Avaya Solution & Interoperability Test Lab

Application Notes for Avaya Aura™ Communication Manager and Avaya Aura™ SIP Enablement Services SIP Trunking with AT&T IP Flexible Reach Service – Issue 1.0

Abstract

These Application Notes describe the steps for configuring Avaya Aura™ Communication Manager and Avaya Aura™ SIP Enablement Services SIP trunking with the AT&T IP Flexible Reach service using MIS-PNT transport service connections.

The AT&T IP Flexible Reach service is one of several SIP-based Voice over IP services offered to enterprises for a variety of voice communications needs. The AT&T IP Flexible Reach service allows enterprises in the U.S.A. to place outbound local, long distance and international calls, receive inbound Direct Inward Dialing calls from the PSTN, and place calls between an enterprise'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 at the Avaya Solution and Interoperability Test Lab.

TABLE OF CONTENTS

1. Introduction

These Application Notes describe the steps for configuring Avaya Aura™ Communication Manager and Avaya Aura™ SIP Enablement Services (SES) SIP trunking with the AT&T IP Flexible Reach service using MIS-PNT transport service connections.

The AT&T IP Flexible Reach service is one of several SIP-based Voice over IP (VoIP) services offered to enterprises for a variety of voice communications needs. The AT&T IP Flexible Reach service allows enterprises in the U.S.A. to place outbound local, long distance and international calls, receive inbound Direct Inward Dialing (DID) calls from the PSTN, and place calls between an enterprise's sites.

1.1. Interoperability Compliance Testing

The interoperability compliance testing focused on verifying outbound and inbound call flows between Avaya Aura™ Communication Manager, Avaya Aura™ SIP Enablement Services and AT&T IP Flexible Reach service.

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

- SIP Trunking.
- T.38 Fax.
- Passing of DTMF events and their recognition by navigating automated menus.
- Call Redirection with Diversion Header. In cases, where the alternate destination is an N11, NPA-555-1212, or 8xx number, AT&T IP Flexible Reach service requires the use of Diversion Header.

1.2. Support

AT&T customers may obtain support for the AT&T IP Flexible Reach service by calling (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, (866) GO-AVAYA (866-462-8292) provides access to overall sales and service support menus. Customers may also use specific numbers (provided on http://support.avaya.com) to directly access specific support and consultation services based upon their Avaya support agreements.

1.3. Known Limitations

1. Direct IP-to-IP media is not supported on calls between SIP 96XX phones in the Avaya site and the AT&T IP Flexible Reach service due to a known issue in Avaya Aura™ Communication Manager Release 5.2.1. The administration shown in **Section 4.6** disables Direct IP-to-IP media for SIP phones in the Avaya site.
2. Avaya one-X™ Communicator does not currently support G.729B codec, therefore Avaya Aura™ Communication Manager renegotiates the call to G.729A to support Direct IP-to-IP media.
3. G.711 faxing is not supported over the AT&T IP Flexible Reach SIP trunk as Communication Manager does not support the protocol negotiation that AT&T requires to have G.711 fax calls work. T.38 faxing is supported.
4. While Group 3 and Super Group 3 fax devices are supported, the AT&T IP Flexible Reach service has a maximum T.38 fax transmission of 9600.
5. AT&T Flexible Reach service support E911/911 capabilities in certain circumstances but there are significant limitations on how these capabilities are delivered. Please review the AT&T IP Flexible Reach Service Guide to understand the limitations and restrictions.
6. An AT&T IP Flexible Reach circuit can support up to 4 C-LAN cards and one SES. If additional C-LANS and SES servers must be supported, an AT&T IP Flexible Reach/Avaya certified Session Border Controller (SBC) will be required.

2. Reference Configuration

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

- Avaya Aura™ Communication Manager provides the Enterprise Voice communications services. In this sample configuration, Avaya Aura™ Communication Manager runs on an Avaya S8710 Server. This solution is extensible to other Avaya S8xxx Servers as well as Communication Manager supported on Midsize Business Enterprise.
- The Avaya Media Gateway provides the physical interfaces and resources for enterprise voice communications. In this sample configuration, an Avaya G650 Media Gateway is used. This solution is extensible to other Avaya Media Gateways.
- Avaya Aura™ SIP Enablement Services runs on an Avaya S8500 server. SES serves as a SIP proxy between Communication Manager and the AT&T IP Flexible Reach service. SES also provides registrar services to SIP phones in the enterprise.
- The Avaya "office" phones include Avaya SIP, H.323, and Digital phones, and Avaya one-X™ Communicator, which is a Softphone that runs on a Desktop (not shown).
- Avaya Modular Messaging system provides the corporate voice messaging capabilities for enterprise users. The provisioning of Modular Messaging system is beyond the scope of this document.
- Enterprise sites will generally have additional or alternate routes to PSTN using analog or digital TDM trunks.
- IP Network Address Translation (NAT) devices, firewalls, Application Layer Gateway (ALG) devices, and Session Border Controller (SBC) devices that may exist between the enterprise site and the AT&T IP Flexible Reach service are not explicitly shown. Those

devices generally must be SIP-aware and configured properly for SIP trunking to function properly. When configured correctly, those devices are transparent to the Avaya communications infrastructure.

- Enterprise networks of sufficient size or complexity may use SES in a separate Home/Edge server configuration, i.e., separate servers for SES Home and Edge roles. This configuration is functionally equivalent to the SES combined Home/Edge server configuration illustrated in these Application Notes. As appropriate, special notes have been included to clarify specific administration details related to the SES separate Home/Edge server configuration.

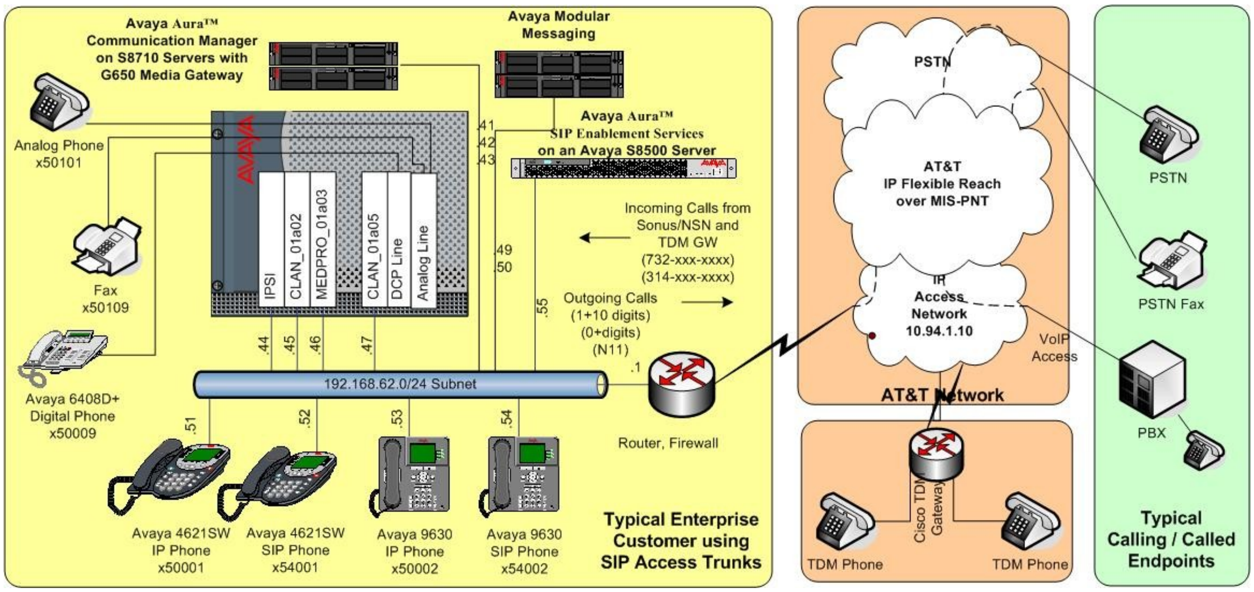

Figure 1: Sample SIP Trunking Configuration

2.1. Illustrative Configuration Information

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

Note - The AT&T IP Flexible Reach service border element IP addresses shown in this document are examples. AT&T Customer Care will provide the actual IP addresses as part of the IP Flexible Reach provisioning process.

Component	Illustrative Value in these Application Notes
Avaya Aura™ SIP Enablement Services	
IP Address	192.168.62.55
Avaya Aura™ Communication Manager	
C-LAN IP Address	192.168.62.45
Avaya Aura™ Communication Manager extensions	5xxxx
Avaya CPE local dial plan	732-320-4xxx 314-332-5xxx
Voice Messaging Pilot Extension	55000
Avaya Modular Messaging	
Messaging Application Server (MAS) IP Address	192.168.62.49
Exchange Server IP Address	192.168.62.50
Pilot Number	17323255000
AT&T IP Flexible Reach Service	
Border Element IP Address	10.94.1.10

Table 1: Illustrative Values Used in these Application Notes

2.2. Call Flows

To understand how outbound and inbound PSTN calls, as well as calls between the Avaya site and TDM gateway sites, are routed, several call flows are described in this section.

2.2.1. Outbound

The outbound call scenario illustrated in **Figure 2** is an outbound call from Communication Manager via SIP trunk to AT&T IP Flexible Reach service to PSTN or TDM Gateway.

1. A Communication Manager Phone or fax originates a call to a PSTN number.
2. Communication Manager applies any necessary origination treatment (verifying permissions, determining the proper route, selecting the outgoing trunk, etc.) and routes the call to SES.
3. SES routes the call to the AT&T IP Flexible Reach service.
4. The AT&T IP Flexible Reach service routes the call to the PSTN or TDM Gateway (4a).
5. The PSTN completes the call to the PSTN or TDM Gateway (5a) phone or fax.

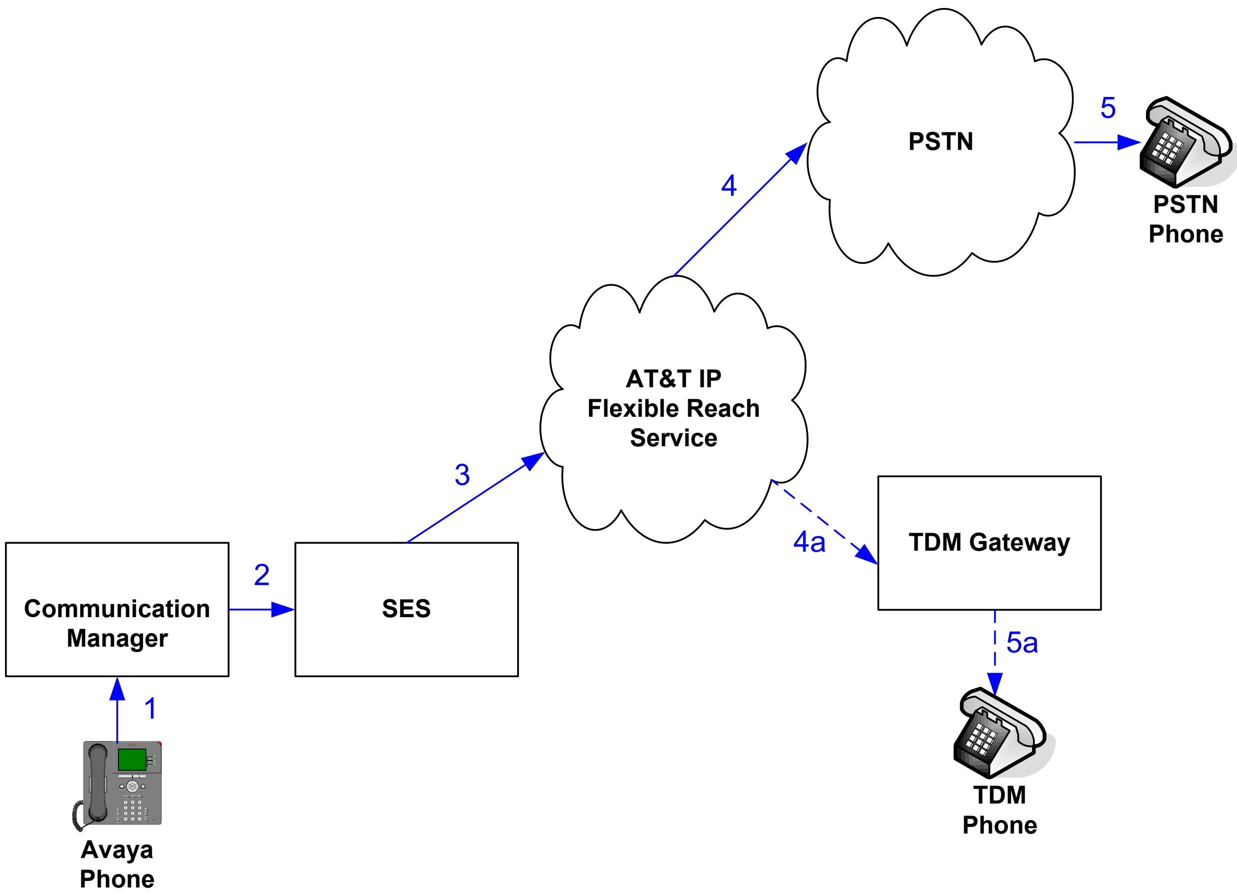

Figure 2: Outbound Call to PSTN or TDM Gateway Scenario

2.2.2. Inbound

The call scenario illustrated in **Figure 3** is an inbound call from a PSTN or TDM Gateway via a SIP trunk between AT&T Flexible Reach Service and Communication Manager.

1. A PSTN or TDM (1a) phone or fax originates a call to a DID number assigned to a Communication Manager site.
2. The PSTN or TDM Gateway (2a) routes the call to the AT&T IP Flexible Reach service.
3. The AT&T IP Flexible Reach Service routes the call to SES.
4. SES routes the call to Communication Manager.
5. Communication Manager applies any necessary termination treatment (examining and, if necessary, modifying the called party number, verifying permissions, etc.) and completes the call to the Communication Manager Phone or fax.

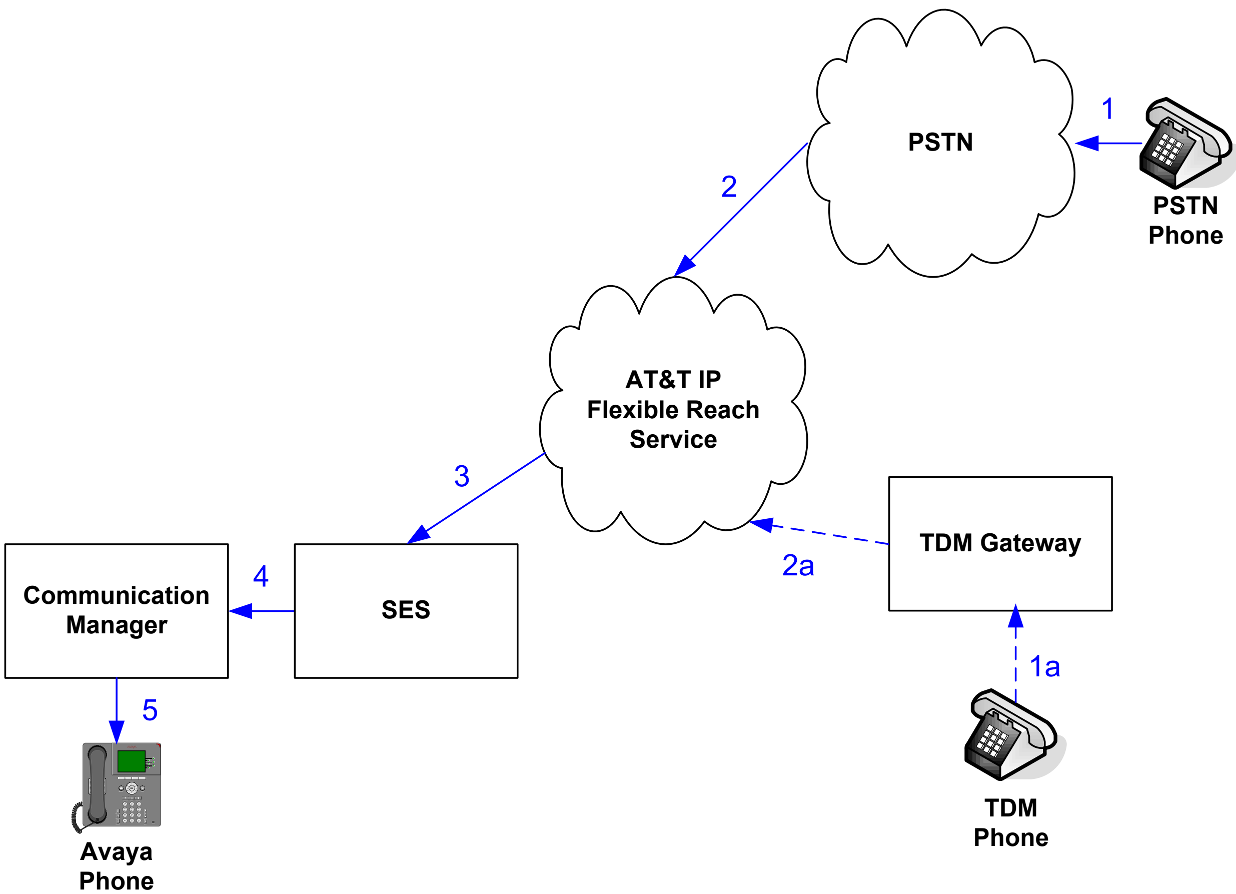

Figure 3: Inbound Call from PSTN or TDM Gateway Scenario

2.2.3. Call Forward Re-direction

The call scenario illustrated in **Figure 4** is an inbound AT&T IP Flexible Reach service call to Communication Manager which routes the call to a destination phone; however the station has set Call Forwarding to an alternate (off-net) destination. Without answering the call, Communication Manager immediately redirects the call back to the AT&T IP Flexible Reach service for routing to the alternate destination.

Note – In cases where the alternate destination is an N11, NPA-555-1212, or 8xx number, then the AT&T IP Flexible Reach service requires the use of SIP Diversion Header for the redirected call to complete (see **Section 4.4.1**, **Step 4**).

1. Same as the first call scenario in **Section 2.2.1**.
2. Because the Communication Manager phone has set Call Forward to another AT&T IP Flexible Reach service number, Communication Manager initiates a new call back out to SES, and to the AT&T IP Flexible Reach service network.
3. The AT&T IP Flexible Reach service places a call to the alternate destination and upon answer, Communication Manager connects the calling party to the target party.

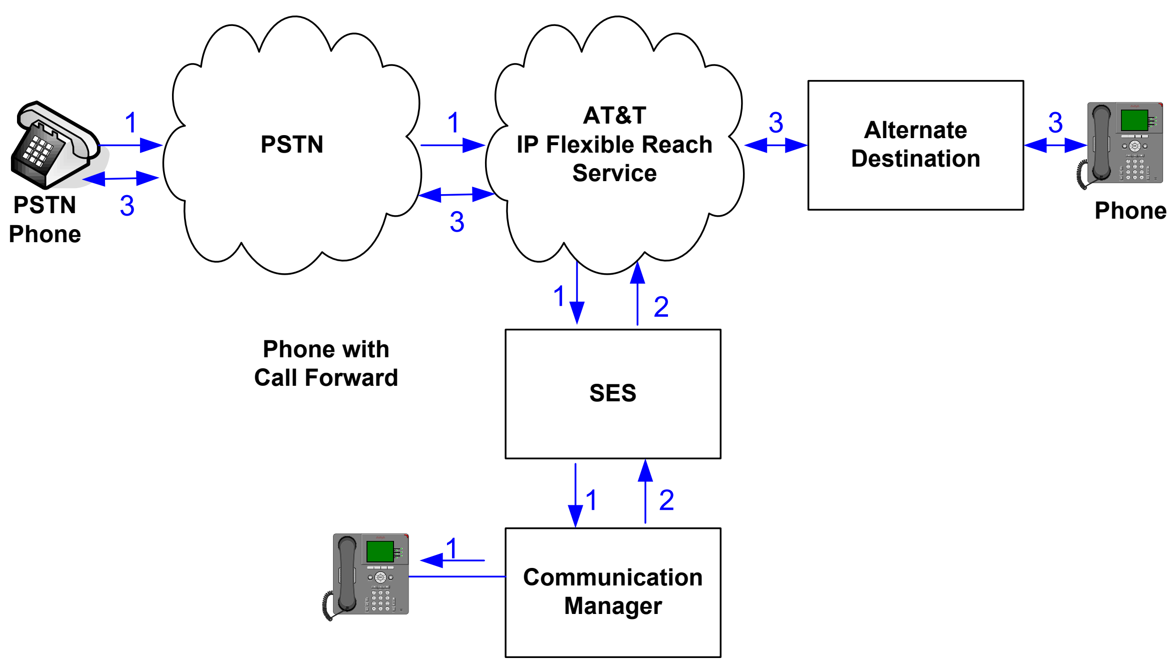

Figure 4: Re-directed (e.g. Call Forward) AT&T IP Flexible Reach Call

2.2.4. Coverage to Voicemail

This call scenario illustrated in **Figure 5** is an inbound AT&T IP Flexible Reach service call to Communication Manager which routes the call to a destination phone that is covered to voicemail. In this scenario, the voicemail system is an Avaya Modular Messaging system connected to SES.

1. Same as the first call scenario in **Section 2.2.1**.
2. The called Communication Manager phone does not answer the call, and the call covers to the phone's voicemail which Communication Manager forwards[1] to SES.
3. SES applies any necessary SIP header manipulations, and routes the call to Modular Messaging. Modular Messaging answers the call and connects the caller to the called phone's voice mailbox. Note that the call[2] continues to go through Communication Manager.

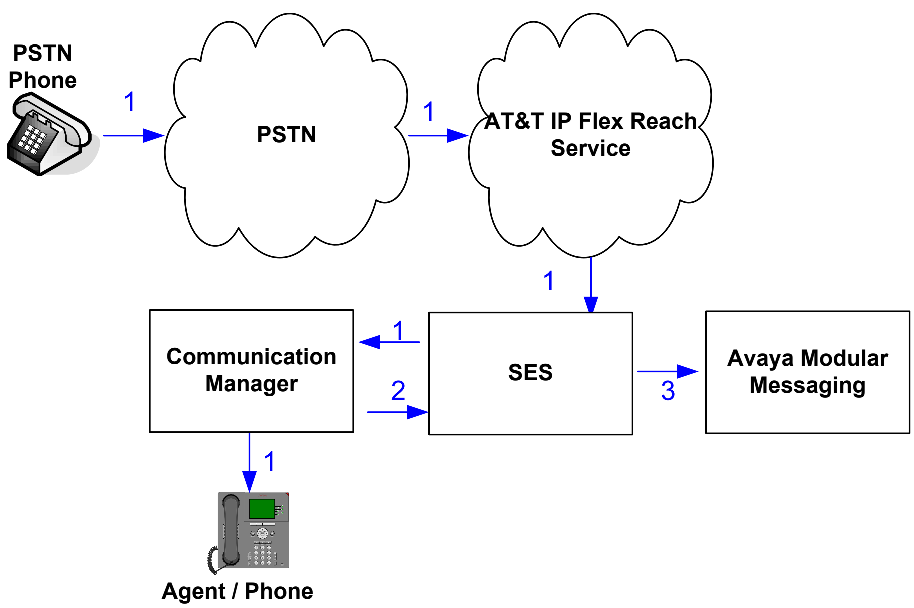

Figure 5: Coverage to Voicemail Scenario

[1] Avaya Aura™ Communication Manager places a call to Avaya Modular Messaging, and then connects the inbound caller to Avaya Modular Messaging. SIP redirect methods, e.g., 302, are not used.
[2] The SIP signaling path still goes through Avaya Aura™ Communication Manager. In addition, since the inbound call and Avaya Modular Messaging use different codecs (G.729 and G.711, respectively), Avaya Aura™ Communication Manager performs the transcoding, and thus the RTP media path also goes through Avaya Aura™ Communication Manager.

3. Equipment and Software Validated

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

Component	Version
Avaya S8710 Server	Avaya Aura™ Communication Manager 5.2.1 (R015x.02.1.016.4)
Avaya G650 Media Gateway	
TN2312BP IP Server Interface (IPSI)	HW06 FW049
TN799DP Control-LAN (C-LAN)	HW01 FW034
TN2602AP IP Media Resource 320 (MedPro)	HW02 FW051
TN2501AP VAL-Announcement	HW02 FW018
TN2224CP Digital Line	HW08 FW015
TN793B Analog Line	HW05
Avaya S8500B Server	Avaya Aura™ SIP Enablement Services 5.2.1 (SES05.2.1.016.4)
Avaya 9630 IP Telephone	Avaya one-X™ Deskphone Edition H.323 Release 3.0
Avaya 9630 IP Telephone	Avaya one-X™ Deskphone Edition SIP Release 2.4.2
Avaya 4610SW H323 Telephone	2.8
Avaya 4621SW SIP Telephone	2.2.2
Avaya 6408D+ Digital Telephone	-
Avaya 6211 Analog Telephone	
Avaya one-X™ Communicator	5.2
AT&T IP Flexible Reach Service over MIS-PNT	VNI 16

Table 2: Equipment and Software Versions

4. Avaya Aura™ Communication Manager

This section describes the administration steps for Communication Manager in support of SIP trunking integration with the AT&T IP Flexible Reach service. The steps are performed from the Communication Manager System Access Terminal (SAT) interface. These Application Notes assume that basic Communication Manager Administration, including stations, C-LAN and Media Processor boards, SIP phone signaling/trunk group(s), etc., has already been performed. Refer to **[1]**, **[2]**, and **[3]** for further details if necessary.

4.1. System Parameters

This section reviews the Communication Manager licenses and features that are required for the sample configuration described in these Application Notes. For required licenses that are not enabled in the steps that follow, contact an authorized Avaya account representative to obtain the licenses.

1. Enter the **display system-parameters customer-options** command. On Page 2 of the **system-parameters customer-options** form, verify that the **Maximum Administered SIP Trunks** number is sufficient for the number of expected SIP trunks.

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

IP PORT CAPACITIES                                              USED
                     Maximum Administered H.323 Trunks: 8000  204
           Maximum Concurrently Registered IP Stations: 12000 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: 5000  150
  Maximum Administered Ad-hoc Video Conferencing Ports: 0     0
   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   0
          Maximum TN2602 Boards with 320 VoIP Channels: 128   2
   Maximum Number of Expanded Meet-me Conference Ports: 0     0
```

Figure 6: System-Parameters Customer-Options Form – Page 2

2. On Pages 3 and 4 of the **system-parameters customer-options** form, verify that the bolded fields in the following screenshots are set to "**y**".

```
display system-parameters customer-options                    Page   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? y
                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
```

Figure 7: System-Parameters Customer-Options Form – Page 3

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

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

           IP Attendant Consoles? n
```

Figure 8: System-Parameters Customer-Options Form – Page 4

4.2. Dial Plan and Feature Access Codes

This section briefly describes the dial plan requirements and feature access codes for the sample configuration described in these Application Notes.

1. Enter the **change dialplan analysis** command to provision the dial plan. Note the following dialed strings administered in Figure 9:
   - 3-digit dial access codes (indicated with a **Call Type** of "**dac**") beginning with the digit "**1**" – Trunk Access Codes (TACs) defined for trunk groups in this sample configuration conform to this format.
   - 5-digit extensions with a **Call Type** of "**ext**" beginning with the digits "**5**" – local extensions for Communication Manager stations, voicemail access, meet-me conferences, etc. in this sample configuration conform to this format.
   - Single-digit ("**9**") feature access code (indicated with a **Call Type** of "**fac**") – This dialed string will be interpreted as a Feature Access Codes (FAC). In this sample configuration, "**9**" is used as the user-dialed prefix for outbound calls to the PSTN.

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

       Dialed   Total  Call    Dialed   Total  Call     Dialed   Total  Call
       String   Length Type    String   Length Type     String   Length Type
     1            3    dac
     5            5    ext
     9            1    fac
```

Figure 9: Dialplan Analysis Form

2. Enter the **change feature-access-codes** command. On Page 1 of the **feature-access-codes** form, provision an access code for **Auto Route Selection (ARS) – Access Code 1** that is valid under the administered dial plan in **Step 1**. In this sample configuration, ARS is used for routing calls to the PSTN, and the access code ("**9**" in **Figure 10**) entered here is used as the user-dialed prefix for outbound PSTN calls.

```
change feature-access-codes                                 Page   1 of   8
                               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:
                       Answer Back Access Code:
                         Attendant Access Code:
      Auto Alternate Routing (AAR) Access Code:
    Auto Route Selection (ARS) - Access Code 1: 9     Access Code 2:
                 Automatic Callback Activation:        Deactivation:
```

Figure 10: Feature-Access-Codes Form

4.3. IP Network Parameters

These Application Notes assume that the appropriate IP network regions and IP codec sets have already been administered to support internal calls, i.e., calls within the Avaya site. For simplicity in this sample configuration, all Communication Manager elements, e.g., stations, C-LAN and MedPro boards, etc., within the Avaya site are assigned to a single IP network region and all internal calls use a single IP codec set. This section describes the steps for administering additional IP network regions to represent the PSTN and the TDM gateway sites, and another IP codec set for external calls, i.e., calls between the Avaya site and the PSTN and the TDM gateway sites.

1. Enter the **change ip-codec-set *ci*** command, where ***ci*** is the number of an IP codec set used only for internal calls. On Page 1 of the **ip-codec-set** form, ensure that "**G.729B**" and "**G.729A**" are included in the codec list as shown in **Figure 11**.

Note - The **Frames Per Pkt** and **Packet Size (ms)** values for **G.729B** and **G.729A** are set according to the requirements of the AT&T IP Flexible Reach service.

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

                          IP Codec Set

    Codec Set: 1

    Audio        Silence      Frames   Packet
    Codec        Suppression  Per Pkt  Size(ms)
 1: G.711MU          n          2        20
 2: G.729B           n          3        30
 3: G.729A           n          3        30
```

Figure 11: IP-Codec-Set Form for Internal Calls – Page 1

- On Page 2 of the **ip-codec-set** form, set **FAX Mode** to "**t.38-standard**".

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

                          IP Codec Set

                              Allow Direct-IP Multimedia? n

                    Mode                 Redundancy
    FAX             t.38-standard         0
    Modem           off                   0
    TDD/TTY         US                    3
    Clear-channel   n                     0
```

Figure 12: IP-Codec-Set Form for Internal Calls – Page 2

- Repeat this step as necessary for each IP codec set used only for internal calls.

2. Enter the **change ip-codec-set *ce*** command, where ***ce*** is the number of an unused IP codec set. This IP codec set will be used for external calls. On Page 1 of the **ip-codec-set** form, provision the codecs in the order shown in **Figure 13**.

Note - The **Frames Per Pkt** and **Packet Size (ms)** values for **G.729B** and **G.729A** are set according to the requirements of the AT&T IP Flexible Reach service.

```
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          3        30
 2: G.729A           n          3        30
 3: G.711MU          n          2        20
```

Figure 13: IP-Codec-Set Form for External Calls – Page 1

On Page 2 of the **ip-codec-set** form, set **FAX Mode** to “**t.38-standard**”.

```
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         US                   3
    Clear-channel   n                    0
```

Figure 14: IP-Codec-Set Form for External Calls – Page 2

3. Enter the **change ip-network-region nrp**, where **nrp** is the number of an unused IP network region. This IP network region will be used to represent the PSTN. On Page 1 of the **ip-network-region** form, set the **UDP Port Min** and **UDP Port Max** to "**16384**" and "**32767**" (this port range is an AT&T IP Flexible Reach service requirement).

```
change ip-network-region 3                                      Page   1 of  19
                               IP NETWORK REGION
  Region: 3
Location:         Authoritative Domain:
    Name: ATT PSTN
MEDIA PARAMETERS                Intra-region IP-IP Direct Audio: yes
      Codec Set: 2              Inter-region IP-IP Direct Audio: yes
   UDP Port Min: 16384                     IP Audio Hairpinning? n
   UDP Port Max: 32767
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 15: IP-Network-Region Form for the Network Region Representing the PSTN

- Repeat **Step 3** to administer another IP network region to represent the TDM gateway site. IP Network Region **4** was setup for calls to TDM Gateway.

4. Enter the **change ip-network-region nrl**, where **nrl** is the number of an IP network region administered for local Communication Manager Elements within the Avaya site. On Page 1 of the **ip-network-region** form, set the **UDP Port Min** and **UDP Port Max** to "**16384**" and "**32767**".

```
change ip-network-region 2                                      Page   1 of  19
                               IP NETWORK REGION
  Region: 2
Location:         Authoritative Domain: devconnect.com
    Name: Local
MEDIA PARAMETERS                Intra-region IP-IP Direct Audio: yes
      Codec Set: 1              Inter-region IP-IP Direct Audio: yes
   UDP Port Min: 16384                     IP Audio Hairpinning? y
   UDP Port Max: 32767
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 Form for a Network Region Administered for Local Communication Manager Elements – Page 1

On Page 3 of the **ip-network-region** form, for each IP network region pair consisting of this IP network region as the **src rgn** and the IP network region assigned to the PSTN or TDM gateway sites as the **dst rgn**, provision the following:

- **codec set** – Set to the codec set administered in **Step 2**.
- **direct WAN** – Set to "**y**".
- **WAN-BW-limits** – Set to the maximum number of calls or bandwidth allowed between the two IP network regions. The setting shown in **Figure 17** was used for testing purposes only.

```
change ip-network-region 2                                    Page   3 of  19

 Source Region:  2     Inter Network Region Connection Management

 dst codec direct   WAN-BW-limits   Video      Intervening   Dyn
 rgn  set   WAN  Units    Total Norm  Prio Shr Regions       CAC IGAR AGL
 1    2     y    NoLimit                                          n   all
 2    1
 3    2     y    NoLimit                                          n   all
 4    2     y    NoLimit                                          n   all
```

Figure 17: IP-Network-Region Form for an IP Network Region Administered for Local Communication Manager Elements – Page 3

Repeat this step as necessary for other IP network regions administered for local Communication Manager Elements.

5. Enter the **list node-names** command, and note the node names and IP addresses of the SES server used in **Section 4.4.1** and **Section 4.5.1** (combined Home/Edge server or Home server in separate SES Home/Edge configurations), as well as of the C-LAN board used in **Section 4.4.1**, **Step 1** and **Section 5.3**, **Step 1**.

```
list node-names

                        NODE NAMES

Type      Name              IP Address
IP        CLAN-1A02         192.168.62.45
IP        Gateway           192.168.62.1
IP        MEDPRO-1B07       192.168.62.47
IP        SES               192.168.62.55
IP        VAL-1B04          192.168.62.60
IP        default           0.0.0.0
IP        procr             .  .  .
```

Figure 18: List Node-Names Form

4.4. Outbound Calls

In this sample configuration, since the AT&T IP Flexible Reach service required sites to send 11-digit called party numbers for outbound calls, the administration steps that follow in this section reflect that requirement. In actual deployment, customers and AT&T must agree on the called party number length to be sent by the customer sites.

4.4.1. Outbound PSTN Calls

This section describes the steps for administering the routing of outbound calls to the PSTN.

1. Enter the **add signaling-group spo** command, where **spo** is the number of an unused signaling group, and provision the following:
   - **Group Type** – Set to "**sip**".

- **Transport Method** – Set to "**tls**". Note that this is only the transport protocol used between Communication Manager and SES. The transport protocol used between SES and the AT&T IP Flexible Reach service is UDP.
- **Near-end Node Name** – Set to the node name of the first C-LAN board noted in **Section 4.3**, **Step 5**.
- **Far-end Node Name** – Set to the node name of the SES server noted in **Section 4.3**, **Step 5**.
- **Near-end Listen Port** and **Far-end Listen Port** – set to default "**5061**" for Transport Method "tls".
- **Far-end Network Region** – Set to the IP network region administered in **Section 4.3**, **Step 3** to represent the PSTN.
- **Far-end Domain** – Enter the IP address of the AT&T IP Flexible Reach service Border Element.
- **DTMF over IP** – Set to "**rtp-payload**" to enable Communication Manager to use DTMF according to RFC 2833.
- **Direct IP-IP Audio Connections** – Set to "**y**", indicating that the RTP paths should be optimized to reduce the use of MedPro resources when possible.

```
add signaling-group 3                                           Page   1 of   1

 Group Number: 3             Group Type: sip
                       Transport Method: tls


   Near-end Node Name: CLAN-1A02              Far-end Node Name: SES
 Near-end Listen Port: 5061                 Far-end Listen Port: 5061
                                         Far-end Network Region: 3
       Far-end Domain: 10.94.1.10

                                            Bypass If IP Threshold Exceeded? n

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

Figure 19: Signaling-Group Form for Outbound PSTN Calls

2. Enter the **add trunk-group tp** command, where **tp** is the number of an unused trunk group. On Page 1 of the **trunk-group** form, provision the following:
   - **Group Type** – Set to "**sip**".
   - **Group Name** – Enter a descriptive name.
   - **TAC** – Enter a trunk access code that is consistent with the dial plan.
   - **Direction** – set to "**outgoing**" so that this trunk group will be used for both outbound and inbound calls with the PSTN.
   - **Service Type** – Set to "**public-ntwrk**".
   - **Signaling Group** – Set to the number of the signaling group administered in **Step 1**.

- **Number of Members** – Enter the maximum number of simultaneous calls permitted on this trunk group.

```
add trunk-group 3                                               Page   1 of  21
                                TRUNK GROUP

Group Number: 3                    Group Type: sip          CDR Reports: y
  Group Name: ATT PSTN                    COR: 1        TN: 1          TAC: 103
   Direction: outgoing        Outgoing Display? n
 Dial Access? n
Queue Length: 0
Service Type: public-ntwrk

                                                       Signaling Group: 3
                                                     Number of Members: 10
```

Figure 20: Trunk-Group Form for Outbound PSTN Calls – Page 1

3. On Page 2 of the **trunk-group** form, set the **Preferred Minimum Refresh Interval(sec)** field to **900**.

```
add trunk-group 3                                               Page   2 of  21
   Group Type: sip

TRUNK PARAMETERS

   Unicode Name: auto

                                          Redirect on OTPIM Failure: 5000

         SSCAN? n                                Digital Loss Group: 18
                    Preferred Minimum Session Refresh Interval(sec): 900
```

Figure 21: Trunk-Group Form for Outbound PSTN Calls – Page 2

4. On Page 4 of the **trunk-group** form,
   - **Send Diversion Header** – Set to "**y**" (See note in **Section 2.2.3**)
   - **Support Request History** – Set to "**n**"
   - **Telephone Event Payload Type -** Set to the RTP payload type required by the AT&T IP Flexible Reach service.

```
add trunk-group 3                                               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
                   Send Diversion Header? y
                 Support Request History? n
            Telephone Event Payload Type: 100
```

Figure 22: Trunk-Group Form for Outbound PSTN Calls – Page 4

5. Repeat **Steps 1–4** for any additional AT&T IP Flexible Reach service Border Elements.

6. Enter the **change ars analysis d** command, where **d** is any digit(s). In the **ars analysis** form, provision an entry for a PSTN destination as follows:

   - **Dialed String –** Enter "**1**" (the country code in this sample configuration) followed by enough leading digits to uniquely identify a PSTN destination number range.
   - **Total Min** and **Max –** Enter "**11**" for Call Type "**natl**" and **10** and **18** for **Call Type** "**intl**".
   - **Route Pattern –** Enter the number of an unused route pattern. The route pattern will specify the trunk group(s) to be used for outbound calls matching this entry.
   - **Call Type –** Enter "**natl**" if the destination is within USA and "**intl**" if destination it is an international call.

In **Figure 23**, entries are shown for outbound calls to **1-303-xxx-xxxx**, and **1-800-xxx-xxxx.** Typical deployments generally require additional entries, or the use of less exact or wildcard matching strings, to cover all permitted local, long distance, international, operator, N11 service, and other PSTN destination numbers, but that is beyond the scope of these Application Notes. Ensure that there are entries to cover all permitted PSTN destination numbers. Refer to **[1]** and **[2]** for further information on ARS administration.

```
Change ars analysis 1732                                          Page   2 of   2
                             ARS DIGIT ANALYSIS TABLE
                                   Location:  all        Percent Full:    2

          Dialed           Total      Route     Call   Node  ANI
          String          Min  Max   Pattern    Type   Num   Reqd
    1303                  11   11     3         natl         n
    1800                  11   11     3         natl         n
    011                   10   18     3         intl         n
```

Figure 23: ARS Analysis Form

7. Enter the **change route-pattern rp** command, where **rp** is the route pattern entered in **Step 6**. Provision an entry as follows:
   - **Pattern Name** – Enter a descriptive name.
   - **Secure SIP** – Set to "**n**".
   - **Grp No** – Enter the number of a trunk group administered in **Steps 2–4**.
   - **FRL** – Enter the minimum Facility Restriction Level necessary to use this trunk group, with 0 being the least restrictive.
   - **Pfx Mrk** – Enter "**1**" so that for outbound 11-digit called party numbers that begin with "**1**", the prefix "**1**" is not removed.
   - **NPA** – [Optional] Enter the "home" area code of the Avaya site so that (in conjunction with Pfx Mrk being set to "**1**") a "**1**" is prefixed to outbound 10-digit called party numbers in the same "home" area code as the Avaya site.

```
Change route-pattern 3                                     Page   1 of   3
                    Pattern Number: 3   Pattern Name: ATT PSTN
                             SCCAN? n     Secure SIP? n
    Grp FRL NPA Pfx Hop Toll No.  Inserted                          DCS/ IXC
    No          Mrk Lmt List Del  Digits                            QSIG
                             Dgts                                   Intw
 1: 3    0  303  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 24: Route-Pattern Form for Outbound PSTN Calls

Provision additional entries as necessary for any additional trunks groups administered in **Step 5**.

4.4.2. Outbound Calls To TDM Gateway Site

Follow steps listed in **Section 4.4.1** to configure for outbound calls to TDM Gateway via the SIP trunk to AT&T IP Flexible Reach service. Following were configured for this test:

- Signaling Group 4
- Trunk Group 4
- Route Pattern 4

Note - Trunk Group configured in Section 4.4.1 can be used for outbound calls to TDM Gateway Sites. This configuration step was performed in the sample testing for debugging purposes.

4.4.3. Calling Party Number

Enter the **change public-unknown-numbering 0** command to specify the calling party numbers sent on outbound calls. In the **public-unknown-numbering** form, for each local extension range assigned to Avaya Communication Manager Phones, provision one entry for outbound PSTN calls and another entry for outbound calls to the TDM gateway sites as follows:

- **Ext Len** – Enter the total number of digits in the local extension range.
- **Ext Code –** Enter enough leading digits to identify the local extension range.
- **Trk Grp(s)** – For outbound PSTN calls enter the number of the trunk group administered in **Section 4.4.1**, **Steps 2–4**. For outbound calls to the TDM gateway sites, enter the number of the trunk group administered in **Section 4.4.2**.
- **CPN Prefix** – If necessary, enter enough prefix digits to form the desired calling party number.
- **CPN Len** – Enter the total length of the calling party number to be sent.

In **Figure 25**, for outbound calls placed by Communication Manager phones with extensions 50xxx, 10-digit calling party numbers 7323250xxx are sent for outbound PSTN calls.

Provision as many entries as necessary to cover all local extension ranges assigned to Communication Manager Phones.

```
Change public-unknown-numbering 0                                Page   1 of   2
                      NUMBERING - PUBLIC/UNKNOWN FORMAT
                                                  Total
Ext Ext             Trk       CPN                  CPN
Len Code            Grp(s)    Prefix               Len
                                                           Total Administered: 23
 5  5                                              5          Maximum Entries: 9999
 5  50              3-4       73232                10
 5  54              3-4       73232                10
```

Figure 25: Public-Unknown-Numbering Form

4.5. Inbound Calls

In this sample configuration, since the AT&T IP Flexible Reach service sends 10-digit and 4-digit called party numbers on inbound calls to the sites, the administration steps that follow in this section reflect that requirement. In actual deployment, customers and AT&T must agree on the called party number length to be sent to customer sites.

4.5.1. Inbound PSTN Calls

This section describes the steps for administering the routing of inbound calls from the PSTN.

1. Enter the **add signaling-group spi** command, where **spi** is the number of an unused signaling group. Administer the signaling group in the same manner as the signaling group administered in **Section 4.4.1**, **Step 1** for outbound calls to the PSTN, with the following exception:

    - **Far-end Domain** – Leave blank to allow this SIP signaling group to handle inbound PSTN calls from any domain. This is a catch all signaling group which means that it will accept calls from any domain or IP address.

Note – To restrict calls to a specific IP address or a domain, that IP address or domain can be entered in the **Far-end Domain** field but in that case for each domain or IP address, a separate Signaling Group and Trunk Group will need to be configured.

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

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


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

                                        Bypass If IP Threshold Exceeded? n

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

Figure 26: Signaling-Group Form for Inbound PSTN Calls

2. Enter the **add trunk-group tpi** command, where **tpi** is the number of an unused trunk group. Administer the trunk group in the same manner as the trunk group administered in **Section 4.4.1**, **Steps 2–4** for outbound calls to the PSTN, with the following exceptions:
   - **Direction** – Set to "**incoming**".
   - **Signaling Group** – Set to the number of the signaling group administered in **Step 1**.

```
add trunk-group 5                                               Page   1 of  21
                                TRUNK GROUP

Group Number: 5                   Group Type: sip          CDR Reports: y
  Group Name: Incoming Trunk             COR: 1        TN: 1        TAC: 105
   Direction: incoming       Outgoing Display? n
 Dial Access? n
Queue Length: 0
Service Type: public-ntwrk

                                                      Signaling Group: 5
                                                    Number of Members: 20
```

Figure 27: Trunk-Group Form for Inbound PSTN Calls – Page 1

3. Enter the **change inc-call-handling-trmt trunk-group tp** command, where **tp** is the number of the trunk group administered in **Steps 1–2**. In the **inc-call-handling-trmt trunk-group** form, provision an entry for a DID number / number range as follows:
   - **Called Len** – Enter the total number of digits in the DID number / number range.
   - **Called Number** – Enter enough leading digits to uniquely match the DID number / number range.
   - **Del** and **Insert** – If necessary, enter the number of leading digits that need to be deleted from the DID number / number range, and the specific leading digits that need to be prefixed to the DID number / number range (after any deletion is performed), respectively, in order to match a local Communication Manager extension / extension range.

   In this sample configuration, the AT&T IP Flexible Reach service sends 10-digit called party numbers in the range 7323204084 on inbound DID calls. Thus the entry in **Figure 28** matches the called party number deletes the all 10 digits to match the local Communication Manager extension 50001.

   Provision as many entries as necessary to cover all expected DID numbers / number ranges.

```
change inc-call-handling-trmt trunk-group 5                     Page   1 of  30
                       INCOMING CALL HANDLING TREATMENT
 Service/      Called    Called      Del Insert
 Feature        Len      Number
 public-ntwrk    10 7323204084       10  50001
```

Figure 28: Inc-Call-Handling-Trmt Trunk-Group Form for Inbound PSTN Calls

4. If more incoming trunk groups are configured, then similar entries can be made to route the calls to the proper extension.

4.5.2. Connected Party Number

The connected party numbers sent on inbound calls can be specified in the **public-unknown-numbering** form. Configure an entry in the **public-unknown-numbering** form as follows:

- **Ext Len** – Enter the total number of digits in the local extension range.
- **Ext Code –** Enter enough leading digits to identify the local extension range.
- **Trk Grp(s)** – Enter the number of the trunk group administered in **Section 4.5.1**, **Step 2**.
- **CPN Prefix** – If necessary, enter enough prefix digits to form the desired connected party number.
- **CPN Len** – Enter the total length of the connected party number to be sent.

Provision as many entries as necessary to cover all local extension ranges assigned to Communication Manager Phones.

```
Change public-unknown-numbering 0                               Page   1 of   2
                      NUMBERING - PUBLIC/UNKNOWN FORMAT
                                         Total
Ext Ext             Trk       CPN             CPN
Len Code            Grp(s)    Prefix          Len
                                                        Total Administered: 23
 5  5               5                         5            Maximum Entries: 9999
```

Figure 29: Public-Unknown-Numbering Form

4.6. SIP Phones

Notes:

1. A separate signaling group and trunk group needs to be administered for SIP phones, but that is beyond the scope of these Application Notes. Refer to **[3]** and **[5]** for further details.
2. The DTMF_PAYLOAD_TYPE parameter in the "46xxsettings.txt" configuration file that Avaya SIP phones download (from a file server) during startup must be configured to match the RTP payload type required by the AT&T IP Flexible Reach service. Contact AT&T or examine a SIP trace of an inbound call from the AT&T IP Flexible Reach service to determine this value. Refer to **[5]** for further details.
3. The rest of this section addresses a known issue in Communication Manager Release 5.2.1 involving Avaya SIP phones and Direct IP-IP media (a.k.a. "shuffling"). This configuration step can be skipped when the issue is resolved.

Set **Direct-IP-IP Audio Connections** to "**n**" on the signaling group administered for SIP phones.

```
Change signaling-group 2                                      Page   1 of   1

 Group Number: 2              Group Type: sip
                        Transport Method: tls


   Near-end Node Name: CLAN-1A05            Far-end Node Name: SES
 Near-end Listen Port: 5061               Far-end Listen Port: 5061
                                       Far-end Network Region: 1
      Far-end Domain: devconnect.com

                                          Bypass If IP Threshold Exceeded? n

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

Figure 30: Signaling-Group Form for SIP Phones

4.7. Optional Features

The reference configuration uses hunt groups, vectors, and Vector Directory Numbers (VDNs), to provide additional functionality during testing:

- Hunt Group 1 – Modular Messaging coverage for Communication Manager extensions.
- VDN 2000/Vector 2 – Auto-attendant.
- VDN 2001/Vector 3 – Meet-me Conference

Note - The administration of Communication Manager Call Center elements – hunt groups, vectors, and Vector Directory Numbers (VDNs) are beyond the scope of these Application Notes. Additional licensing may be required for some of these features. Refer to **[1]**, **[2]**, **[7]**, and **[8]** for further details if necessary. The samples that follow are provided for reference purposes only.

4.7.1. Hunt Group for Station Coverage to Modular Messaging

Hunt group 2 is used in the reference configuration to verify the Send-All-Calls functionality. The hunt group (e.g. **2**) is defined with the 5 digit Modular Messaging pilot number (e.g. **55000** in **Figure 32**). The hunt group is associated with a coverage path (e.g.**h2** in **Figure 33**) and the coverage path is assigned to a station (e.g. **50001** in **Figure 34**).

```
display hunt-group 1                                            Page   1 of  60
                                  HUNT GROUP
            Group Number: 2                                 ACD? n
              Group Name: MM Voicemail                    Queue? n
         Group Extension: 55000                          Vector? n
              Group Type: ucd-mia                  Coverage Path:
                      TN: 1          Night Service Destination:
                     COR: 1                     MM Early Answer? n
           Security Code:                Local Agent Preference? n
 ISDN/SIP Caller Display: mbr-name
```

Figure 31: Hunt Group 1Form – Page 1

```
display hunt-group 1                                            Page   2 of  60
                                  HUNT GROUP
                      Message Center: sip-adjunct
    Voice Mail Number        Voice Mail Handle         Routing Digits
                                                (e.g., AAR/ARS Access Code)
    55000                    55000                     8
```

Figure 32: Hunt Group 1 Form – Page 2

```
display coverage path 1
                                 COVERAGE PATH
                   Coverage Path Number: 2
     Cvg Enabled for VDN Route-To Party? n        Hunt after Coverage? n
                       Next Path Number:          Linkage
COVERAGE CRITERIA
    Station/Group Status    Inside Call     Outside Call
             Active?            n               n
               Busy?            y               y
       Don't Answer?            y               y        Number of Rings: 3
                All?            n               n
 DND/SAC/Goto Cover?            y               y
   Holiday Coverage?            n               n
COVERAGE POINTS
    Terminate to Coverage Pts. with Bridged Appearances? n
  Point1: h2              Rng: 2  Point2:
  Point3:                         Point4:
  Point5:                         Point6:
```

Figure 33: Coverage Path 1 Form

```
display station 50001                                          Page   1 of   5
                                     STATION

Extension: 50001                         Lock Messages? n              BCC: 0
     Type: 9620                          Security Code: xxxxxx          TN: 1
     Port: S00000                      Coverage Path 1: 2              COR: 1
     Name: H323-9630                   Coverage Path 2:                COS: 1
                                       Hunt-to Station:
STATION OPTIONS
                                           Time of Day Lock Table:
              Loss Group: 19        Personalized Ringing Pattern: 1
                                                 Message Lamp Ext: 50001
            Speakerphone: 2-way               Mute Button Enabled? y
        Display Language: english
 Survivable GK Node Name:
          Survivable COR: internal              Media Complex Ext:
   Survivable Trunk Dest? y                          IP SoftPhone? n
                                              Customizable Labels? y
```

Figure 34: Station 50001 Form

4.7.2. Auto Attendant

Basic auto-attendant functionality is defined in the reference configuration for DTMF testing. The auto-attendant is defined by a VDN (e.g. **2001**) and a Vector (e.g. **3**). As with other inbound calls from the AT&T Flexible Reach service, calls may be directed to the auto-attendant VDN extension via SES.

```
display vdn 2001                                              Page   1 of   2
                            VECTOR DIRECTORY NUMBER

                             Extension: 50010
                                 Name*: AutoAttend
                           Destination: Vector Number        3

                  Meet-me Conferencing? n
                    Allow VDN Override? n
                                   COR: 1
                                   TN*: 1
                              Measured: none
```

Figure 35: Auto Attendant VDN

```
display vector 3                                              Page   1 of   6
                                  CALL VECTOR

    Number: 3                Name: AutoAttendant
                                            Meet-me Conf? n           Lock? n
    Basic? y   EAS? n   G3V4 Enhanced? y   ANI/II-Digits? y   ASAI Routing? y
Prompting? y   LAI? n  G3V4 Adv Route? n   CINFO? n   BSR? n   Holidays? n
 Variables? n   3.0 Enhanced? n
01 wait-time    4   secs hearing ringback
02 collect      5    digits after announcement 33017
03 route-to     digits with coverage n
04 wait-time    5   secs hearing silence
05 stop
06
07
```

Figure 36: Auto Attendant Vector

4.7.3. Meet-me Conference

A basic meet-me conference functionality is defined in the reference configuration for DTMF testing. The meet-me conference functionality is defined by a VDN (e.g. **2000**) and a Vector (e.g. **2**). As with other inbound calls from the AT&T Flexible Reach service, calls may be directed to the meet-me conference VDN extension via SES.

```
display vdn 2000                                                Page   1 of   2
                            VECTOR DIRECTORY NUMBER

                             Extension: 2000
                                  Name: MeetMe Conference
                           Destination: Vector Number        2

                  Meet-me Conferencing? y
                                   COR: 1
                                    TN: 1
```

Figure 37: Meet-me Conference VDN – Page 1

```
display vdn 2000                                                Page   2 of   2
                            VECTOR DIRECTORY NUMBER

                  MEET-ME CONFERENCE PARAMETERS:

                    Conference Access Code: xxxxxx
                     Conference Controller: 50001
                           Conference Type: 6-party
```

Figure 38: Meet-me Conference VDN – Page 2

```
display vector 2                                                  Page   1 of   6
                                  CALL VECTOR
    Number: 2                  Name: MeetMe
                                             Meet-me Conf? y           Lock? y
     Basic? y   EAS? n   G3V4 Enhanced? y   ANI/II-Digits? y   ASAI Routing? y
 Prompting? y   LAI? n  G3V4 Adv Route? n   CINFO? n   BSR? n   Holidays? n
 Variables? n   3.0 Enhanced? n
01 wait-time    5   secs hearing ringback
02 collect      6    digits after announcement 33011
03 goto step    5             if digits          =      meet-me-access
04 goto step    2             if unconditionally
05 announcement 33014
06 route-to     meetme
07 stop
08
```

Figure 39: Meet-me Conference Vector

5. Avaya Modular Messaging

In this reference configuration, Avaya Modular Messaging is used to verify DTMF, Message Wait Indicator (MWI), as well as basic call coverage functionality. Avaya Modular Messaging used in the reference configuration is provisioned for Multi-Site mode. Multi-Site mode allows Avaya Modular Messaging to server subscribers in multiple locations. The administration for Modular Messaging is beyond the scope of these Application Notes. Refer to **[10]** and **[11]** for further details.

6. Configure Avaya Aura™ SIP Enablement Services

This section describes the administration steps for SES in support of SIP trunking with the AT&T IP Flexible Reach service. These Application Notes assume that the necessary SES licenses have been installed and basic SES administration has already been performed. Refer to **[3]**, **[4]**, and **[6]** for further details.

6.1. Background

The sample configuration described in these Application Notes explicitly show SES combined Home/Edge server configuration. In this case, a single SES server supports both the Home and Edge roles. Multiple SES servers may exist using a separate Home/Edge configuration as warranted by capacity considerations (predominately for the support of SIP phones). In the separate Home/Edge server configuration:

- The SIP signaling relationship with the AT&T services exists between the AT&T Border Element and the SES Edge server.
- The Communication Manager SIP signaling group relationship exists between a C-LAN (or equivalent) interface and a specific SES Home server.
- SIP message routing between the Home and Edge servers is performed automatically and transparently. However, administration is required on the Home and Edge servers; refer to **[3]**, **[4]**, and **[6]** for further details.
- Only one SES Edge server exists within a given SIP domain. Multiple Home servers may exist as required.
- All 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.

6.2. Host Configuration

1. Launch a web browser, enter http://<IP address of the SES master administrator>/admin in the URL, and log in with the appropriate credentials. Click on **Adminstration->SIP Enablement Services**.

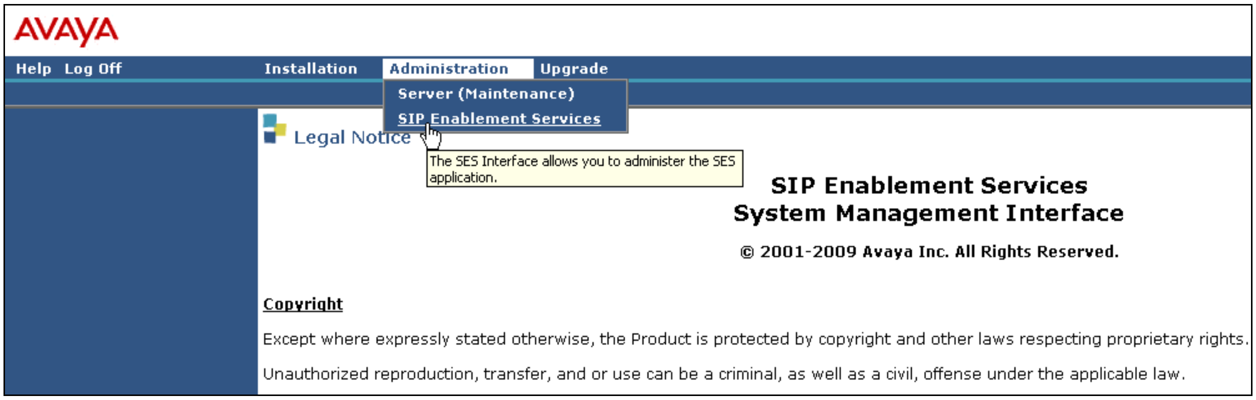

Figure 40: SES Web Interface

2. In the left pane of the SES Administration Interface, expand **Hosts**, and click on "**List**". In the **List Hosts** page, click on "**Edit**".

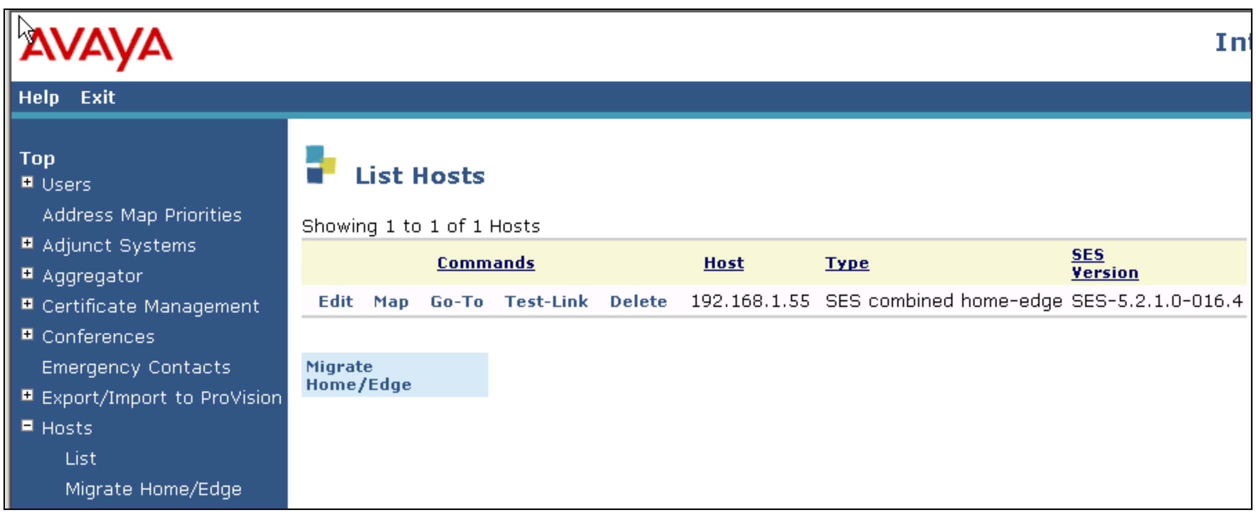

Figure 41: List Hosts Page

3. In the **Edit Host** page, ensure that the following are provisioned:
   - **Listen Protocols** – The "**UDP**", "**TCP**", and "**TLS**" checkboxes are checked.
   - **Link Protocols** – "**TLS**" is selected.
   - **Outbound Proxy** and **Outbound Direct Domains** fields are blank.

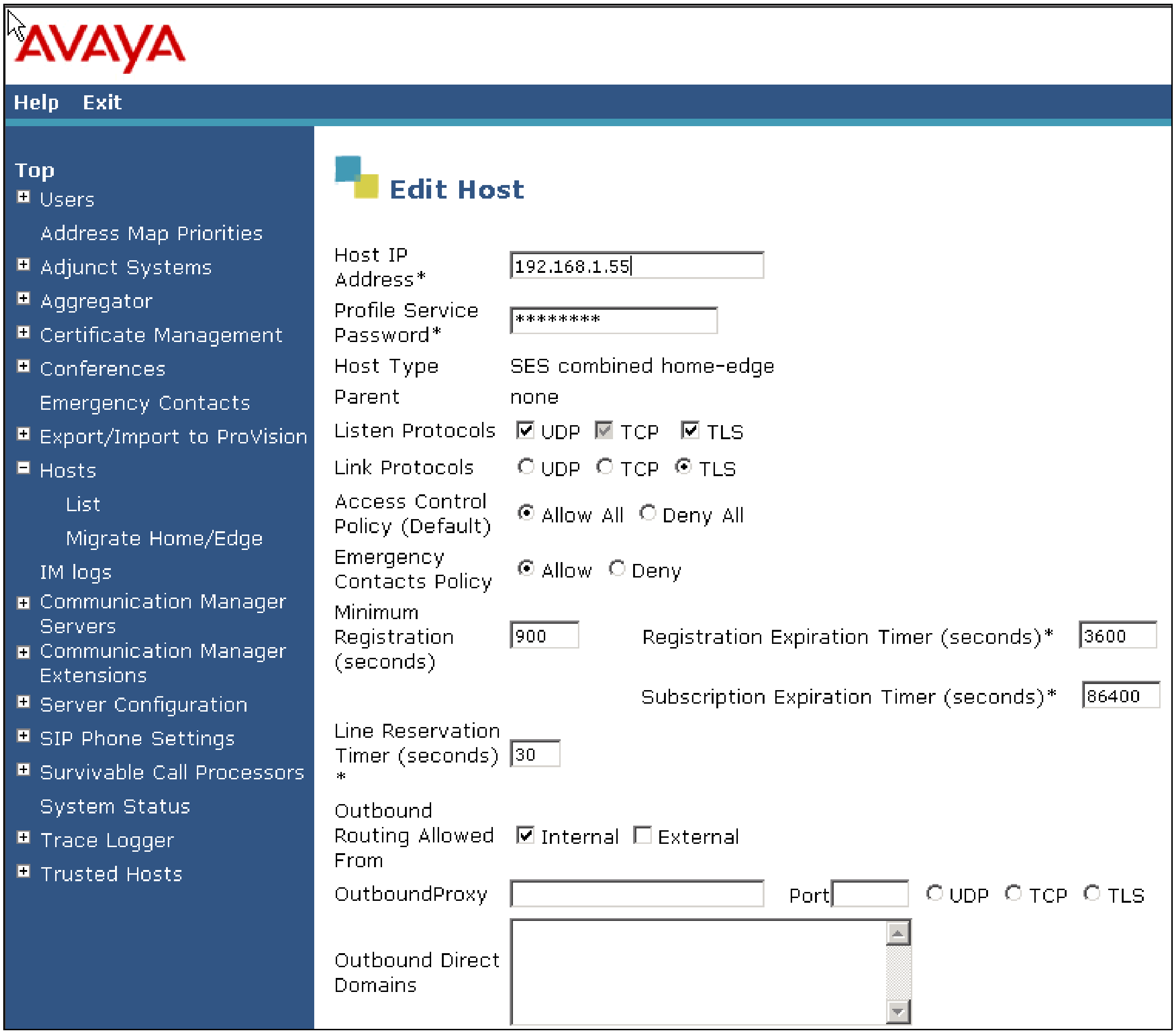

Figure 42: Edit Host Page

Separate Home/Edge Configuration Note:
In the SES separate Home/Edge server configuration, there will be at least two SES hosts listed (Home and Edge servers).

- **SES Edge server** – Configure in the same manner as shown for the combined Home/Edge server above.
- **SES Home servers** – On each SES Home server, set the **Outbound Proxy** to the IP address of the SES Edge server, with **Port** set to "**5061**" and "**TLS**" selected.

6.3. Interfaces to Avaya Aura™ Communication Manager

1. In the left pane of the SES Administration Interface, expand **Communication Manager Servers**, and click on "**Add**". In the **Add Communication Manager Server Interface** page, provision the following:
   - **Communication Manager Server Interface Name** – Enter a descriptive name.
   - **Host** – Select the IP address of the SES combined Home/Edge server.
   - **SIP Trunk Link Type** – Select "**TLS**".
   - **SIP Trunk IP Address** – Enter the IP address of the first C-LAN board noted in **Section 4.3**, **Step 5**.

   Scroll down to the bottom of the page and click on "**Add**" (not shown). Click on "**OK**" and then "**Continue**" in the subsequent confirmation pages (not shown).

   Separate Home/Edge Configuration Note:
   For SES separate Home/Edge server configurations, select the IP address of an SES Home server for **Host**.

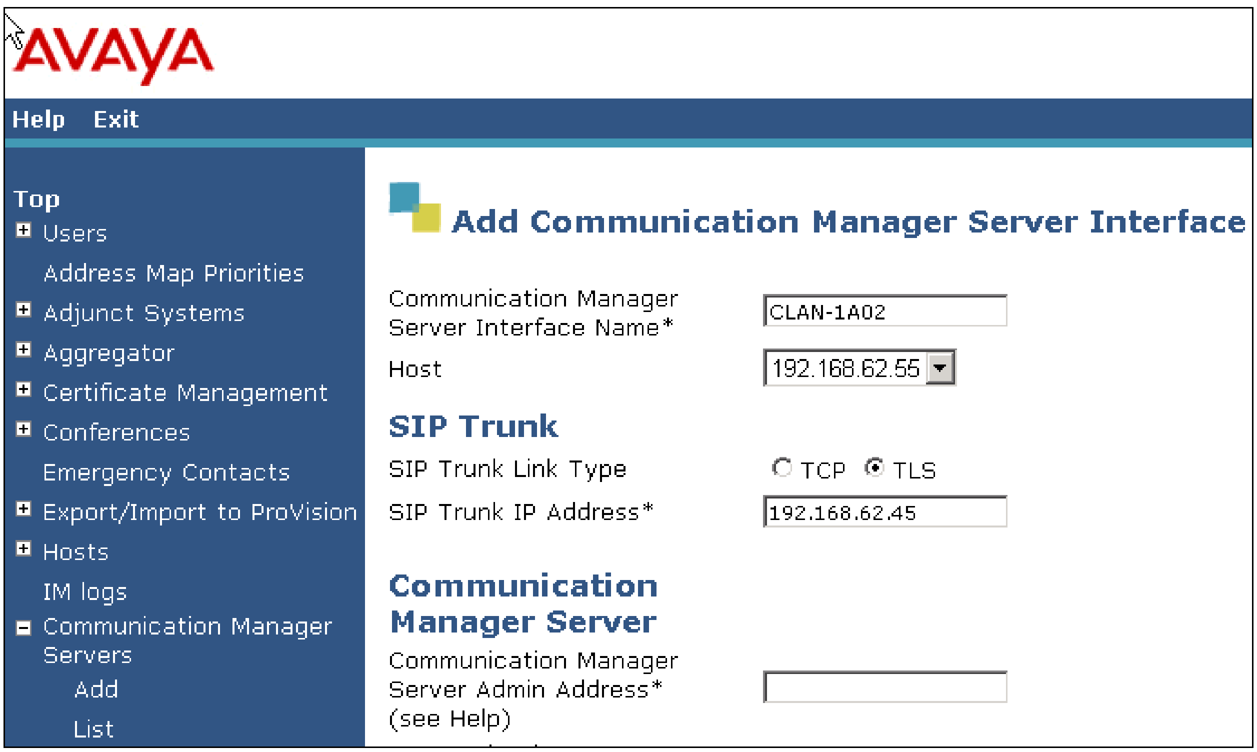

Figure 43: Add Communication Manager Server Interface Page – For First C-LAN

2. Repeat **Step 1** if additional interfaces to Communication Manager need to be configured.

6.4. Call Routing

6.4.1. Background

SES functions as a SIP proxy for the SIP trunking with the AT&T IP Flexible Reach service. SES examines the Request-URI of a received SIP INVITE message (from Communication Manager for outbound calls, and from the AT&T IP Flexible Reach service for inbound calls), modifies the Request-URI and certain SIP headers, and then forwards the message to the appropriate destination.

The Request-URI generally takes the form of *sip:user@domain*, where the *user* part typically contains the called party number digits identifying the phone number being called and the *domain* part can contain a Fully Qualified Domain Name (FQDN) or an IP address. Address maps may be administered in SES to match SIP INVITE messages with certain Request-URI *user* part patterns, i.e., match certain called party numbers; if a match is found, the matching address map can modify the *user* and/or *domain* parts if necessary. The address map patterns are designed to match a range of called party numbers that require identical SIP message routing and must be specific enough to direct each unique called party number to the proper signaling contact. If the resulting *domain* part of a SIP INVITE message (whether the *user* part has been matched and possibly modified by an address map, or not matched at all) does not match the SIP domain configured on SES and the outbound proxy is not specified on SES, then SES routes the message to the IP address (FQDNs are resolved to IP addresses via DNS) indicated in the *domain* part.

In this sample configuration, for outbound calls to the AT&T IP Flexible Reach service, Communication Manager already inserts the appropriate called party number in the Request-URI *user* part and the IP address of the AT&T IP Flexible Reach service Border Element in the *domain* part. Therefore, SES address maps are not required for such outbound calls. For inbound calls from the AT&T IP Flexible Reach service, the Request-URI *domain* part contains the IP address of the SES server. Therefore, one or more SES address maps are required to match the *user* part of such inbound calls and modify the *domain* part in order to properly route the calls to Communication Manager.

Note that with the AT&T IP Flexible Reach service, all calls must be routed to the AT&T IP Flexible Reach service core. Therefore, SES will route all outbound PSTN and TDM gateway site calls to the AT&T IP Flexible Reach service Border Element(s). Similarly, SES will expect to receive all inbound PSTN and TDM gateway site calls from the same AT&T IP Flexible Reach service Border Element(s).

6.4.2. Outbound Calls to AT&T IP Flexible Reach Service

The domain part of the Request-URI for outbound calls to the AT&T IP Flexible Reach service is specified by the Far-end Domain field of the Communication Manager SIP signaling groups administered in **Section 4.4.1**, **Step 1**. In this sample configuration, since the IP address of the AT&T IP Flexible Reach service Border Element is specified in that Far-end Domain field, no additional SES administration is necessary for routing such outbound calls.

6.4.3. Inbound Calls from AT&T IP Flexible Reach Service

SES address maps are used to route inbound calls from the PSTN and the TDM gateway sites to Communication Manager.

1. In the left pane of the SES Administration Interface, expand **Communication Manager Servers**, and click on "**List**". In the **List Communication Manager Servers** page, click on "**Map**" in the row corresponding to the Communication Manager Server Interface administered in **Section 6.3**, **Step 1**. The Communication Manager Server Map to be added will match inbound PSTN calls with a certain called party number / number range and route those calls to Communication Manager.

Figure 44: List Communication Manager Server Page

2. In the **List Communication Manager Server Address Map (not shown)** page, click on "**Add Map in New Group**".

3. In the **Add Communication Manager Server Address Map** page, provision as follows:

   - **Name** – Enter any descriptive name.
   - **Pattern** – Enter a Linux regular expression that matches the number in the user part of the Request-URI, i.e., the called party number, of inbound SIP INVITE messages for PSTN calls from the AT&T IP Flexible Reach service. In this sample configuration, for inbound calls from the PSTN, the AT&T IP Flexible Reach service inserts 10-digit numbers beginning with "7323204xxx" in the user part of the Request-URI to reach local extensions in the Avaya site. Thus, the pattern "**^sip:(732320){0,1}[0-9]{4}**" shown in **Figure 45**, matches SIP INVITE messages with a Request-URI that may begin with "sip:732320" followed by any four consecutive digits or just have four digits.
   - Click on "**Add**"
   - Click on "**Continue**" in the subsequent confirmation page (not shown).

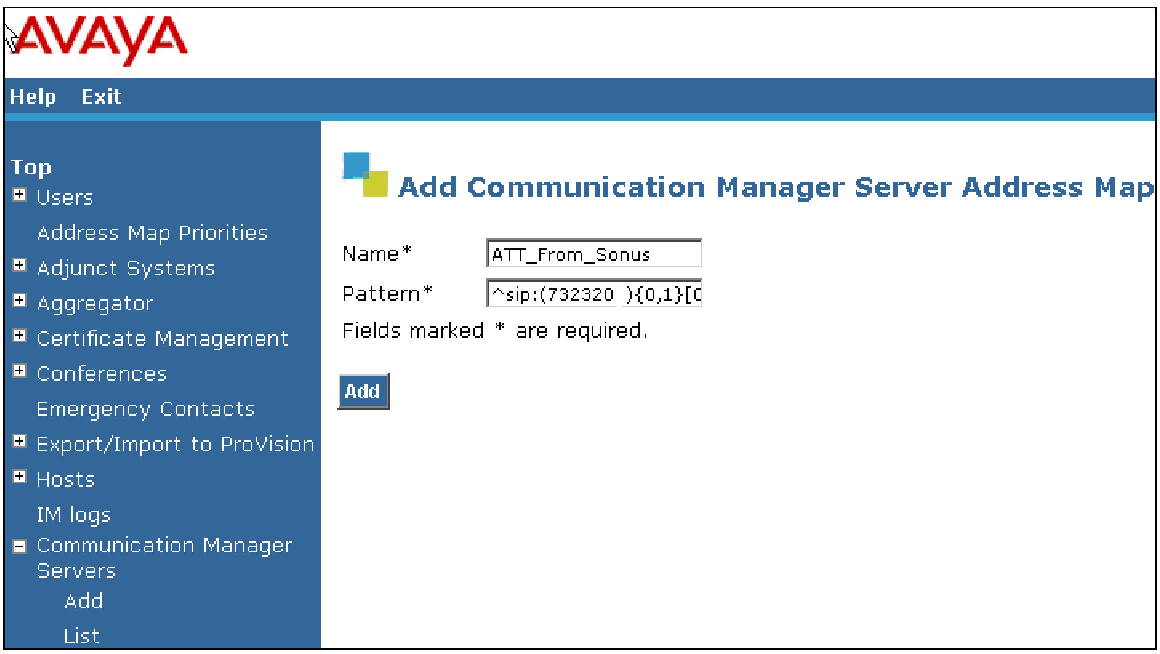

Figure 45: Add Communication Manager Server Address Map Page – Match 10-Digit Numbers

4. A Contact is automatically created after creating the first Communication Manager Server Address Map (see **Figure 46**). The **Contact** specifies that the SIP messages matched by the Communication Manager Server Address Map(s) administered in **Step 3** are to be routed to the IP address of the Communication Manager Server Interface administered in **Section 6.3**, **Step 1**. The "**$(user)**" string in the Contact is substituted with the user part of the original inbound Request-URI.

   - Click on "**Add Another Map**" and repeat **Step 3** for each called party number / number range as necessary.

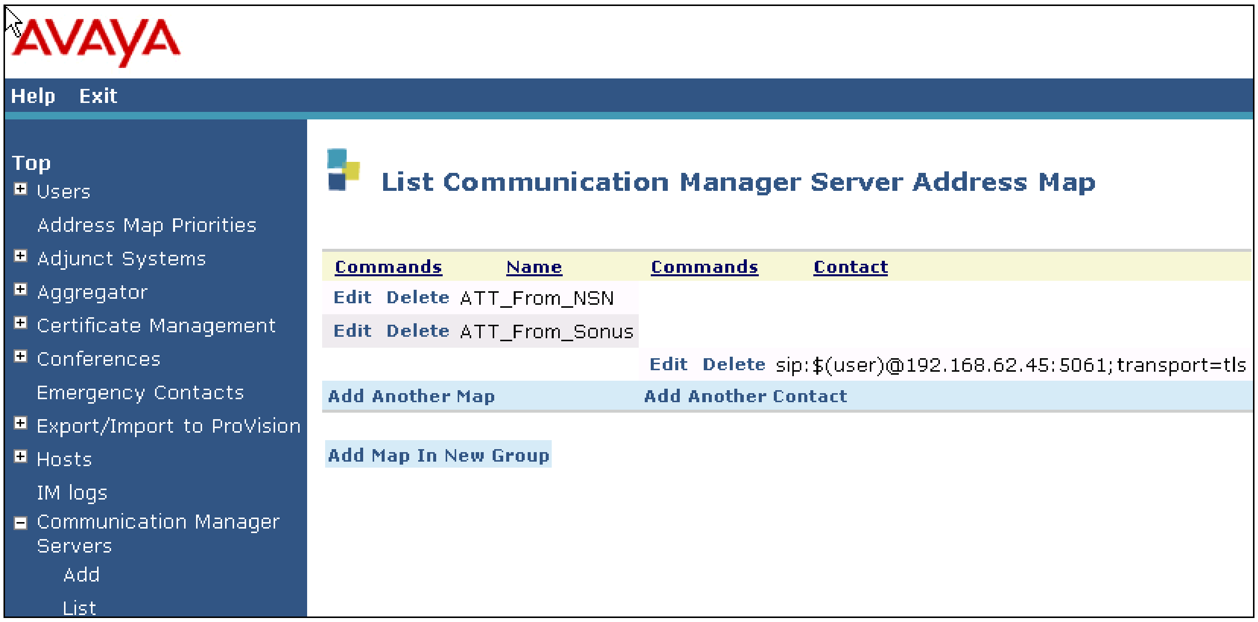

Figure 46: List Communication Manager Server Address Map Page – First C-LAN

6.5. Trusted Host

The AT&T IP Flexible Reach service Border Element IP address must be added as a trusted host entry in SES. SES will not attempt to authenticate incoming requests from trusted hosts.

In the left pane of the SES Administration Interface, expand **Trusted Hosts**, and click on "**Add**". In the **Add Trusted Host** page, provision the following:

- **IP Address** – Enter the IP address of the AT&T IP Flexible Reach service Border Element.
- **Host** – Select the IP address of the SES combined Home/Edge server.
- **Comment** – Enter a description of the trusted host.
- Click on "**Continue**" in the subsequent confirmation page (not shown).
- Repeat the above administration steps for any other AT&T IP Flexible Reach service Border Elements provided.

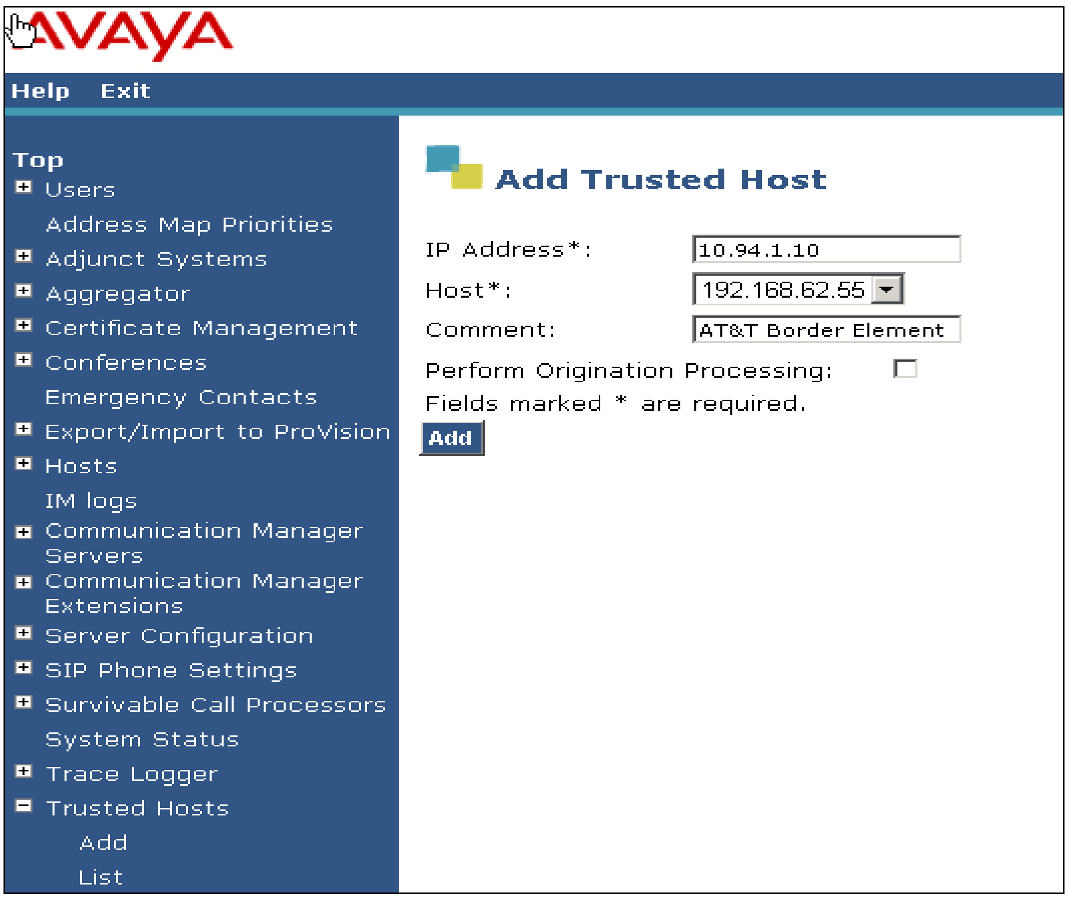

Figure 47: Add Trusted Host Page

Separate Home/Edge Configuration Note:
In the SES separate Home/Edge server configuration, there will be at least two SES hosts listed (Home and Edge servers).

- SES Edge server – Configure the trusted host relationship on the SES Edge server.
- SES Home servers – Configure the trusted host relationship on each SES Home server.

7. General Test Approach and Test Results

The test environment consisted of:

- A simulated enterprise with Avaya Aura™ SIP Enablement Services, Avaya Aura™ Communication Manager, Avaya phones, fax machines (Ventafax application), and Avaya Modular Messaging.
- A laboratory version of the AT&T IP Flexible Reach service, to which the simulated enterprise was connected.

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

- Inbound AT&T IP Flexible Reach service calls to Communication Manager Phones and VDNs/Vectors.
- Inbound and Outbound calls and two-way talk path establishment between PSTN or TDM Gateway Phones and Communication Manager Phones via the AT&T Flexible Reach service.
- Basic supplementary telephony features such as hold, resume, transfer, and conference.
- G.729 codecs for voice calls with PSTN and TDM Gateway endpoints
- T.38 fax calls between Communication Manager and AT&T IP Flexible Reach (PSTN or TDM Gateway) service in G3 and SG3 fax mode.
- DTMF tone transmission using RFC 2833 between Communication Manager and AT&T IP Flexible Reach service (PSTN or TDM Gateway) automated access systems.
- Inbound AT&T IP Flexible Reach service call to Communication Manager that is routed to stations, and if unanswered, covered to Avaya Modular Messaging.
- Long duration calls.
- Direct IP-to-IP media (shuffling).

The test objectives stated in **Section 7**, with limitations as noted in **Section 1.3**, were verified.

8. Verification Steps

8.1. Verification Tests

The following steps may be used to verify the configuration:

1. Verify the outbound PSTN call routing administration on Communication Manager by using the "**list ars route-chosen n**" command, where **n** is a PSTN number (including the leading country code "1" in this sample configuration but not including the ARS code). As shown below, the **ars route-chosen** form shows the **Route Pattern** that would be selected for a given dialed PSTN number.

```
List ars route-chosen 13035381760

                           ARS ROUTE CHOSEN REPORT

     Location:  1                      Partitioned Group Number:  1

      Dialed             Total        Route    Call      Node
      String           Min    Max    Pattern   Type     Number    Location

 1303                  11     11        3      natl                   all
```

Figure 48: ARS Route-Chosen Form

2. Place outbound and inbound calls, and verify that two-way talkpath exists, and that the calls remain stable for several minutes and disconnect properly.

3. For trunk calls established on Avaya 4600 and 9600 Series IP Phones (H.323), enter the "**status station e**" command, where **e** is the extension of the Avaya phone. On Page 5 of the **status station** form, verify that "**G.729AB**" is displayed and the **Audio Connection Type** is "**ip-direct**".

```
Status station 50001                                         Page   5 of   8

                          AUDIO CHANNEL Port: S00000
G.729AB       Switch-End Audio Location:
            IP Address                                  Port  Node Name
Rgn
Other-End:  10. 94.  1. 10                              28588                14
  Set-End: 192.168. 62. 52                              28838                1
Audio Connection Type: ip-direct
```

Figure 49: Status Station Form

8.2. Troubleshooting Tools

The 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 interoperability issues.

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

9. Conclusion

These Application Notes described the steps for configuring Avaya Aura™ Communication Manager and Avaya Aura™ SIP Enablement Services SIP trunking with the AT&T IP Flexible Reach service. The AT&T IP Flexible Reach service allows enterprises in the U.S.A. to place outbound local and long distance and international calls, receive inbound Direct Inward Dialing (DID) calls from the PSTN, and place calls between an enterprise's sites.

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

10. References

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

[1] *Administering Avaya Aura™ Communication Manager*, Issue 5.0, Release 5.2, May 2009, Document Number 03-300509
[2] *Avaya Aura™ Communication Manager Feature Description and Implementation*, Issue 7, Release 5.2, May 2009, Document Number 555-245-205
[3] *SIP Support in Avaya Aura™ Communication Manager Running on the Avaya S8xxx Servers*, Issue 9, May 2009, Document Number 555-245-206
[4] *Installing, Administering, Maintaining, and Troubleshooting Avaya Aura™ SIP Enablement Services*, Issue 7.0, May 2009, Document Number 03-600768
[5] *Avaya one-X™ Deskphone Edition for 9600 Series SIP IP Telephones Administrator Guide Release 2.5*, Issue 5, November 2009, Document Number 16-601944
[6] *Avaya Aura™ SIP Enablement Services (SES) Implementation Guide*, Issue 6, May 2009, Document Number 16-300140
[7] *Avaya Aura™ Call Center 5.2 Call Vectoring and Expert Agent Selection (EAS) Reference*, Release 5.2, April 2009, Document Number 07-600780
[8] *Avaya Aura™ Call Center 5.2 Automatic Call Distribution Reference*, Release 5.2, April 2009, Document Number 07-602568
[9] *Modular Messaging Multi-Site Guide Release 5.2*, November 2009
[10] *Modular Messaging for Microsoft Exchange Release 5.2 Installation and Upgrades*, Issue 1.0, November 2009

AT&T IP Flexible Reach and AT&T VPN Service Descriptions:
[11] *AT&T IP Flexible Reach*
http://www.business.att.com/enterprise/Service/business-voip-enterprise/network-based-voip-enterprise/ip-flexible-reach-enterprise/

Please e-mail any questions or comments pertaining to these Application Notes along with the full title name and filename, located in the lower right corner, directly to the Avaya DevConnect program at devconnect@avaya.com.