Avaya Solution & Interoperability Test Lab

Application Notes for Configuring Avaya Communication Manager SIP Trunking with the Telcordia Service Interconnection Registry – Issue 1.0

Abstract

These Application Notes describe the steps for configuring SIP trunking with the Telcordia Service Interconnection Registry and an Avaya SIP trunking solution using Avaya Communication Manager and Avaya SIP Enablement Services.

The Telcordia Service Interconnection Registry uses SIP trunking to perform directory lookups to determine if a telephone number can be reached using wide-area Internet connections. If the number being called is registered in the Telcordia SIR, a SIP redirection is returned allowing the telephone call to be completed via the SIP trunk. Otherwise, when no match is found, the Telcordia SIR response permits Avaya Communication Manger to complete the telephone call using alternate routing such as a PSTN trunk group.

Telcordia is a member of the Avaya DevConnect Service Provider program. Information in these Application Notes has been obtained through DevConnect compliance testing and additional technical discussions. Testing was conducted via the DevConnect Program at the Avaya Solution and Interoperability Test Lab with remote access to the Telcordia Service Interconnection Registry.

1. Introduction

These Application Notes describe the steps for configuring SIP trunking with the Telcordia Service Interconnection Registry (SIR) and an Avaya SIP trunking solution using Avaya Communication Manager and Avaya SIP Enablement Services.

SIP (Session Initiation Protocol) is a standards-based communications protocol designed to provide a common framework to support multimedia communication. RFC 3261 [8] is the primary specification governing this protocol. SIP manages the establishment and termination of connections and the transfer of related information such as the desired codec, calling party identity, etc. Within these Application Notes, SIP is used as the signaling protocol between the Avaya components and the SIR offered by Telcordia.

The Telcordia Service Interconnection Registry uses SIP trunking to perform directory lookups to determine if a telephone number can be reached using wide-area Internet connections. If the number being called is registered in the Telcordia SIR, a SIP redirection is returned allowing the telephone call to be completed via the SIP trunk. Otherwise, when no match is found, the Telcordia SIR response permits Avaya Communication Manger to complete the telephone call using alternate routing such as a PSTN trunk group.

1.1. Illustrative Telcordia Service Interconnection Registry Solution

Figure 1 illustrates a representative customer location using an Avaya Communication Manager based SIP trunking solution with the Telcordia Service Interconnection Registry.

This representative configuration includes:

- A simulated "Acme Corp." customer location using the Avaya SIP trunking solution in addition to ISDN PRI trunks to their local PSTN carrier. This configuration consists of:
    - Avaya Communication Manager providing the communication services for this customer location.
    - Avaya SIP Enablement Services (in the combined home/edge configuration) serving as the SIP proxy with the Telcordia SIR via the Internet.
    - An ISDN PRI trunk to the local PSTN.
    - Various Avaya telephones and other endpoints.
- Several other simulated locations connected with either SIP trunks to the Internet and/or traditional PSTN trunks.
    - XYZ Corp – another business that has subscribed to the Telcordia SIR. They are able to send or receive SIP trunk calls via the Internet with other subscribed businesses such as Acme Corp. XYZ can have PSTN connections for other communication purposes but they are not shown.
    - Unknown Inc. – another business that has NOT subscribed to the Telcordia SIR. Their SIP trunking connections are unknown to Acme Corp. Thus, while they may be technically able to send and receive SIP trunk calls, all calls will use the PSTN.

- Traditional Inc. and Smith Home represent locations that have no SIP trunking connections and all calls must use the PSTN.

- The Telcordia Service Interconnection Registry providing a SIP directory lookup and redirection functions necessary to permit subscribers to reach each other using SIP trunking.

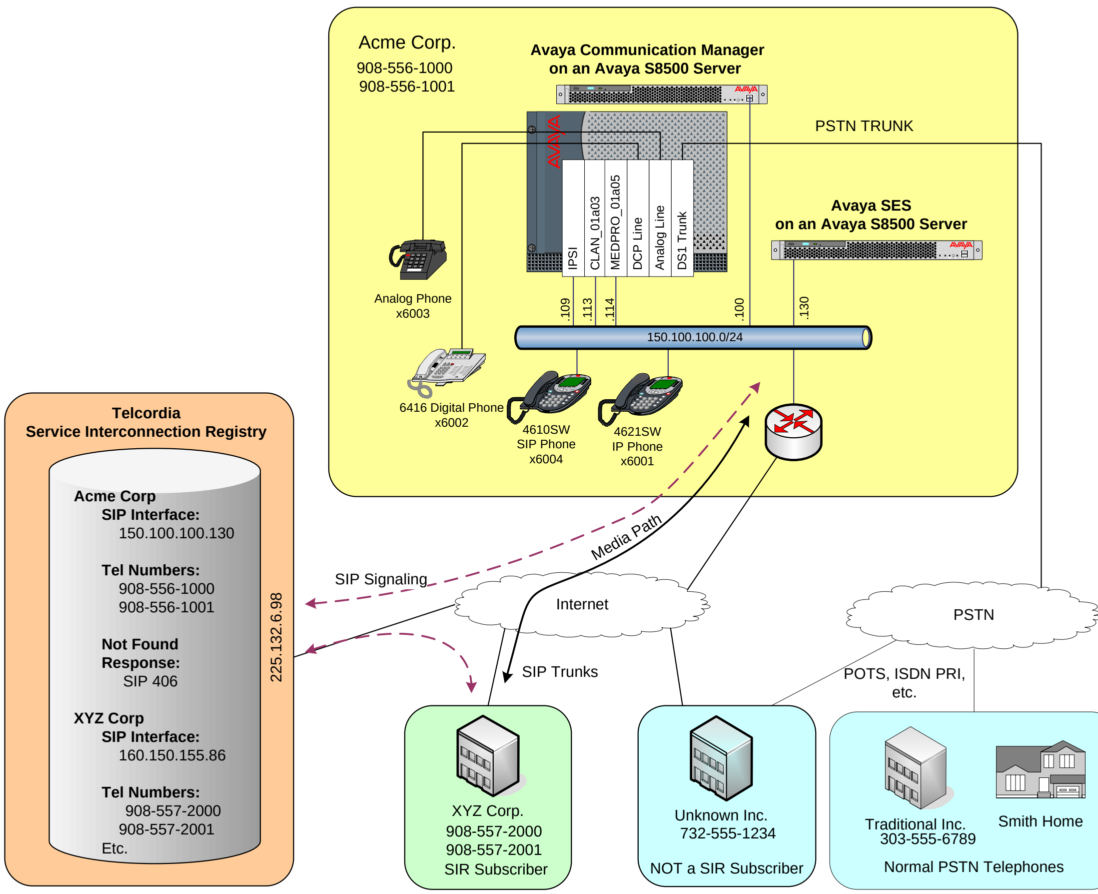

Figure 1 – Illustrative SIP Trunking Configuration Using the Telcordia SIR

1.2. Telcordia SIR Configuration Information

These Application Notes provide an illustrative example of how the Avaya SIP trunking solution is configured to work with the Telcordia Service Interconnect Registry.

The specific values provided below are illustrative only. *Each customer uses their specific values obtained from Telcordia at the time of service provisioning.*

Telcordia Service Interconnection Registry Information	Value Used in these Application Notes
Telcordia – SIR IP Address	225.132.6.98

Acme Corp. – Avaya SES IP Address	150.100.100.130
Acme Corp. – PSTN Numbers Registered in SIR	908-556-1000 908-556-1001
Acme Corp. – Number Not Found SIP Response	406
For testing purposes only, the following information is known but not used in the Avaya Sip trunking solution configuration	
XYZ Corp. – PSTN Numbers Registered in SIR	908-557-2000 908-557-2001
Unknown Inc. – PSTN Number Not Registered In SIR	732-555-1234

2. Equipment and Software Validated

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

Component	Version
Avaya	
Avaya S8500 Server	Avaya Communication Manager 4.0.1 (R014x.00.0.730.5)
Avaya G650 Media Gateway	
TN2312BP IP Server Interface (IPSI) TN799DP Control-LAN (C-LAN)	HW03 FW022 HW01 FW017
TN2602AP IP Media Processor (Medpro)	HW02 FW031
TN2224CP Digital Line	HW08 FW015
TN793CP Analog Line	HW09 FW09
TN464GP DS1 Interface	HW06 FW017
Avaya 4621SW IP (H.323) Telephone	Release 2.8.3
Avaya 4621 SIP Telephone	Release 2.2.2
Avaya 6416D+M Digital Telephone	n/a
Avaya S8500B Server	Avaya SIP Enablement Services 4.0 (SES-4.0.0.0-033.6)
Telcordia	
Service Interconnection Registry	Version 2.0.1

Table 1 – Equipment and Version

3. Configure Avaya Communication Manager

Avaya Communication Manager was installed and configured for basic station to station calling prior to beginning the configuration shown in these Application Notes. In addition, this configuration assumes that an existing PSTN trunk group (e.g., trunk group 3) exists that is capable of placing outbound calls and receiving inbound Direct Inward Dialed calls to the assigned numbers.

These basic configuration details are outside the scope of this SIP trunking application and are not included here.

3.1. SIP Trunk Configuration

3.1.1. Verify System Capacity and Required Features

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

Verify that there is sufficient remaining Avaya Communication Manager SIP trunk capacity available for the SIP Trunks using the Telcordia SIR, taking into consideration other applications that may require Avaya Communication Manager SIP trunk resources.

This is done by displaying Page 2 of the **System-Parameters Customer-Options** form. The number of SIP trunks available to add to new or existing trunk groups is the difference between the **Maximum Administered SIP Trunks** and the **USED** value.

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

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

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

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

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

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

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

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



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

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

3.1.2. Determine Node Names

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

- “ses” and “150.100.100.130” are the **Name** and **IP Address** respectively of the Avaya SIP Enablement Services server interface where Avaya Communication Manager SIP trunk messages are sent.

- “clan_01a03” and “150.100.100.113” are the **Name** and **IP Address** respectively of the TN799DP C-LAN interface used for the SIP signaling group.

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

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

Figure 4: IP Node Names

3.1.3. Define IP Codec Set for SIP Trunk Calls

This configuration uses IP codec set 2 to assign G.729B, G729A and G.711mu codecs (in that priority) for voice calls. T.38 will be used for group 3 fax calls to PSTN connected fax machines.

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

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

                          IP Codec Set

    Codec Set: 2

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

Figure 5: IP Codec Set 2 – Audio Codec Settings

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

                          IP Codec Set

                              Allow Direct-IP Multimedia? n


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

Figure 6: IP Codec Set 2 – Fax, Modem, and TTD/TTY Mode Settings

3.1.4. Verify Near End IP Network Region

These Application Notes use IP network region 1 (the normal default) for the G650 Media Gateway, the IP telephones, the C-LAN (in slot 01a03) used for IP telephone registration and the SIP trunk signaling group to the Telcordia SIR. Verify this using the “display cabinet 1” command.

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


                 Location: 1         IP Network Region: 1

 Rack: row6          Room: sitl         Floor:             Building:

 CARRIER DESCRIPTION
   Carrier       Carrier Type        Number

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

3.1.5. Verify the C-LAN IP Network Region Assignment

In these Application Notes, the C-LAN was previously installed as part of the initial Avaya Communication Manager basic installation (using the procedures as described in [2]) and assigned the Node Name shown in **Figure 4**.

Using the “display ip-interface 01a03” command (where 01 is the cabinet, a is the carrier, and 03 is the slot of the respective C-LAN), verify the C-LAN is assigned to **Network Region** 1.

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


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

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

Figure 7: IP Interface of C-LAN 01a03 used for SIP Signaling Group 3

3.1.6. Define IP Network Region

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

In these Application Notes,

- IP network region 1 defines the properties for the main Avaya Communication Manager site previously configured during installation.
- IP network region 11 is assigned to Telcordia SIR to allow codec preferences different from network region 1 to be used.
- IP network region 1 and 11 are defined to be directly connected with 512 Kbps of bandwidth.
- IP codec-set 2 (defined in Section 3.1.3) will be used for calls between IP network region 1 and 11.

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

- **Name:** a descriptive string such as "Avaya CM Main Location".
- **Authoritative Domain:** the SIP domain of the Avaya SES (in this case "customer-sipdomain.com" as defined in Section 4.1.2).
- **Codec Set:** the value "1" corresponding to the ip-codec-set (defined during initial configuration) for local calls between telephones on Avaya Communication Manager.
- **Intra-region IP-IP Direct Audio**: the value "yes" (the default).
- **Inter-region IP-IP Direct Audio:** the value "yes" (the default).

The IP-IP Direct Audio settings ensure the most efficient use of TN2602AP Media Processor resources.

Defaults for the remaining values are used.

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

Figure 8: IP Network Region 1 – Page 1

Page 3 of the IP network region form is used to define the codec set and connectivity characteristics between IP network regions.

On Page 3, configure the "**src rgn 1 dst rgn 11**" row as follows:

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

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

                   Inter Network Region Connection Management

 src dst codec direct   WAN-BW-limits   Video                           Dyn
 rgn rgn  set   WAN  Units    Total Norm  Prio Shr Intervening-regions  CAC IGAR
 1   1    1
 1   2
 1   3


 1   9
 1   10
 1   11   2     y    Kbits     512     0     0  y                             n
 1   12
```

Figure 9: IP Network Region 1 – Page 3

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

- **Name:** a descriptive string such as "SIP PSTN Trks"
- **Authoritative Domain:** the SIP domain of the Avaya SES (in this case "customer-sipdomain.com" as defined in Section 4.1.2).
- **Codec Set:** the value "2" corresponding to the ip-codec-set defined in Section 3.1.3.
- **Intra-region IP-IP Direct Audio:** the value "yes" (the default).
- **Inter-region IP-IP Direct Audio:** the value "yes" (the default).

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

Figure 10: IP Network Region 11 – Page 1

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

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

                   Inter Network Region Connection Management

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

Figure 11: IP Network Region 11 – Page 3

3.1.7. Define Outbound SIP Trunk Group

One SIP trunk group is defined for outbound calls using the Telcordia SIR (routed via the Avaya SES). This SIP trunk group requires a corresponding SIP signaling group to define the characteristics of the signaling relationship.

3.1.7.1 Establish the SIP Signaling Group

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

- **Group Type**: set to “sip”.
- **Transport Method**: automatically set to “tls”. The Transport Layer Security (TLS) transport protocol is used between Avaya Communication Manager and the Avaya SES. Note this is not the transport protocol used to communicate between the Avaya SES and the Telcordia SIR.
- **Near-end Node Name**: set to the C-LAN node name (defined in Section 3.1.2) used for the respective signaling group. In these Application Notes, “clan_01a03” is used for signaling group 11.
- **Far-end Node Name**: set to the Avaya SES. In these Application Notes, the node name “ses” is used as defined in Section 3.1.2
- **Near-end Listen Port**: set to “5061”, the default port for SIP signaling using tls transport.
- **Far-end Listen Port**: set to “5061”.
- **Far-end Network Region**: set to “11”, the network region defined in Section 3.1.6.
- **Far-end Domain**: set to the IP address provided by Telcordia as their SIR IP address. In these Application Notes, the IP address “225.132.6.98” will be used.
- **Direct IP-IP Audio Connections**: set to “y”, indicating the RTP paths should be optimized to reduce the use of media processing resources when possible.
- **DTMF over IP:** set to “rtp-payload”. This value enables Avaya Communication Manager to send DTMF transmissions using RFC 2833 [9].

The default values for the other fields are used.

The resulting form for signaling group 11 is shown below.

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

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






  Near-end Node Name: clan_01a03            Far-end Node Name: ses
Near-end Listen Port: 5061                Far-end Listen Port: 5061
                                       Far-end Network Region: 11
      Far-end Domain: 225.132.6.98

                                            Bypass If IP Threshold Exceeded? n

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

Figure 12: Signaling Group 11

3.1.7.2 Establish the SIP Trunk Group

Using the "add trunk-group 11" command, configure trunk group 11 as follows.

On Page 1 of the Trunk Group form:

- **Group Type**: set to "sip".
- **Group Name**: enter a descriptive string such as "SIP OB PSTN TRKS".
- **TAC**: enter a trunk access code such as "#011".
- **Service Type**: set to "public-ntwrk" for trunks to the PSTN.
- **Signaling Group**: set to "11" as defined within Section 3.1.7.1.
- **Number of Members**: set to the maximum number of simultaneous calls permitted for each trunk group. Within these Application Notes, "10" was used.

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

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

Group Number: 3                    Group Type: sip           CDR Reports: y
  Group Name: SIP OB PSTN TRKS            COR: 1        TN: 1        TAC: #011
   Direction: two-way        Outgoing Display? n
 Dial Access? n                                   Night Service:
Queue Length: 0
Service Type: public-ntwrk          Auth Code? n

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

Figure 13: SIP Trunk Group 11

3.1.8. Define Inbound SIP Trunk Group

A second SIP trunk group is defined for incoming calls from other customers who used the Telcordia SIR to determine routing information. These customers may have an unknown far-end domain.

The SIP trunk group configured here implements the recommended Avaya practice of configuring at least one signaling-group with a blank "Far-end Domain" as specified in the SIP trunk engineering notes section of Reference [4]. This permits Avaya Communication Manager Release 4.0 and later to accept incoming SIP calls from unknown domains.

3.1.8.1 Establish the SIP Signaling Group

Using the "add signaling-group 1" command, configure signaling group 1 as follows:

- **Group Type**: set to "sip".
- **Transport Method**: automatically set to "tls". The Transport Layer Security (TLS) transport protocol is used between Avaya Communication Manager and the Avaya SES. Note this is not the transport protocol used to communicate between the Avaya SES and the Telcordia SIR.

- **Near-end Node Name**: set to the C-LAN node name (defined in Section 3.1.2) used for the respective signaling group. In these Application Notes, "clan_01a03" is used for signaling group 1.
- **Far-end Node Name**: set to the Avaya SES. In these Application Notes, the node name "ses" is used as defined in Section 3.1.2
- **Near-end Listen Port**: set to "5061", the default port for SIP signaling using tls transport.
- **Far-end Listen Port**: set to "5061".
- **Far-end Network Region**: set to "11", the network region defined in Section 3.1.6.
- **Far-end Domain**:leave blank. This permits incoming calls from unknown domains to be accepted.
- **Direct IP-IP Audio Connections**: set to "y", indicating the RTP paths should be optimized to reduce the use of media processing resources when possible.
- **DTMF over IP:** set to "rtp-payload". This value enables Avaya Communication Manager to send DTMF transmissions using RFC 2833 [9].

The default values for the other fields are used.

The resulting form for signaling group 1 is shown below.

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

Group Number: 1             Group Type: sip
                       Transport Method: tls







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

                                           Bypass If IP Threshold Exceeded? n

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

Figure 14: Signaling Group 11

3.1.8.2 Establish the SIP Trunk Group

Using the "add trunk-group 1" command, configure trunk group 1 as follows.

On Page 1 of the Trunk Group form:

- **Group Type**: set to "sip".
- **Group Name**: enter a descriptive string such as "SIP IB TRK".
- **TAC**: enter a trunk access code such as "#001".

- **Service Type**: set to “public-ntwrk” for trunks to the PSTN.
- **Signaling Group**: set to “1” as defined within Section 3.1.8.1.
- **Number of Members**: set to the maximum number of simultaneous calls permitted for each trunk group. Within these Application Notes, “10” was used.

The default values are used on the remaining pages of the trunk-group form. The resulting form for trunk-group 1 is shown below.

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

Group Number: 1                   Group Type: sip           CDR Reports: y
  Group Name: SIP IB TRKS                COR: 1        TN: 1        TAC: #001
   Direction: two-way        Outgoing Display? n
 Dial Access? n                                 Night Service:
Queue Length: 0
Service Type: public-ntwrk          Auth Code? n

                                                     Signaling Group: 1
                                                   Number of Members: 10
```

Figure 15: SIP Trunk Group 1

3.1.9. Configure Calling Party Number Information

The SIP “From” header shown below contains information about the calling party. The header contains the calling party number (e.g., “19085561000”) in the userinfo segment and a domain (or IP address) in the hostname segment separated by an “@” sign.

```
From: "Jane Smith" <sip:19085561000@customer-sipdomain.com>;tag=80f839da25
```

The “public-unknown-numbering” command controls the calling party number sent in the userinfo segment. The **Authoritative Domain** field of the near-end IP Network Region form completed in **Figure 10** associated with the SIP trunk group sets the hostname segment.

Public-unknown-numbering must always be setup. In these Application Notes the public-unknown-numbering is configured to send an 11 digit number corresponding to the assigned DID numbers.

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

- **Ext Len**: set to “4”, the length of the extensions used.
- **Ext Code**: set to the leading digit of the extension used. In these Application Notes “60” is entered to cover the assigned extensions of 60xx.
- **Trk Grp(s):** by default, leave blank to perform the same conversion across all SIP (and ISDN) trunk groups.

- **CPN Prefix**: set to the leading digits (e.g., "190855610") that are to be sent as the calling party number.
- **Total CPN Len**: set to the total length (e.g., "11") of the calling party number to be sent. The extension number will be appended to the **CPN Prefix** to form complete calling party number of **Total CPN Len** digits.

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

```
change public-unknown-numbering 4                               Page   1 of   2
                    NUMBERING - PUBLIC/UNKNOWN FORMAT
                                         Total
Ext Ext             Trk       CPN          CPN
Len Code            Grp(s)    Prefix       Len
                                                    Total Administered: 2
 4  60                        190855610    11          Maximum Entries: 9999
```

Figure 16: Public Unknown Numbering

3.1.10. Configure Call Routing

3.1.10.1 Outbound Calls

In these Application Notes, Automatic Route Selection (ARS) is used to route outbound calls via the SIP trunk group to the Telcordia SIR. The Telcordia SIR (and the subscribed XYZ Corp.) expects to receive the following digits in the SIP INVITE message.

Type of Call	Digits Sent
Local and Long Distance in North American Numbering Plan	Digit 1 plus any 10 digits

Table 2: Outbound Dialing Rule

Here, the configuration of one outbound calling pattern supporting calls to 1-908-xxx-xxx is shown. A typical installation will generally require additional ARS dial string and route pattern entries but that is beyond the scope of these Application Notes. Further information on ARS administration is discussed in References [1] and [3].

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

Following the verification, use the "change dialplan analysis" command to create a feature access code (fac) for ARS use.

- **Dialed String**: enter "9" that will become the user dialed prefix for outbound calls.
- **Total Length**: enter "1" as the length of the prefix.
- **Call Type**: enter "fac" as the type of prefix.

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

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

Figure 17: Dial Plan Analysis

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

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

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

Figure 18: ARS Feature Access Code

Use the "change ars analysis nn" command to configure the ARS route pattern selection rules as follows.   Here "nn" is "19", the first two digits of the dialed number after the ARS access code.

- **Dialed String**: enter the leading digits (e.g., "1908") necessary to uniquely select the desired route pattern.
- **Total Min**: enter the minimum number of digits (e.g., "11") expected for this PSTN number.
- **Total Max**: enter the maximum number of digits (e.g., "11") expected for this PSTN number.

- **Route Pattern**: enter the route pattern number (e.g., “3”) to be used. The route pattern (to be defined next) will specify the trunk group(s) to be used for calls matching the dialed number.
- **Call Type**: enter “fnpa”, the call type for North American 1+10 digit calls.

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

          Dialed            Total     Route     Call   Node  ANI
          String           Min  Max  Pattern    Type   Num   Reqd
    1908                  11   11    3          fnpa         n
```

Figure 19: ARS Digit Analysis Entries

Next the route pattern used for the Telcordia SIR calls is specified. This route pattern must contain at least two choices:

- the SIP trunk group used to reach the Telcordia SIR
- a second trunk group used as a next choice in the event the dialed number is not subscribed to the Telcordia SIR.

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

In the form:

- **Pattern Name**: enter a descriptive string such as “Telcordia SIR LD” to describe the routing pattern.
- **Secure SIP?**: leave as “n”, the default.
- **Grp No**: enter the trunk groups to be used in priority order. Trunk group 11 is the first choice SIP trunk group to the Telcordia SIR.   A second trunk group 7 (not shown) is used to provide a second choice route when the Telcordia SIR returns the subscriber not found status.
- **FRL**: enter the minimum facility restriction level (e.g., 1) necessary to use this trunk group, with 0 being the least restrictive. The FRL within the Class of Restriction (COR) assigned to the station must be greater than or equal to 1 in this case to use these trunk groups.
- **Pfx Mrk**: enter “1”, to always send the prefix 1 on 10 digit calls.

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

```
change route-pattern 3                                          Page   1 of   3
                    Pattern Number: 3   Pattern Name: Telcorida SIR LD
                                              Secure SIP? n
    Grp FRL NPA Pfx Hop Toll No.  Inserted                            DCS/ IXC
    No          Mrk Lmt List Del  Digits                              QSIG
                             Dgts                                     Intw
 1: 11   1       1                                                     n   user
 2: 7    1       1                                                     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                                         next
 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 20: Route Pattern 3

Use the "change locations" command to designate the SIP trunk route pattern (route pattern 3 below) in the **Proxy Sel. Rte. Pat.** field if not previously set.

```
change locations                                                Page   1 of  16
                                   LOCATIONS

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

 Loc  Name           Timezone Rule  NPA  ARS  Atd      Disp   Prefix    Proxy Sel
 No                   Offset             FAC  FAC      Parm               Rte Pat
 1:  Main            + 00:00   0                        1                   3
 2:                      :
```

Figure 21: Location Form Administration

3.1.10.2 Incoming Calls

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

In these Application Notes, the incoming PSTN DID numbers 1-908-556-1000 through 1001 are configured. They are assigned to extensions as shown in **Table 3**.

Dialed PSTN Number	Digits Received (within SIP INVITE message)	Extension Assigned
908 556 1000	908 556 1000	6000
908 556 1001	908 556 1001	6001

Table 3 - Incoming Number Assignments

Use the “change inc-call-handling-trmt trunk-group n” command (where “n” is the SIP trunk group number “1”) to administer the incoming number routing. This administration must be done for each incoming trunk group.

- **Called Len**: enter the total number of incoming digits received (e.g., “10”).
- **Called Number**: enter the specific digit pattern to be matched.
- **Del**: enter the number of leading digits that should be deleted
- **Insert**: enter the specific digits to be inserted at the beginning of the adjusted incoming digit string (to form what should be the complete number).

The completed inc-call-handling-trmt form for trunk group 1 is shown below.

```
change inc-call-handling-trmt trunk-group 1                     Page   1 of  30
                       INCOMING CALL HANDLING TREATMENT
 Service/      Called    Called      Del Insert
 Feature        Len      Number
 public-ntwrk    10 9085561000       10  6000
 public-ntwrk    10 9085561001       10  6001
```

3.1.11. Save Avaya Communication Manager Changes

This completes the configuration of Avaya Communication Manager.

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

4. Configure Avaya SIP Enablement Services

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

For reference purposes, **Figure 22** illustrates the IP Network Configuration entered into the Avaya SES during installation.

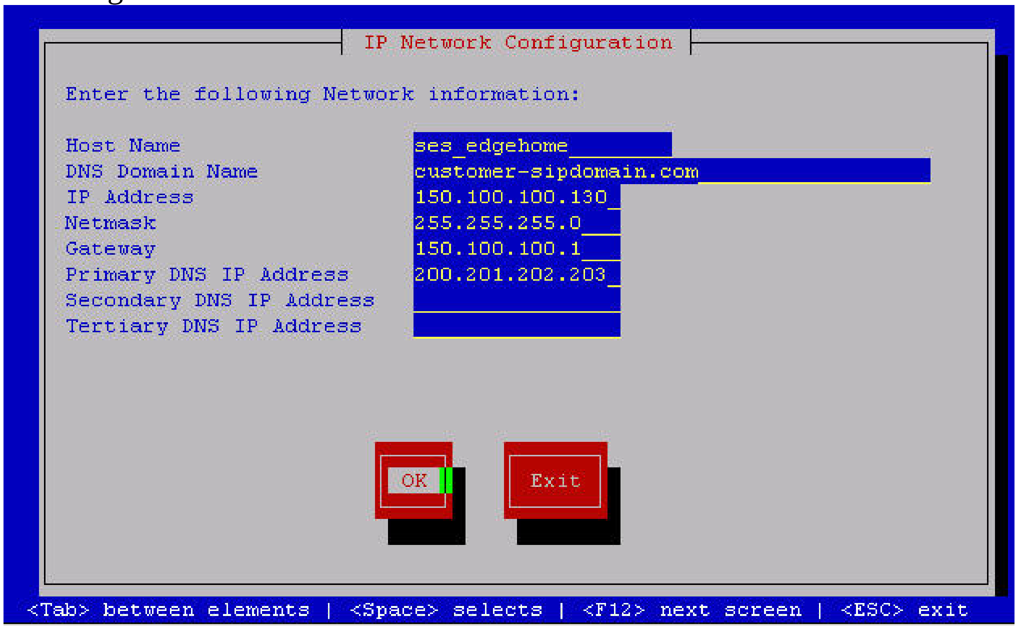

Figure 22 - Avaya SES Initial Setup Information

4.1. SIP Connection to the Telcordia SIR

4.1.1. Log in to Avaya SIP Enablement Services

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

Log in with the appropriate credentials and then select the **Launch Administration Web Interface** link from the main page as shown in **Figure 23**.

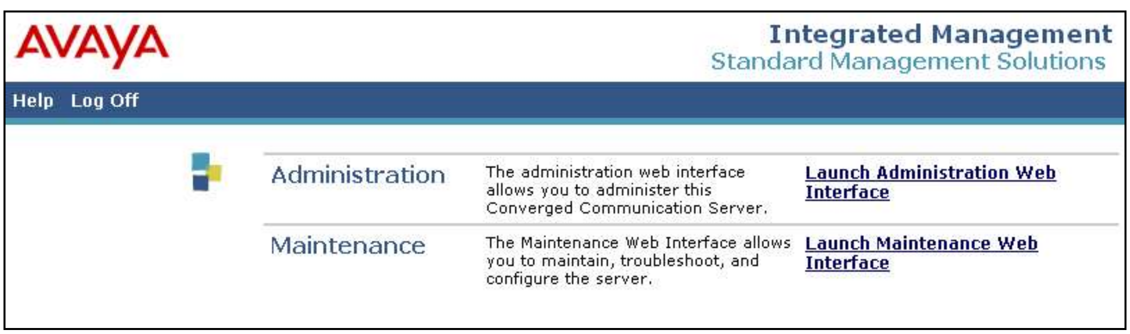

Figure 23: Avaya SES Main Page

The Avaya SES administration home page shown in **Figure 24** is displayed.

Figure 24: Avaya SES Administration Home Page

4.1.2. Verify System Properties

From the left pane of any SIP Server Management page, expand the **Server Configuration** option and select **System Properties**. This page displays the Avaya **SES Version** and the **Network Properties** entered via the install script during the installation process.

In the **Edit System Properties,** page note the **SIP Domain was** entered during the initial installation. The **SIP Domain** “customer-sipdomain.com” is used in these Application Notes.

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

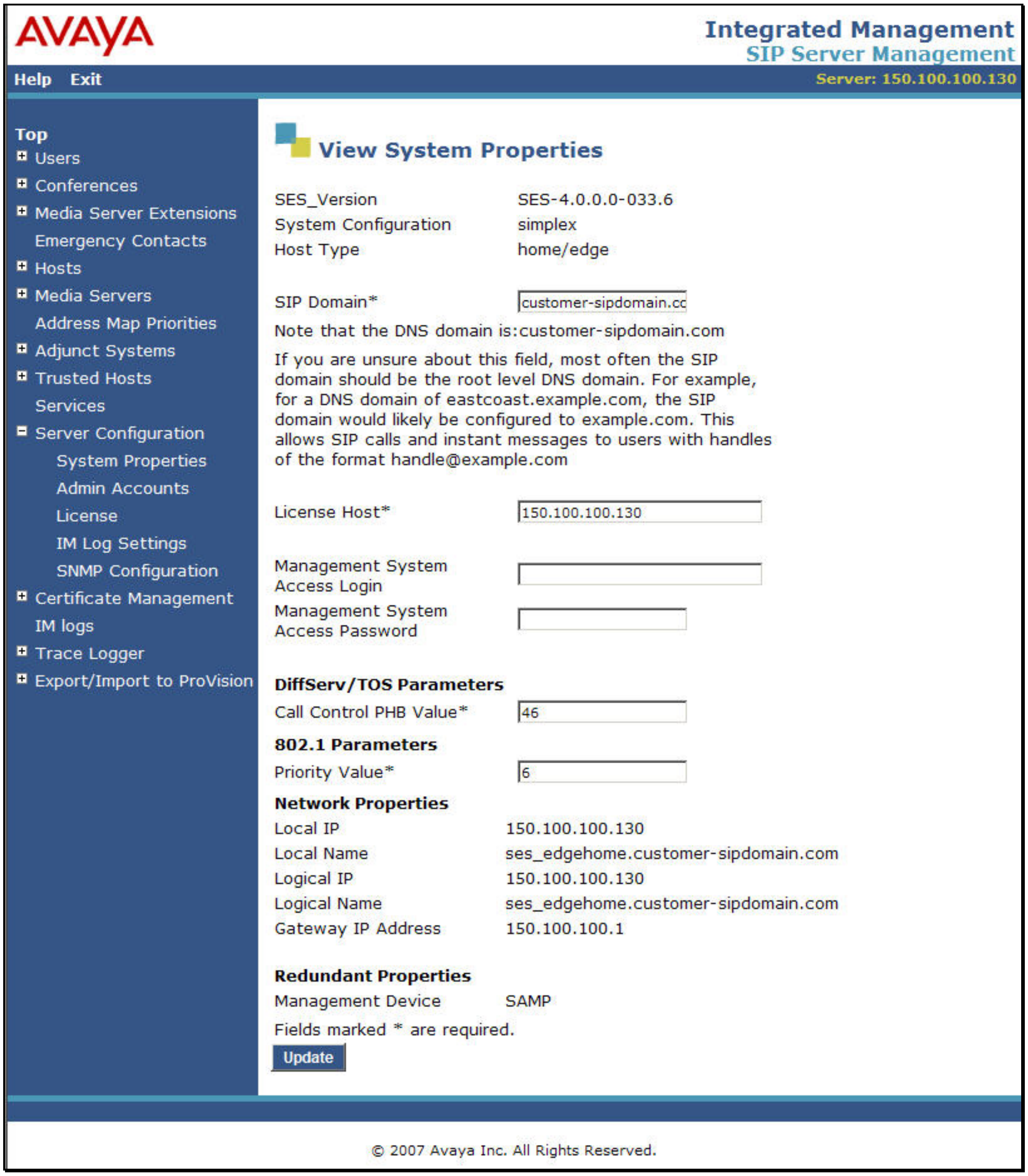

Figure 25: System Properties

4.1.3. Verify the Avaya SES Host Information

Verify the Avaya SES Host information using the **Edit Host** page. In these Application Notes the Avaya SES **Host Type** is a combined "home/edge"**.** This means that a single Avaya SES routes SIP messages between the Telcordia SIR and Avaya Communication Manager. (Note that separate Avaya SES home and edge servers may exist in other configurations. Communications with the Telcordia SIR will always occur via the edge Avaya SES.)

Navigate to the **Edit Host** page (**Figure 26**) by following the **Hosts** link in the left navigation pane and then clicking on the **Edit** option under the **Commands** section of the **List Hosts** screen.

On the **Edit Host** screen:

- Verify that the IP address of this combined Avaya SES Home/Edge server is in the **Host IP Address** field.
- Do not modify the **DB Password** or **Profile Service Password** fields. If these fields are changed, exit the form without using the **Update** button. These values must match the values entered during the Avaya SES installation; incorrect changes may disable the Avaya SES.
- Verify that the **UDP**, **TCP** and **TLS** checkboxes are enabled as **Listen Protocols**.
- Verify that **TLS** is selected as the **Link Protocol**.
- Ensure that the **Outbound Proxy** and **Outbound Direct Domains** fields are left blank.
- Default values for the remaining fields may be used.
- Click the **Update** button only if changes are necessary. Otherwise, exit the **Edit Host** page by selecting the **Top** link on the left navigation bar.

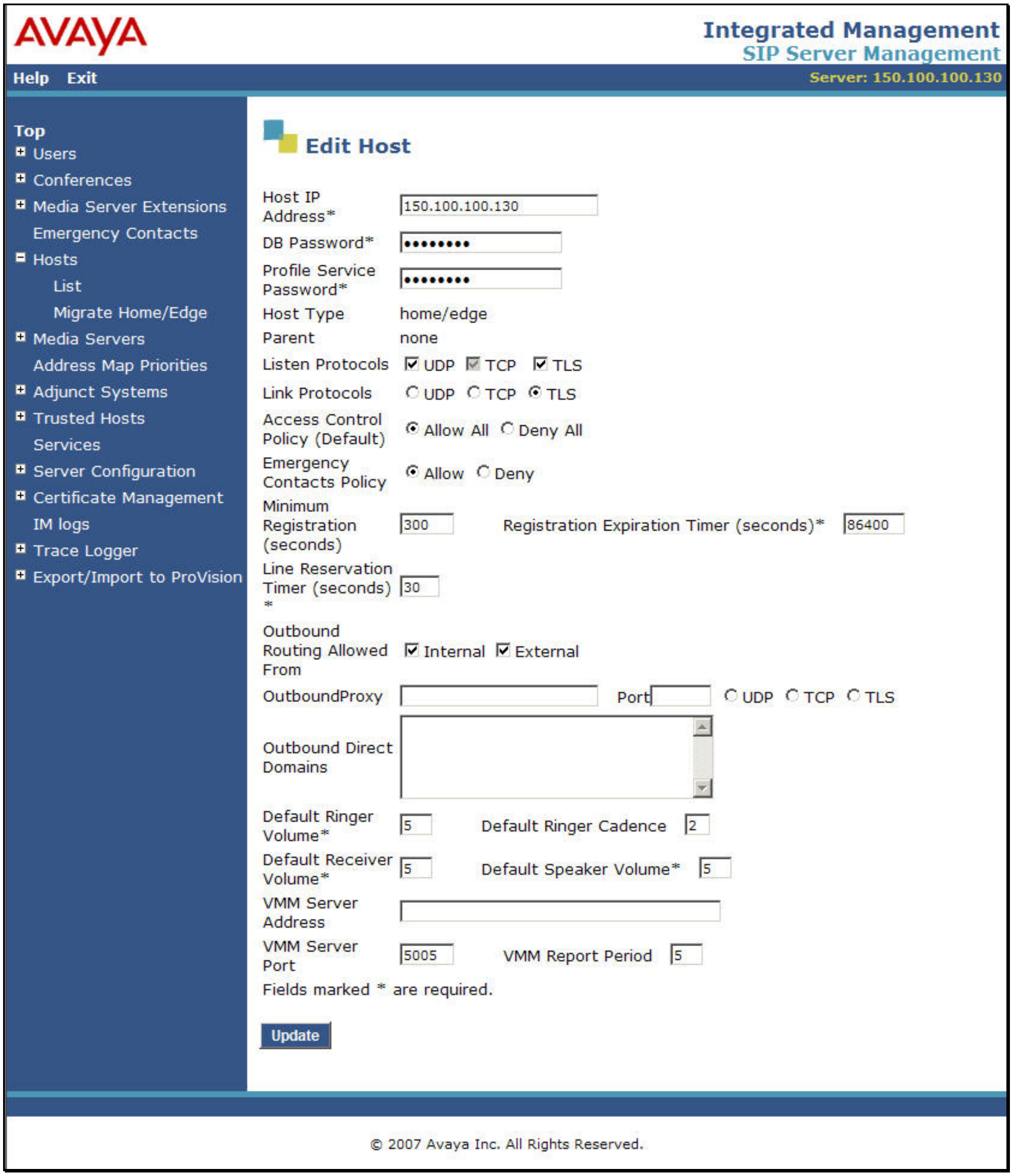

Figure 26: Edit Host

4.1.4. Add Avaya Communication Manager Media Server Interface

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

Expand the **Media Servers** option within any Avaya SES SIP Server Management page, and select **Add** to display the Add Media Server page (**Figure 27**).

In the Add Media Server Interface page, enter information corresponding to the signaling group "11" entry performed in Section 3.1.7.1.

- Enter "CLAN-1A03-5061**"** as the descriptive name in the **Media Server Interface Name** field.
- Select the Avaya SES home/edge IP address in the **Host** field.
- Select "TLS" (Transport Link Security) for the **SIP Trunk Link Type**. TLS provides encryption at the transport layer between Avaya Communication Manager and the Avaya SES.
- Enter the IP address of the "clan-1a03" interface in the **SIP Trunk IP Address** field as defined in **Figure 4**. Note: This may be the IP address of the "Processor Ethernet" interface in other Avaya Communication Manager configurations.

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

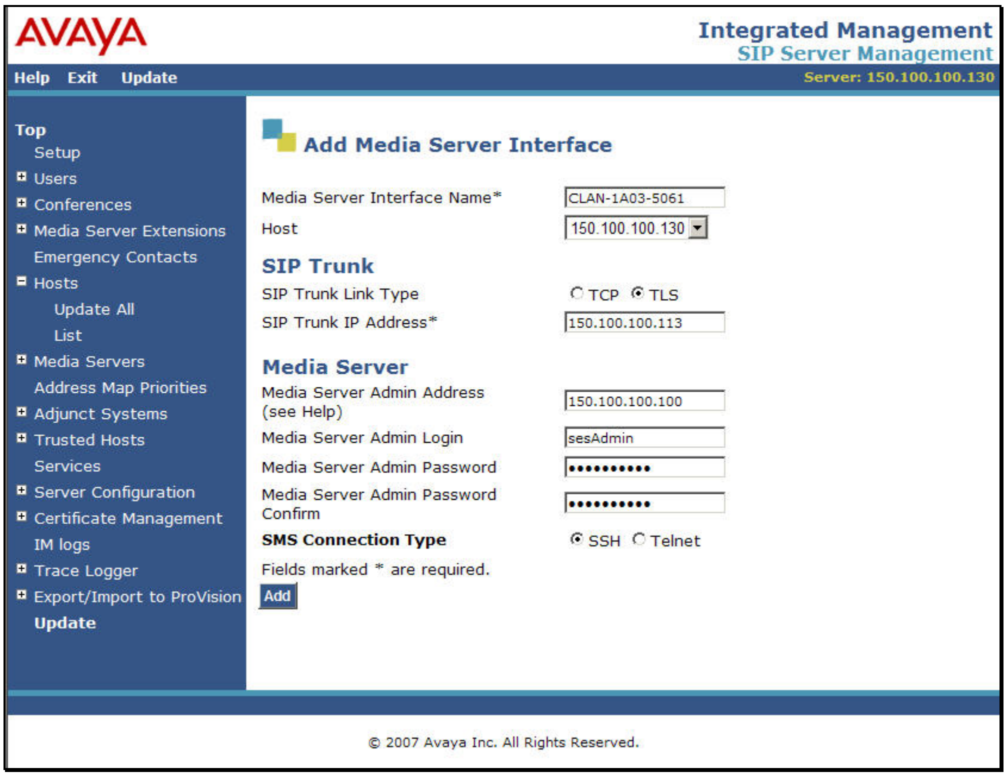

Figure 27: Add Media Server Interface for Voice Calls

When these operations are completed, the List Media Servers page will appear as shown in **Figure 28**.

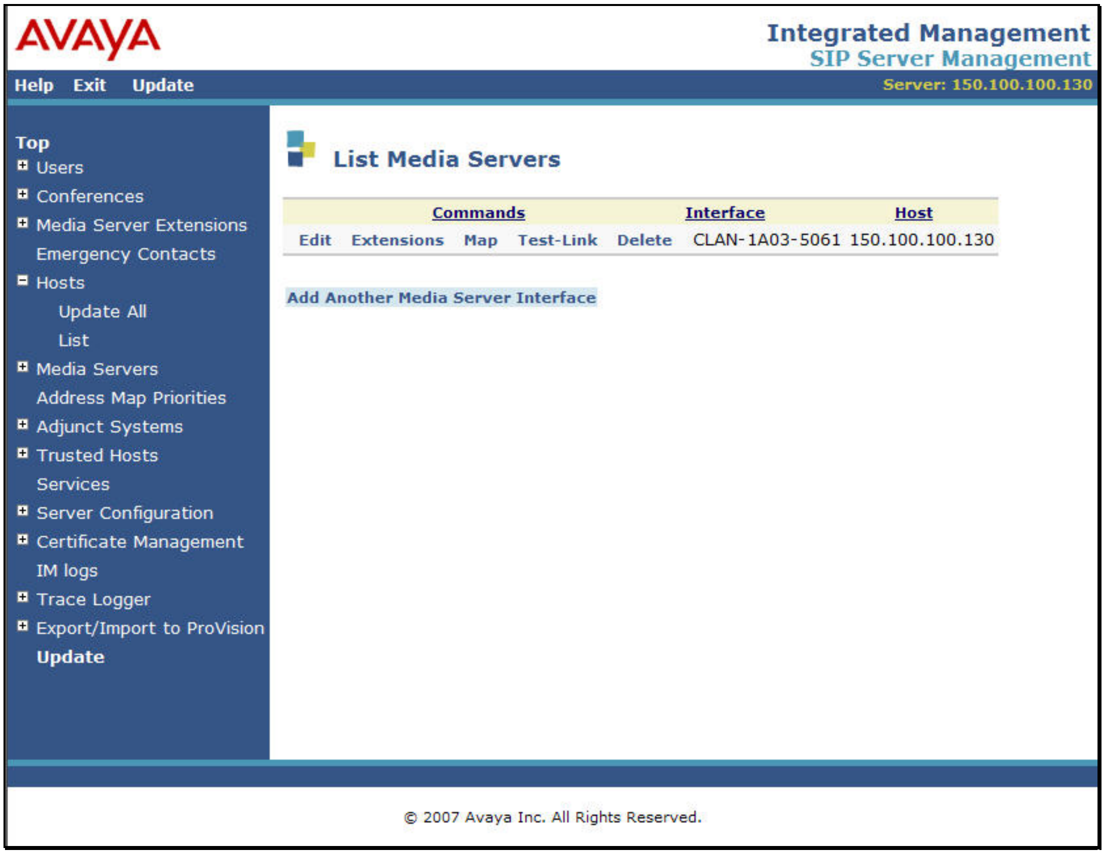

Figure 28: Completed List Media Servers

4.1.5. Configure Call Routing

4.1.5.1 Background

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

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

The SIP messages are routed to the IP address associated with the *domain* part when the *domain* does not match the SIP domain of the Avaya SES (**Figure 25**) and the outbound proxy is not specified. This rule applies to all outbound calls to the Telcordia SIR in these Application Notes.

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

The media server address maps are used to route all incoming SIP trunk calls to Avaya Communication Manager in these Application Notes.

4.1.5.2 Outbound Calls Using the Telcordia SIR

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

Since this domain value specifies the Telcordia SIR, no additional administration is necessary in the Avaya SES for outbound calls.

4.1.5.3 Inbound Direct Inward Dialed Calls

SIP messages for incoming calls directed by the Telcordia SIR are sent to the Avaya SES. The Avaya SES then routes these messages to the appropriate Avaya Communication Manager using Avaya SES media server address maps.

In these Application Notes, the incoming PSTN calls use media server address map patterns matching the 10-digit called number in the user part of the SIP Request URI.

An example of a SIP Request URI in an INVITE message received for the DID number 908-556-1000 is:

sip:9085561000@150.100.100.130

The user part in this case is the 10-digit number “9085561000”.

Table 4 below summarizes the media server address map strategy used in these Application

Notes for incoming calls.

Dialed PSTN Number	SIP Request URI User Part	SES Media Server Address Map Pattern	Media Server Interface
908-556-1000 through 908-556-1001	9085561000 through 9085561001	^sip:908556100[01]	CLAN-1A03-5061

Table 4: Incoming DID Address Map Rules Used

To configure the media server address map for voice calls:

- Expand the **Media Servers** link in the left navigation menu of any SIP Server Management page. Select **List** to display the List Media Servers screen shown in **Figure 29**.
- Click on the **Map** link to display the List Media Server Address Map screen associated with the “CLAN-1A03-5061” **Interface**.

Figure 29: List Media Servers

- Since no previous media server address map exists, click on the **Add Map In New Group** link as shown in **Figure 30**.

Figure 30 - List Media Server Address Map

The Add Media Server Address Map page shown in **Figure 31** is displayed.

- Enter a descriptive name in the **Name** field, such as “DID-908556100x”.

- Enter the **Address Map Pattern** for incoming DID calls (from **Table 4**) into the **Pattern** field.

  The pattern specification for these DID numbers is:

  ^sip:908556100[01]

  This means that URIs beginning with “sip:908556100” followed either 0 or 1 will match the pattern and be routed to the interface defined as “CLAN-1A03-5061”.

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

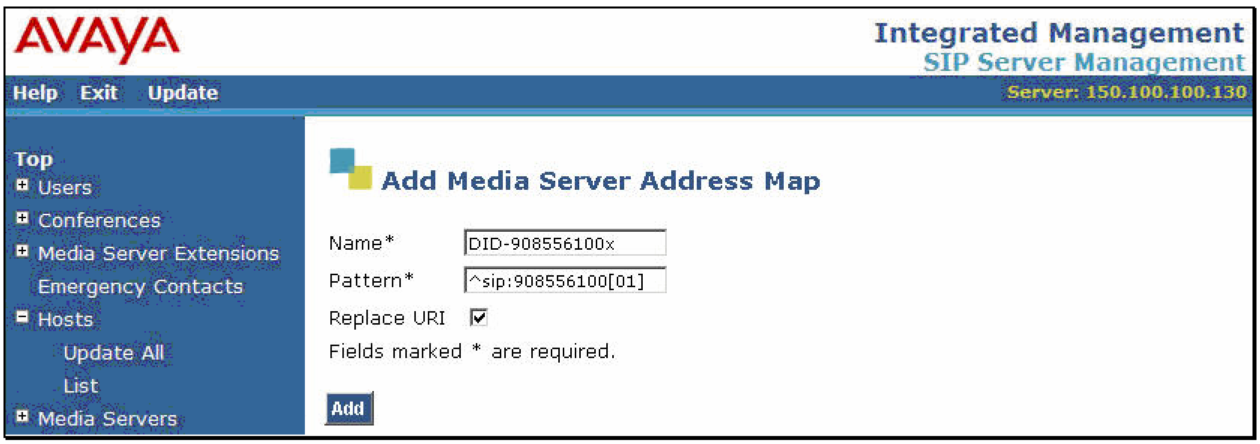

Figure 31: Incoming DID Calls - Media Server Address Map

After configuring the media server address maps, the **List Media Server Address Map** page appears as shown in **Figure 32**.

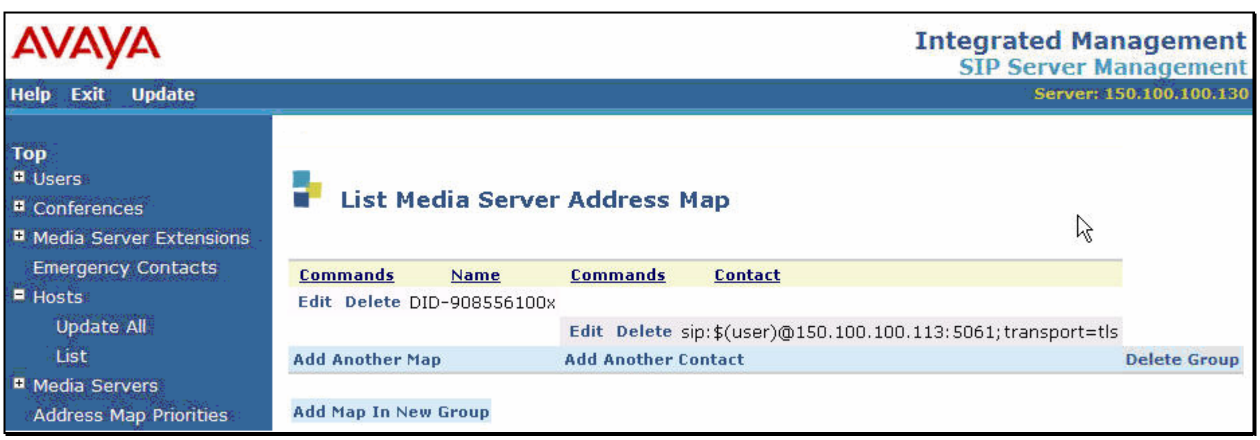

Figure 32: List Media Server Address Map

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

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

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

4.1.6. Specify the Telcordia SIR as a Trusted Host

The Telcordia SIR IP address (e.g., 225.132.6.98) must be added as a trusted host entry in the Avaya SES. As a trusted host, the Avaya SES will not attempt to authenticate incoming requests from the designated IP address.[1]

To add the trusted host entry:

- Expand the **Trusted Hosts** link in the left navigation menu of any SIP Server Management page. Select **Add** to display the Add Trusted Host screen shown in **Figure 33**.
- Enter the Telcordia SIR IP address (e.g., "225.132.6.98") into the **IP Address** field.
- Select "150.100.100.130" (the address of the Avaya SES) as the value of the **Host** field.
- Enter a description (e.g. "Telcordia SIR)" in the **Comment** field.
- Click the **Add** button once the form is completed.

Figure 33 - Add Trusted Host

The resulting List Trusted Hosts page should appear as shown in **Figure 34.**

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

Figure 34 - List Trusted Hosts

4.1.7. Save the Avaya SES Changes

After making any changes within Avaya SES, commit the database changes by using the **Update** link.

Perform this step by clicking on either **Update** link on any SIP Server Management page as shown in **Figure** 35.

Figure 35: Update Displayed After an Avaya SES Administrative Change

5. Verification Steps

5.1. Verification Tests

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

- Outbound Calls
    - Verify that calls placed to a telephone number registered in the Telcordia SIR are routed via the assigned SIP outbound trunk group. Verify that the talk-path exists in both directions and that calls remain stable and disconnect properly.

      Examining the SIP traces of this call should indicate a SIP INVITE message sent to Telcordia SIR with a resulting 302 redirect. A "list trace station" should show the call is using the first choice SIP trunk group.

    - Verify that a call placed to a telephone number not subscribed within the Telcordia SIR is routed via the second choice trunk group.

      Examining the SIP traces of this call should indicate a SIP INVITE message sent to Telcordia SIR with a resulting 406 final status returned. A "list trace station" should show the overflow to the second choice route trunk group.

- Inbound Calls
    - Verify that an incoming call from another known Telcordia SIR subscriber (who attempts to call via the Internet) is successfully received via the incoming SIP trunk group.

      Examining a SIP trace should show an incoming call from the far end subscriber. A "list trace station" or "list trace tac" should show the expected SIP trunk group being used.

    - Verify that an incoming call from a normal PSTN telephone (not a Telcordia SIR subscriber) is received over the expected trunk group.

- Direct IP-IP Connections – This applies if IP and/or SIP telephones and Direct IP-IP are used. Verify that stable calls are using Direct IP-IP talk paths using the "status station" or "status trunk-group" commands. When Direct IP-IP is used, the **Audio Connection** field will indicate "ip-direct" instead of "ip-tdm".

5.2. Troubleshooting Tools

The Avaya Communication Manager "list trace station", "list trace tac", "status station" and/or "status trunk commands are helpful diagnostic tools to verify correct operation and to troubleshoot problems. Diagnostic traces (performed by Avaya support) can be helpful in understanding the specific SIP interoperability issues.

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

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

6. Conclusion

These Application Notes describe the steps for configuring SIP trunking between an Avaya Communication Manager based SIP telephony solution and Telcordia SIR.

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

Further information on the Avaya DevConnect program can be found at http://www.avaya.com/devconnect.

7. References

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

[1] *Administrator Guide for Avaya Communication Manager*, February 2007, Issue 3.1, Document Number 03-300509.
[2] *Adding New Hardware for Avaya Media Servers and Gateways*, February 2007, Issue 2, Release 4.0, Document Number 03-300684
[3] *Feature Description and Implementation for Avaya Communication Manager*, Issue 5, Document Number 555-245-205
[4] *SIP Support in Avaya Communication Manager Running on the Avaya S8300, S8400, S8500 series and S8700 series Media Server*, March 2007, Issue 6.1, Document Number 555-245-206.
[5] *4600 Series IP Telephone Release 2.6 LAN Administrator Guide*, August 2006, Issue 4, Document Number 555-233-507
[6] *Installing and Administering SIP Enablement Services*, March 2007, Issue 2.1, Document Number 03-600768
[7] *Avaya Communication Manager Network Region Configuration Guide*, October 2005, Document ID 103244

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

[8] RFC 3261 - *SIP (Session Initiation Protocol)*, June 2002, Proposed Standard
[9] RFC 2833 - *RTP Payload for DTMF Digits, Telephony Tones and Telephony Signals*, May 2000, Proposed Standard

APPENDIX A: Specifying Pattern Strings in Address Maps

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

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

In the pattern matching string used in the Avaya SES:

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

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

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

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

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

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

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.