Avaya Solution & Interoperability Test Lab

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 1.1

Abstract

These Application Notes describe the steps for configuring SIP trunking between the AT&T IP Flexible Reach and IP Toll Free services and an Avaya SIP trunking solution using Avaya Communication Manager (R5.0), Avaya SIP Enablement Services (R5.0) and various Avaya telephony endpoints.

AT&T IP Flexible Reach and IP Toll Free are managed Voice over IP communication solutions using SIP trunks to provide inbound and outbound local, long distance, international and toll-free services for U.S. sites.

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 AT&T's Virtual Interoperability Test Lab.

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1. Introduction

These Application Notes describe the steps for configuring SIP trunking between the AT&T IP Flexible Reach and IP Toll Free services and an Avaya SIP trunking solution using Avaya Communication Manager (Release 5.0), Avaya SIP Enablement Services (SES) (Release 5.0) and various Avaya telephony endpoints.

SIP (Session Initiation Protocol) is a standards-based communications approach designed to provide a common framework to support multimedia communication. RFC 3261 [11] is the primary specification governing this protocol. 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 services offered by AT&T.

The AT&T IP Flexible Reach and IP Toll Free are managed Voice over IP communication services using SIP trunks to provide inbound and outbound local, long distance, international and toll-free services for U.S. sites. In the remainder of this document, "AT&T services" will be used to generically refer to both AT&T IP Flexible Reach and AT&T IP Toll Free services.

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 AT&T's 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 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.

For simplicity, aspects that may exist in customer configurations but are beyond the scope of these Application Notes are not addressed. Specifically,

- Avaya Communication Manager operates on the entire family of Avaya S8xxx servers and supports configurations with all of Avaya's Gxxx Media Gateways. The concepts

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presented in these Application Notes apply to all valid configurations but the specifics may vary with different S8xxx Servers and Gxxx Media Gateways.

- Customer locations will generally have additional or alternate routes to the PSTN using analog or digital TDM trunks.
- The configuration of Avaya 9600 and 4600 series SIP telephones.
- IP Network Address Translation (NAT), firewalls, Application Layer Gateway (ALG), and Session Border Controller (SBC) devices that may exist between the AT&T services and the Enterprise's communications infrastructure are not explicitly shown.   These devices generally must be SIP-aware and configured properly for SIP trunking to function properly. When configured correctly, they are transparent to the Avaya communications infrastructure.
- Enterprise networks often have multiple geographic locations tied together via an IP network. These configurations may be supported by one or more Avaya Communication Manager systems and/or multiple media gateways. An Avaya SES edge serves as the SIP proxy gateway interfacing with SIP trunking to the AT&T services. (Note this SES edge may be a high availability duplex server configuration).
- Enterprise networks of sufficient size or complexity may use the Avaya SIP Enablement Services in a separate home and edge server configuration. This configuration is functionally equivalent to the SES combined home-edge server solution illustrated in these Application Notes. As appropriate, special notes have been included to clarify specific administration details related to the SES separate home and edge server configuration.

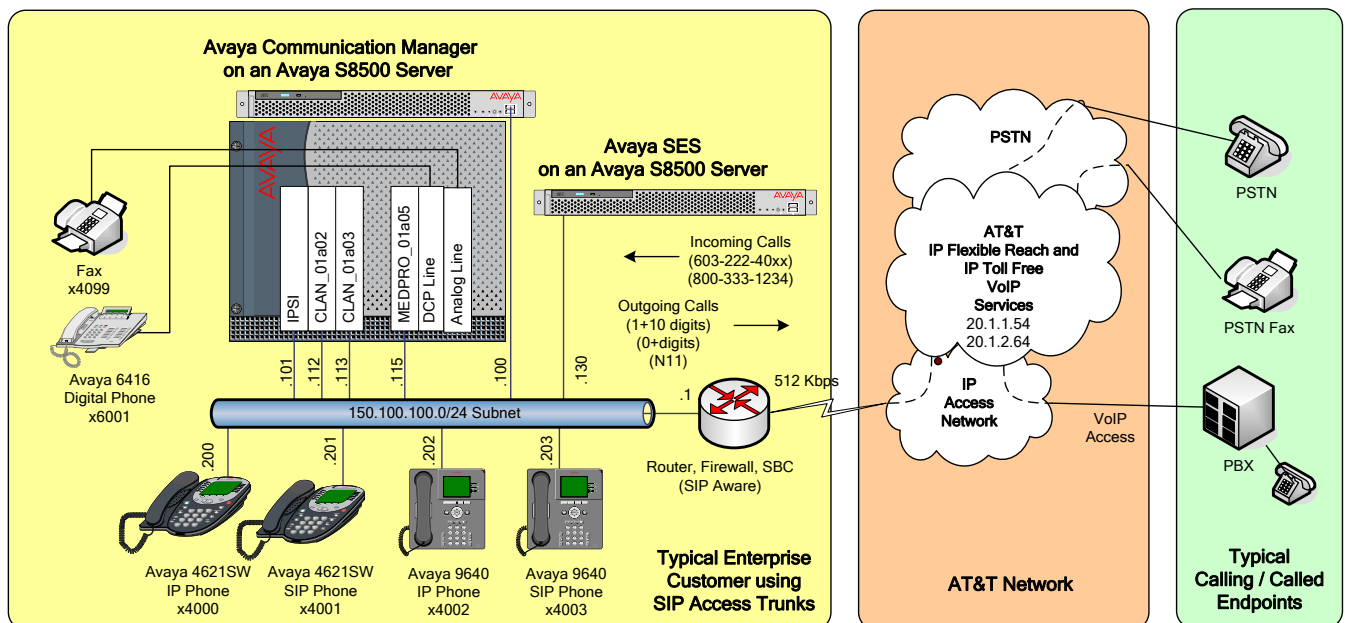## SIP Trunking with AT&T IP Flexible Reach and IP Toll Free Services

Figure 1 – Typical SIP Trunking Configuration

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1.2. AT&T Services Configuration Information

These Application Notes provide **an illustrative example** of how the Avaya SIP trunking solution is configured with the AT&T IP Flexible Reach and IP Toll Free services. These Application Notes do not address the configuration used for the Avaya SIP Network Call Redirection feature with the AT&T IP Transfer Connect service option (of the AT&T IP Toll Free service).

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 Flexible Reach or IP Toll Free services.*

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
Assigned Direct Inward Dial (DID) Numbers	1-603-222-40xx (where x is any digit)
DID Digits Passed in SIP Request URI	60322240xx
DID Digits Passed in SIP To Header	Same as SIP Request URI
Incoming Toll-Free Number	800-333-1234
Incoming Toll-Free Digits Passed in SIP Request URI	0000000034
Incoming Toll-Free Digits Passed in SIP To Header	0004152100034

1.3. Call Flows

To understand how calls are routed between the PSTN and the enterprise site shown in **Figure 1** using SIP trunks, two call flows are described in this section.

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.

1. A user on the PSTN dials a DID (or toll free) number assigned by the AT&T IP Flexible Reach (or IP Toll Free) service to the customer location.   The PSTN routes the call to the AT&T service.

2. Based on the number, the AT&T service offers the call to Avaya SES using SIP signaling messages sent over the IP access facility. Note that the assignment of the DID number and the address of the Avaya SES server were established during the ordering and provisioning of the service.

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3. Avaya SES routes the call to the Avaya S8500 Server running Avaya Communication Manager over a SIP trunk.

4. Avaya Communication Manager terminates the call to the Avaya telephone as shown in **Figure 2**.
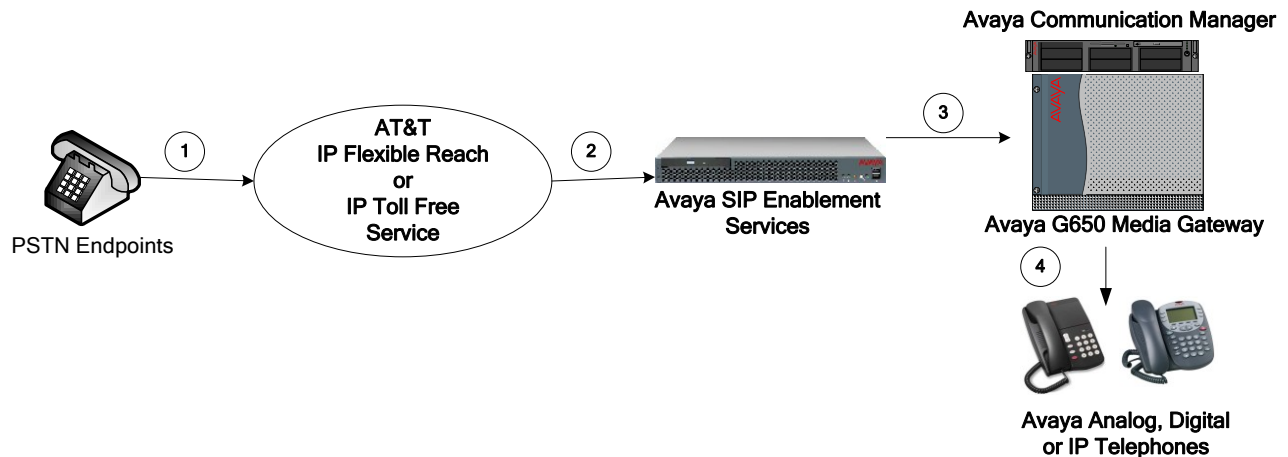

Figure 2: Incoming PSTN Calls to Avaya Communication Manager

The second call scenario illustrated in **Figure 3** is an outgoing call from an Avaya telephone at the enterprise site to the PSTN via the SIP trunk to the AT&T IP Flexible Reach service.

1. An Avaya telephone served by Avaya Communication Manager originates a call to a user on the PSTN.

2. The call request is handled by Avaya Communication Manager where origination treatment such as Class of Restrictions (COR) and Automatic Route Selection (ARS) is performed.   Avaya Communication Manager selects the SIP trunk and sends the SIP signaling messages to Avaya SES.

3. Avaya SES routes the call to the AT&T IP Flexible Reach service.

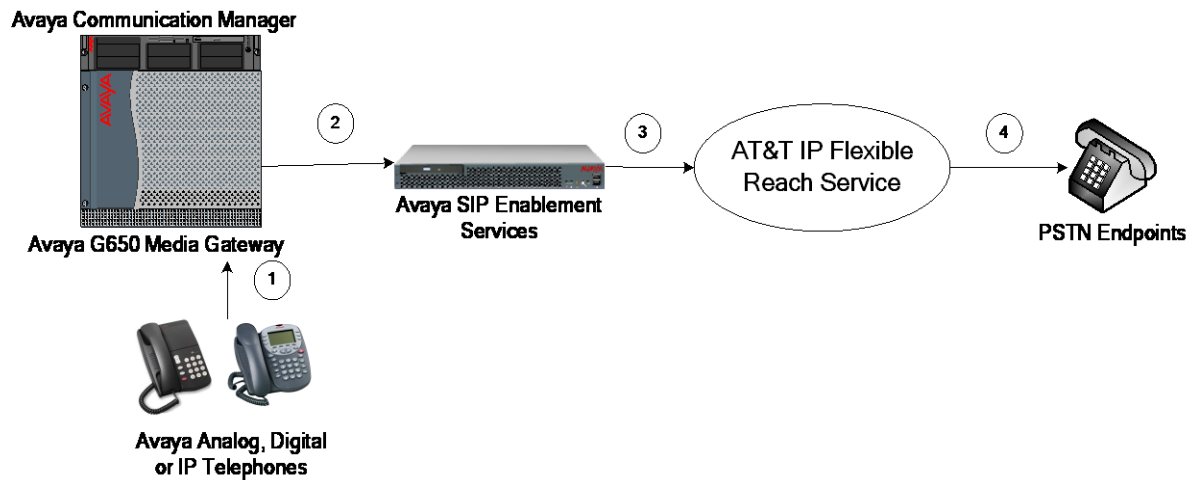4. The AT&T IP Flexible Reach service completes the call to the PSTN.

Figure 3: Outgoing Calls from Avaya Communication Manager to the PSTN

Appendix A illustrates examples of the SIP INVITE messages sent on the SIP trunk to begin each call.

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 15653 – see Product Support Notice 1842U)
Avaya G650 Media Gateway	
TN2312BP IP Server Interface (IPSI)	HW03 FW042
TN799DP Control-LAN (C-LAN)	HW01 FW026
TN2602AP IP Media Processor (Medpro)	HW02 FW033
TN2224CP Digital Line	HW08 FW015
TN793CB Analog Line	000005
Avaya 9640 one-X™ Deskphone Edition Telephone	Release S1.5
Avaya 4621SW IP (H.323) Telephone	Release 2.8.3
Avaya 9640 one-X™ Deskphone SIP Telephone	Release 2.0.3
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 Flexible Reach Service	Network Version VNI 11
IP Toll Free Service	Network Version VNI 11

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 Flexible Reach and AT&T IP Toll Free services 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

Avaya Communication Manager was installed and configured for basic station to station calling prior to beginning the configuration shown in these Application Notes. These basic configuration details are outside the scope of the SIP trunking application and not included here.

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 to the AT&T services, 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 4: 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 5: System-Parameters Customer-Options Form – Page 3

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 6: 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 via 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 **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 7: 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 8: 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 trunk signaling group to AT&T. 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 9: 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 [2]) and assigned the Node Name shown in **Figure 6**.

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 10: 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 region 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 region 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" as defined in Section 4.2.2). (Note: In previous versions of the AT&T services, this value must be the IP address of the Avaya SES instead of the SIP domain. When an IP address is used, SIP endpoints are not supported for use with the AT&T services.)
- **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 11: 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.

On Page 3, configure the "**src rgn 1 dst rgn 11**" row as follows:
- **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   9
 1   10
 1   11   2     y   Kbits    512    0     0   y                          n
 1   12
```

Figure 12: 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 IP Flexible Reach service.
- **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 13: 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 14: IP Network Region 11 – Page 3

3.1.7. Define SIP Trunk Groups

A SIP trunk group is defined for calls with the AT&T services (routed via the Avaya SES) to/from the first AT&T Border Element IP address. Both incoming Direct Inward Dialed (DID) calls and outbound calls to the PSTN use this trunk group. This SIP trunk group requires a corresponding SIP signaling group to define the characteristics of the signaling relationship. A

similar process will be followed to establish a SIP trunk group with the second AT&T Border Element IP address provided.

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 the 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 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 [12].

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 15: Signaling Group 11

3.1.7.2 Establish the SIP Trunk Group

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 SIP TRKS".
- **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 SIP TRKS                 COR: 1      TN: 1       TAC: #011
   Direction: two-way      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 16: SIP Trunk Group 11 – Page 1

On Page 4 of the Trunk Group form:
- **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 17: SIP Trunk Group 11 – Page 4

The second SIP trunk group 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.

3.1.8. Configure Calling Party Number Information

The SIP "From" header shown below contains information about the calling party. The header contains the calling party number (e.g., "16032224001") in the userinfo segment and a domain (or IP address) in the hostname segment separated by an "@" sign.

```
From: "Jane Smith" <sip:16032224001@customer-sipdomain.com>;tag=80f839da25
```

The "public-unknown-numbering" command controls the calling party number sent in the userinfo segment. The **Authoritative Domain** field of the near-end IP Network Region form completed in **Figure 11** associated with the SIP trunk group sets the hostname segment.

Public-unknown-numbering must always be configured. In these Application Notes the public-unknown-numbering is configured to send an 11 digit number corresponding to the DID or 800 number assigned by AT&T.

Using the "change public-unknown-numbering n" command (where "n" is the leading digit of the extension range), specify the calling party number information as follows:

- **Ext Len**: set to "4", the length of the extensions used.
- **Ext Code**: set to the leading digit of the extension used. In these Application Notes "40" and "6001" are entered to cover the assigned extensions of 40xx and 6001.
- **Trk Grp(s):** by default, leave blank to perform the same conversion across all SIP (and ISDN) trunk groups.
- **CPN Prefix**: set to the leading digits (e.g., "1603222") that are to be sent as the calling party number.
- **Total CPN Len**: set to the total length (e.g., "11") of the calling party number to be sent. The extension number will be appended to the **CPN Prefix** to form the complete calling

party number of **Total CPN Len** digits.

The completed public-unknown-numbering form is shown below.

```
change public-unknown-numbering 4                        Page   1 of   2
                     NUMBERING - PUBLIC/UNKNOWN FORMAT
                                          Total
Ext Ext          Trk       CPN            CPN
Len Code         Grp(s)    Prefix         Len
                                                Total Administered: 2
  4  40                    1603222        11       Maximum Entries: 9999
  4  6001                  18003331234    11
```

Figure 18: Public Unknown Numbering

3.1.9. Configure Call Routing

3.1.9.1 Outbound Calls

In these Application Notes, Automatic Route Selection (ARS) is used to route outbound calls via the SIP trunk group(s) to AT&T. AT&T expects to receive the following digits in the SIP INVITE message.

Type of Call	Digits Sent
Local and Long Distance in North American Numbering Plan	Digit 1 plus any 10 digits
Operator and International Calling	Digit 0 with or without additional digits; overall length indeterminate
N11 Service Calls	Digits 2 thru 9 followed only by the digits 11

Table 2: AT&T Outbound Dialing Rules

Here, the configuration of one outbound calling pattern supporting calls to 1-733-xxx-xxx is shown. A typical installation will generally require additional ARS dial string and route pattern entries but that is beyond the scope of these Application Notes. Further information on ARS administration is discussed in References [1] and [3].

ARS administration begins by verifying the availability of the feature as shown in Section 3.1.1.

Following the verification, use the "change dialplan analysis" command to create a feature access code (fac) for ARS use.
- **Dialed String**: enter "9" that will become the user dialed prefix for outbound calls.
- **Total Length**: enter "1" as the length of the prefix.
- **Call Type**: enter "fac" as the type of prefix.

```
change dialplan analysis                                        Page   1 of  12
                          DIAL PLAN ANALYSIS TABLE
                                                     Percent Full:    1

      Dialed   Total  Call    Dialed   Total  Call    Dialed   Total  Call
      String   Length Type    String   Length Type    String   Length Type
        0        1    attd
        4        4    ext
        5        4    ext
        6        4    ext
        9        1    fac
        *        3    dac
        #        4    dac
```

Figure 19: Dial Plan Analysis

Use the "change feature-access-codes" command to assign the feature access code "9" to **Auto Route Selection (ARS) - Access Code 1** as shown below.

```
change feature-access-codes                                     Page   1 of   7
                          FEATURE ACCESS CODE (FAC)
         Abbreviated Dialing List1 Access Code:
         Abbreviated Dialing List2 Access Code:
         Abbreviated Dialing List3 Access Code:
Abbreviated Dial - Prgm Group List Access Code:
                   Announcement Access Code: *71
                   Answer Back Access Code:

     Auto Alternate Routing (AAR) Access Code:
     Auto Route Selection (ARS) - Access Code 1: 9      Access Code 2:
              Automatic Callback Activation:          Deactivation:
Call Forwarding Activation Busy/DA: *61     All: *62    Deactivation: *60
  Call Forwarding Enhanced Status:          Act:        Deactivation:
                    Call Park Access Code:
                    Call Pickup Access Code:
CAS Remote Hold/Answer Hold-Unhold Access Code:
              CDR Account Code Access Code:
                   Change COR Access Code:
              Change Coverage Access Code:
                   Contact Closure   Open Code:        Close Code:
```

Figure 20: ARS Feature Access Code

Use the "change ars analysis nn" command to configure the ARS route pattern selection rules as follows.   Here "nn" is "17", the first two digits of the dialed number after the ARS access code.

- **Dialed String**: enter the leading digits (e.g., "1733") necessary to uniquely select the desired route pattern.
- **Total Min**: enter the minimum number of digits (e.g., "11") expected for this PSTN number.
- **Total Max**: enter the maximum number of digits (e.g., "11") expected for this PSTN number.
- **Route Pattern**: enter the route pattern number (e.g., "11") to be used. The route pattern (to be defined next) will specify the trunk group(s) to be used for calls matching the dialed number.
- **Call Type**: enter "fnpa", the call type for North American 1+10 digit calls.

```
change ars analysis 17                                          Page   1 of   2
                        ARS DIGIT ANALYSIS TABLE
                          Location:  all        Percent Full:    1

        Dialed              Total      Route    Call  Node ANI
        String            Min  Max   Pattern    Type  Num  Reqd
   1733                     11   11      11      fnpa       n
```

Figure 21: ARS Digit Analysis Entries

Use the "change route-pattern n" command (where "n" is the **Route Pattern** number used above) to specify the SIP trunk groups selected for the outbound call.

In the form:
- **Pattern Name**: enter a descriptive string such as "AT&T PSTN LD" to describe the routing pattern.
- **Secure SIP?**: leave as "n", the default.
- **Grp No**: enter the trunk groups to be used in priority order. Trunk group 11 is the first (and only) route choice in this case.
- **FRL**: enter the minimum facility restriction level (e.g., 1) necessary to use this trunk group, with 0 being the least restrictive. The FRL within the Class of Restriction (COR) assigned to the station must be greater than or equal to 1 in this case to use this trunk group.
- **Pfx Mrk**: enter "1", to always send the prefix 1 on 10 digit calls as specified for this Service Provider.

The defaults values for the remaining fields are used. The completed route pattern form is shown below.

```
change route-pattern 11                                         Page   1 of   3
                    Pattern Number: 11   Pattern Name: AT&T PSTN LD
                                    Secure SIP? n
    Grp FRL NPA Pfx Hop Toll No.  Inserted                         DCS/ IXC
    No          Mrk Lmt List Del  Digits                           QSIG
                         Dgts                                       Intw
 1: 11   1       1                                                  n   user
 2:                                                                 n   user
 3:                                                                 n   user
 4:                                                                 n   user
 5:                                                                 n   user
 6:                                                                 n   user

     BCC VALUE  TSC CA-TSC    ITC BCIE Service/Feature PARM  No. Numbering LAR
     0 1 2 M 4 W    Request                                  Dgts Format
                                                               Subaddress
 1: y y y y y n  n             rest                                       none
 2: y y y y y n  n             rest                                       none
 3: y y y y y n  n             rest                                       none
 4: y y y y y n  n             rest                                       none
 5: y y y y y n  n             rest                                       none
 6: y y y y y n  n             rest                                       none
```

Figure 22: Route Pattern 3

Use the "change locations" command to designate the SIP trunk route pattern (route pattern "11" below) in the **Proxy Sel. Rte. Pat.** Field if not previously set.

```
change locations                                            Page   1 of  16
                              LOCATIONS

              ARS Prefix 1 Required For 10-Digit NANP Calls? y

Loc  Name            Timezone Rule  NPA  ARS  Atd      Disp   Prefix   Proxy Sel
No                    Offset             FAC  FAC      Parm            Rte Pat
 1:  Main            + 00:00   0                        1              11
 2:                        :
```

Figure 23: Location Form Administration

3.1.9.2 Incoming Calls

This step configures the routing of incoming DID numbers to the associated Avaya Communication Manager extensions.

In these Application Notes, the incoming PSTN DID numbers 1-603-222-4000 through 4099 and the toll-free number 1-800-333-1234 are assigned by AT&T. They are assigned to extensions as shown in **Table 3**.

Dialed PSTN Number	Digits Received (within SIP INVITE message)	Extension Assigned
1 603 222 40xx	603 222 40xx	40xx
1 800 333 1234	000 000 0034	6001

Table 3 - Incoming Number Assignments

Use the "change inc-call-handling-trmt trunk-group n" command (where "n" is the SIP trunk group number "11") to administer the incoming number routing. This administration must be done for each incoming trunk group.
- **Called Len**: enter the total number of incoming digits received (e.g., "10").
- **Called Number**: enter the specific digit pattern to be matched.
- **Del**: enter the number of leading digits that should be deleted
- **Insert**: enter the specific digits to be inserted at the beginning of the adjusted incoming digit string (to form what should be the complete number).

The completed inc-call-handling-trmt form for trunk group 11 is shown below.

```
change inc-call-handling-trmt trunk-group 11                Page   1 of  30
                  INCOMING CALL HANDLING TREATMENT
 Service/      Called    Called     Del Insert
 Feature       Len       Number
 public-ntwrk  10 60322240          6
 public-ntwrk  10 0000000034        10  6001
```

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

3.1.10.   Save Avaya Communication Manager Changes

Use the "save translation" command to make the changes permanent.

4. Configure Avaya SIP Enablement Services

4.1. Background

These Application Notes explicitly show the administration of Avaya SIP Enablement Services using the combined home-edge server configuration. In this case, a single Avaya SES server simultaneously functions in both the home and edge roles.
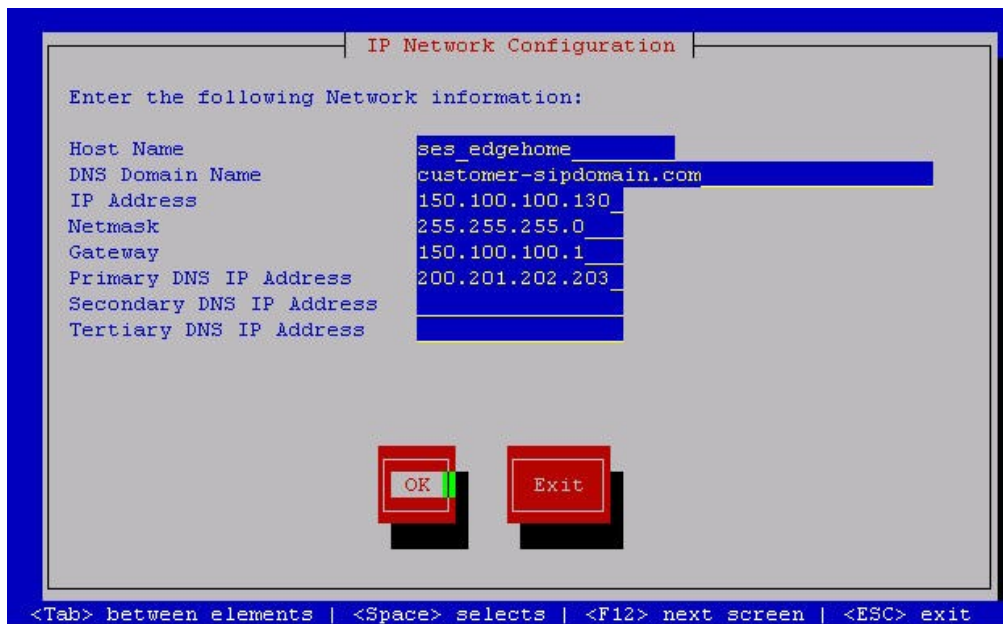
Note however that multiple Avaya SES servers may exist using a separate home / edge configuration as warranted by capacity considerations (predominately for the support of SIP telephones). In the separate home / edge server configuration:
- The SIP signaling relationship with the AT&T services exist between the AT&T Border Element and the Avaya SES edge server.
- The Avaya Communication Manager SIP signaling group relationship exists between a C-LAN (or equivalent) interface and a specific Avaya SES home server.
- SIP message routing between the home and edge servers is performed automatically and transparently.
- Only one Avaya SES edge server exists within a given sip domain (e.g., customer-sipdomain.com specified in these Application Notes.). Multiple home servers may exist as required.
- All Avaya SES administration is performed from a single SES server designated during installation as the master administrator.

The SIP trunking administration is generally the same for the combined and separate home/edge configurations. Any specific clarifications will be noted within the individual sections as necessary.

Avaya SIP Enablement Services is configured via an Internet browser using SIP Server Management screens. It is assumed that Avaya SIP Enablement Services software and the license file have been previously installed. During the software installation, the "initial_setup" installation script was run to specify the IP network properties of the server including DNS server address(es). For additional information on these installation tasks, refer to Reference [6].

For reference purposes, **Figure 24** illustrates the IP Network Configuration entered into the Avaya SES during installation.

Figure 24 - Avaya SES Initial Setup Information

4.2. SIP Connection to the AT&T Services

4.2.1. Log in to Avaya SIP Enablement Services

Access the Avaya SES SIP Server Management pages by entering "http://*<ip-addr>*/admin" as the URL in an Internet browser, where "*<ip-addr>*" is the IP address of Avaya SIP Enablement Services server. In these Application Notes, the URL "http://150.100.100.130/admin" is used.

Log in with the appropriate credentials and then select the **Launch Administration Web Interface** link from the main page as shown in **Figure 25**.
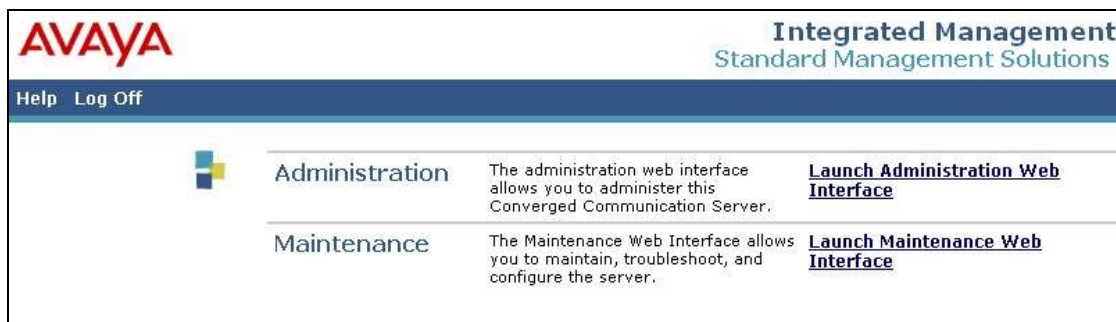
Figure 25: Avaya SES Main Page

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The Avaya SES administration home page shown in **Figure 26** is displayed.

Figure 26: Avaya SES Administration Home Page

4.2.2. Verify System Properties

From the left pane of any SIP Server Management page, expand the **Server Configuration** option and select **System Properties**. This page displays the Avaya **SES Version** and the

Network Properties entered via the install script during the installation process.

In the **View System Properties** page note the **SIP Domain** entered during the initial installation. The **SIP Domain** "customer-sipdomain.com" is used in these Application Notes.

Note that throughout the SES administration screens, the Help link may be used at anytime for further information regarding the meaning of any field.

Figure 27: System Properties

4.2.3. Verify the Avaya SES Host Information

Verify the Avaya SES Host information using the **Edit Host** page. In these Application Notes the Avaya SES **Host Type** is an "SES combined home-edge". This means that a single Avaya SES routes SIP messages between AT&T and Avaya Communication Manager.

Navigate to the **Edit Host** page (**Figure 28**) by following the **Hosts** link in the left navigation pane and then clicking on the **Edit** option under the **Commands** section of the **List Hosts** screen.

On the **Edit Host** screen:
- Verify that the IP address of this combined Avaya SES home-edge server is in the **Host IP Address** field.
- Verify that the **UDP**, **TCP** and **TLS** checkboxes are enabled as **Listen Protocols**.
- Verify that **TLS** is selected as the **Link Protocol**.
- Ensure that the **Outbound Proxy** and **Outbound Direct Domains** fields are left blank.
- Default values for the remaining fields may be used.
- Click the **Update** button only if changes are necessary. Otherwise, exit the **Edit Host** page by selecting the **Top** link on the left navigation bar.

Figure 28: Edit Host

Separate Home / Edge Configuration Note:
In the Avaya SES separate home / edge server configuration, there will be at least two SES hosts listed (home and edge servers).
- The <u>SES edge</u> server is configured as shown above for the combined home-edge.
- The <u>SES home</u> server **OutboundProxy** settings are different from the SES edge. The **Edit Hos**t page for each separate home server will specify the **Host Type** as "SES home".

  Configure all <u>SES home</u> servers as follows:
  - **OutboundProxy**: set to the IP address of the Avaya SES edge server.
  - **Port**: set to "5061".
  - **TLS** selected.

4.2.4. Add Avaya Communication Manager Media Server Interface

In these Application Notes, one media server signaling interface named "CLAN-1A03-5061" is used with Avaya Communication Manager.

Expand the **Media Servers** option within any Avaya SES SIP Server Management page, and select **Add** to display the Add Media Server page (**Figure 29**).

In the Add Media Server Interface page, enter information corresponding to the signaling group "11" entry performed in Section 3.1.7.1.
- Enter "CLAN-1A03-5061**"** as the descriptive name in the **Media Server Interface Name** field.
- Select the Avaya SES combined home-edge IP address in the **Host** field.
- Select "TLS" (Transport Link Security) for the **SIP Trunk Link Type**. TLS provides encryption at the transport layer between Avaya Communication Manager and the Avaya SES.
- Enter the IP address of the "clan-1a03" interface in the **SIP Trunk IP Address** field as defined in **Figure 6**. Note: This may be the IP address of the "procr" interface in other Avaya Communication Manager configurations.

Separate Home / Edge Configuration Note:
In the Avaya SES separate home / edge server configuration, there will be at least two SES hosts listed (home and edge servers).
- An <u>SES home</u> IP address is selected as the value used in the **Host** field on the Add Media Server Interface page.

After completing the Add Media Server Interface page, click on the **Add** button.

Figure 29: Add Media Server Interface for Voice Calls

When these operations are completed, the List Media Servers page will appear as shown in **Figure 30**.

Figure 30: Completed List Media Servers

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4.2.5. Configure Call Routing

4.2.5.1 Background
Avaya SIP Enablement Services functions as a SIP proxy for the SIP trunking with AT&T. The Avaya SES examines the SIP Request URI of an incoming SIP INVITE message, modifies the SIP Request URI and certain SIP headers and then forwards the message to the appropriate destination.

The SIP Request URI generally takes the form of *sip:user@domain*, where *domain* can be a fully qualified domain name or an IP address. The *user* part for SIP trunking in these Application Notes contains the called number digits identifying the telephone number being called.

The SIP messages are routed to the IP address associated with the *domain* part when the *domain* does not match the SIP domain of the Avaya SES (**Figure 27**) and the outbound proxy is not specified. This rule applies to all outbound calls to the AT&T services in these Application Notes.

The Avaya SES address maps are used to route SIP Messages when the *domain* part matches the Avaya SES SIP domain. In this case, the *user* part is compared to address map patterns specified in the Avaya SES and when a pattern match is found the SIP messages are routed to the corresponding signaling contact address destination. The address map patterns are specified using Linux regular expression syntax. Patterns are designed to match a collection of *called numbers* that require identical SIP message routing and must be specific enough to direct each unique *called number* to the proper signaling contact. **Appendix B** provides a detailed description of the Linux regular expression syntax used within the address map patterns.

The media server address maps are used to route all incoming calls from AT&T to Avaya Communication Manager in these Application Notes.

4.2.5.2 Outbound Calls to AT&T
In these Application Notes, the domain value of the SIP Request URI for outbound calls to AT&T is specified by the Far-end Domain field of the signaling group 11 form (Section 3.1.7.1) of the outbound SIP trunk

Since this domain value specifies the AT&T network, no additional administration is necessary in the Avaya SES for outbound calls to AT&T.

4.2.5.3 Inbound Direct Inward Dialed Calls
SIP messages for incoming calls from AT&T are sent to the Avaya SES. The Avaya SES then routes these messages to the appropriate Avaya Communication Manager using Avaya SES media server address maps.

In these Application Notes, the incoming PSTN calls use media server address map patterns

matching the 10-digit called number in the user part of the SIP Request URI.

An example of a SIP Request URI in an INVITE message received from AT&T for the DID number 603-222-4009 is:

sip:6032224009@150.100.100.130

The user part in this case is the 10-digit number "6032224009".

Table 4 below summarizes the media server address map strategy used in these Application Notes for incoming calls. Note that AT&T user part for toll-free numbers typically do not match the dialed toll-free number.

Dialed PSTN Number	SIP Request URI User Part	SES Media Server Address Map Pattern	Media Server Interface
1-603-222-4000 through 4099	6032224000 through 6032224099	^sip:60322240[0-9]{2}	CLAN-1A03-5061
1-800-333-1234	0000000034	^sip:0000000034	CLAN-1A03-5061

Table 4: Incoming DID and Toll-Free Address Map Rules Used

To configure the media server address map for voice calls:
- Expand the **Media Servers** link in the left navigation menu of any SIP Server Management page. Select **List** to display the List Media Servers screen shown in **Figure 31**.
- Click on the **Map** link to display the List Media Server Address Map screen associated with the "CLAN-1A03-5061" **Interface**.

Figure 31: List Media Servers

- Since no previous media server address map exists, click on the **Add Map In New Group** link as shown in **Figure 32**.

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Figure 32 - List Media Server Address Map

The Add Media Server Address Map page shown in **Figure 33** is displayed.

- Enter a descriptive name in the **Name** field, such as "DID-60322240xx".
- Enter the **Address Map Pattern** for incoming DID calls (from **Table 4**) into the **Pattern** field.

    In this case, the DID numbers provided by the AT&T IP Flexible Reach service are 603-222-4000 thru 4099. The pattern specification for these DID numbers is:

    ^sip:60322240[0-9]{2}

    This means that URIs beginning with "sip:60322240" followed any other 2 digits will match the pattern and be routed to the interface defined as "CLAN-1A03-5061".

- Click the **Add** button once the form is completed.

Figure 33: Incoming DID Calls - Media Server Address Map

A second media server address map is created for the assigned 800 number found in **Table 4**. The Name and Pattern will be entered as shown in **Figure 34**.

Figure 34: Incoming 800 Call - Media Server Address Map

After configuring the media server address maps, the **List Media Server Address Map** page appears as shown in **Figure 35**.

Figure 35: List Media Server Address Map

Note that after the first media server address map is created, a corresponding media server **Contact** entry is created automatically.

sip:$(user)@150.100.100.113:5061;transport=tls

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This **Contact** entry contains the IP address of the "CLAN-1A03" interface on Avaya Communication Manager, the port (5061) and the transport protocol (tls) to be used. The incoming digits sent in the user part of the original request URI will replace the $(user) string when the message is sent.

4.2.6. Specify the AT&T Network as a Trusted Host

The Border Element IP addresses provided by AT&T (e.g., 20.1.1.54 and 20.1.2.64) must be added as a trusted host entry in the Avaya SES. As a trusted host, the Avaya SES will not attempt to authenticate incoming requests from the designated IP address.[1] To add the trusted host entry:

- Expand the **Trusted Hosts** link in the left navigation menu of any SIP Server Management page. Select **Add** to display the Add Trusted Host screen shown in **Figure 36**.
- Enter "20.1.1.54" into the **IP Address** field. This is the address provided by AT&T of their system that sends SIP messages to the Avaya SES and is the same address as the Far-end Domain value entered in the signaling group form in Section 3.1.7.1.
- Select "150.100.100.130" (the address of the Avaya combined home-edge SES) as the value of the **Host** field.
- Enter "ATT Network" in the **Comment** field.
- Click the **Add** button once the form is completed.

Figure 36 - Add Trusted Host

---

[1] If the trusted host step is not done, authentication challenges to incoming SIP messages (such as INVITEs and BYEs) will be issued by Avaya SES. This may cause call setup to fail, active calls to be disconnected after timeout periods, and/or SIP protocol errors.

Repeat for any other AT&T Border Element addresses provided.

Separate Home / Edge Configuration Note:
In the Avaya SES separate home / edge server configuration, there will be at least two SES hosts listed (home and edge servers).
- Configure the trusted host relationship with each <u>SES home</u> server in the **Host** list to each AT&T Border Element.
- The SES edge server does not require a trusted host relationship.

The resulting List Trusted Hosts page should appear as shown in **Figure 37. (**Note that trusted host changes may take several minutes to take effect due to caching characteristics.)

Figure 37 - List Trusted Hosts

4.2.7. Saving the Avaya SES Changes

Beginning with Avaya SES Release 5.0 it is no longer necessary to explicitly save the administrative changes performed above. These changes are automatically committed to the Avaya SES database after each operation. Note that the "Update" link no longer appears in the navigation menu.

5. Interoperability Compliance Testing

This section describes the interoperability compliance testing used to verify SIP trunking interoperability between the AT&T IP Flexible Reach and IP Toll Free services and an Avaya SIP trunking solution. 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 Services via simulated managed access facilities. The enterprise site was configured as if using the generally available service provided by AT&T.

The following features and functionality were covered during the SIP trunking interoperability compliance test.
- Outgoing calls to PSTN telephones
- Incoming calls to AT&T provided DID and Toll Free numbers from PSTN telephones
- Calls using Avaya 4600 and 9600 Series IP Telephones with the H.323 firmware configurations
- Calls using Avaya 4600 and 9600 Series Telephones with the SIP firmware configurations
- G.729B and G.711MU codecs for voice calls
- T.38 codecs for fax calling
- DTMF tone transmission using RFC 2833 with successful Voice Mail / IVR navigation
- Telephone features such as hold, transfer, conference, and voice mail
- Trunk to trunk call forwarding, transfers and EC-500 feature operation
- Direct IP-to-IP media (also known as "shuffling") with IP and SIP telephones

The following feature is not supported by Avaya with the AT&T services and was not tested.
- Fax detection during G.729 voice with codec change to G.711mu. (T.38 fax is supported instead).

5.2. Test Results

Interoperability testing of the sample configuration was completed with successful results.

The following compatibility issues described in **Table 5** were observed.

Item	Issue Observed	Discussion / Workaround
Trunk to trunk transfer of an incoming PSTN call to an AT&T supported VoIP-TDM gateway fails	Call drops after transfer completed. AT&T issues incorrect BYE. Issue under investigation by AT&T.	Avoid SIP trunk to SIP trunk transfers to locations using AT&T supported VoIP –TDM gateway devices.
Access to the AT&T network teleconferencing service fails if calling number restriction used.	The use of calling number restriction causes calls to the AT&T Teleconferencing Service to fail.	Avoid using calling number restriction and/or avoid using AT&T SIP trunking for their Teleconference service access.
Unreliable T.38 fax transmission with AT&T HIPCS service endpoints and/or some SG3 mode fax machines.	Fax calls (using T.38) to higher speed "super group 3 (SG3) fax machines on AT&T's HIPCS service failed due to known AT&T issues. T.38 mode fax transmission has been observed to be variable and/or unreliable with higher speed "super group 3" (SG3) fax machines. Operating at normal group 3 speeds was satisfactory.	Avoid SG3 fax machine use via SIP trunks if problems observed. Use alternate trunking or operate fax machine at lower speeds.
Emergency 911/E911 Services Limitations	Discussion item added per AT&T request. Contact AT&T for additional information. Note: Explicit 911/E911configuration and testing was not performed during this Avaya DevConnect interoperability testing.	While AT&T IP Flexible Reach services support E911/911 calling capabilities in certain circumstances, there are significant limitations on how these capabilities are delivered. Please review the AT&T IP Flexible Reach Service Guide[2] in detail to understand these limitations and restrictions.

Table 5: Summary of Issues Identified During Interoperability Testing

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 Calls – Verify that calls placed from a PSTN telephone to the DID number assigned are properly routed via the SIP trunk group(s) 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.

---

[2] Access the AT&T IP Flexible Reach Service Guide at
http://new.serviceguide.att.com/portals/sgportal.portal?_nfpb=true&_pageLabel=voip_page

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- Outbound Calls – Verify that calls placed to a PSTN telephone are properly routed via the SIP trunk group(s) defined in the ARS route patterns. Verify that the talk-path exists in both directions and that calls remain stable and disconnect properly.

- Direct IP-IP Connections – This applies if IP and/or SIP telephones and Direct IP-IP are used. 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".

6.2. Troubleshooting Tools

The Avaya Communication Manager "list trace station", "list trace tac", "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 may obtain support for the AT&T IP Flexible Reach Service by calling 1-877-288-8362. Support for the AT&T IP Toll Free Service should be directed to 1-800-325-5555.

Avaya customers may obtain documentation and support for Avaya products by visiting http://support.avaya.com. The "Connect with Avaya" section provides the worldwide support directory. In the United States, 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 between an Avaya Communication Manager based SIP telephony solution and AT&T IP Flexible Reach and / or 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.

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

[2] *Adding New Hardware for Avaya Media Servers and Gateways*, January 2008, Issue 3, Release 5.0, Document Number 03-300684

[3] *Feature Description and Implementation for Avaya Communication Manager,* January 2008, Issue 6, Document Number 555-245-205

[4] *SIP Support in Avaya Communication Manager Running on the Avaya S8xxx Servers*, January 2008, Issue 8, Document Number 555-245-206.

[5] *4600 Series IP Telephone LAN Administrator Guide,* October 2007, Issue 7, Document Number 555-233-507

[6] *Installing, Administering, Maintaining and Troubleshooting SIP Enablement Services,* January 2008, Issue 5.0, Document Number 03-600768

[7] *Avaya Communication Manager Network Region Configuration Guide*, October 2005, Document ID 103244

[8] *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

[9] *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*

[10] *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.

[11] RFC 3261 - *SIP (Session Initiation Protocol)*, June 2002, Proposed Standard

[12] RFC 2833 - *RTP Payload for DTMF Digits, Telephony Tones and Telephony Signals*, May 2000, Proposed Standard

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APPENDIX A: Sample SIP INVITE Messages

This section displays the format of typical SIP INVITE messages sent between AT&T and the Avaya SES at the enterprise site. These INVITE messages may be used for comparison and troubleshooting purposes. Differences in these messages may indicate that different configuration options were selected.

Sample SIP INVITE Message from the AT&T services to the Avaya SES:

```
INVITE sip:6032224042@150.100.100.130:5060 SIP/2.0
Via: SIP/2.0/UDP 20.1.1.54:5060;branch=z9hG4bK3f3m5s30dgegrbkqj7c0.1
From: "OUT_OF_AREA" <sip:+17328521637@20.1.1.54:5060;user=phone>;tag=SDnq8e501-dsd104dd6
To: <sip:6032224042@150.100.100.130;user=phone>
Call-ID: SDnq8e501-2bf8da9ee68658807f7737f12a6efc1a-fms3e43
CSeq: 1 INVITE
Max-Forwards: 68
Contact: <sip:+17328521637@20.1.1.54:5060;transport=udp>
P-Asserted-Identity: <sip:7328521637@20.1.1.54:5060>
Allow: INVITE,ACK,CANCEL,BYE,INFO,PRACK
Accept: application/sdp, application/isup, application/dtmf, application/dtmf-relay,
multipart/mixed
Accept-Language: en; q=0.0
Content-Length: 277
Content-Disposition: session; handling=required
Content-Type: application/sdp

v=0
o=Sonus_UAC 20784 10712 IN IP4 20.1.1.54
s=SIP Media Capabilities
c=IN IP4 20.1.1.54
t=0 0
m=audio 22728 RTP/AVP 2 18 0 96
a=rtpmap:2 G726-32/8000
a=rtpmap:18 G729/8000
a=rtpmap:0 PCMU/8000
a=rtpmap:96 telephone-event/8000
a=fmtp:96 0-15
a=sendrecv
```

Sample SIP INVITE Message from Avaya SES to the AT&T services:

```
INVITE sip:17328521637@20.1.1.54 SIP/2.0
Accept-Language: en
Call-ID: 0b02d53eae2dc154f479c652e00
CSeq: 1 INVITE
From: "9640 IP H.323" <sip:6032224043@customer-
sipdomain.com:5061>;tag=0b02d53eae2dc153f479c652e00
Record-Route: <sip:150.100.100.130:5060;lr>,<sip:150.100.100.112:5061;lr;transport=tls>
To: "17328521637" <sip:17328521637@20.1.1.54>
Via: SIP/2.0/UDP 150.100.100.130:5060;branch=z9hG4bK030303262626030303a8cb.0,SIP/2.0/TLS
150.100.100.112;psrrposn=2;received=150.100.100.112;branch=z9hG4bK0b02d53eae2dc155f479c652e00
Content-Length: 217
Content-Type: application/sdp
Contact: "9640 IP H.323" <sip:6032224043@150.100.100.112:5061;transport=tls>
Max-Forwards: 68
User-Agent: Avaya CM/R015x.00.0.825.4
Allow: INVITE,CANCEL,BYE,ACK,PRACK,SUBSCRIBE,NOTIFY,REFER,OPTIONS,INFO,PUBLISH
Supported: 100rel,timer,replaces,join,histinfo
Alert-Info: <cid:internal@20.1.1.54>;avaya-cm-alert-type=internal
Min-SE: 1200
Session-Expires: 1200;refresher=uac
P-Asserted-Identity: "9640 IP H.323" <sip:6032224043@customer-sipdomain.com:5061>
History-Info: <sip:17328521637@20.1.1.54>;index=1,"17328521637"
<sip:17328521637@20.1.1.54>;index=1.1

v=0
o=- 1 1 IN IP4 150.100.100.112
s=-
c=IN IP4 150.100.100.116
b=AS:64
t=0 0
m=audio 2148 RTP/AVP 18 0 8 101
a=rtpmap:18 G729/8000
a=rtpmap:0 PCMU/8000
a=rtpmap:8 PCMA/8000
a=rtpmap:101 telephone-event/8000
```

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APPENDIX B: Specifying Pattern Strings in Address Maps

The syntax for the pattern matching used within the Avaya SES is a Linux regular expression used to match against the URI string found in the SIP INVITE message.

Regular expressions are a way to describe text through pattern matching. The regular expression is a string containing a combination of normal text characters, which match themselves, and special *metacharacters*, which may represent items like quantity, location or types of character(s).

In the pattern matching string used in the Avaya SES:
- Normal text characters and numbers match themselves.
- Common metacharacters used are:
  - A period `.` matches any character once (and only once).
  - An asterisk `*` matches zero or more of the preceding characters.
  - Square brackets enclose a list of any character to be matched. Ranges are designated by using a hyphen. Thus the expression `[12345]` or `[1-5]` both describe a pattern that will match any single digit between 1 and 5.
  - Curly brackets containing an integer 'n' indicate that the preceding character must be matched exactly 'n' times. Thus `5{3}` matches '555' and `[0-9]{10}` indicates any 10 digit number.
  - The circumflex character `^` as the first character in the pattern indicates that the string must begin with the character following the circumflex.

Putting these constructs together as used in this document, the pattern to match the SIP INVITE string for any valid 1+ 10 digit number in the North American dial plan would be:

`^sip:1[0-9]{10}`

This reads as: "Strings that begin with exactly **sip:1** and having any 10 digits following will match.

A typical INVITE request below uses the shaded portion to illustrate the matching pattern.

```
INVITE sip:17325551638@20.1.1.54:5060;transport=udp SIP/2.0
```

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