Avaya Solution & Interoperability Test Lab

Application Notes for Configuring SIP Trunking between the Telenet SIP Trunking service and an Avaya Aura™ Communication Manager Telephony Solution – Issue 1.0

Abstract

These Application Notes describe the steps to configure trunking using the Session Initiation Protocol (SIP) between the Telenet SIP Trunking service and Avaya Aura™ Communication Manager. The Avaya solution consists of Avaya Aura™ Communication Manager, Avaya Aura™ SIP Enablement Services and various IP Telephones.

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.

1. Introduction

These Application Notes describe the steps for configuring Session Initiation Protocol (SIP) trunking between the Telenet (Belgium) SIP Trunking Service and an Avaya Aura™ SIP telephony solution consisting of Avaya Aura™ SIP Enablement Services (SES), Avaya Aura™ Communication Manager and Avaya IP telephones using H.323 and SIP protocols as endpoints.

SIP is a standards-based communications approach designed to provide a common framework to support multimedia communication. RFC 3261 [4] is the primary specification governing this protocol. In the configuration described in these Application Notes, SIP is used as the signaling protocol between the Avaya components and the network service offered by Telenet. SIP manages the establishment and termination of connections and the transfer of related information such as the desired codec, calling party identity, etc.

1.1. Telenet SIP Trunking Service - IP Trunking Overview

The Telenet SIP Trunking Service with IP trunking used within these Application Notes serves as an interface between Avaya Telephones and ISDN, GSM and analog endpoints communicating with the Telenet IP Multimedia Subsystem (IMS) network.
The Telenet SIP Trunking standard service uses UDP as the transport protocol.

The following features were tested:

- Incoming & outgoing basic calls, including no answer, calling party hang-up, called party hang-up
- Outbound calls to domestic and international PSTN and GSM endpoints
- Codec support and priority selection
- DTMF tone generation and recognition using RFC 2833
- Calling Party Number and Called Party Number presentation and restriction for incoming and outgoing calls
- Call Hold / Resume
- Call Forwarding unrestricted / Busy / No Answer
- Supervised Call Transfer / Blind Call Transfer
- Conference Call
- Fax Send / Receive using T.38, using both the G.711 and G.729 codecs.
- Simultaneous Calls
- Long Calls
- Extension to Cellular (EC500)

Wherever possible, the tests were performed with combinations of local extensions, PSTN telephones, and GSM handsets registered with various providers.

1.2. Support

Support is available at:

www.telenet.be

2. Reference Configuration

The following diagram illustrates the configuration used for testing:

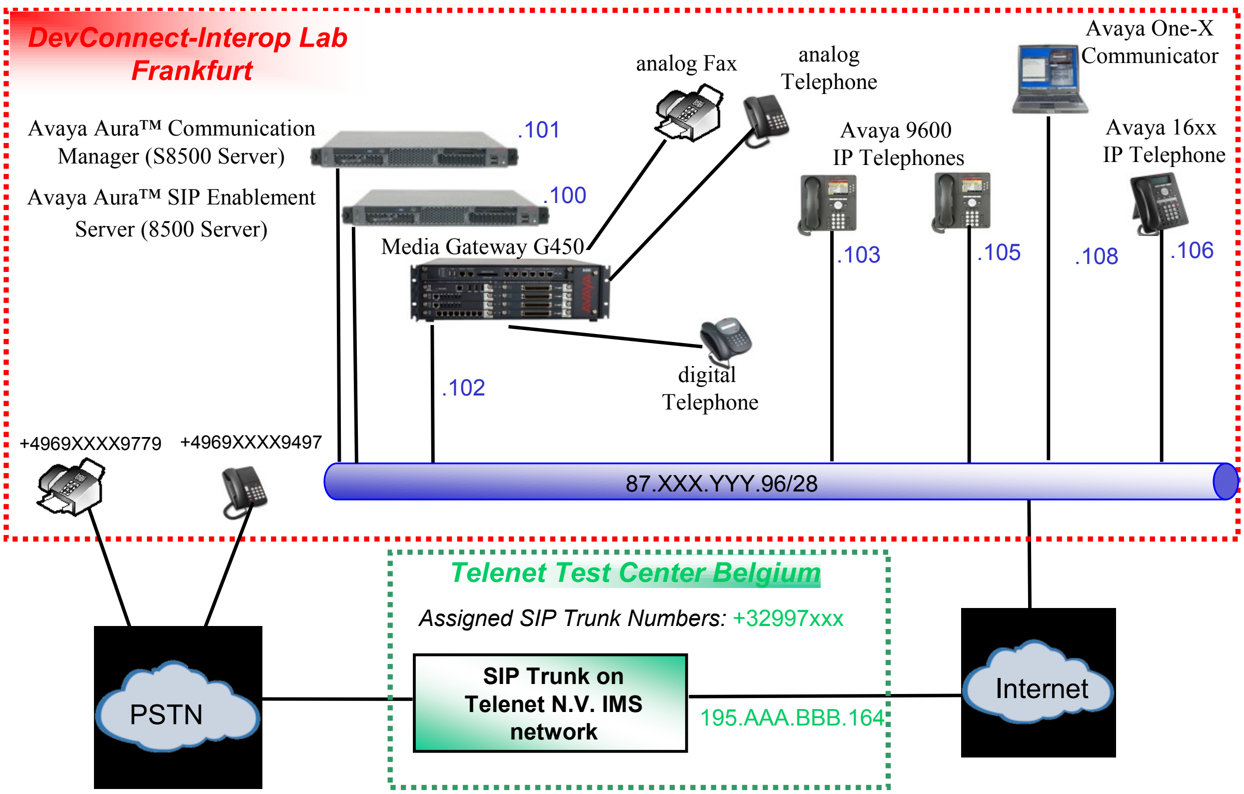

Figure 1: Avaya Aura™ SIP Telephony Solution using Telenet SIP Trunking Service

NOTE: All public IP addresses in these Application Notes are purposely concealed because of security reasons.

The simulated customer configuration included:

- Avaya S8500C Media Server running Avaya Aura™ Communication Manager

- Avaya G450 Media Gateway
- Avaya Aura™ SIP Enablement Services operating on a SES Combined Home/Edge server (S8500C).
- Avaya 96xx Series IP telephones using the H.323 / SIP software bundle.
- Avaya Digital phones
- Avaya Analog phones
- Fax machines
- PC running the Avaya One-X Communicator softphone

In the above diagram, Avaya IP Telephones are attached to the Avaya S8500 Server running Avaya Aura™ Communication Manager via Processor Ethernet.

The Avaya Aura™ Communication Manager / Telenet SIP Trunking service configuration used for testing are configured to support T.38 fax transmission.

3. Equipment and Software validated

The following equipment and software were used for the sample configuration provided:

Component	Version
Avaya	
Avaya Aura™ SIP Enablement Services (SES)	**Version 5.2:** Release String: SES-5.2.0.0-947.3b **SP2:** SES-02.0.947.3-SP2a
Avaya Aura™ Communication Manager	**Version 5.2:** Release String: S8500-015-02.1.016.4
Avaya G450 Media Gateway Media Module: MM711AP (analog stations)	Firmware Version: 28 .17 .0 Hardware Vintage: 1 Hardware Vintage:27 Firmware Version:072
Avaya 9630 IP Telephones (SIP)	2.4.1 *Application file:* SIP96xx_2_4_1_0.bin *Boot file:* hb96xxua1_50.bin

Component	Version
Avaya 9640 IP Telephones (H.323)	3.0 SP1 *Application file:* ha96xxua3_0_02.bin *Boot file:* hb96xxua3_00.bin
Avaya 2420 Digital Telephones	-
Avaya 6211 Analog Telephones	-
Avaya 1616 H.323 IP Telephones	1.2 *Application file:* ha1616ua1_2000.bin Bm32a2_1.hex *Boot file:* hb1616ua1_2000.bin
Avaya one-X™ Communicator	*Product Version*: 5.2.0.10 *Signaling Protocol*: H.323 *Version*: R.5200-GA-18347
Service Provider	
SIP server / Network Border Switch: *SONUS NBS GSX9000*	*Software version: V07.01.03 R000*

Table 1: Equipment and Software validated

4. Configuration

4.1. Avaya Aura™ Communication Manager

The Communication Manager configuration was performed using the System Access Terminal (SAT) and the Web interface to Avaya Aura™ Communication Manager

4.1.1. Verify system-parameters customer-options

Use the **display system-parameters customer-options** command to verify that Avaya Aura™ Communication Manager is licensed to meet the minimum requirements to interoperate with the Telenet SIP Trunking service. Those items shown in bold indicate required values or minimum capacity requirements. If these are not met in the configuration, please contact an Avaya representative for further assistance.

Verify that the parameters are set as shown in the following table:

Parameter	Usage
Maximum Off-PBX Telephones – EC500 (p.1)	This parameter must be large enough to support the number of stations which are paired with cell phones.
Maximum Concurrently Registered IP Stations (p.2)	This parameter must be large enough to support the number of IP stations to be attached.
Maximum Administered SIP Trunks (p.2)	This parameter must be large enough to support the number of SIP trunks to be attached.
ARS (p.3)	This parameter must be set to “y”.
Enhanced EC500 (p.4)	This parameter must be set to “y”.
Extended Cvg/Fwd Admin (p.4)	This parameter must be set to “y”.
IP Trunks (p.4)	This parameter must be set to “y”.
ISDN-PRI (p.4)	This parameter must be set to “y”.

Table 2: Optional Features Parameters

```
change system-parameters customer-options                       Page   1 of  10
                                OPTIONAL FEATURES

     G3 Version: V15                            Software Package: Standard
       Location: 1                           RFA System ID (SID): 1
       Platform: 12                          RFA Module ID (MID): 1

                                                             USED
                                Platform Maximum Ports: 44000 71
                                      Maximum Stations: 36000 9
                              Maximum XMOBILE Stations: 0     0
                    Maximum Off-PBX Telephones - EC500: 10    1
                    Maximum Off-PBX Telephones -   OPS: 100   3
                    Maximum Off-PBX Telephones - PBFMC: 0     0
                    Maximum Off-PBX Telephones - PVFMC: 0     0
                    Maximum Off-PBX Telephones - SCCAN: 0     0
```

Figure 2: Optional Features Form, Page 1

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

IP PORT CAPACITIES                                           USED
                     Maximum Administered H.323 Trunks: 1000  70
           Maximum Concurrently Registered IP Stations: 18000 3
             Maximum Administered Remote Office Trunks: 0     0
Maximum Concurrently Registered Remote Office Stations: 0     0
              Maximum Concurrently Registered IP eCons: 10    0
  Max Concur Registered Unauthenticated H.323 Stations: 0     0
                        Maximum Video Capable Stations: 0     0
                   Maximum Video Capable IP Softphones: 1000  0
                       Maximum Administered SIP Trunks: 100   60
  Maximum Administered Ad-hoc Video Conferencing Ports: 0     0
   Maximum Number of DS1 Boards with Echo Cancellation: 10    0
                             Maximum TN2501 VAL Boards: 10    1
                     Maximum Media Gateway VAL Sources: 100   1
           Maximum TN2602 Boards with 80 VoIP Channels: 128   0
          Maximum TN2602 Boards with 320 VoIP Channels: 128   0
   Maximum Number of Expanded Meet-me Conference Ports: 0     0
```

Figure 3: Optional Features Form, Page 2

```
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? y
        Analog Trunk Incoming Call ID? n                       CAS Branch? n
 A/D Grp/Sys List Dialing Start at 01? n                         CAS Main? n
Answer Supervision by Call Classifier? n                Change COR by FAC? n
                                 ARS? y  Computer Telephony Adjunct Links? y
                 ARS/AAR Partitioning? n   Cvg Of Calls Redirected Off-net? n
          ARS/AAR Dialing without FAC? y                      DCS (Basic)? n
          ASAI Link Core Capabilities? y                DCS Call Coverage? n
          ASAI Link Plus Capabilities? y               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? y
                  Attendant Vectoring? n
```

Figure 4: Optional Features Form, Page 3

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

   Emergency Access to Attendant? y                              IP Stations? y
           Enable 'dadmin' Login? y
           Enhanced Conferencing? n                        ISDN Feature Plus? n
                  Enhanced EC500? y        ISDN/SIP Network Call Redirection? n
    Enterprise Survivable Server? n                          ISDN-BRI Trunks? n
       Enterprise Wide Licensing? n                                 ISDN-PRI? y
              ESS Administration? n               Local Survivable Processor? n
          Extended Cvg/Fwd Admin? y                     Malicious Call Trace? n
     External Device Alarm Admin? n                 Media Encryption Over IP? n
  Five Port Networks Max Per MCC? n     Mode Code for Centralized Voice Mail? n
                Flexible Billing? n
   Forced Entry of Account Codes? n                 Multifrequency Signaling? y
      Global Call Classification? n        Multimedia Call Handling (Basic)? 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: Optional Features Form, Page 4

4.1.2. Set system-parameters features

Use the **change system-parameters features** command to set the parameters as shown in the following table:

Parameter	Usage
Trunk-to-Trunk Transfer	Set this value to “all”.

Table 3: Feature-Related System Parameters

```
change system-parameters features                               Page   1 of  18
                            FEATURE-RELATED SYSTEM PARAMETERS
                               Self Station Display Enabled? n
                                    Trunk-to-Trunk Transfer: all
                Automatic Callback with Called Party Queuing? n
    Automatic Callback - No Answer Timeout Interval (rings): 3
                       Call Park Timeout Interval (minutes): 10
        Off-Premises Tone Detect Timeout Interval (seconds): 20
                                 AAR/ARS Dial Tone Required? y
                             Music/Tone on Hold: music Type:
               Music (or Silence) on Transferred Trunk Calls? no
                       DID/Tie/ISDN/SIP Intercept Treatment: attd
    Internal Auto-Answer of Attd-Extended/Transferred Calls: transferred
                  Automatic Circuit Assurance (ACA) Enabled? n




             Abbreviated Dial Programming by Assigned Lists? n
       Auto Abbreviated/Delayed Transition Interval (rings): 2
                    Protocol for Caller ID Analog Terminals: Bellcore
    Display Calling Number for Room to Room Caller ID Calls? n
```

Figure 6: Feature-Related System Parameters Form, Page 1

4.1.3. Dial Plan

Use the **change dialplan analysis** command to configure the dial plan as shown in the following table.

Parameter	Usage
Dialed string: "00"	Use a "00" as Facilities Access Code (FAC) to access external telephone numbers.
Dialed string: "2"	Three digit numbers starting with "2" are for local extensions.
Dialed string: "*2"	The dialed string "*2" is the Trunk Access Code (TAC) for the SIP trunk.

Table 4: Dial Plan Analysis Parameters

```
change dialplan analysis                                        Page   1 of  12
                             DIAL PLAN ANALYSIS TABLE
                                   Location:  all          Percent Full:    0

       Dialed    Total  Call     Dialed    Total  Call      Dialed    Total  Call
       String    Length Type     String    Length Type      String    Length Type
     00            2    fac
     2             3    ext
     4             2    ext
     5             2    ext
     6             3    ext
     77            2    fac
     9             3    ext
     *01           3    fac
     *2            2    dac
     *3            2    dac
     *7            2    dac
     *8            2    fac
     *9            2    dac
     #8            2    fac
```

Figure 7: Dialplan Analysis Table Form

4.1.4. SIP Interface to Telenet SIP Trunking Service

Use the **change node-names ip** command to assign the names and the corresponding IP addresses of the Avaya components used:

"G450-GW-MAIN" for the G450-gateway,
"SES" for the Avaya Aura™ SIP Enablement Services and
"procr" for the S8500C server, running Avaya Aura™ Communication Manager

```
change node-names ip                                            Page   1 of   2
                                  IP NODE NAMES
    Name              IP Address
G450-GW-MAIN        87.XXX.YYY.102
SES                 87.XXX.YYY.100
SESS-MGR-SM100      192.168.0.111
default             0.0.0.0
procr               87.XXX.YYY.101
```

Figure 8: IP Node Names Form

Use the **add signaling-group** command to allocate a signaling group for the interface to the Avaya Aura™ SIP Enablement Services using the following parameters:

Parameter	Usage
Group Type	Enter "sip".
Transport Method	Enter "tls". (**Note**: During the test "tcp" was used for being able to take traces on the trunk between CM and SES)
Near-end Node Name	Enter "procr" to designate the S8500 processor as the near end node name.
Far-end Node Name	Enter "SES".
Near-end Listen Port	Enter "5060".
Far-end Listen Port	Enter "5060".
Far-end Network Region	Enter "1".
Far-end Domain	Leave the **Far-end-Domain** field blank. When an incoming SIP call is received by the SES and sent to the Communication Manager, the Communication Manager will look for a trunk on which to receive the call. This is done by looking at the domain/IP address of the SIP URI in the "FROM"-header and then by looking for a match in the Far-end Domain field on one of the SIP signaling groups configured on Communication Manager. When the Far-end-Domain field of a trunk is left blank, all calls that do not match any other signaling group will go over this trunk. (Note: This domain is specified in the Uniform Resource Identifier (URI) of the "SIP To Address" in the INVITE message. Configuring this field incorrectly may prevent calls from being successfully established to other SIP endpoints or to the PSTN.)
DTMF over IP	Enter "rtp-payload". This value is used to have Avaya Aura™ Communication Manager send DTMF transmissions using RFC 2833.
Direct IP-IP Audio Connections	Enter "n" to disallow direct IP-IP endpoint connections (shuffling).

Table 5: Signaling-Group Parameters

```
add signaling-group 2                                          Page   1 of   1
                                SIGNALING GROUP

 Group Number: 2              Group Type: sip
                        Transport Method: tls
  IMS Enabled? n





   Near-end Node Name: procr                Far-end Node Name: SES
 Near-end Listen Port: 5060               Far-end Listen Port: 5060
                                       Far-end Network Region: 1
Far-end Domain:

                                             Bypass If IP Threshold Exceeded? n
Incoming Dialog Loopbacks: eliminate                  RFC 3389 Comfort Noise? n
         DTMF over IP: rtp-payload            Direct IP-IP Audio Connections? n
Session Establishment Timer(min): 3                     IP Audio Hairpinning? n
         Enable Layer 3 Test? n
                                                 Alternate Route Timer(sec): 20
```

Figure 9: Signaling Group Form

Use the **add trunk-group** <**n**> command, were **n** is an unused trunk number, to allocate a trunk group to be used as an interface to the Avaya Aura™ SIP Enablement Services. Use the parameters shown in the following table.

Parameter	Usage
Group Type (p.1)	Enter "sip".
Group Name (p.1)	Assign a name for identification purposes.
TAC (p.1)	Enter the Trunk Access Code "*2", as defined in the dialplan analysis table (**Figure 4**)
Service Type (p.1)	Enter "public-ntwrk".
Signaling Group (p.1)	Enter the number of the signaling group allocated in **Figure 9**.
Number of Members (p.1)	Enter a number large enough to support the maximum number of anticipated simultaneous calls to be handled by the SIP trunk.
Redirect On OPTIM Failure (p.2)	Enter a timeout value, in milliseconds, to recover from failed responses for EC500. For the tested configuration, a value of 5000ms was chosen.
Preferred Minimum Session Refresh Interval (p.2)	Enter "900" seconds. This should be half of the Session Refresh Interval which is configured for the Telenet SIP Trunking Service.
Send Diversion Header (p.4)	Enter "y".
Support Request History (p.4)	Enter "y".
Telephone Event Payload Type (p.4)	Enter "101". This is the same value used on the Telenet side.

Table 6: Trunk Group Parameters

```
add trunk-group 2                                         Page   1 of  21
                                TRUNK GROUP

Group Number: 2                   Group Type: sip          CDR Reports: y
  Group Name: SIP-TRUNK-TO-SES …..COR: 1        TN: 1        TAC: *2
   Direction: two-way        Outgoing Display? n
 Dial Access? n                                 Night Service:
Queue Length: 0
Service Type: public-ntwrk          Auth Code? n

                                                       Signaling Group: 2
                                                     Number of Members: 30
```

Figure 10: Trunk Group Form, p.1

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

TRUNK PARAMETERS

     Unicode Name: auto

                                            Redirect On OPTIM Failure: 5000

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

Figure 11: Trunk Group Form, p.2

```
change trunk-group 86                                          Page   4 of  21
                               PROTOCOL VARIATIONS

                      Mark Users as Phone? n
            Prepend '+' to Calling Number? n
      Send Transferring Party Information? n
                 Network Call Redirection? n
                    Send Diversion Header? y
                  Support Request History? y
             Telephone Event Payload Type:101
```

Figure 12: Trunk Group Form, p.4

4.1.5. Outgoing Call Routing

For the test configuration, outgoing dialed numbers have the format: *00<country code><national number><number>*. Use the **change feature-access-codes** command to assign dialed digit strings to feature access codes. Use a "00" as the leading digits of ARS numbers which provide access to the SIP trunk.

```
change feature-access-codes                                     Page   1 of   6
                               FEATURE ACCESS CODE (FAC)
         Abbreviated Dialing List1 Access Code:
         Abbreviated Dialing List2 Access Code:
         Abbreviated Dialing List3 Access Code:
Abbreviated Dial - Prgm Group List Access Code:
                      Announcement Access Code:
                       Answer Back Access Code:
                         Attendant Access Code:
      Auto Alternate Routing (AAR) Access Code:
    Auto Route Selection (ARS) - Access Code 1: 00    Access Code 2:
                 Automatic Callback Activation:        Deactivation:
Call Forwarding Activation Busy/DA:        All:        Deactivation:
   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 13: Feature Access Code Form

Use the **change ars analysis** command to designate that all ARS calls to any numbers beginning with any digits ("x") with a minimum length of "3" digits and a maximum length of "20" digits be routed via route pattern "1" using public numbering format ("pubu").

```
change ars analysis 0                                          Page   1 of   2
                             ARS DIGIT ANALYSIS TABLE
                                   Location:  all           Percent Full:    0

          Dialed           Total      Route     Call   Node  ANI
          String          Min  Max  Pattern    Type   Num   Reqd
    x                      3    20     1       pubu         n
```

Figure 14: ARS Digit Analysis Table Form

Use the **change route-pattern** command to designate that calls routed via route pattern "1" be routed via trunk group "2".

```
change route-pattern 1                                        Page   1 of   3
                    Pattern Number: 1   Pattern Name: SIP-TRUNK-OUTG
                                          Secure SIP? n
    Grp FRL NPA Pfx Hop Toll No.  Inserted                           DCS/ IXC
    No          Mrk Lmt List Del  Digits                             QSIG
                             Dgts                                    Intw
 1: 2    0       1                                                    n   user
 2:                                                                   n   user
 3:                                                                   n   user
 4:                                                                   n   user
 5:                                                                   n   user
 6:                                                                   n   user

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

Figure 15: Route Pattern Form

Use the **change public-unknown-numbering** command to designate the calling party number sent to the far-end. The entries below indicate that 3-digit extensions beginning with "2" will prefix the digits "32997" to the calling party number on calls using trunk-group "2".

```
                      NUMBERING - PUBLIC/UNKNOWN FORMAT
                                        Total
Ext Ext           Trk       CPN          CPN
Len Code          Grp(s)    Prefix       Len
                                                  Total Administered: 2
 3  2             2         32997        8           Maximum Entries: 9999
```

Figure 16: Public Unknown Numbering Form

4.1.6. Incoming Call Routing

Use the **change inc-call-handling-trmt trunk-group** command to map calls arriving from trunk group "2" from public numbering format to the extensions of the locally attached endpoints shown in **Figure 1**.

```
change change inc-call-handling-trmt trunk-group 2                  Page   1 of  30
                       INCOMING CALL HANDLING TREATMENT
 Service/      Number   Number      Del Insert
 Feature        Len      Digits
public-ntwrk    8        32997       5
```

Figure 17: Incoming Call Handling Treatment Form

4.1.7. Configure Codec Sets

Use the **change ip-codec-set** command to designate a codec set to be used for communication with the Telenet SIP Trunking Service. Testing was done with the G.729B, G.729A and G.711A codecs, using the default of 2 frames per packet and a packet size of 20ms in all cases.

Parameter	Usage
Audio Codec (p. 1)	Enter "G.729A", "G.729B" and "G.711A" as the codecs to be used for communication with the Telenet SIP Trunking Service. The Telenet IMS network supports G.729A, G.729B, and G.711A for all calls.

Table 7: IP Codec Set Parameters

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

Figure 18: IP Codec Set Form, p.1

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

                          IP Codec Set

                              Allow Direct-IP Multimedia? n

                    Mode                Redundancy
    FAX             off                  0
    Modem           off                  0
    TDD/TTY         off                  3
    Clear-channel   n                    0
```

Figure 19: IP Codec Set Form, p.2

4.1.8. Configure IP Network Region

Use the **change ip-network-region <x>** command to designate a network region to be used for the Telenet SIP Trunking service using the parameters shown in the following table, where <x> is the network region assigned to the procr IP interface. In this case "1" is used, as the procr IP interface is assigned to a default network region of "1".

Parameter	Usage
Location	Enter "1".
Authoritative Domain	Enter an appropriate domain name to be assigned to the SIP trunk.
Name	Enter a name to identify the region.
Codec Set	Enter the number of the codec set defined in **Figure 18**.

Table 8: IP Network Region Parameters

```
change change ip-network-region 1                                      Page   1 of  19
                               IP NETWORK REGION
  Region: 1
Location: 1       Authoritative Domain: ffm.com
    Name: BT-Voice
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 20: IP Network Region Form, p.2

4.1.9. Configure Telephone Stations

Use the **add station** command to allocate an IP station using the parameters shown in the following table. Repeat this for each of the locally attached stations shown in **Figure 1**.

Parameter	Usage
Type (p. 1)	Enter the type identifier of local telephone.
Security Code (p. 1)	Enter the security code to be assigned to the station for security purposes.
Name (p. 1)	Enter a name to identify the station or its user.
EC500 (p. 4)	Add an EC500 button to activate/deactivate EC500.
extnd-call (p. 4)	Add that button to be able to extend an EC500 call.

Table 9: Station Parameters for IP Telephones

```
change station 254                                              Page   1 of   5
                                     STATION

Extension: 254                          Lock Messages? n               BCC: 0
     Type: 9630                         Security Code: 000000           TN: 1
     Port: S00001                     Coverage Path 1:                 COR: 1
     Name: IP-254-PH                  Coverage Path 2:                 COS: 1
                                      Hunt-to Station:
STATION OPTIONS
                                          Time of Day Lock Table:
              Loss Group: 19        Personalized Ringing Pattern: 1
                                                Message Lamp Ext: 254
            Speakerphone: 2-way              Mute Button Enabled? y
        Display Language: english                 Button Modules: 0
 Survivable GK Node Name:
          Survivable COR: internal               Media Complex Ext:
   Survivable Trunk Dest? y                           IP SoftPhone? n




                                             Customizable Labels? y
```

Figure 21: Station Form for IP Telephones, page 1

```
change station 254                                              Page   4 of   5
                                     STATION
 SITE DATA
       Room:                                           Headset? n
       Jack:                                           Speaker? n
      Cable:                                          Mounting: d
      Floor:                                       Cord Length: 0
   Building:                                         Set Color:

ABBREVIATED DIALING
     List1:                   List2:                   List3:




BUTTON ASSIGNMENTS
 1: call-appr   Auto-A/D? n        Rg:r  5:
 2: call-appr   Auto-A/D? n        Rg:r  6:
 3: call-appr   Auto-A/D? n        Rg:r  7: ec500      Timer? n
 4: call-appr   Auto-A/D? n        Rg:r  8: extnd-call

    voice-mail Number:
```

Figure 22: Station Form for IP Telephones, page 4

Use the **change cor 1** command to allow local stations to make external calls by setting “Calling Party Restriction” to “none”. This Class of Restriction is assigned to the stations which have access to the Telenet SIP Trunking Service, as shown in **Figure 1**.

Parameter	Usage
Calling Party Restriction	Enter “none” to allow local stations to make external calls.

Table 10: Class of Restriction Parameters

```
change cor 1                                                Page   1 of  23
                              CLASS OF RESTRICTION

               COR Number: 1
          COR Description:

                      FRL: 0                                APLT? y
  Can Be Service Observed? n           Calling Party Restriction: none
Can Be A Service Observer? n            Called Party Restriction: none
 Partitioned Group Number: 1       Forced Entry of Account Codes? n
         Priority Queuing? n                Direct Agent Calling? n
     Restriction Override: none       Facility Access Trunk Test? n
     Restricted Call List? n                 Can Change Coverage? n

            Access to MCT? y            Fully Restricted Service? n
Group II Category For MFC: 7
         Send ANI for MFE? n
            MF ANI Prefix:              Automatic Charge Display? n
Hear System Music on Hold? y   PASTE (Display PBX Data on Phone)? n
                        Can Be Picked Up By Directed Call Pickup? n
                                    Can Use Directed Call Pickup? n
                                    Group Controlled Restriction: inactive
```

Figure 23: Class of Restriction Form

Use the **change cos** command with the parameters shown in the following table for service class "1", which is assigned to the stations which forward calls via the Telenet SIP Trunking Service. This Class of Service is assigned to the stations which have access to the Telenet SIP Trunking Service, as shown in **Figure 1**.

Parameter	Usage
Restrict Call Fwd-Off Net	Enter "n" to allow calls to be forwarded via the SIP trunk.

Table 11: Class of Service Parameters

```
change cos                                                      Page   1 of   2
CLASS OF SERVICE

                               0  1  2  3  4  5  6  7  8  9 10 11 12 13 14 15
 Auto Callback                 n  y  y  n  y  n  y  n  y  n  y  n  y  n  y  n
 Call Fwd-All Calls            n  y  n  y  y  n  n  y  y  n  n  y  y  n  n  y
 Data Privacy                  n  y  n  n  n  y  y  y  y  n  n  n  n  y  y  y
 Priority Calling              n  y  n  n  n  n  n  n  n  y  y  y  y  y  y  y
 Console Permissions           n  n  n  n  n  n  n  n  n  n  n  n  n  n  n  n
 Off-hook Alert                n  n  n  n  n  n  n  n  n  n  n  n  n  n  n  n
 Client Room                   n  n  n  n  n  n  n  n  n  n  n  n  n  n  n  n
 Restrict Call Fwd-Off Net     y  n  y  y  y  y  y  y  y  y  y  y  y  y  y  y
 Call Forwarding Busy/DA       n  n  n  n  n  n  n  n  n  n  n  n  n  n  n  n
 Personal Station Access (PSA) n  n  n  n  n  n  n  n  n  n  n  n  n  n  n  n
 Extended Forwarding All       n  y  n  n  n  n  n  n  n  n  n  n  n  n  n  n
 Extended Forwarding B/DA      n  y  n  n  n  n  n  n  n  n  n  n  n  n  n  n
 Trk-to-Trk Transfer Override  n  n  n  n  n  n  n  n  n  n  n  n  n  n  n  n
 QSIG Call Offer Originations  n  n  n  n  n  n  n  n  n  n  n  n  n  n  n  n
 Contact Closure Activation    n  n  n  n  n  n  n  n  n  n  n  n  n  n  n  n
```

Figure 24: Class of Service Form

4.1.10. Configure FAX Devices

Use the **add station** command to add the fax device show in **Figure 1** using the parameters shown in the following table.

Parameter	Usage
Type	Enter "2500" to assign an analog device.
Port	Enter the identifier for the analog port to which the FAX is to be attached.
Name	Enter a name to identify the FAX or its user.

Table 12: Station Parameters for FAX Device

```
add station 252                                             Page   1 of   4
                                     STATION

Extension: 252                         Lock Messages? n               BCC: 0
     Type: 2500                          Security Code:                  TN: 1
     Port: 001V801                     Coverage Path 1:                 COR: 1
     Name: ANALOG-42                   Coverage Path 2:                 COS: 1
                                       Hunt-to Station:               Tests? y
STATION OPTIONS
      XOIP Endpoint type: auto             Time of Day Lock Table:
              Loss Group: 1             Message Waiting Indicator: none
    Off Premises Station? n



          Survivable COR: internal
   Survivable Trunk Dest? y
```

Figure 25: Station Form for FAX Device

4.1.11. SIP Endpoint Configuration

This section describes the administration of optional SIP endpoints and requires the SIP trunk configuration described in the chapters above.

Step1: Use the **add station** command to allocate a SIP station using the parameters shown in the following table. Repeat this for each of the locally attached stations.

Parameter	Usage
Type (p. 1)	Enter the SIP-type identifier of local telephone.
Security Code (p. 1)	Enter the security code to be assigned to the station for security purposes.
Name (p. 1)	Enter a name to identify the station or its user.
Per Station CPN – Send Calling Number (p. 2)	Allow calling party number information to be sent to the far-end when placing outgoing calls from this station by setting this field to “Y”.

Table 13: Station Parameters for IP Telephones

```
add station 256                                                   Page   1 of   6
                                     STATION

Extension: 256                          Lock Messages? n              BCC: 0
     Type: 9630SIP                      Security Code: 123456          TN: 1
     Port: S00003                     Coverage Path 1:                COR: 1
     Name: 96_30_sip                  Coverage Path 2:                COS: 1
                                      Hunt-to Station:
STATION OPTIONS
                                          Time of Day Lock Table:
              Loss Group: 19
                                                Message Lamp Ext: 256

        Display Language: english                 Button Modules: 0

          Survivable COR: internal
   Survivable Trunk Dest? y                         IP SoftPhone? n
```

Figure 26: Add Station – Page 1

```
change station 256                                              Page   2 of   6
                                     STATION
FEATURE OPTIONS
           LWC Reception: spe
          LWC Activation? y                      Coverage Msg Retrieval? y
                                                            Auto Answer: none
             CDR Privacy? n                            Data Restriction? n
                                              Idle Appearance Preference? n
                                            Bridged Idle Line Preference? n
   Bridged Call Alerting? n
  Active Station Ringing: single

        H.320 Conversion? n          Per Station CPN - Send Calling Number? y
                                                       EC500 State: enabled

    MWI Served User Type:
              AUDIX Name:
                                                Coverage After Forwarding? s

                                              Direct IP-IP Audio Connections? y
  Emergency Location Ext: 256          Always Use? n IP Audio Hairpinning? n
```

Figure 27: Add Station – Page 2

Step2: Configure the **Off-PBX Telephone station-mapping** form so that calls destined for a SIP telephone at the enterprise site are routed to SES, which will then forward the call to the intended SIP telephone.

Parameter	Usage
Application (p.1)	Enter "OPS" for Off-PBX station.
Phone Number (p. 1)	Enter the SIP extensions configured in SES that match the extensions of the corresponding stations in Communication Manager.
Trunk (p. 1)	Enter the number of the trunk between CM and SES.
Per Station CPN – Send Calling Number (p. 2)	Allow calling party number information to be sent to the far-end when placing outgoing calls from this station by setting this field to "Y".
Call Limit (p.2)	Set the Call Limit field to the maximum number of calls that may be active simultaneously at the station. In this example, the call limit is set to 3, which corresponds to the number of call appearances configured on the station form.

```
change off-pbx-telephone station-mapping 256                    Page   1 of   3
                  STATIONS WITH OFF-PBX TELEPHONE INTEGRATION

 Station       Application Dial   CC  Phone Number    Trunk       Config  Dual
 Extension                 Prefix                     Selection   Set     Mode
 256              OPS           -     256              2          1
                                -
```

Figure 28: Stations with Off-PBX Telephone Integration – Page 1

```
change off-pbx-telephone station-mapping 210                  Page   2 of   2
                  STATIONS WITH OFF-PBX TELEPHONE INTEGRATION

Station       Appl    Call        Mapping    Calls       Bridged       Location
 Extension     Name    Limit        Mode       Allowed     Calls
 6050          OPS      3         both         all         none
```

Figure 29: Stations with Off-PBX Telephone Integration – Page 2

4.1.12. Configure EC500

Enter the **change off-pbx-telephone configuration-set** command to specify that “Cellular Voice Mail Detection” is not to be used.

```
change off-pbx-telephone configuration-set 1                    Page   1 of   1



                                     CONFIGURATION SET: 1

                         Configuration Set Description:
                                  Calling Number Style: network
                                   CDR for Origination: phone-number
                    CDR for Calls to EC500 Destination? y
                           Fast Connect on Origination? n
                          Post Connect Dialing Options: dtmf
                         Cellular Voice Mail Detection: none
                                         Barge-in Tone? n
                           Calling Number Verification? y
             Call Appearance Selection for Origination: primary-first
                                      Confirmed Answer? n

 Use Shared Voice Connections for Second Call Answered? n
Use Shared Voice Connections for Second Call Initiated? n
```

Figure 30: Off-PBX-Telephone Configuration-Set Form

Enter the **change off-pbx-telephone station mapping”** and assign the following parameters:

Parameter	Usage
Station Extension	Enter the extension that is to be assigned with EC500.
Application	Enter EC500.
Phone Number	Enter the number of the PSTN-phone (e.g. a number of a mobile) that should be paired with the extension.
Config Set	Enter the configuration set, defined in **Figure 30**.
Trunk select	Enter the number of SIP trunk group:“2”.

```
change off-pbx-telephone station-mapping

                    STATION TO OFF-PBX TELEPHONE MAPPING

Station        Appl   CC  Phone Number     Config Trunk  Mapping      Calls
Extension                                  Set    Select Mode         Allowed

254            EC500      491719682165     1  /   2      both         all
255            OPS        255              1  /   2      both         all
256            OPS        256              1  /   2      both         all
257            OPS        257              1  /   2      both         all
```

Figure 31: Off-pbx-telephone station-mapping Form, page 1

```
change off-pbx-telephone station-mapping 254                    Page   2 of   3
                  STATIONS WITH OFF-PBX TELEPHONE INTEGRATION

 Station       Appl    Call        Mapping    Calls       Bridged       Location
 Extension     Name    Limit       Mode       Allowed     Calls
 254           EC500    3        both         all         both
```

Figure 32: Off-pbx-telephone station-mapping Form, page 2

4.2. Avaya IP Telephones

All Avaya IP Telephones must be configured such that the default gateway is assigned to the IP address of the access router which provides access to the Telenet SIP Trunking Service. Since Processor Ethernet is used for the tested configuration, the server address must be assigned to the Processor Ethernet address of the Avaya S8500 server. This IP address has to be used in the H.323 phones for the "Call Server" entry. In SIP phones the IP address of the SES has to be entered in the "SIP-Proxy Server" menu item. These values can either be assigned manually to each telephone, or automatically via DHCP.

4.3. Configure Avaya SIP Enablement Services

This section covers the administration of Avaya Aura™ SIP Enablement Services (SES). SES is configured via an Internet browser using **SIP Server Management** screens. It is assumed that SES software and the license file have already been installed. For additional information on these installation tasks, refer to Reference [7].

4.4. SIP trunking to Telenet SIP Trunking service

4.4.1. Log in to Avaya SIP Enablement Services

Access the SES **SIP Server Management** pages **(Figure 34)** by entering "**https://<ip-addr>/admin**" as the URL in an Internet browser, where **<ip-addr>** is the IP address of the SES server.
Log in with the appropriate credentials and then select the "**Administration ->SIP Enablement Services**" link from the main page as shown in **Figure 33**:

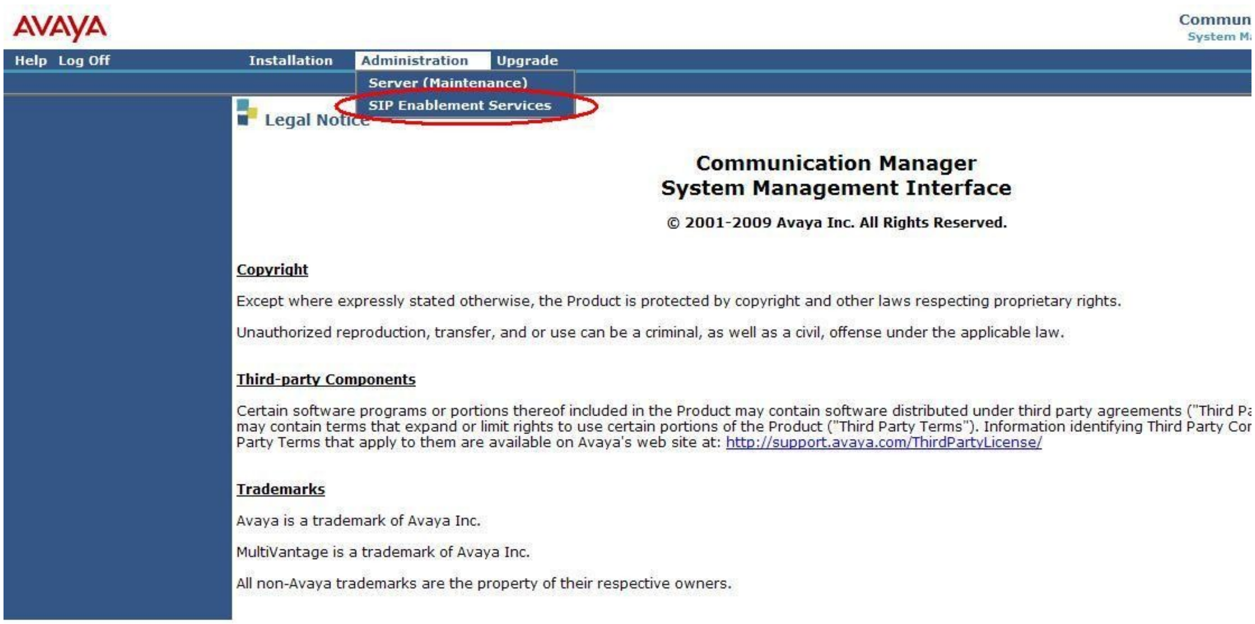

Figure 33: SES Main Page

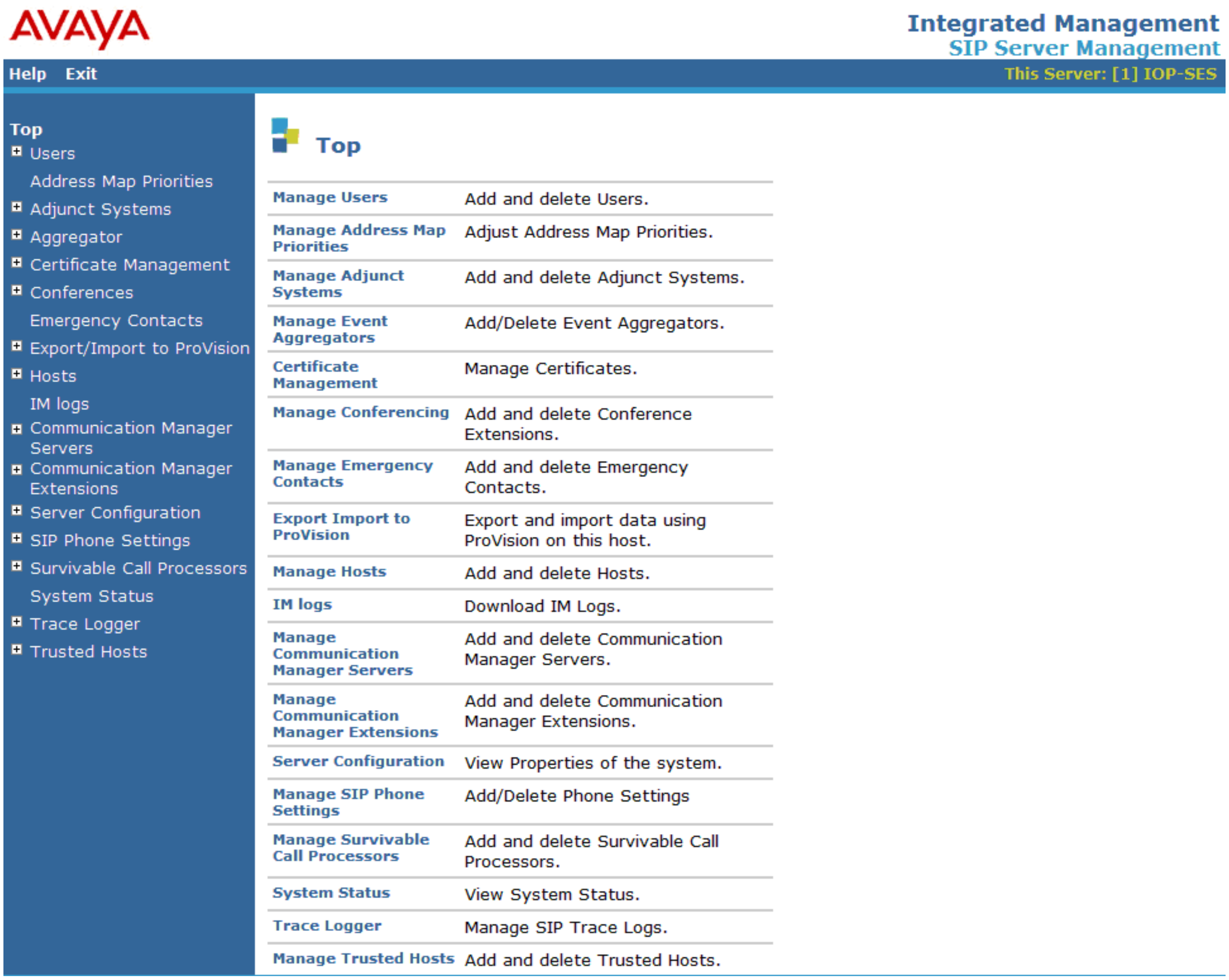

Figure 34: SES SIP Server Management pages

4.4.2. Verify System Properties

From the left pane of any **SIP Server Management** page, expand the **Server Configuration** option and select **System Properties**. This page **(Figure 35)** displays the **SES Version** and the **Network Properties** entered via the install script during the installation process.

In the **Edit System Properties** page, enter the same **SIP Domain** that has been entered in the network region of Communication Manager **(Figure 20).** The **SIP Domain** "`ffm.com`" is used in these Application Notes.

The License Host is running on the SES in this test environment. Hence enter the IP address of the SES in the **SIP License Host** field.

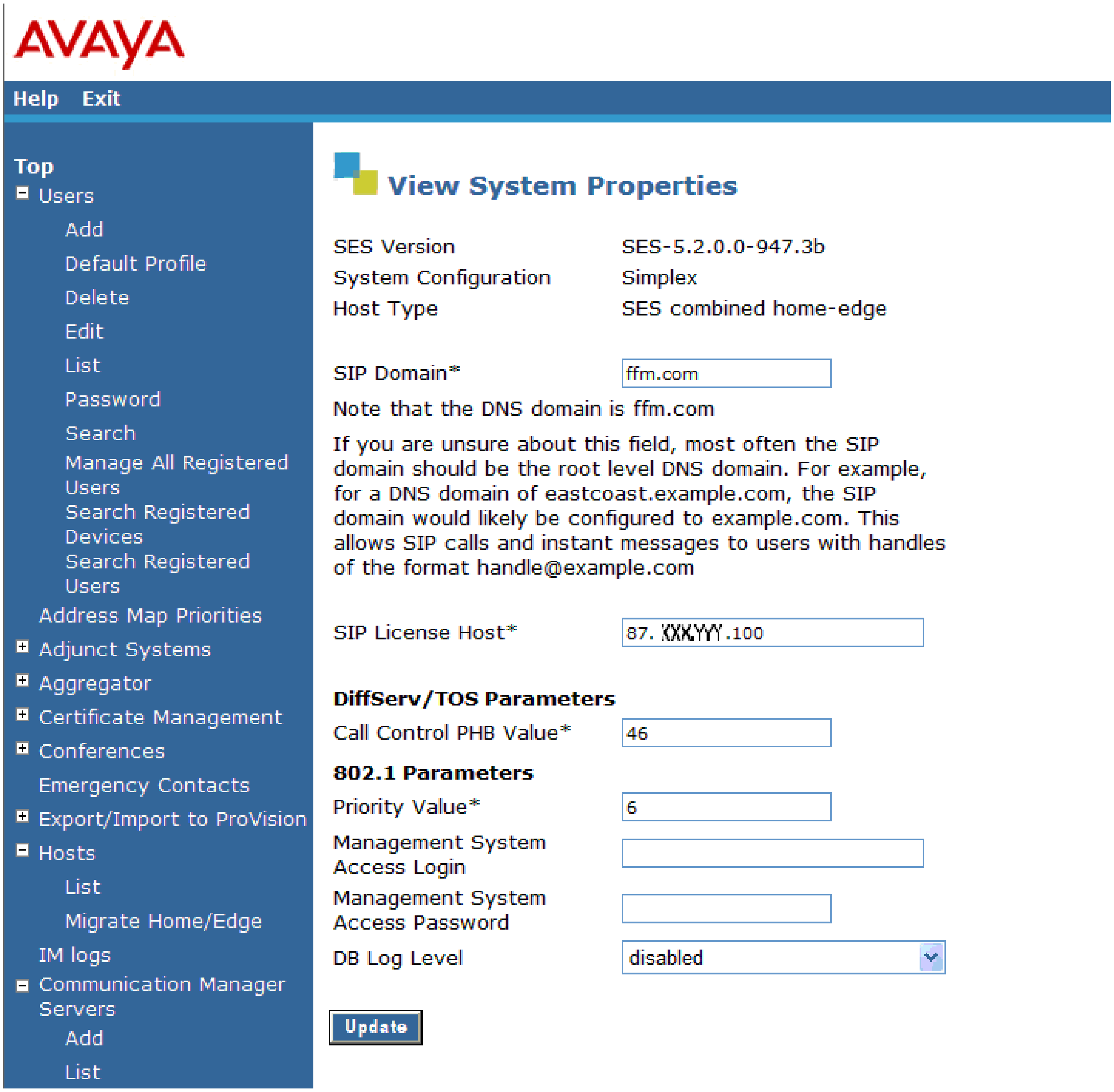

Figure 35: System Properties

4.4.3. Verify the Avaya SES Host Information

Verify the SES Host information using the **Edit Host** page. In these Application Notes the SES **Host Type** is a "Combined Home-Edge SES".

Display the **Edit Host** page by following the **Hosts** link in the left navigation pane and then click 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 SES Home/Edge server is in the **Host IP Address** field.
- Do not modify the **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 SES installation; incorrect changes may disable the SES.
- Verify that the **UDP**, **TCP** and **TLS** checkboxes are enabled as **Listen Protocols**.
- Verify that **TLS** is selected as the **Link Protocol**. (**Note**: During the test "TCP" was used for being able to take traces on the trunk between CM and SES)
- Ensure that the Telenet SIP Trunking Service Network Border Switch (e.g., `Telenet-sip.net`) is not in the **Outbound Proxy** or **Outbound Direct Domains** fields.
- Default values for the remaining fields may be used.
- After completing the **Edit Host** screen, click on the **Update** button.

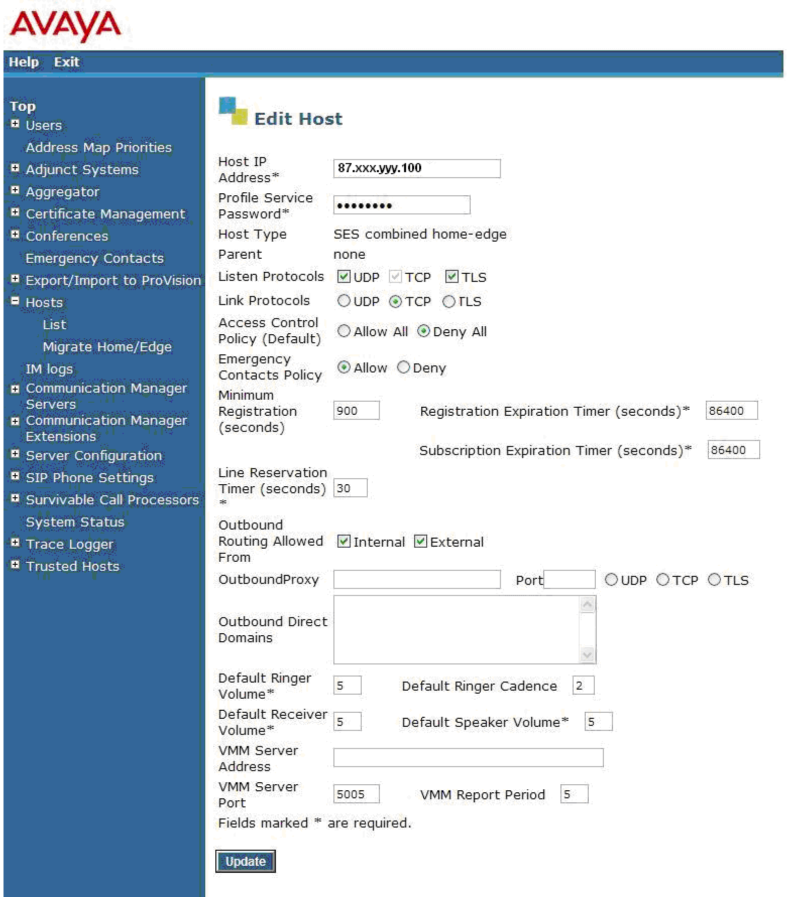

Figure 36: Edit Host

4.4.4. Add Communication Manager Media Server Interfaces

Under the **Communication Manager Servers** link from the left pane of the **SIP Server Management** page, select **Add (/Edit)** to add (/change)the Communication Manager on the enterprise site.

In the **Add (/Edit) Communication Manager Server Interface** screen, enter the following information:

- A descriptive name in the **Communication Manager Server Interface Name** field (e.g. "`procr`").
- Select the SES home/edge IP address in the **Host** field.
- Select ***TLS*** for the **SIP Trunk Link Type**. That is the protocol used on the trunk between SES and Communication Manager. (**Note**: During the test "tcp" was used for being able to take traces on the trunk between CM and SES)
- Enter the IP address of the Communication Manager in the **SIP Trunk IP Address** field.[1]
- After completing the **Add Media Server** screen, click on the **Update** button.

[1] Depending on the platform of the media server and gateway, this field may be set to the IP address of a C-LAN board or a media server.

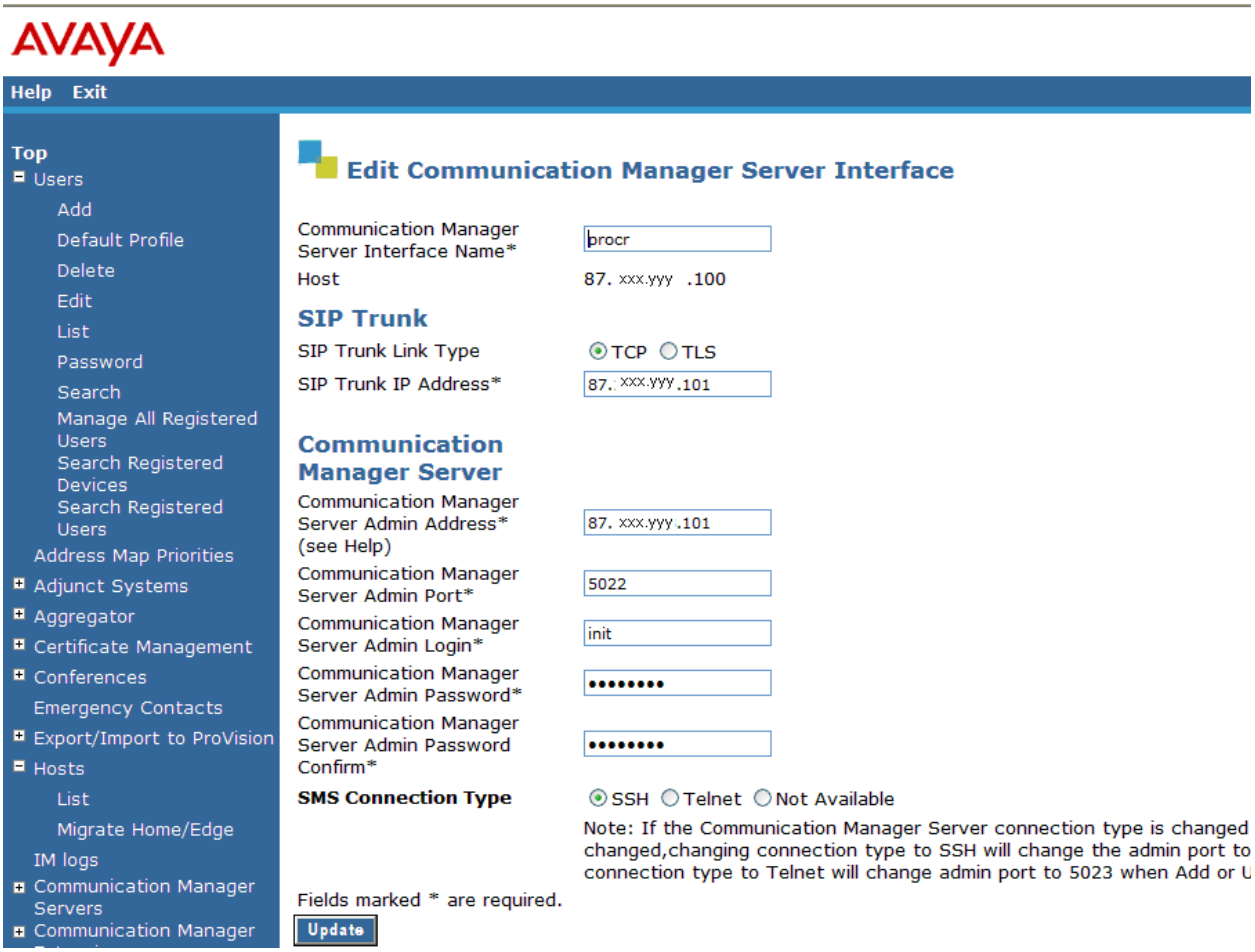

Figure 37: Add Media Server Interface

4.4.5. Configure Call Routing

SES operates as a SIP proxy server. In this role SES must direct SIP messages originating from Communication Manager to the Telenet IMS network for outbound calls. In a similar manner incoming SIP messages during an incoming call must be routed from SES to the proper signaling interface to Communication Manager.

In these Application Notes, the SIP message routing will be done for both outbound and inbound calls using Address Maps that examine some or all of the *called number* (using a *"Pattern"*) and route to a specific predetermined destination (called a *"Contact"*). The outbound proxy and direct domain routing feature is not used due to interactions with the trusted host capabilities.

The *called number* is contained within the *user* part of the Uniform Resource Identifier (URI) of an incoming SIP INVITE message. The URI usually 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 will only contain digits[2].

The Address Map Patterns are specified using Linux regular expression syntax. Patterns are generally designed to match a collection of *called numbers* that require identical SIP message routing. However, each Pattern must also be specific enough to direct each unique *called number* to the proper signaling Contact**.** The Address Map Patterns must also be mutually exclusive (non-overlapping) from all other Address Map Patterns used in the SES to ensure proper operation.

Appendix B provides a detailed description of the Linux regular expression syntax used within the address map patterns.

4.4.5.1 Outbound PSTN Calls

SIP signaling for outbound calls is directed to SES based upon rules configured for Automatic Route Selection in Communication Manager. The choice of trunk group (made within ARS) determines the codec that will be used for voice (or fax or modem) calls. For outbound calls, no further differentiation of voice vs. fax / modem is required within the SES address maps.

Outbound calls require SIP signaling messages to be routed to the Telenet Network Border Switch using Host Address Maps within SES. Calls matching Host Address Map Patterns will be directed to the Telenet Network Border Switch contained within the corresponding Contact information.

More specific pattern matches would be used if necessary to selectively route SIP traffic to different destinations (such as multiple service providers serving different geographic regions). Also note that a user dialed access code (such as “9” or “00” to place a PSTN or ISDN call) has been previously deleted (by ARS) prior to seizing the outbound SIP trunk.

---

[2] SIP does permit mnemonic addressing such as “sip:john.doe@customer.com”. However, his convention is not used in these Application Notes for SIP Trunking. Further discussion of this topic is beyond the scope of this document.

Begin the outbound routing configuration by navigating to the **Add Host Address Map** pages.

- From any **SIP Server Management** page, expand the **Hosts** link and choose the **List** link causing the **List Host** page to appear.
- Select the **Map** link on the **List Hosts** page **(Figure 38)** causing the **List Host Address Map** page to appear (**Figure 39**).

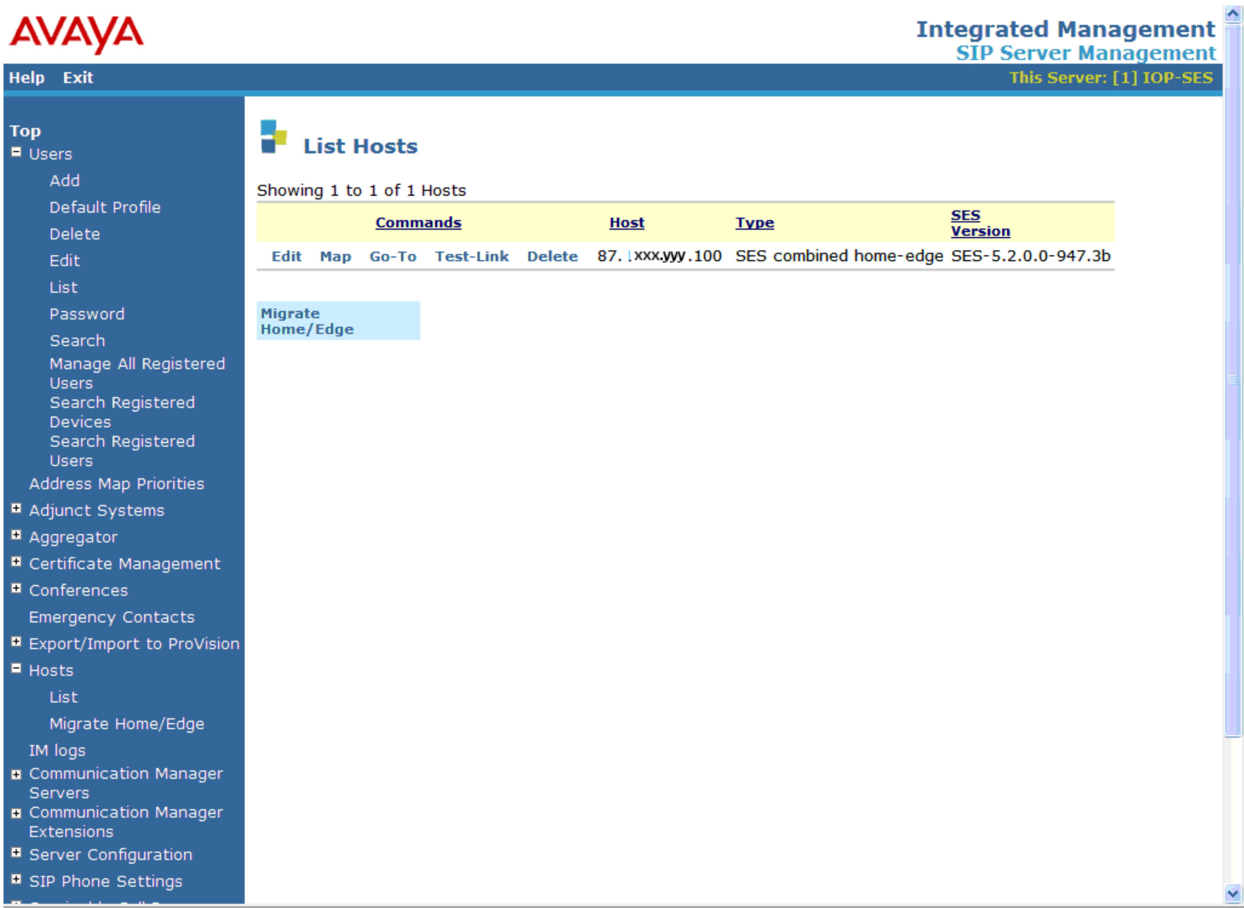

Figure 38: Accessing Host Address Maps from List Hosts Screen

- Select the **Add another Map** or the **Edit** link on the **List Host Address Map** page (**Figure 39**). This will display the **Add Host** Map Entry or **Edit Host Map Entry** screen (**Figure 40**).

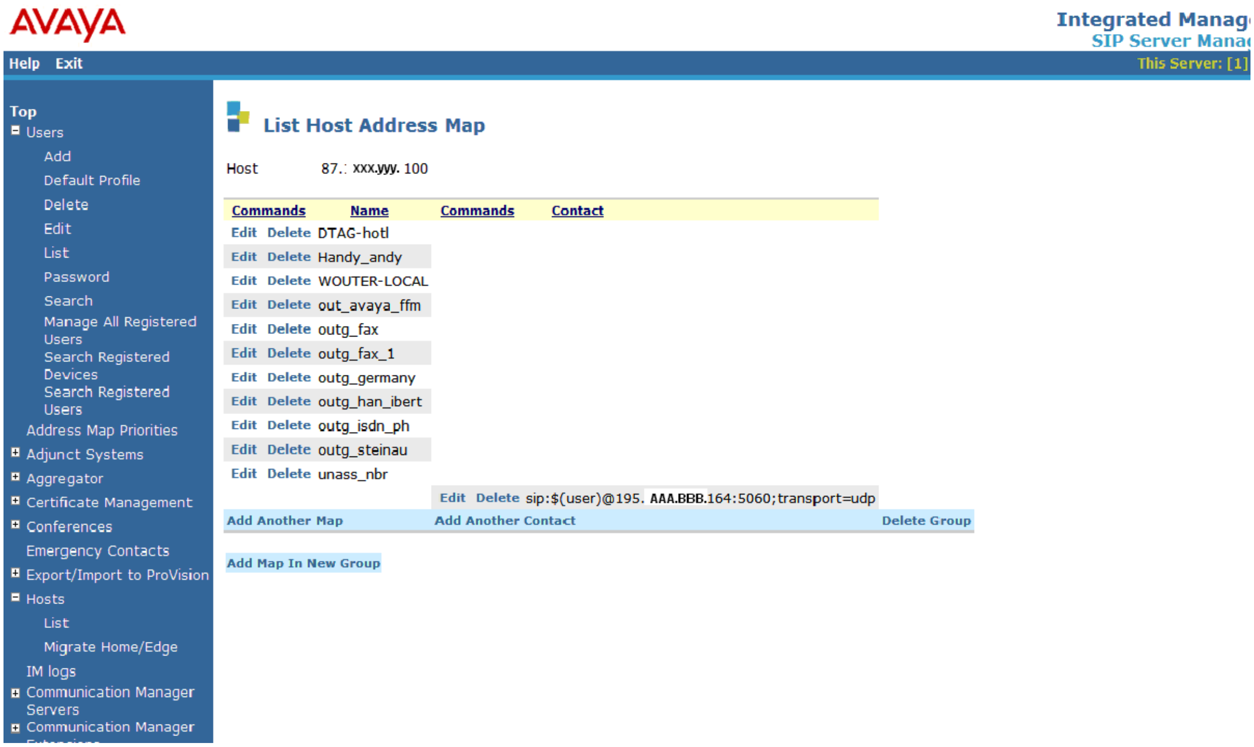

Figure 39: Adding a Host Address Map Group

The configuration of the **Host Address Map** for outgoing calls to Germany is shown in **Figure 40**:

- Enter a descriptive **Name** for the map, such as "`outg_germany`".
- Enter the appropriate pattern for the call type. In this example, the pattern is used for international calls to the German ISDN network: "`^sip:0049`"
- Select the **Replace URI** checkbox.
- Click the **Update** button.

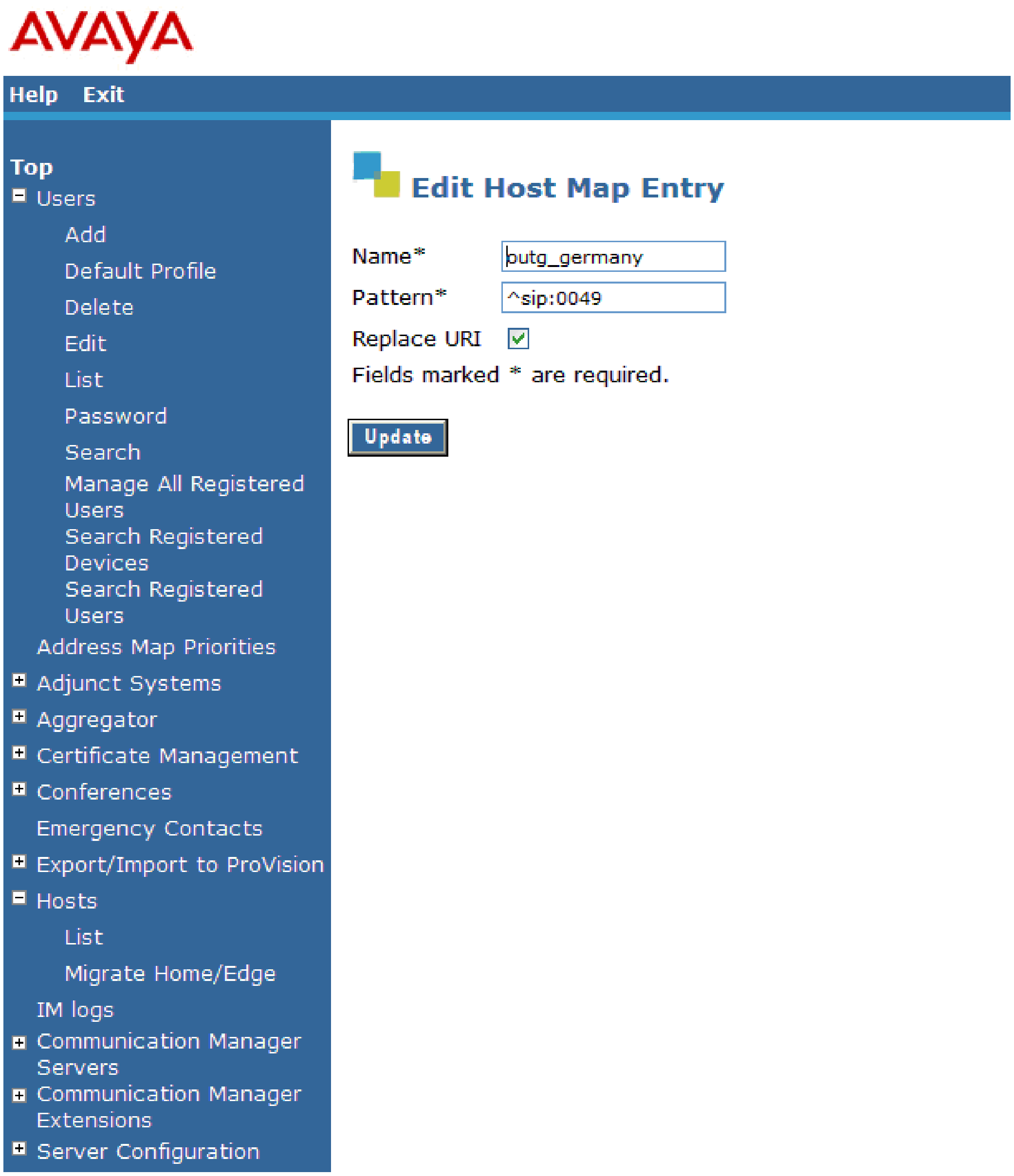

Figure 40: Address Map for outgoing calls to Germany

Note that in these Application Notes some other Maps were used, that are not explained here in detail.

To enter the remote contact data of the Telenet SIP Trunking Service, access the **Host Address Map** page by expanding the **Hosts** link in the left pane of any **SIP Server Management** page and select **List.** Then click on the **Map** link associated with the appropriate **Host**.

The **List Host Address Map** page is shown in **Figure 41**.

Note: Should an entry already exist due to prior administration, the entry should be edited or deleted instead of using **Add Another Contact**.

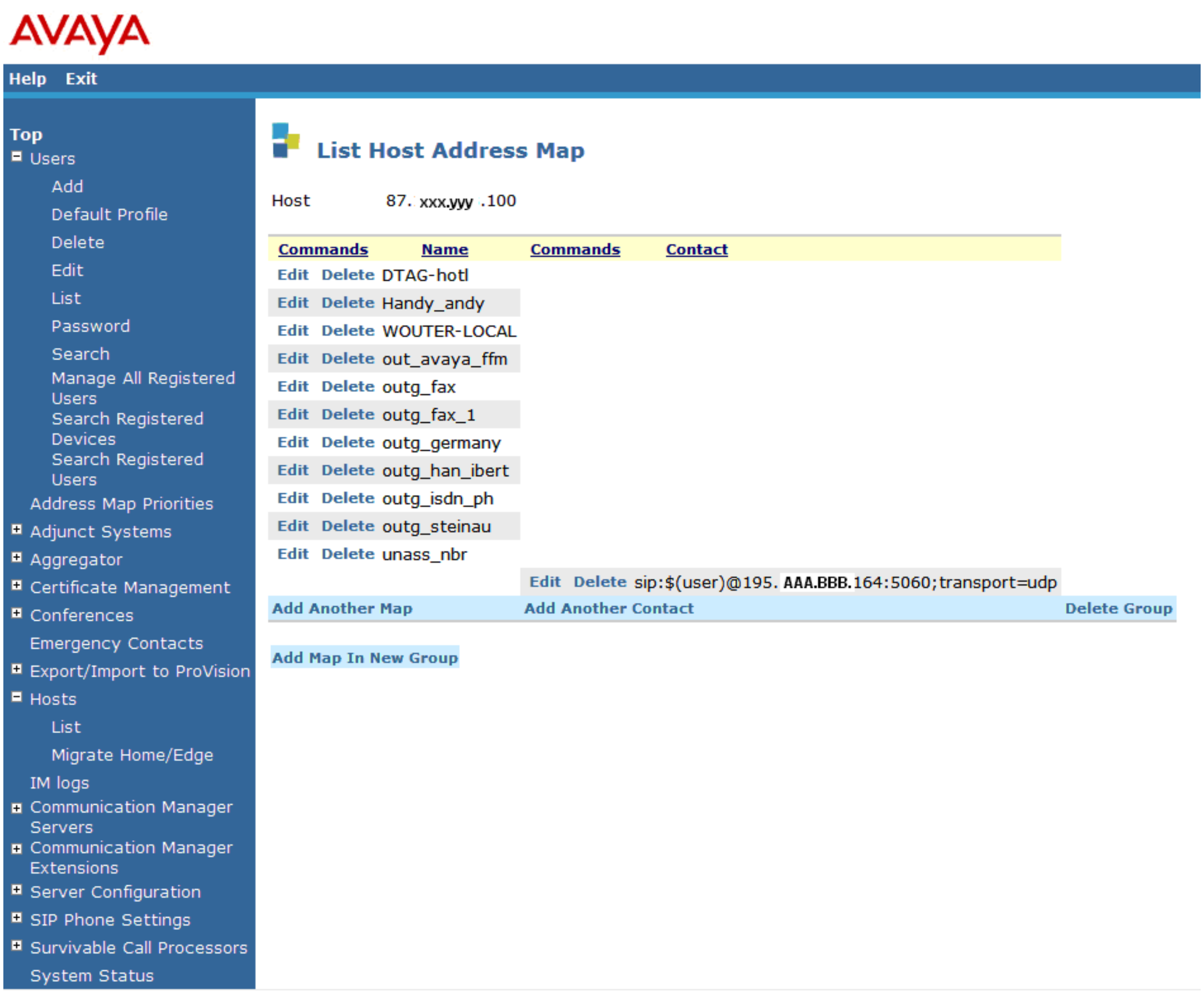

Figure 41: List Host Address Map

- Click on the **Add Another Contact / Edit Host Contact)** link to open the **Add Host Contact / Edit Host Contact** page (**Figure 42**).
- In the **Add/Edit Host Contact** page, the **Contact** field specifies the destination for the call:

  “`sip:$(user)@195.AAA.BBB.164;transport=udp`”

  where *“195.AAA.BBB.164”* is the IP address of the Network Border Switch of Telenet.

- Click the **Submit** button when completed.

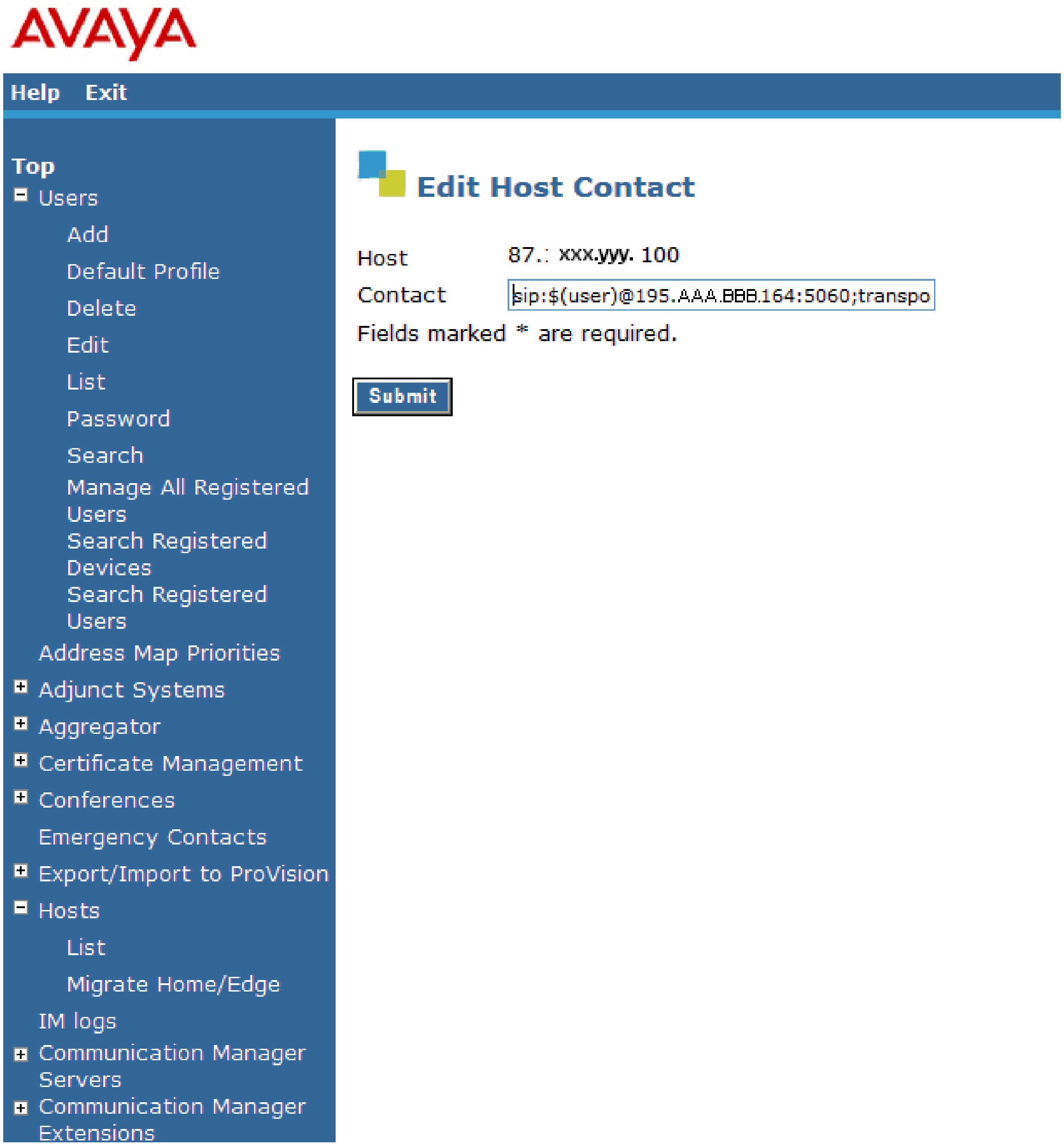

Figure 42: Edit (/Add) Host Contact Entry

4.4.5.2 Inbound Direct Inward Dialed Calls

SIP messages of incoming calls from the Telenet SIP Trunking Service are sent to the SES server. SES then routes these messages to the appropriate Communication Manager using an SES **Communication Manager Server Address Map**.

An example of a SIP URI in an INVITE message received from the Telenet SIP Trunking Service for the number ***32997252*** is:

sip: 32997252@87.XXX.YYY.100

The ***user*** part in this case is the 9-digit number "***32997252***".
To configure the **Communication Manager Address Map**:

- Expand the **Communication Manager Servers** link in the left navigation menu of any **SIP Server Management** page. Select **List** to display the **List Communication Manager Servers** page as shown in **Figure 43**.
- Click on the **Map** link of the "`procr`" Interface to display the **List Communication Manager Server Address Map** (shown in **Figure 44**) screen associated with this signaling interface.

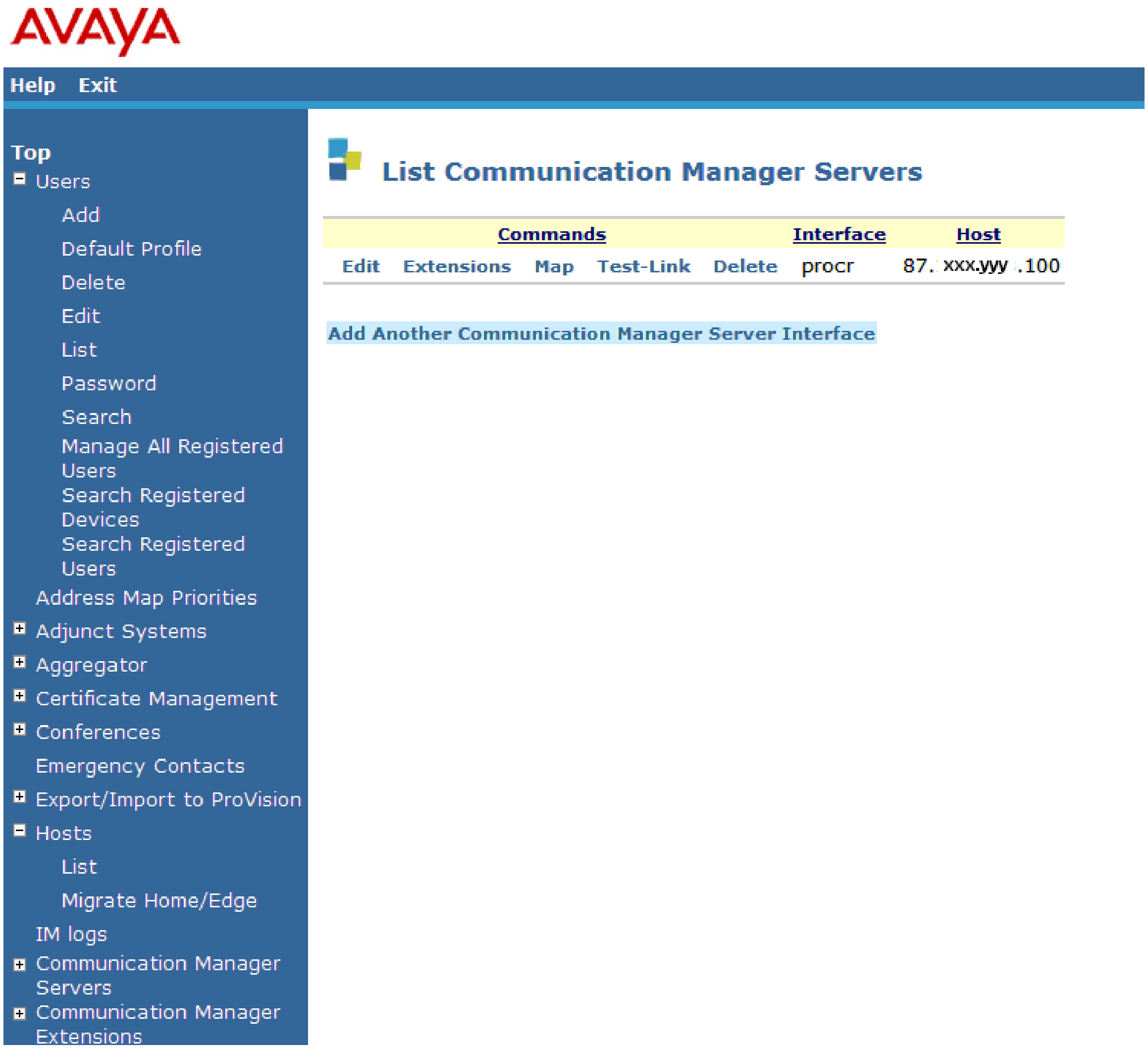

Figure 43: List Communication Manager Servers

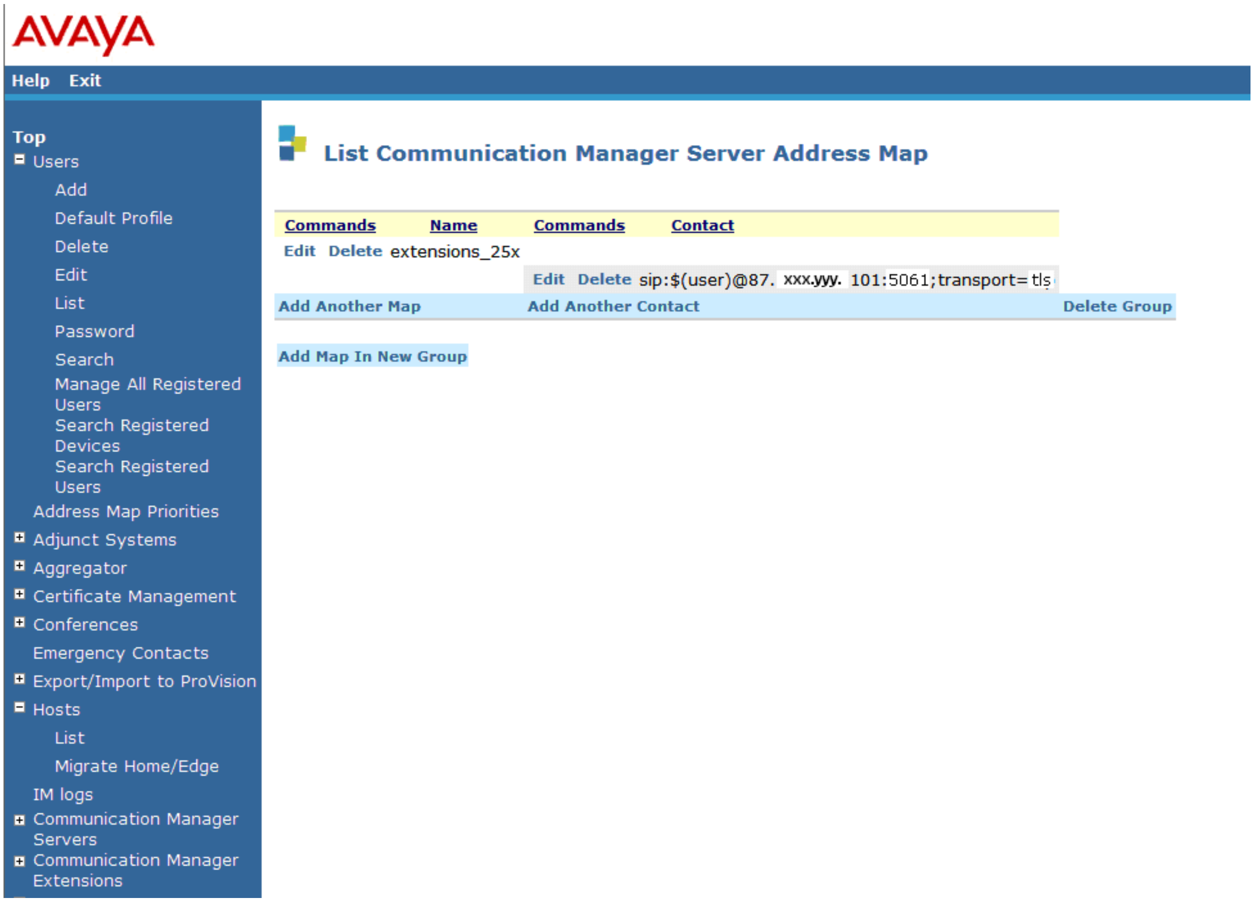

Figure 44: List Communication Manager Server Address Map

- Click on the **Add Another Map** (/**Edit** in the Command column) link. The page shown in **Figure 45** is displayed. The **Host** field in **Figure 43** displays the name of the media server to which this map applies.
- Enter a descriptive name in the **Name** field, such as "`extensions_25x`".
- Enter the **Address Map Pattern** for incoming calls into the **Pattern** field.

In this case, the pattern would match the numbers provided by the provider Telenet. The range for extensions is from *32997250* until *32997259*. The corresponding pattern specification for this DID number range is:

"^sip:3299725[0-9]{1}"

This means that URIs beginning with *"sip:3299725"* followed the single digit 0, 1, 2, 3, 4, 5,6,7,8 or 9 will match the pattern and be routed to the Communication Manager server.

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

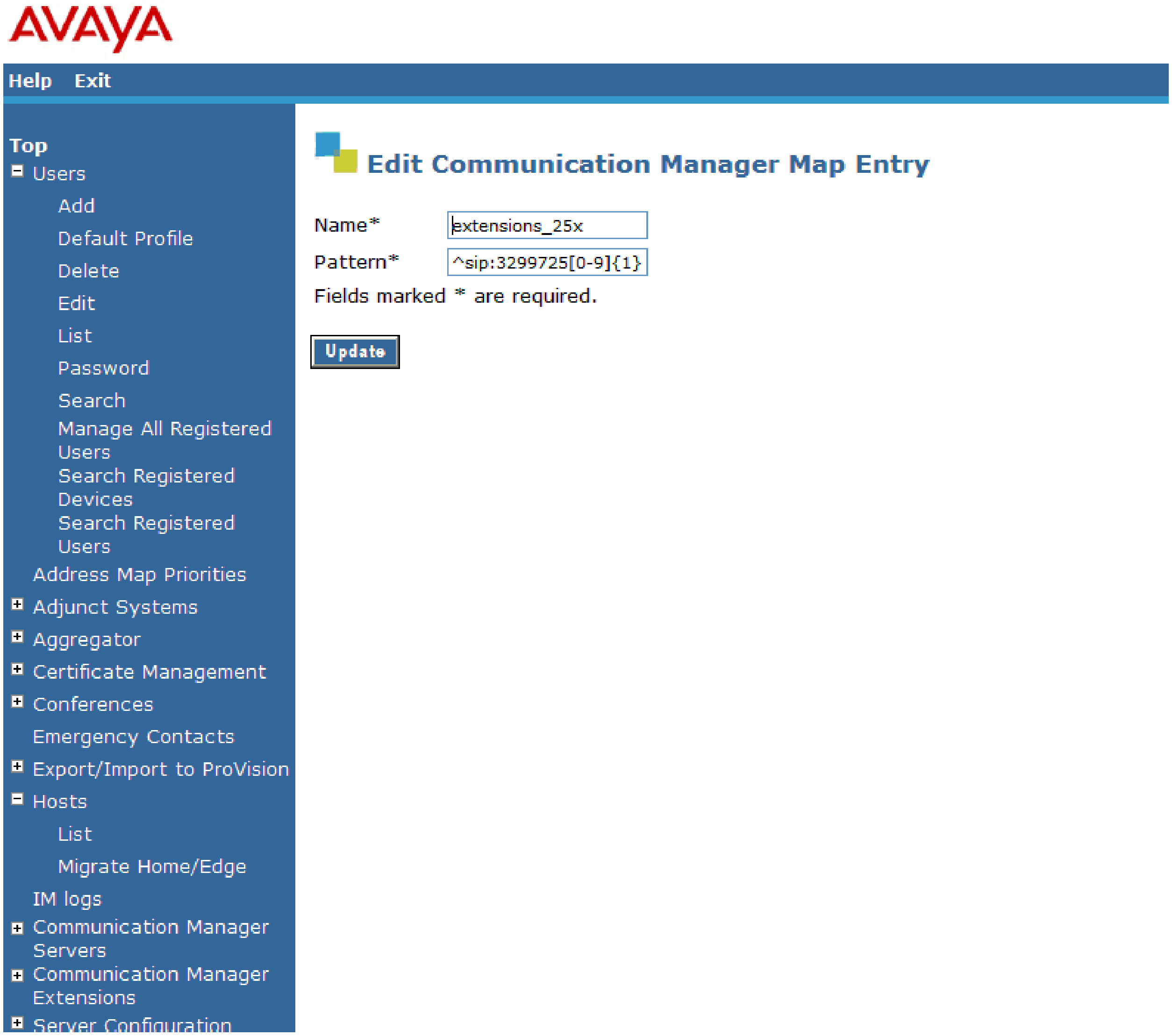

Figure 45: Incoming Calls - Communication Manager Server Address Map

Note that after the first **Communication Manager Server Address Map** is created, a corresponding media server **Contact** entry is created automatically.

"sip:$(user)@87.XXX.YYY.101:5061;transport=tls"

This **Contact** entry contains the IP address of the Communication Manager Server ("procr"), the port (5060) and the transport protocol (TCP) to be used to send SIP signaling messages. The incoming digits sent in the ***user*** part of the original request URI will replace the $(user) string when the message is sent to the **Contact**.

4.4.6. Specify the Telenet SIP Trunking Network Border Switch as a Trusted Host

The final step to complete the SIP trunk administration on SES is to designate the IP address of the Telenet SIP Trunking Service as a trusted host. As a trusted host, SES will not issue SIP authentication challenges for incoming requests from the designated IP address.[3]
To configure a trusted host:

- Expand the **Trusted Hosts** link in the left navigation menu of any **SIP Server Management** page. Select **List** to display the **List Trusted Hosts** page as shown in **Figure 46.**
- Click on the **Add Another Trusted Host** link. The page shown in **Figure 47** is displayed. The **Host** field displays the IP address of the SES server:*"87.XXX.YYY.100"*.
- Enter the IP address of the Network Border Switch of the Telenet SIP Trunk Service in the **IP Address** field, such as *"195.AAA.BBB.164"*
- Enter a descriptive name in the **Comment** field such as *"Telenet NL"*.
- Click the **Update** button once the form is completed.

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

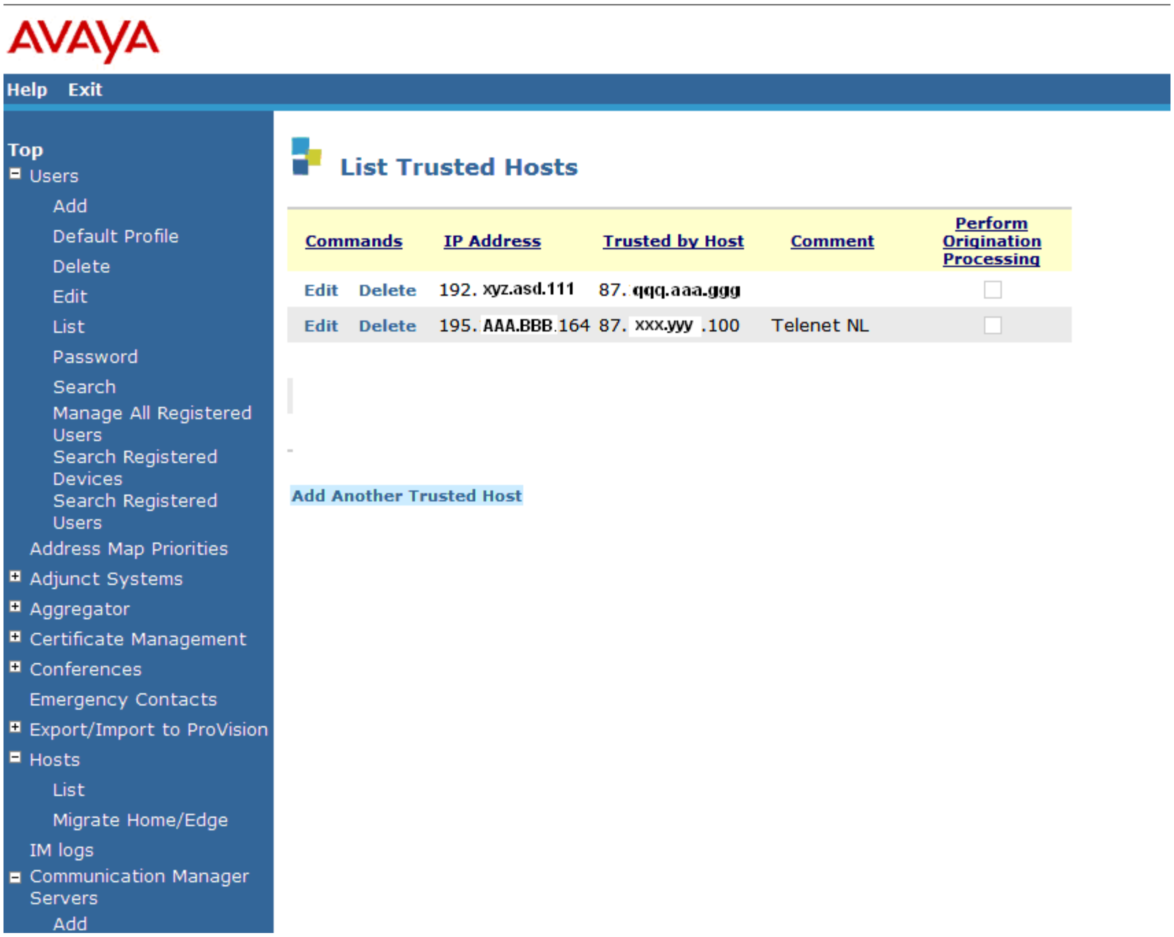

Figure 46: List Trusted Hosts Map

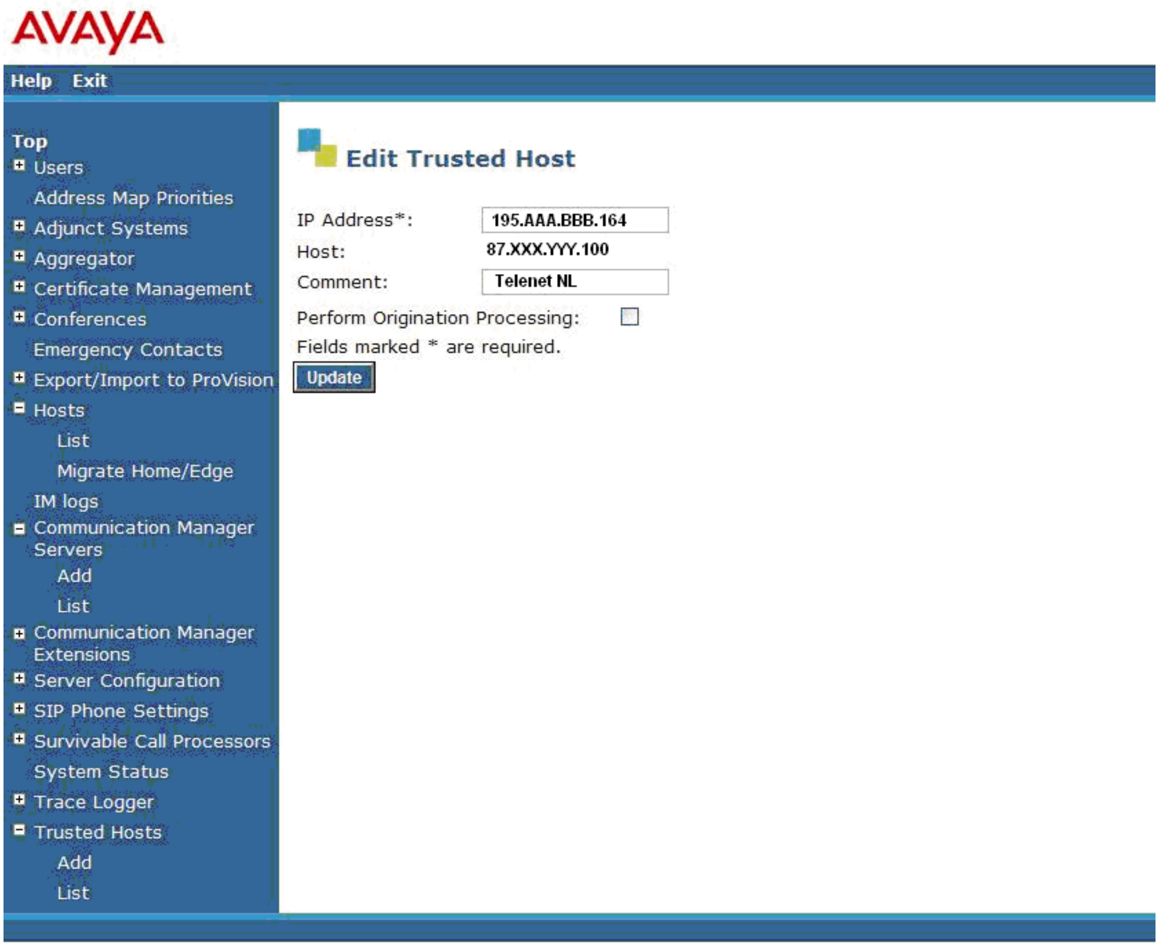

Figure 47: Edit Trusted Hosts Map

4.5. Configuration for SIP Telephones

This section provides basic instructions for completing the administration to use Avaya 9600 Series SIP telephones with the described configuration. Additional features, such as the use of mnemonic addressing, are beyond the scope of these Application Notes.

As noted previously, SIP telephones are optional; it is not necessary to have SIP telephones to use SIP trunking to the Telenet SIP Trunking service.

The steps below are repeated for each SIP telephone provisioned.

4.5.1. Add a SIP User

Add a SIP User to the SES as follows:

- In SES **SIP Server Management**, expand the **Users** link in the left side blue navigation bar and click on the **Add** link.
- In the **Add User** page **(Figure 48)** enter the extension number for the SIP telephone in the **Primary Handle** field.
- Enter a user password in the **Password** and **Confirm Password** fields. This password will be used when logging into the user's SIP telephone.
- In the **Host** field, select the SES server hosting the domain for this user.
- Enter the **First Name** and **Last Name** of the user.
- Select the **Add Communication Manager Extension** checkbox. This associates a Communication Manager extension with this SIP User. Calls from this user will be provided features and routing via Communication Manager.
- Click the **Add** button. This will cause a confirmation screen to appear.
- Click **Continue** on the confirmation screen.

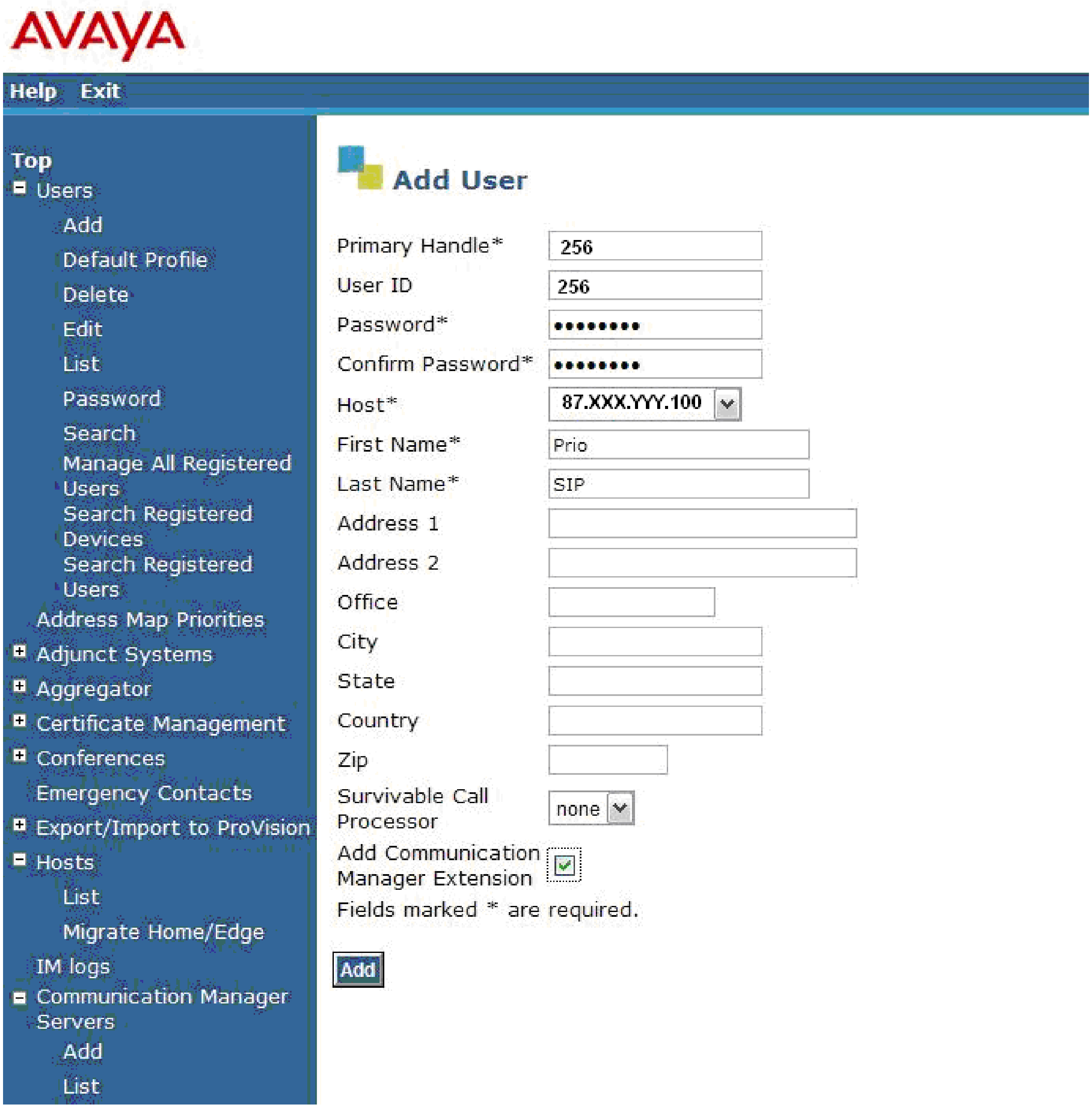

Figure 48: Add User

4.5.2. Specify Corresponding Communication Manager Extension

The SIP phone handle must now be associated with the corresponding extension on Communication Manager.

If the **Add Media Server Extension** checkbox was not selected, complete this step:

- In the **Add Communication Manager Extension** page, enter the extension number (e.g., "*256*") corresponding to the extension previously configured on Communication Manager **Figure 26**. The Communication Manager Extension and the SIP Primary Handle are usually the same, but are not required to be.
- Select the **Communication Manager Server** (e.g. *"87.XXX.YYY.100"*) interface assigned to this extension.
- Click the **Add** button.
- After that choose the **"Assign"** link on the **"List Communication Manager Extensions"** page for assigning a CM extension to the user.

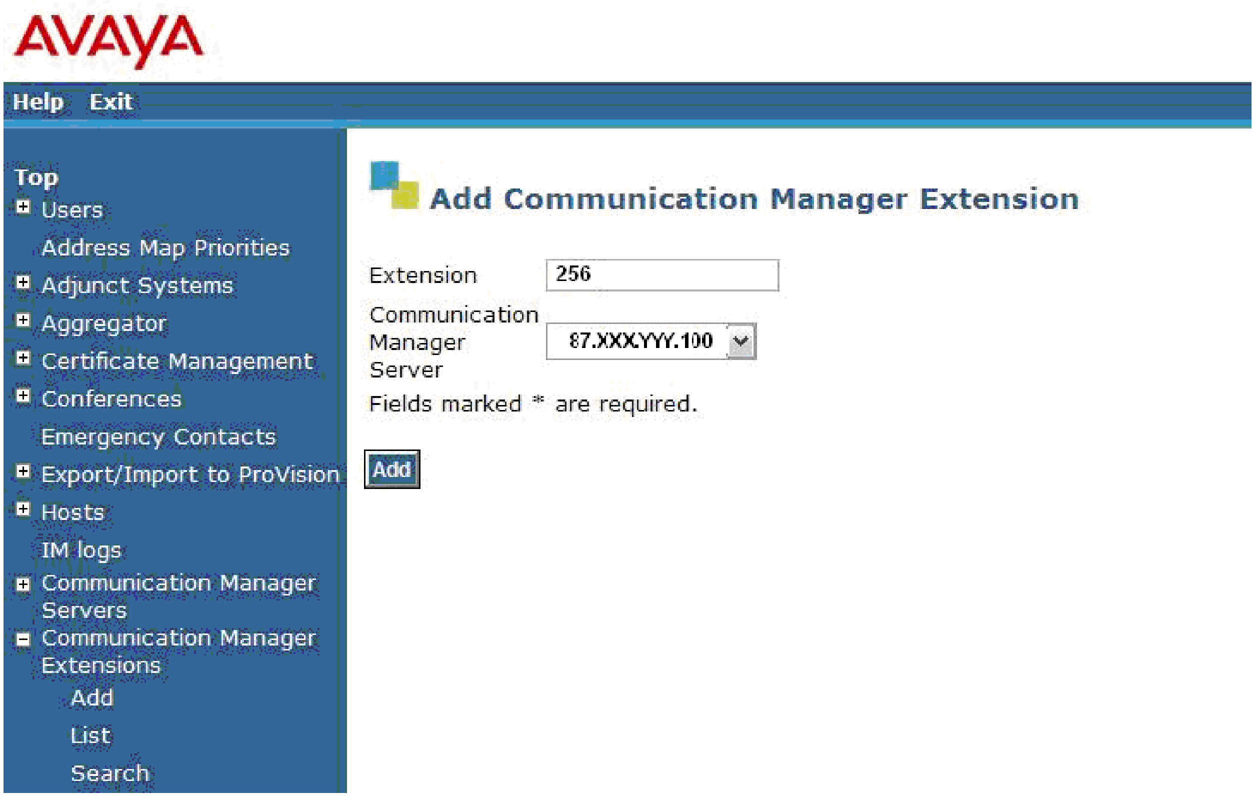

Figure 49: Add Media Server Extension

5. General Test Approach and Test Results

The following issues were encountered during testing:

- The Telenet SIP Trunking Service does not support shuffling.
- The Telenet SIP Trunking Service does not support T.38 fax.

6. Verification Steps

- Use the "status signaling-group 2" command from the SAT terminal to verify that the "Group State" has a value of "in-service", where "2" is the number of the SIP trunk attached to the Telenet SIP Trunking Service.

```
status signaling-group 2
                         STATUS SIGNALING GROUP

        Group ID: 2                              Active NCA-TSC Count: 0
      Group Type: sip                              Active CA-TSC Count: 0
  Signaling Type: facility associated signaling
     Group State: in-service
```

Figure 50: Signaling-Group Status

- Verify that local extensions can call to and receive calls from endpoints attached to the PSTN and mobile networks.
- Verify the calling party number is presented correctly at the called endpoint for both incoming and outgoing calls.
- Verify that unanswered incoming calls can be dialed via the call log of the called endpoint.

7. Conclusion

These Application Notes contain instructions for configuring Avaya Aura™ Communication Manager and Avaya Aura™ SIP Enablement Services to connect to the Telenet SIP Trunking service. All test cases passed with exceptions noted in **Section 5**.

8. Additional References

This section references documentation relevant to these Application Notes. The Avaya product documentation is available at http://support.avaya.com.

[1] *Administering Avaya Aura™ Communication Manager*, January 2009, Issue 5.0, Document Number 03-300509.
[2] *Avaya Aura™ Communication Manager Feature Description and Implementation,* May 2009, Issue 7, Document Number 555-245-205.
[3] *Avaya Extension to Cellular User Guide Avaya Aura™ Communication Manager,* April 2009, Issue 12, Document Number 210-100-700

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

[4] RFC 3261 - *SIP (Session Initiation Protocol)*, June 2002, Proposed Standard
[5] RFC 2833 - *RTP Payload for DTMF Digits, Telephony Tones and Telephony Signals*, May 2000, Proposed Standard
[6] RFC 3555 - *MIME Type Registration of RTP Payload Formats*, July 2003, IETF Standard
[7] *Installing, Administering, Maintaining and Troubleshooting Avaya* Aura™ *SIP Enablement Services,* May 2009, Issue 0.0, Document Number 03-600768

Appendix A: Sample SIP INVITE Messages

These traces were captured using a port which mirrored the connection between the Avaya SES Ethernet interface and the Telenet SIP Trunking Service IMS network.

Incoming call:

```
Session Initiation Protocol
    Request-Line: INVITE sip:32997254@87.XXX.YYY.100:5060 SIP/2.0
        Method: INVITE
        [Resent Packet: False]
    Message Header
        Via: SIP/2.0/UDP 195.AAA.BBB.164:5060;branch=z9hG4bK02B10716d18d542aa3d
            Transport: UDP
            Sent-by Address: 195.AAA.BBB.164
            Sent-by port: 5060
            Branch: z9hG4bK02B10716d18d542aa3d
        From: <sip:4969xxxx9497@195.AAA.BBB.164;pstn-params=808382808883>;tag=gK0225436a
            SIP from address: sip:4969xxxx9497@195.AAA.BBB.164
            SIP tag: gK0225436a
        To: <sip:32997254@87.XXX.YYY.100>
            SIP to address: sip:32997254@87.XXX.YYY.100
        Call-ID: 318940777_13634980@195.AAA.BBB.164
        CSeq: 1322 INVITE
            Sequence Number: 1322
            Method: INVITE
        Max-Forwards: 104
        Allow:
INVITE,ACK,CANCEL,BYE,REGISTER,REFER,INFO,SUBSCRIBE,NOTIFY,PRACK,UPDATE,OPTIONS,MESSAGE,PUBLISH
        Accept: application/sdp, application/isup, application/dtmf, application/dtmf-relay,
multipart/mixed
        Contact: <sip:4969xxxx9497@195.AAA.BBB.164:5060>
            Contact Binding: <sip:4969xxxx9497@195.AAA.BBB.164:5060>
                URI: <sip:4969xxxx9497@195.AAA.BBB.164:5060>
                    SIP contact address: sip:4969xxxx9497@195.AAA.BBB.164:5060
        P-Asserted-Identity: <sip:4969xxxx9497@195.AAA.BBB.164:5060>
        Supported: timer,100rel
        Session-Expires: 1800
        Min-SE: 90
        Content-Length:  268
        Content-Disposition: session; handling=optional
        Content-Type: application/sdp
    Message Body
        Session Description Protocol
            Session Description Protocol Version (v): 0
            Owner/Creator, Session Id (o): Sonus_UAC 26691 15858 IN IP4 195.AAA.BBB.164
                Owner Username: Sonus_UAC
                Session ID: 26691
                Session Version: 15858
                Owner Network Type: IN
                Owner Address Type: IP4
                Owner Address: 195.AAA.BBB.164
            Session Name (s): SIP Media Capabilities
            Connection Information (c): IN IP4 195.AAA.BBB.165
                Connection Network Type: IN
                Connection Address Type: IP4
                Connection Address: 195.AAA.BBB.165
            Time Description, active time (t): 0 0
                Session Start Time: 0
                Session Stop Time: 0
            Media Description, name and address (m): audio 12936 RTP/AVP 8 18 101
                Media Type: audio
                Media Port: 12936
                Media Proto: RTP/AVP
                Media Format: ITU-T G.711 PCMA
```

```
            Media Format: ITU-T G.729
            Media Format: DynamicRTP-Type-101
        Media Attribute (a): rtpmap:8 PCMA/8000
            Media Attribute Fieldname: rtpmap
            Media Format: 8
            MIME Type: PCMA
        Media Attribute (a): rtpmap:18 G729/8000
            Media Attribute Fieldname: rtpmap
            Media Format: 18
            MIME Type: G729
        Media Attribute (a): rtpmap:101 telephone-event/8000
            Media Attribute Fieldname: rtpmap
            Media Format: 101
            MIME Type: telephone-event
        Media Attribute (a): fmtp:101 0-15
            Media Attribute Fieldname: fmtp
            Media Format: 101 [telephone-event]
            Media format specific parameters: 0-15
        Media Attribute (a): sendrecv
        Media Attribute (a): maxptime:20
            Media Attribute Fieldname: maxptime
            Media Attribute Value: 20
```

Outgoing call:

```
Session Initiation Protocol
    Request-Line: INVITE sip:004969xxxx9497@195.AAA.BBB.164:5060;lr SIP/2.0
        Method: INVITE
        [Resent Packet: False]
    Message Header
        Call-ID: 80ba9718aff6de14444b18992900
        CSeq: 1 INVITE
            Sequence Number: 1
            Method: INVITE
        From: "IP-254-PH" <sip:32997254@ffm.com>;tag=80ba9718aff6de14344b18992900
            SIP Display info: "IP-254-PH"
            SIP from address: sip:32997254@ffm.com
            SIP tag: 80ba9718aff6de14344b18992900
        Record-Route: <sip:87.XXX.YYY.100:5060;lr>,<sip:87.XXX.YYY.101;lr;transport=tcp>
        To: "004969xxxx9497" <sip:004969xxxx9497@87.XXX.YYY.100>
            SIP Display info: "004969xxxx9497"
            SIP to address: sip:004969xxxx9497@87.XXX.YYY.100
        Via: SIP/2.0/UDP 87.XXX.YYY.100:5060;branch=z9hG4bK8383830303032626263cfa.0,SIP/2.0/TCP
87.XXX.YYY.101;psrrposn=2;received=87.XXX.YYY.101;branch=z9hG4bK80ba9718aff6de14544b18992900
            Transport: UDP
            Sent-by Address: 87.XXX.YYY.100
            Sent-by port: 5060
            Branch: z9hG4bK8383830303032626263cfa.0,SIP/2.0/TCP
        Content-Length: 169
        Content-Type: application/sdp
        Contact: "IP-254-PH" <sip:32997254@87.XXX.YYY.101>
            Contact Binding: "IP-254-PH" <sip:32997254@87.XXX.YYY.101>
                URI: "IP-254-PH" <sip:32997254@87.XXX.YYY.101>
                    SIP Display info: "IP-254-PH"
                    SIP contact address: sip:32997254@87.XXX.YYY.101
        Max-Forwards: 70
        User-Agent: Avaya CM/R015x.02.1.016.4
        Allow: INVITE,CANCEL,BYE,ACK,PRACK,SUBSCRIBE,NOTIFY,REFER,OPTIONS,INFO,PUBLISH
        Supported: timer,replaces,join,histinfo,100rel
        Alert-Info: <cid:internal@invalid.unknown.domain>;avaya-cm-alert-type=internal
        Min-SE: 1800
        Session-Expires: 1800;refresher=uac
        P-Asserted-Identity: "IP-254-PH" <sip:32997254@ffm.com>
        History-Info: <sip:004969xxxx9497@87.XXX.YYY.100>;index=1,"004969xxxx9497"
<sip:004969xxxx9497@87.XXX.YYY.100>;index=1.1
    Message Body
        Session Description Protocol
            Session Description Protocol Version (v): 0
            Owner/Creator, Session Id (o): - 1 1 IN IP4 87.XXX.YYY.101
                Owner Username: -
                Session ID: 1
                Session Version: 1
                Owner Network Type: IN
                Owner Address Type: IP4
                Owner Address: 87.XXX.YYY.101
            Session Name (s): -
            Connection Information (c): IN IP4 87.XXX.YYY.102
                Connection Network Type: IN
                Connection Address Type: IP4
                Connection Address: 87.XXX.YYY.102
            Bandwidth Information (b): AS:64
                Bandwidth Modifier: AS [Application Specific (RTP session bandwidth)]
                Bandwidth Value: 64 kb/s
            Time Description, active time (t): 0 0
                Session Start Time: 0
                Session Stop Time: 0
            Media Description, name and address (m): audio 2052 RTP/AVP 18 101
                Media Type: audio
                Media Port: 2052
                Media Proto: RTP/AVP
                Media Format: ITU-T G.729
```

```
            Media Format: DynamicRTP-Type-101
        Media Attribute (a): rtpmap:18 G729/8000
            Media Attribute Fieldname: rtpmap
            Media Format: 18
            MIME Type: G729
        Media Attribute (a): rtpmap:101 telephone-event/8000
            Media Attribute Fieldname: rtpmap
            Media Format: 101
            MIME Type: telephone-event
```

APPENDIX B: Specifying Pattern Strings in Address Maps

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

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

In the pattern matching string used in the Avaya Aura™ 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.