Avaya Solution & Interoperability Test Lab

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

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 Toll Free service using MIS-PNT transport service connections.

The AT&T IP Toll Free service is one of several SIP-based Voice over IP services offered to enterprises for a variety of voice communications needs. Note that these Application Notes do NOT cover the AT&T IP Transfer Connect service option of the AT&T IP Toll Free service.

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

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1. Introduction

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

The AT&T IP Toll Free service is one of several SIP-based Voice over IP (VoIP) services offered to enterprises for a variety of voice communications needs. **Note that these Application Notes do NOT cover the AT&T IP Transfer Connect service option of the AT&T IP Toll Free service**. Avaya Aura™ Communication Manager and Avaya Aura™ SIP Enablement Services SIP trunking with AT&T IP Transfer Connect service option will be addressed in separate Application Notes.

1.1. Interoperability Compliance Testing

The interoperability compliance testing focused on verifying inbound call flows (see Section 2.2) from AT&T IP Toll Free service to Avaya Aura™ SIP Enablement Services and Avaya Aura™ SIP Communication Manager.

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

- SIP Trunking
- T.38 Fax
- Legacy Transfer Connect
- Alternate Destination Routing

1.2. Support

AT&T customers may obtain support for the AT&T IP Toll Free service by calling (888)325-5555.

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

1.3. Known Limitations

1. SIP phones were only tested for calls delivered to stations and not to agents as part of the reference configuration used to validate this solution. Calls to SIP phones were not shuffled due to a known issue with Aura™ Communication Manager 5.2.1.
2. If Avaya Aura™ Communication Manager receives an SDP offer with multiple codecs, where at least two of the codecs are supported in the codec set provisioned on Avaya Aura™ Communication Manager, then Avaya Aura™ Communication Manager selects a codec according to the priority order specified in the Avaya Aura™ Communication Manager codec set, not the priority order specified in the SDP offer. For example, if the AT&T IP Toll Free service offers G.711, G.729A, and G.726 in that order, but the Avaya Aura™ Communication Manager codec set contains G.729B, G729A, and G.711 in that order, then Avaya Aura™ Communication Manager selects G.729A, not G.711. The practical resolution is to provision the Avaya Aura™ Communication Manager codec set to match the expected codec priority order in AT&T IP Toll Free SDP offers.
3. G.711 faxing is not supported over the AT&T IP Toll Free 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 by Avaya Aura™ Communication Manager, the maximum T.38 fax transmission was limited to 9600 in this Reference Configuration.
5. G.726 codec is not supported between Avaya Aura™ Communication Manager and the AT&T IP Toll Free service.
6. An AT&T IP Toll Free 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 Toll Free/Avaya certified Session Border Controller (SBC) will be required.

2. Reference Configuration

The sample configuration used in these Application Notes is shown below 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 S8720 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 Toll Free service. SES also provides registrar services to SIP phones in the enterprise.
- Avaya "office" phones include Avaya H.323, Analog and Digital phones, and Avaya one-X™ Agent, which is a Softphone that runs on a Desktop (not shown). Avaya SIP phones were only used for call delivery to a phone extension and not to an agent.

- Avaya Modular Messaging system provides the corporate voice messaging capabilities for enterprise users. The provisioning of Modular Messaging system is beyond the scope of this document.
- IP Network Address Translation (NAT) devices, firewalls, Application Layer Gateway (ALG) devices, and Session Border Controller (SBC) devices that may exist between the enterprise site and the AT&T IP Toll Free 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.

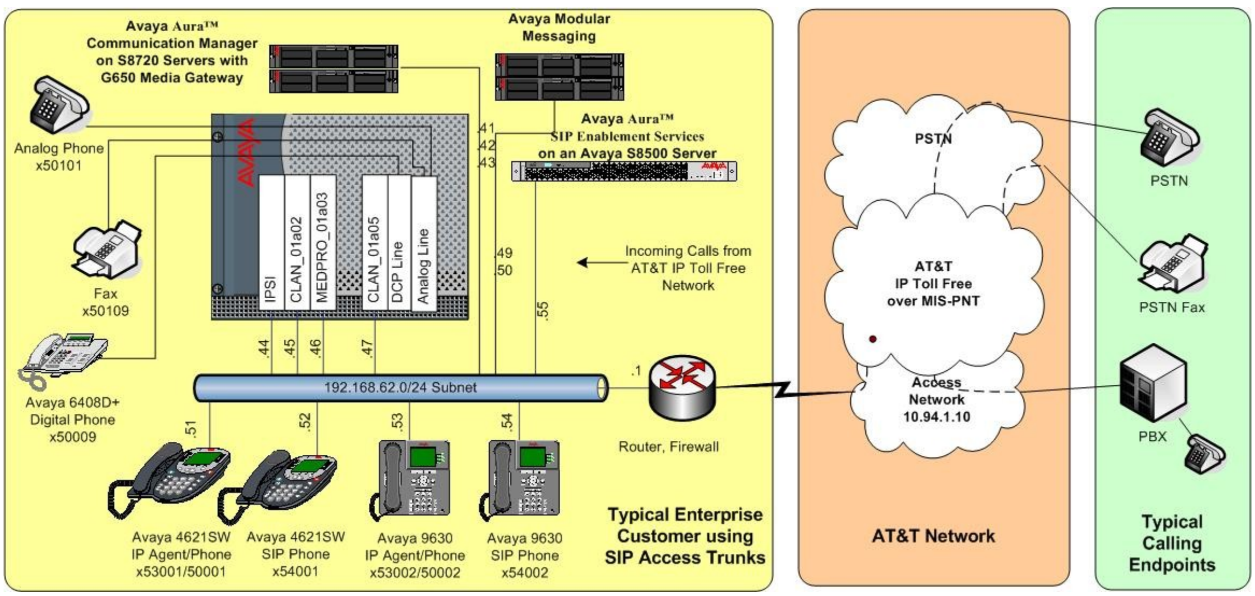

Figure 1: Reference SIP Trunking Configuration

2.1. Illustrative Configuration Information

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

Note - The AT&T IP Toll Free service border element IP address and DNIS digits (Destination digits specified in the SIP Request URIs sent by AT&T Toll Free service), shown in this document are examples. AT&T Customer Care will provide the actual IP addresses as part of the IP Toll Free provisioning process.

Component	Illustrative Value in these Application Notes
Avaya Aura™ SIP Enablement Services	
IP Address	192.168.62.55
Avaya Aura™ Communication Manager	
C-LAN IP Address	192.168.62.45
VDN	2010, 2011, 2012, 2013
Skill (Hunt Group)	11, 12, 13
Agent Extensions	53001, 53002, 53003, 53004, 53005
Hunt Group Extensions	53011 (11), 53012 (12), 53013 (13)
Phone Extensions	50001, 50002, 50003, 50009, 50101, 54001, 54002
Voice Messaging Pilot Extension	55000
Avaya Modular Messaging	
Messaging Application Server (MAS) IP Address	192.168.62.49
Exchange Server IP Address	192.168.62.50
Pilot Number	17323255000
AT&T IP Toll Free Service	
Border Element IP Address	10.94.1.10
Digits Passed in SIP Request-URI	00000104x, 00000105x

Table 1: Illustrative Values Used in these Application Notes

2.2. Call Flows

To understand how inbound ATT&T IP Toll Free service calls are handled by SES and Communication Manager, following call flows are described in this section.

The call scenario illustrated figure below is an inbound call from a PSTN via a SIP trunk from AT&T IP Toll Free service to SES and Communication Manager.

1. A PSTN phone or fax originates a call to an AT&T Toll Free service number.
2. The PSTN routes the call to the AT&T IP Toll Free service network.
3. The AT&T IP Toll Free service routes the call to SES.
4. SES routes the call to Communication Manager.
5. Depending on the DNIS, Communication Manager routes the call to
   - Vector, which in turn, routes the call to an agent
   - Directly to an agent or a phone/fax extension.

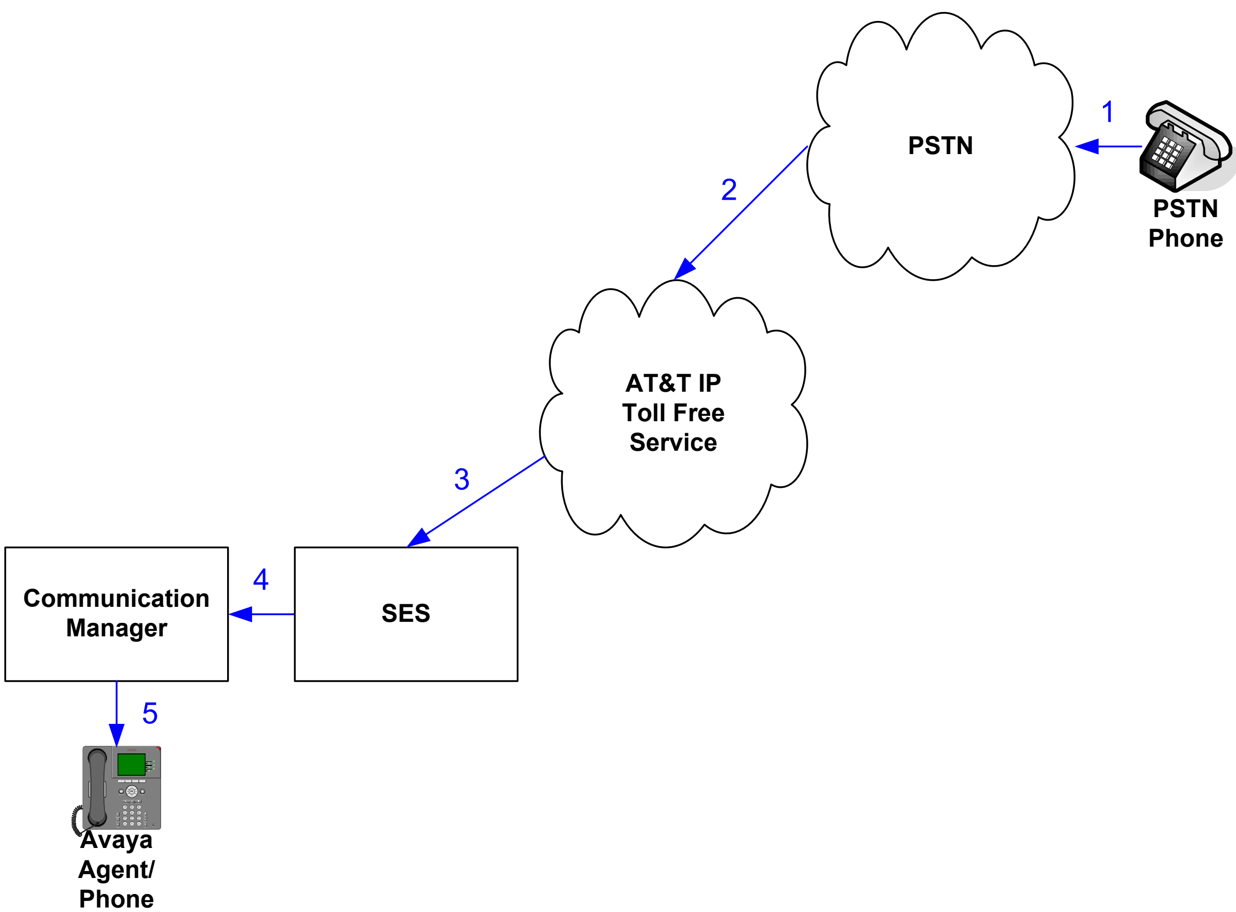

Figure 2: AT&T IP Toll Free Service Call to VDN/Agent/Phone/Fax

This call scenario illustrated in figure below is an inbound call from PSTN over AT&T IP Toll Free service network to SES and Communication Manager that is covered to voicemail. In this scenario, the voicemail system is an Avaya Modular Messaging system connected to SES configured in MultiSite mode.

1. Same as the call scenario in **Figure 2**.
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 agent or phone's voice mailbox. Note that the call[2] continues to go through Communication Manager.

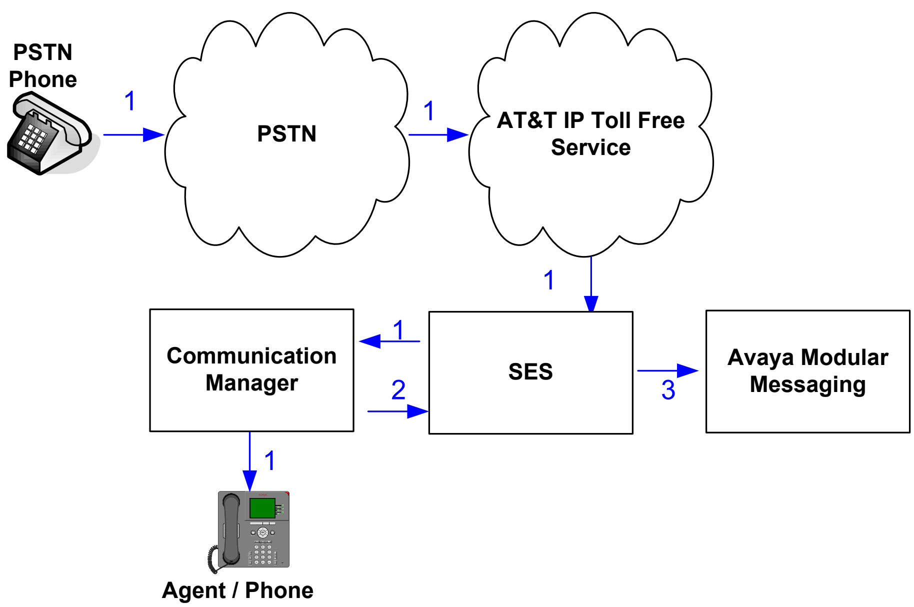

Figure 3: AT&T Toll Free service call to Agent/Phone covered to Modular Messaging

[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 S8720 Server	Avaya Aura™ Communication Manager 5.2.1, Service Pack 2 (R015x.02.1.016.4-18111)
Avaya G650 Media Gateway	
TN2312BP IP Server Interface (IPSI)	HW03 FW050
TN799DP Control-LAN (C-LAN)	HW00 FW037
TN2602AP IP Media Resource 320 (MedPro)	HW02 FW054
TN2501AP VAL-Announcement	HW02 FW018
TN2224B Digital Line	HW03
TN793B Analog Line	HW05
Avaya S8500B Server	Avaya Aura™ SIP Enablement Services 5.2.1, Service Pack 1b (SES05.2.1.016.4-SP1b)
Avaya 9650 IP Telephone	Avaya one-X™ Deskphone Edition H.323 Release S3.0
Avaya 9630 IP Telephone	Avaya one-X™ Deskphone Edition SIP Release 2.5
Avaya 4610SW H323 Telephone	2.8
Avaya 4621SW SIP Telephone	2.2.2
Avaya 6408D+ Digital Telephone	-
Avaya 6211 Analog Telephone	
Avaya one-X Agent	2.0
Fax	VentaFax 6.2 (Home Version)
AT&T IP Toll Free 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 Toll Free 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.

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

4.1. System Parameters

This section reviews the 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 4: System-Parameters Customer-Options Form – Page 2

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

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

   Emergency Access to Attendant? y                               IP Stations? y
           Enable 'dadmin' Login? y
           Enhanced Conferencing? y                         ISDN Feature Plus? n
                  Enhanced EC500? y       ISDN/SIP Network Call Redirection? 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 5: System-Parameters Customer-Options Form – Page 4

4.2. Dial Plan

This section briefly describes the dial plan requirements and feature access codes for the sample configuration described in these Application Notes. Enter the **change dialplan analysis** command to provision the dial plan as shown below.

- 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, agents, voicemail access, etc. in this sample configuration conform to this format.
- 4-digit dial access codes (indicated with a **Call Type** of "**ext**") beginning with the digit "**2**" – VDN extensions conform to this format.

```
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
     2             4    ext
```

Figure 6: Dialplan Analysis 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 an additional IP network region to represent the AT&T IP Toll Free service, and another IP codec set for external calls, i.e., inbound AT&T IP Toll Free calls.

1. Enter the **change ip-codec-set *ci*** command, where ***ci*** is the number of an IP codec set used only for internal calls and configure as follows:

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

Figure 7: 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 8: 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 *ib*** command, where ***ib*** is the number of an unused IP codec set. This IP codec set will be used for inbound ATT&T IP Toll Free calls. On Page 1 of the **ip-codec-set** form, provision the codecs in the order shown in below.

```
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.729A           n          3        20
 2: G.711MU          n          3        20
 3: G.726A-32K       n          2        20
```

Figure 9: IP-Codec-Set Form for Inbound 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 10: IP-Codec-Set Form for Inbound 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 AT&T IP Toll Free Service. On Page 1 of the **ip-network-region** form:
    - **Codec Set** – Set to codec set configured in **Step 2**
    - **UDP Port Min** – Set to **16384**
    - **UDP Port Max** – Set to **32767**

Note: This port range is an AT&T IP Toll Free service requirement.

```
change ip-network-region 3                                    Page   1 of  19
                               IP NETWORK REGION
  Region: 3
Location:         Authoritative Domain:
    Name: ATT IP Toll Free
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 11: IP-Network-Region Form for AT&T IP Toll Free service – 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 IP Toll Free service network as the **dst rgn**, provision the following:

- **codec set** – Set to the codec set configured 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 below was used for testing purposes only

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

 Source Region:  3     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    2     y    NoLimit                                          n   all
 3    2
```

Figure 12: IP-Network-Region Form for AT&T IP Toll Free service – Page 3

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**". Also, set the **Authoritative Domain** field to the domain name already configured in SES.

```
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 13: IP-Network-Region Form 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 below 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
```

Figure 14: IP-Network-Region Form 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 6.3**, **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-1A03        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 15: Node-Names Form

4.4. Inbound Calls

In this sample configuration, since the AT&T IP Toll Free service sends 9-digit DNIS on inbound calls, the administration steps that follow in this section reflect that requirement. In actual deployment, this may be different.

4.4.1. Inbound IP Toll Free 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 and configure as follows:
   - Group Type – Set to **sip**
   - **Transport Method** – Set to **tls** for this testing
   - **Near-end Node Name** – Set to the CLAN node name configured in **Section 4.3**, **Step 5**
   - **Far-end Node Name** – Set to the SES node name configured in **Section 4.3**, **Step 5**
   - Near-end and Far-end Listen Ports are set to a default value of **5061** for **Transport Method** of **tls**
   - **Far-end Domain** – Leave blank to allow this SIP signaling group to handle inbound IP Toll Free calls from any domain. This is a catch all signaling group which means that it will accept calls from any domain or IP address
   - **DTMF over IP** - Set to "**rtp-payload**" to enable Avaya Aura™ 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 media processing resources when possible

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 16: Signaling-Group Form for AT&T IP Toll Free Calls

2. Enter the **add trunk-group tpi** command, where **tpi** is the number of an unused trunk group and configure as follows:
   - **Direction** – Set to **incoming**
   - **TAC** – Set to any value as per the dial plan
   - **Service Type** – Set to **public-ntwrk**
   - **Signaling Group** – Set to the number of the signaling group administered in **Step 1**
   - **Number of Members** – Set to a value large enough to accommodate the call volume. For this testing, this value was set to **20**.

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

Group Number: 5                  Group Type: sip          CDR Reports: y
  Group Name: IPTF Incoming Trunk          R: 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 17: Trunk-Group Form for AT&T IP Toll Free Calls – Page 1

3. Enter the **change inc-call-handling-trmt trunk-group t** command, where **t** 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 DNIS in the SIP Invite message as follows:
   - **Called Len** – Enter the total number of digits in the DNIS
   - **Called Number** – Enter enough leading digits to uniquely match the DNIS range
   - **Del** and **Insert** – If necessary, enter the number of leading digits that need to be deleted from the DNIS, and the specific leading digits that need to be prefixed to the DNIS (after any deletion is performed), respectively, in order to match a local Communication Manager VDN / extension range.

   In this sample configuration, AT&T IP Toll Free service sends 9-digit DNIS in the SIP Invite 000001049 on inbound calls. Thus the entry in the figure below matches the DNIS and deletes all the 9 digits to match the local Communication Manager VDN 2010. VDNs used in this testing are described in **Section 4.5**.

   Provision as many entries as necessary to cover all expected DNIS numbers sent by AT&T IP Toll Free service.

```
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     9 000001049         9   2010
```

Figure 18: Inc-Call-Handling-Trmt Trunk-Group Form for AT&T IP Toll Free Calls

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

4.4.2. 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 4.4.1**, **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.

```
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: 1
 5  5                5                        5           Maximum Entries: 9999
```

Figure 19: Public-Unknown-Numbering Form

4.5. 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 2010/Vector 10 – VDN and vectors used to select the agent skill
- VDN 2011/Vector 11/Hunt Group 11 – Route call to Agent with Skill 11
- VDN 2012/Vector 12/Hunt Group 12 – Route call to Agent with Skill 12
- VDN 2012/Vector 13/Hunt Group 13 – Route call to Agent with Skill 13

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.

4.5.1. 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 21**). The hunt group is associated with a coverage path (e.g.**h2** in **Figure 22**) and the coverage path is assigned to a station/agent.

```
display hunt-group 2                                            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 20: Hunt Group Form – Page 1

```
display hunt-group 2                                            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 21: Hunt Group Form – Page 2

```
display coverage path 2
                                 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 22: Coverage Path Form

4.5.2. Call Center Provisioning

For provisioning the call center functionality, verify that the call center parameters are enabled as shown below. Verify that an agent login id is created with an appropriate skill. Verify the skill (hunt group) for that agent is in place. Make sure that a VDN as per the dial plan is in place along with the vector which lists the steps to be executed when an inbound call is received from AT&T IP Toll Free service.

In the reference configuration below, an inbound call from AT&I IP Toll Free service is handled using the incoming call handling treatment configured in **Section 4.4.1**, **Step 3**, using the VDN 2010 (**Figure 28**) which routes the call to Vector 10 (**Figure 29**) and based upon the digit inputted by the caller, the call is directed to an appropriate skill. Skill 11 (**Figure 30**) is shown for reference purposes and additional skills can be similarly added.

```
display system-parameters customer-options                        Page   6 of  11
                         CALL CENTER OPTIONAL FEATURES

                          Call Center Release: 5.0

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

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

Figure 23: Call Center Optional Features Form

```
display agent-loginID 53001                                     Page   1 of   2
                                AGENT LOGINID

                Login ID: 53001                                      AAS? n
                    Name: Agent1                                   AUDIX? n
                      TN: 1                              LWC Reception: spe
                     COR: 1                     LWC Log External Calls? n
           Coverage Path: 2                   AUDIX Name for Messaging:
           Security Code:
                                         LoginID for ISDN/SIP Display? n
                                                              Password:
                                                Password (enter again):
                                                           Auto Answer: station
                                                     MIA Across Skills: system
                                             ACW Agent Considered Idle: system
                                             Aux Work Reason Code Type: system
                                               Logout Reason Code Type: system
                         Maximum time agent in ACW before logout (sec): system
                                              Forced Agent Logout Time:   :

     WARNING:  Agent must log in again before changes take effect
```

Figure 24: Agent Form – Page 1

```
display agent-loginID 53001                                     Page   2 of   2
                                AGENT LOGINID
      Direct Agent Skill:                                  Service Objective? n
Call Handling Preference: skill-level                  Local Call Preference? n

    SN   RL SL           SN   RL SL           SN   RL SL           SN   RL SL
 1: 11      1       16:                 31:                 46:
 2:                 17:                 32:                 47:
 3:                 18:                 33:                 48:
```

Figure 25: Agent Form – Page 2

```
display hunt-group 11                                           Page   1 of   3
                                 HUNT GROUP

            Group Number: 2                                ACD? y
              Group Name: Skill-11                       Queue? y
         Group Extension: 53011                         Vector? y
              Group Type: ead-mia
                      TN: 1
                     COR: 1                     MM Early Answer? n
           Security Code:                 Local Agent Preference? n
 ISDN/SIP Caller Display:

             Queue Limit: unlimited
 Calls Warning Threshold:      Port:
  Time Warning Threshold:      Port:
```

Figure 26: Skill (Hunt Group) Form – Page 1

```
display hunt-group 11                                          Page   2 of   3
                                  HUNT GROUP

                    Skill? y      Expected Call Handling Time (sec): 180
                      AAS? n
                 Measured: none
     Supervisor Extension:

      Controlling Adjunct: none

Interruptible Aux Threshold: none
                                 Redirect on No Answer (rings):
                                               Redirect to VDN:
                   Forced Entry of Stroke Counts or Call Work Codes? n
```

Figure 27: Skill (Hunt Group) Form – Page 2

```
display vdn 2010                                               Page   1 of   3
                            VECTOR DIRECTORY NUMBER

                             Extension: 2010
                                  Name: To SelectSkill
                           Destination: Vector Number        10

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


                            1st Skill*:
                            2nd Skill*:
                            3rd Skill*:

* Follows VDN override rules
```

Figure 28: SelectSkill VDN

```
display vector 10                                                   Page   1 of   6
                                  CALL VECTOR
    Number: 2                  Name: RouteToSkill
                                               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    2   secs hearing ringback
02 collect      1    digits after announcement 33002    for none
03 goto vector  11   @step 2 if digits          =      1
04 goto vector  12   @step 2 if digits          =      2
05 goto vector  13   @step 2 if digits          =      3
06
```

Figure 29: RouteToSkill Vector

```
display vector 11                                              Page   1 of   6
                                  CALL VECTOR
    Number: 2                Name: Skill 11
                                             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    2   secs hearing ringback
02 announcement 33003
03 queue-to     skill 11   pri m
04 announcement 33006
05 goto step    3             if unconditionally
06
```

Figure 30: Skill-11 Vector

5. Avaya Modular Messaging

In this sample configuration, Avaya Modular Messaging 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 Toll Free 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**.

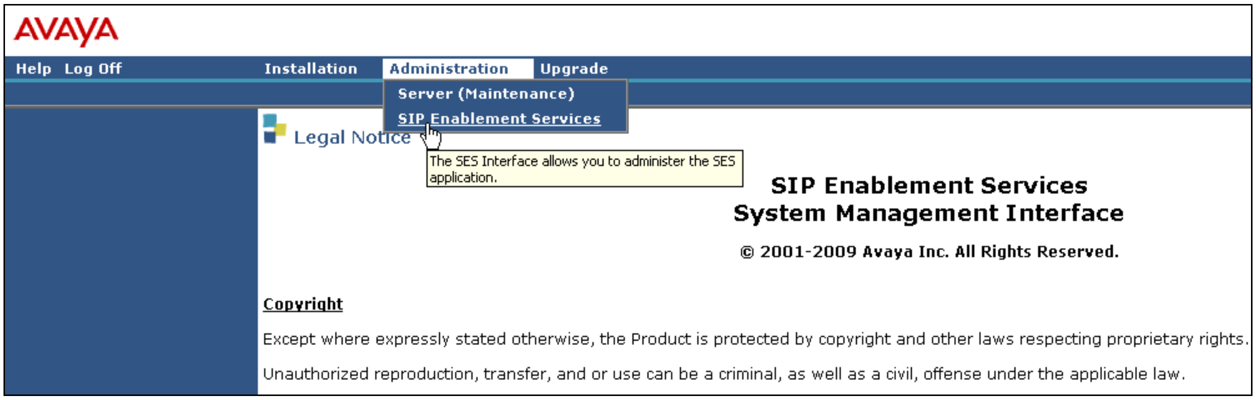

Figure 31: SES Web Interface

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

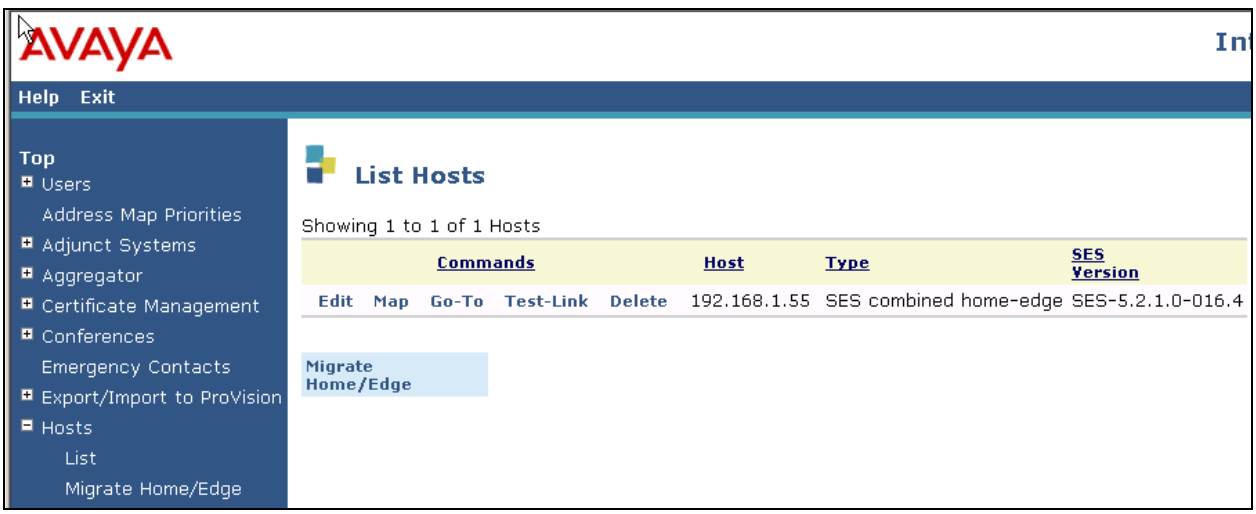

Figure 32: List Hosts Page

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.

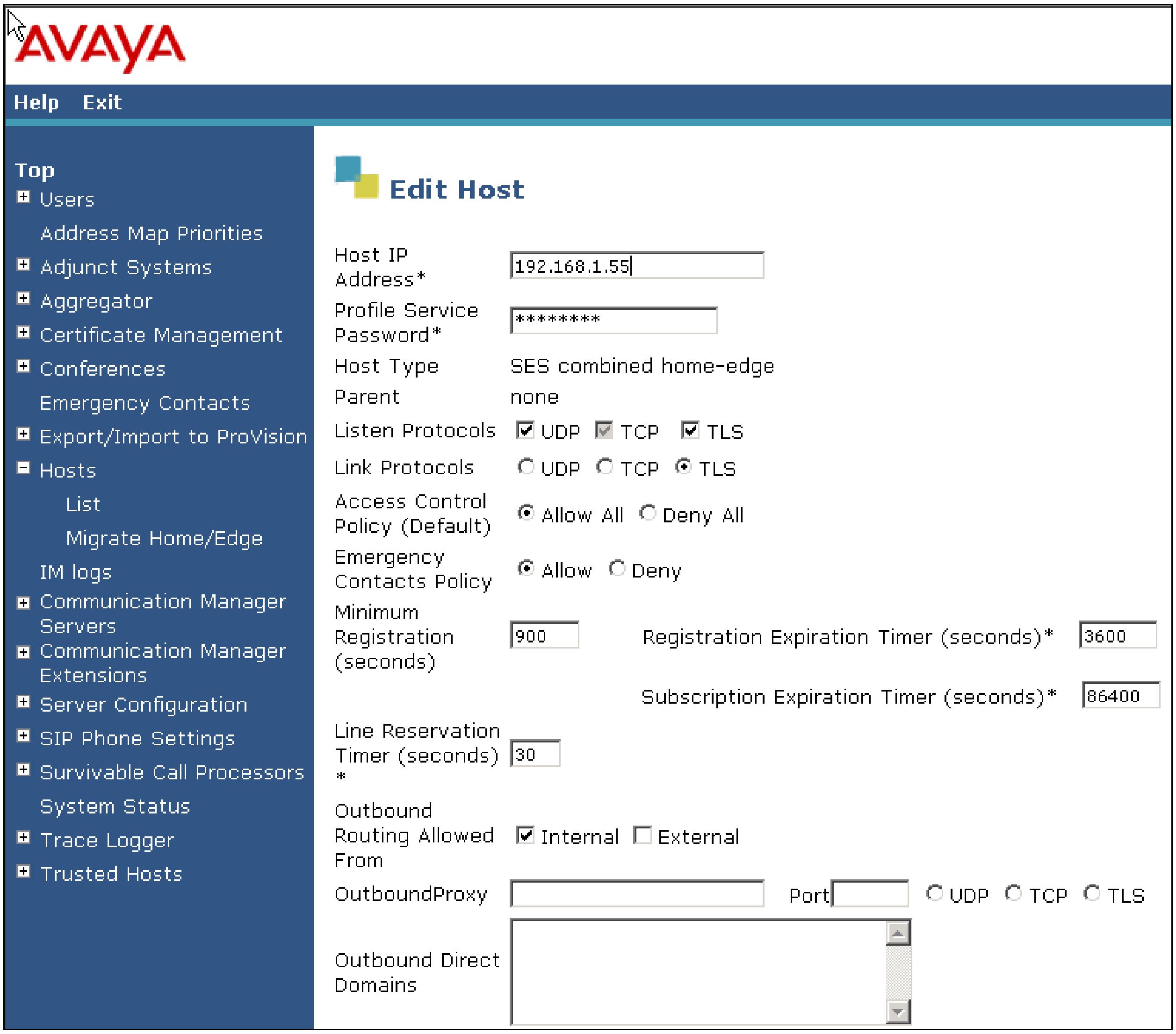

Figure 33: Edit Host Page

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 <u>Home server</u> for **Host**.

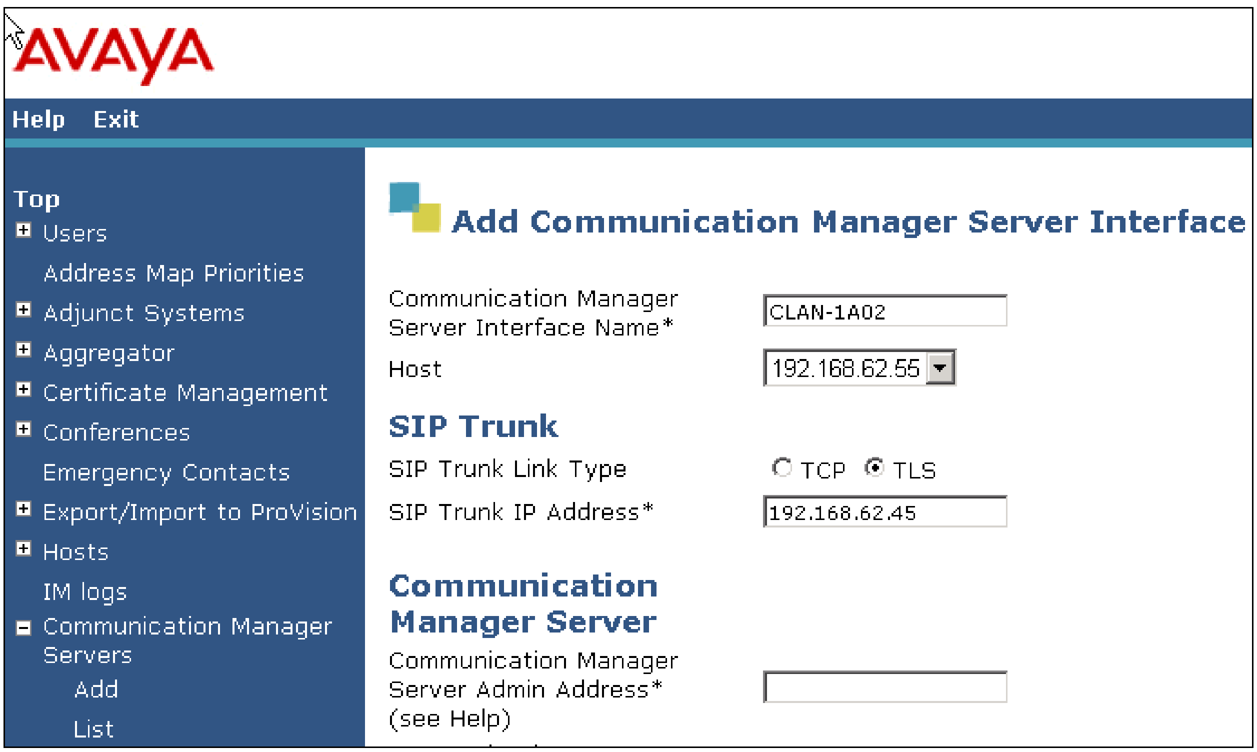

Figure 34: Add Communication Manager Server Interface Page – For First C-LAN

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 Toll Free service. SES examines the Request-URI of a received SIP INVITE message (from the AT&T IP Toll Free service for inbound calls), modifies the Request-URI and certain SIP headers, and then forwards the message to the appropriate destination.

For inbound calls from the AT&T IP Toll Free 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 Toll Free service, all calls must be routed from the AT&T IP Toll Free service core. Therefore, SES will expect to receive all inbound PSTN calls from the same AT&T IP Toll Free service Border Element(s).

6.4.2. Inbound Calls from AT&T IP Toll Free Service

SES address maps are used to route inbound calls from the PSTN 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.

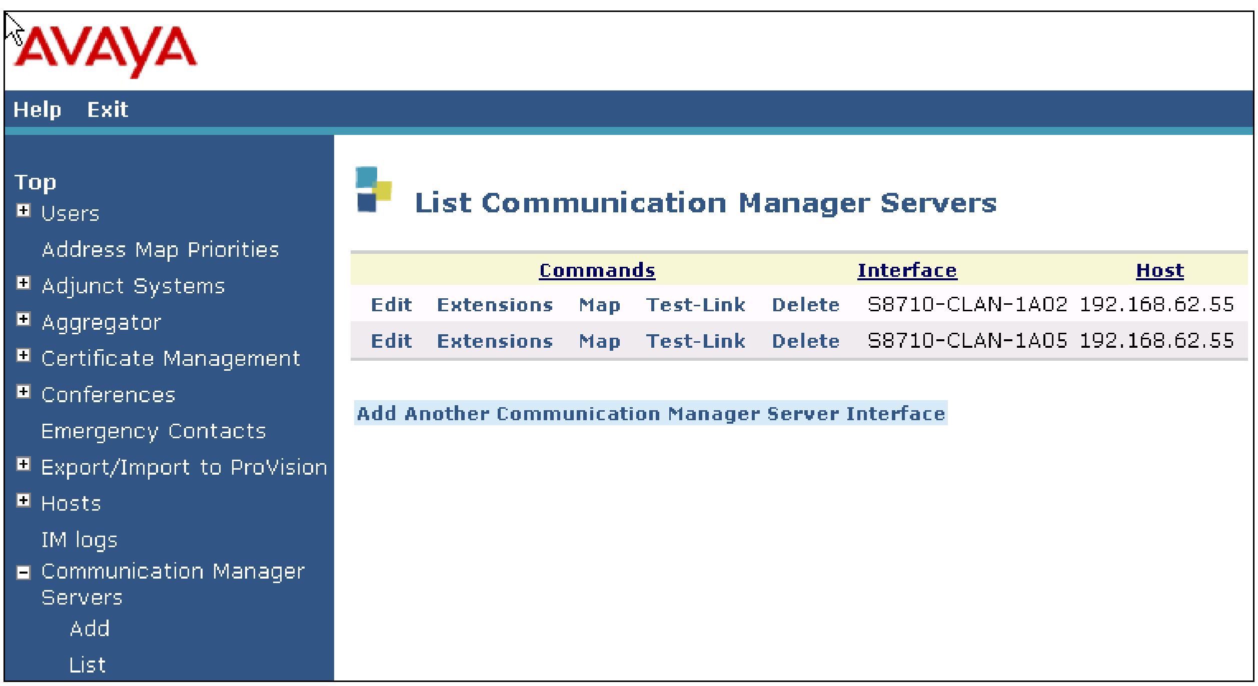

Figure 35: List Communication Manager Server Page

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 Toll Free service. In this sample configuration, for inbound calls from the PSTN, the AT&T IP Toll Free service inserts 9-digit DNIS beginning with "0000010xx" in the user part of the Request-URI to reach an agent or local extensions in the Avaya site. Thus, the pattern "**^sip:(0000010){4,5}[0,1,2,39]**" shown below, matches SIP INVITE messages with a Request-URI that may begin with "sip:0000010" followed by digit **4** or **5** and the last digit being any of **0**, **1**, **2**, **3** or **9**.
   - Click on "**Add**"
   - Click on "**Continue**" in the subsequent confirmation page (not shown).

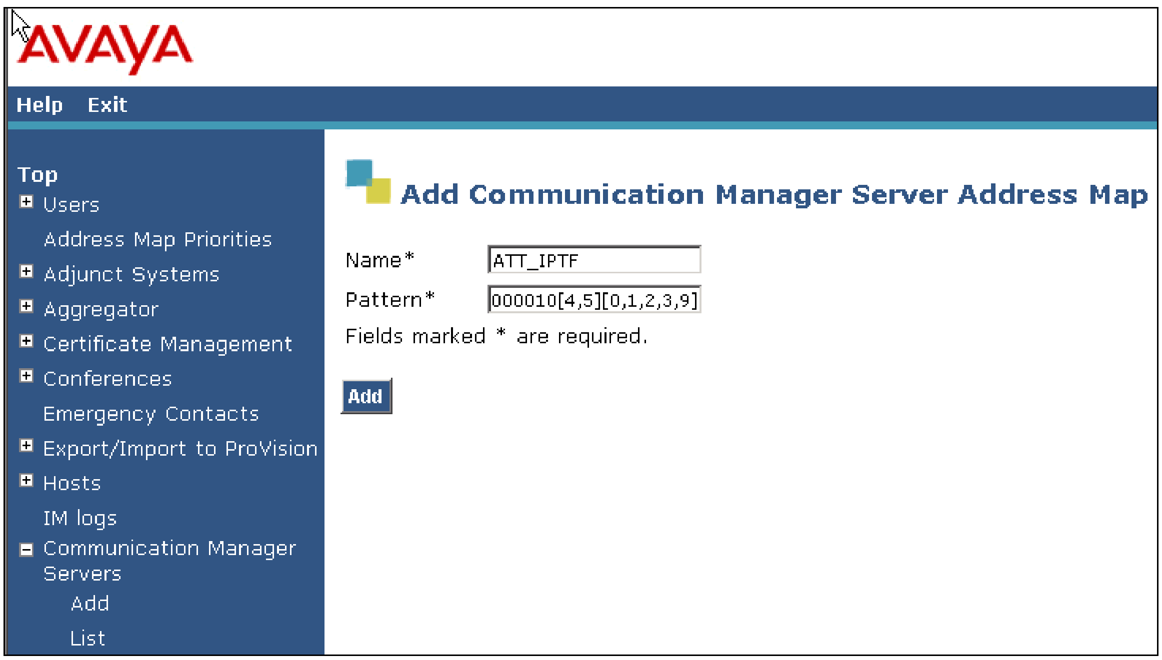

Figure 36: Add Communication Manager Server Address Map Page – Match 9-Digit DNIS

4. A Contact is automatically created after creating the first Communication Manager Server Address Map. 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**"to configure additional address maps to route the calls to Communication Manager.

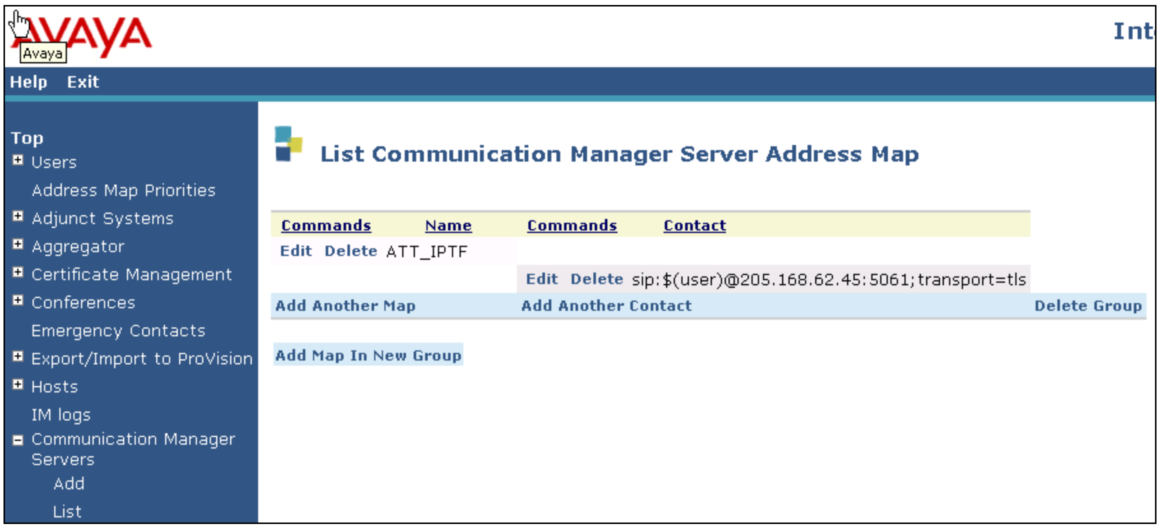

Figure 37: List Communication Manager Server Address Map Page – First C-LAN

6.5. Trusted Host

The AT&T IP Toll Free 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 Toll Free 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 Toll Free service Border Elements provided.

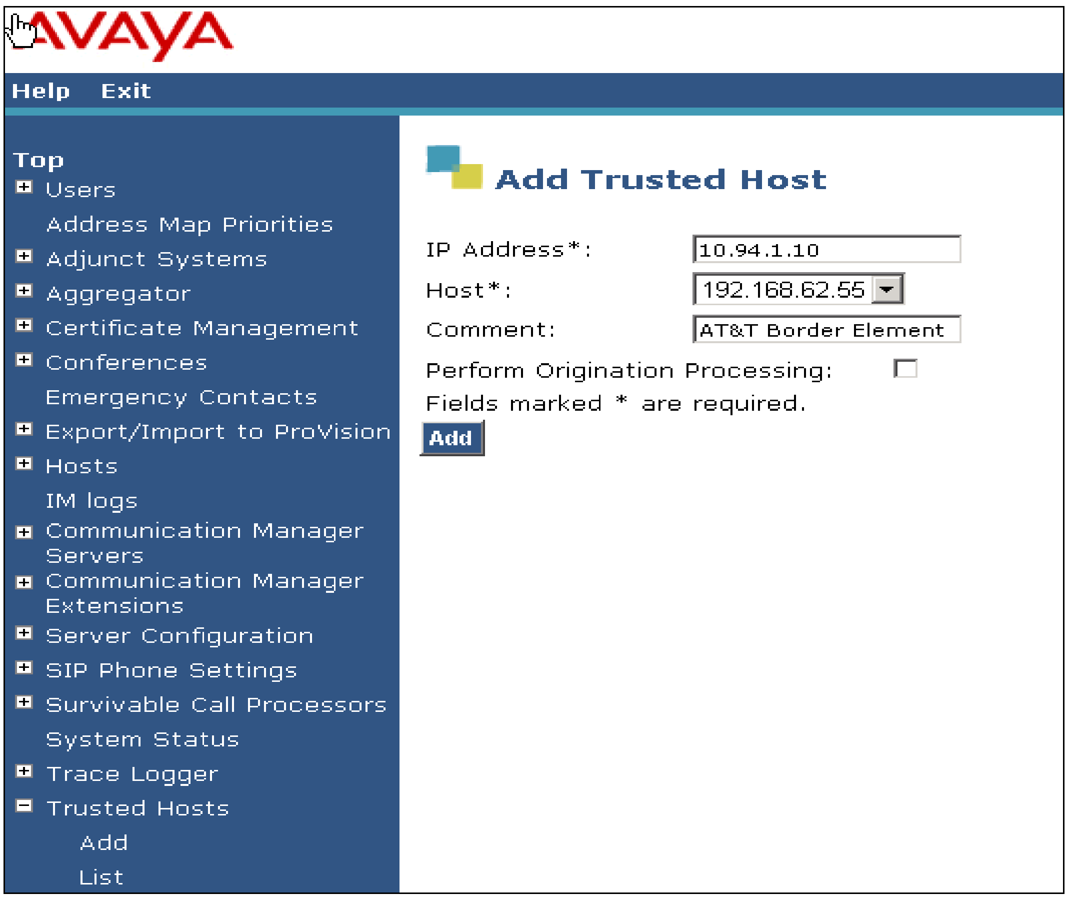

Figure 38: Add Trusted Host Page

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 Toll Free service via MIS/PNT, to which the simulated enterprise was connected.

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

- Inbound AT&T IP Toll Free service calls to Communication Manager VDNs, Agents and Phones.
- Inbound calls and two-way talk path establishment between AT&T IP Toll Free callers and Communication Manager agents/phones.
- Basic supplementary telephony features such as hold, resume, transfer, and conference.
- G.729 and G.711 codecs for voice calls with PSTN
- T.38 fax calls between Communication Manager and AT&T IP Toll Free (PSTN or TDM Gateway) service in G3 and SG3 fax mode.
- DTMF tone transmission using RFC 2833 in both directions.
- Avaya Aura™ Communication Manager phones sending DTMF to the AT&T IP Toll Free to invoke AT&T IP Toll Free Legacy Transfer Connect features, and Avaya Aura™ Communication Manager processing the resulting DTMF responses from the AT&T IP Toll Free service.
- Inbound AT&T IP Toll Free service calls to Avaya Aura™ Communication Manager that are directly routed to agents and unanswered can be covered to Avaya Modular Messaging.
- Long duration calls.

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:

- Place an inbound call, and verify that two-way talkpath exists, and that the calls remain stable for several minutes and disconnect properly.
- Place an inbound call to an agent or a phone and verify that the call goes to coverage if it is not answered.

8.2. Troubleshooting Tools

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

The **traceSES** function within the SES may be used to capture SIP traces between SES and the AT&T IP Toll Free 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 Toll Free service. The AT&T IP Toll Free service allows enterprises to receive inbound Toll Free calls from the PSTN.

Note that these Application Notes did NOT cover the AT&T IP Transfer Connect service option of the AT&T IP Toll Free service.

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 Toll Free Service Description:

[11] *AT&T IP Toll Free Service Description*
http://www.business.att.com/enterprise/Service/business-voip-enterprise/network-based-voip-enterprise/ip-toll-free-enterprise/

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