttSIPTrkcm5.pdf

[2] *Avaya Call Center Release 5.0 Call Vectoring and Expert Agent Selection (EAS) Guid*e, January 2008, Document ID 07-600780

[3] *Administrator Guide for Avaya Communication Manager*, January 2008, Issue 4.0, Release 5.0, Document Number 03-300509.

[4] *Adding New Hardware for Avaya Media Servers and Gateways*, January 2008, Issue 3, Release 5.0, Document Number 03-300684
[5] *Feature Description and Implementation for Avaya Communication Manager,* January 2008, Issue 6, Document Number 555-245-205
[6] *SIP Support in Avaya Communication Manager Running on the Avaya S8xxx Servers*, January 2008, Issue 8, Document Number 555-245-206.
[7] *4600 Series IP Telephone LAN Administrator Guide,* October 2007, Issue 7, Document Number 555-233-507
[8] *Installing, Administering, Maintaining and Troubleshooting SIP Enablement Services,* January 2008, Issue 5.0, Document Number 03-600768
[9] *Avaya Communication Manager Network Region Configuration Guide*, October 2005, Document ID 103244
[10] *Avaya one-X™ Deskphone Edition for 9600 Series IP Telephones Administrator Guide,* May 2007, Issue 4, Release 1.5, Document ID 16-300698

AT&T IP Flexible Reach and IP Toll Free Service Descriptions
[11] *AT&T IP Flexible Reach Overview,* *http://www.business.att.com/service_fam_overview.jsp?repoid=ProductSub-Category&repoitem=eb_ip_flexreach&serv_port=eb_voip&serv_fam=eb_ip_flexreach&segment=ent_biz*
[12] *AT&T IP Toll Free Overview,* *http://www.business.att.com/service_fam_overview.jsp?repoid=ProductSub-Category&repoitem=ip_toll-free&serv_port=eb_voip&serv_fam=ip_toll-free&segment=ent_biz*

Several Internet Engineering Task Force (IETF) standards track RFC documents were referenced within these Application Notes. The RFC documents may be obtained at: http://www.rfc-editor.org/rfcsearch.html.

[13] RFC 3261 - *SIP (Session Initiation Protocol)*, June 2002, Proposed Standard
[14] RFC 2833 - *RTP Payload for DTMF Digits, Telephony Tones and Telephony Signals*, May 2000, Proposed Standard
[15] RFC 3515 – *The Session Initiation Protocol (SIP) Refer Method*, April 2003, Proposed Standard
[16] Transporting User to User Call Control Information in SIP for ISDN Interworking, Working Draft – SIPPING Working Group, January 18, 2008 http://tools.ietf.org/id/draft-johnston-sipping-cc-uui-03.txt

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.