Avaya Solution & Interoperability Test Lab

Application Notes for Configuring SIP Trunking between the Belgacom VoIP Network and an Avaya Aura™ IP Telephony Solution – Issue 1.0

Abstract

These Application Notes describe the steps to configure trunking using the Session Initiation Protocol (SIP) between the Belgacom VoIP Access SIP Service and an Avaya IP telephony solution. The Avaya solution consists of Avaya Aura™ SIP Enablement Services, Avaya Aura™ Communication Manager, and various Avaya 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 procedure for configuring SIP trunking between the Belgacom VoIP Access SIP trunking network and Avaya Aura^{TM} SIP Enablement Services. The Belgacom VoIP Access service allows customers to connect their Communication Manager to the PSTN via an IP network.

SIP is a standards-based communications approach designed to provide a common framework to support multimedia communication. RFC 3261 [3] 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 Belgacom. SIP manages the establishment and termination of connections and the transfer of related information such as the desired codec, calling party identity, etc.

Note that FAX transmission was done without T.38 using in-band transmission via the G.711A codec.

1.1. Interoperability Compliance Testing

The following features were tested:

- Incoming & outgoing basic calls, including busy, no answer, calling party hang-up, called party hang-up.
- Outbound calls to domestic and international PSTN and GSM national and international endpoints.
- Codec support and priority selection.
- DTMF tone generation and recognition using RFC 2833 [4].
- Calling Party Number and Called Party Number presentation and restriction for incoming and outgoing calls.
- Call forwarding unrestricted / busy /no answer to local extension, PSTN, and GSM endpoints.
- Call forwarding to busy endpoint.
- Supervised Call Transfer / Blind Call Transfer to local extension, PSTN, and GSM endpoints.
- Conference Call with local and PSTN endpoints.
- Fax Send / Receive, using the G.711 codec.
- Simultaneous Calls.
- Long Calls.
- Calls with both ends muted.

1.2. Support

Support is available at:
http://www.belgacom.be/private/hbsres/jsp/dynamic/homepage.jsp

Prior registration is required, which can be done at:
http://www.belgacom.be/private/en/jsp/dynamic/productCategory.jsp?dcrName=hbsres_cockpit

2. Reference Configuration

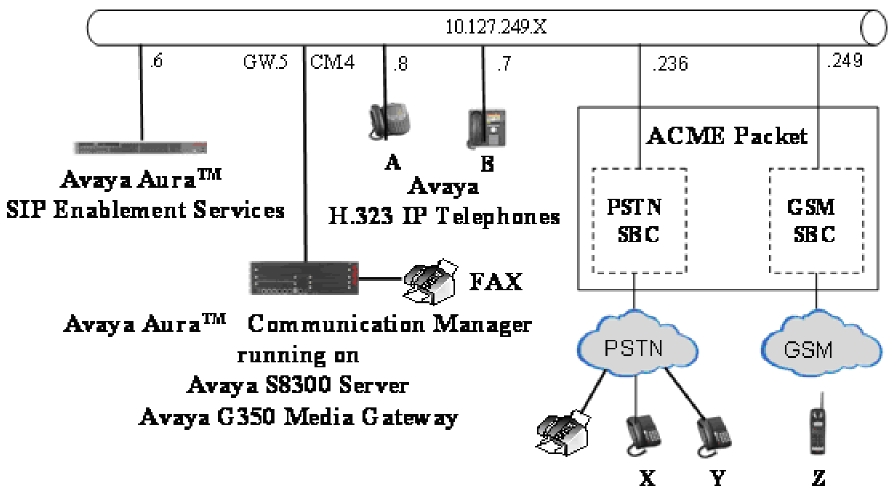

Figure 1: Reference Configuration

In the above diagram, Avaya IP Telephones and a FAX machine are attached to an Avaya S8300 Server running Communication Manager and an Avaya G350 Media Gateway. The Avaya Aura™ SIP Enablement Services server provides the interface to the Belgacom SIP trunk.

The FAX machine and each of the Avaya IP Telephones registered with Communication Manager is assigned a DID number by Belgacom.

Communication Manager and the Belgacom SIP network are configured to support direct IP connections, thus avoiding the necessity to route voice streams thorough the Avaya G350 Media Gateway. Communication Manager and the Belgacom SIP network are not configured to support T.38 FAX transmission, but instead use in-band FAX transmission using the G.711A codec.

Only abbreviated testing was done via the SIP trunk connected to the GSM network consisting of incoming and outgoing basic calls.

The telephone numbers used for testing are shown in the following table.

Endpoint	Ext	PSTN Number	Station Type
A	9682	02xxx 9682	4621
B	9681	02xxx 9681	9630
FAX	9689	02xxx 9689	FAX
X	--	02xxx 9040	PSTN
Y	--	02xxx 3025	PSTN
Z	--	04xxx 1618	Local GSM

Table 1: Extensions Used for Testing

3. Equipment and Software Validated

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

Item	Version
Avaya S8300 Server running Avaya Aura™ Communication Manager	R015x.02.1.016.4 Update 02.1.016.4-17774
Avaya G350 Media Gateway	30.10.4
Avaya Aura™ SIP Enablement Services	SES-5.2.1.0-016.4
Avaya 4621 IP Telephone	2.9.1 (H.323)
Avaya 9630 IP Telephone	3.1 (H.323)
ACME Packet SD4250 SBC	6.1.0 M3P1

Table 2: Equipment and Software Validated

4. Configure Communication Manager

The Communication Manager configuration was performed using the System Access Terminal (SAT).

Belgacom does not use T.38 for FAX transmission. FAX traffic is sent in-band. For this reason, separate trunks have been allocated for PSTN voice and FAX calls, with individual network regions, to enable the network regions to be assigned to different codec sets. For the configuration shown in these application notes, each of these network regions was assigned to the same codec set, but this can be changed if desired.

A separate network region has also been allocated for the trunk to the GSM network.

4.1. Verify system-parameters customer-options

Use the **display system-parameters customer-options** command to verify that Communication Manager is licensed to meet the minimum requirements to interoperate with the Belgacom SIP network. 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 Concurrently Registered IP Stations (Page 2)	This parameter must be large enough to support the number of IP stations to be attached.
Maximum Administered SIP Trunks (Page 2)	This parameter must be large enough to support the number of SIP trunks to be attached.
Enhanced EC500? (Page 4)	This parameter must be set to “y”.
IP Trunks? (Page 4)	This parameter must be set to “y”.
ISDN-PRI? (Page 4)	This parameter must be set to “y”.

Table 3: System-Parameters Customer-Options Parameters

```
display system-parameters customer-options                      Page   2 of  11
                                OPTIONAL FEATURES

IP PORT CAPACITIES                                             USED
                     Maximum Administered H.323 Trunks: 450   0
           Maximum Concurrently Registered IP Stations: 450   2
             Maximum Administered Remote Office Trunks: 450   0
Maximum Concurrently Registered Remote Office Stations: 450   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: 450   8
  Maximum Administered Ad-hoc Video Conferencing Ports: 0     0
   Maximum Number of DS1 Boards with Echo Cancellation: 80    0
                             Maximum TN2501 VAL Boards: 0     0
                     Maximum Media Gateway VAL Sources: 50    1
           Maximum TN2602 Boards with 80 VoIP Channels: 0     0
          Maximum TN2602 Boards with 320 VoIP Channels: 0     0
   Maximum Number of Expanded Meet-me Conference Ports: 0     0
```

Figure 2: System-Parameters Customers-Options Form, Page 2

```
display system-parameters customer-options                      Page   4 of  11
                                OPTIONAL FEATURES

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

           IP Attendant Consoles? n
```

Figure 3: System-Parameters Customers-Options Form, Page 4

4.2. 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 4: System-Parameters Features 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 (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 4: System-Parameters Features Form, Page 1

4.3. SIP Interfaces to SIP Enablement Services

Use the **change node-names ip** command to assign the name "ses" to the IP address of the SIP Enablement Services server.

```
change node-names ip                                            Page   1 of   2
                                  IP NODE NAMES
    Name              IP Address
default             0.0.0.0
procr               10.127.249.4
ses                 10.127.249.6
```

Figure 5: Node-Names Ip Form

4.3.1. SIP Interface for PSTN Voice Access

Use the **add signaling-group** command to allocate a signaling group for the SIP interface to SIP Enablement Services to be used for voice calls using the following parameters:

Parameter	Usage
Group Type	Enter "sip".
Near-end Node Name	Enter "procr" do designate the G350 processor as the near end node name.
Far-end Node Name	Enter "ses" to assign the SES server as the far end node name.
Near-end Listen Port	Specify an otherwise unused port to be used to listen for incoming voice traffic. Note that this listen port cannot be shared by other SIP signaling groups. This must be the same as the SIP Enablement Services port allocated in **Figure 47**.
Far-end Network Region	Enter the number of the network region which is to be used for voice traffic, as defined in **Figure 18**.
Direct IP-IP Audio Connections	Enter "y" to allow direct IP-IP endpoint connections (shuffling).

Table 5: PSTN Voice 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: 5062                 Far-end Listen Port: 5061
                                         Far-end Network Region: 2
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? y
Session Establishment Timer(min): 120                   IP Audio Hairpinning? y
         Enable Layer 3 Test? n                     Direct IP-IP Early Media? n
H.323 Station Outgoing Direct Media? n            Alternate Route Timer(sec): 10
```

Figure 6: PSTN Voice Signaling-Group Form

Use the **add trunk-group <n>** command, where <n> is an available trunk number, to create a trunk group to be used as an interface to SIP Enablement Services for voice calls. Use the parameters shown in the following table.

Parameter	Usage
Group Type (Page 1)	Enter "sip".
Group Name (Page 1)	Assign a name for identification purposes.
TAC (Page 1)	Enter the Trunk Access Code allocated in **Figure 15**.
Service Type (Page 1)	Enter "tie".
Signaling Group (Page 1)	Enter the number of the signaling group allocated in **Figure 6**.
Number of Members (Page 1)	Enter a number large enough to support the maximum number of anticipated simultaneous calls to be made via the SIP trunk.
Preferred Minimum Session Refresh Interval (Page 2)	Enter "900" seconds, as required for the Belgacom SIP trunk interface. This should be half of the Session Refresh Interval which is configured for the Belgacom VoIP Access SIP Service.

Table 6: PSTN Voice Trunk-Group Parameters

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

Group Number: 2                    Group Type: sip           CDR Reports: r
  Group Name: Voice                       COR: 1        TN: 1        TAC: *02
   Direction: two-way        Outgoing Display? n
 Dial Access? n                                   Night Service:
Queue Length: 0
Service Type: tie                    Auth Code? n

                                                    Signaling Group: 2
                                                  Number of Members: 10
```

Figure 7: PSTN Voice Trunk-Group Form, p.1

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

TRUNK PARAMETERS

     Unicode Name: yes

                                            Redirect On OPTIM Failure: 5000

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

Figure 8: PSTN Voice Trunk-Group Form, p.2

4.3.2. SIP Interface for PSTN FAX Access

Use the **add signaling-group** command to allocate a signaling group for the SIP interface to SIP Enablement Services to be used for Fax calls using the following parameters:

Parameter	Usage
Group Type	Enter “sip”.
Near-end Node Name	Enter “procr” do designate the G350 processor as the near end node name.
Far-end Node Name	Enter “ses” to assign the SES server as the far end node name.
Near-end Listen Port	Specify an otherwise unused port to be used to listen for incoming voice traffic. Note that this listen port cannot be shared by other SIP signaling groups. This must be the same as the SIP Enablement Services port allocated in **Figure 51.**
Far-end Network Region	Enter the number of the network region which is to be used for FAX traffic, as defined in **Figure 19.**
Direct IP-IP Audio Connections	Enter “y” to allow direct IP-IP endpoint connections (shuffling).

Table 7: PSTN FAX Signaling-Group Parameters

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

 Group Number: 3             Group Type: sip
                        Transport Method: tls
  IMS Enabled? n




   Near-end Node Name: procr                 Far-end Node Name: ses
 Near-end Listen Port: 5063                Far-end Listen Port: 5061
                                        Far-end Network Region: 3
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? y
Session Establishment Timer(min): 120                    IP Audio Hairpinning? n
         Enable Layer 3 Test? n                      Direct IP-IP Early Media? n
H.323 Station Outgoing Direct Media? n             Alternate Route Timer(sec): 10
```

Figure 9: PSTN FAX Signaling-Group Form

Use the **add trunk-group <n>** command, were <n> is an unused trunk number, to create a trunk group to be used as an interface to SIP Enablement Services for FAX calls. Use the parameters show in the following table.

Parameter	Usage
Group Type (Page 1)	Enter "sip".
Group Name (Page 1)	Assign a name for identification purposes.
TAC (Page 1)	Enter the Trunk Access Code allocated in **Figure 15**.
Service Type (Page 1)	Enter "tie".
Signaling Group (Page 1)	Enter the number of the signaling group allocated in **Figure 9**
Number of Members (Page 1)	Enter a number large enough to support the maximum number of anticipated simultaneous calls to be made via the SIP trunk.
Preferred Minimum Session Refresh Interval (Page 2)	Enter "900" seconds, as required for the Belgacom SIP trunk interface. This should be half of the Session Refresh Interval which is configured for the Belgacom VoIP Access SIP Service.

Table 8: PSTN FAX Trunk-Group Parameters

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

Group Number: 3                  Group Type: sip           CDR Reports: r
  Group Name: SIP FAX / Modem             COR: 1        TN: 1        TAC: *03
   Direction: two-way       Outgoing Display? n
 Dial Access? n                                   Night Service:
Queue Length: 0
Service Type: tie                  Auth Code? n

                                                       Signaling Group: 3
                                                     Number of Members: 4
```

Figure 10: PSTN FAX Trunk-Group Form, p.1

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

TRUNK PARAMETERS

     Unicode Name: yes

                                            Redirect On OPTIM Failure: 5000

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

Figure 11: PSTN FAX Trunk-Group Form, p.2

4.3.3. SIP Interface for GSM Access

Use the **add signaling-group** command to allocate a signaling group for the SIP interface to SIP Enablement Services to be used for GSM calls using the following parameters:

Parameter	Usage
Group Type	Enter "sip".
Near-end Node Name	Enter "procr" do designate the G350 processor as the near end node name.
Far-end Node Name	Enter "ses" to assign the SES server as the far end node name.
Near-end Listen Port	Specify an otherwise unused port as a placeholder. This port will not actually be used to receive incoming traffic, as incoming GSM traffic arrives via the same trunk as PSTN voice traffic.
Far-end Network Region	Enter the number of the network region which is to be used for GSM traffic, as defined in **Figure 20**.
Direct IP-IP Audio Connections	Enter "y" to allow direct IP-IP endpoint connections (shuffling).

Table 9: GSM Signaling-Group Parameters

```
change signaling-group 4                                        Page   1 of   1
                                SIGNALING GROUP

 Group Number: 4              Group Type: sip
                        Transport Method: tls
  IMS Enabled? n




   Near-end Node Name: procr                  Far-end Node Name: ses
 Near-end Listen Port: 5064                 Far-end Listen Port: 5061
                                         Far-end Network Region: 4
Far-end Domain: 10.127.249.249

                                              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? y
Session Establishment Timer(min): 3                      IP Audio Hairpinning? n
         Enable Layer 3 Test? n                        Direct IP-IP Early Media? n
H.323 Station Outgoing Direct Media? n              Alternate Route Timer(sec): 6
```

Figure 12: GSM Signaling-Group Form

Use the **add trunk-group <n>** command, were <n> is an available trunk number, to allocate a trunk group to be used as an interface to SIP Enablement Services to be used for GSM calls Use the parameters show in the following table.

Parameter	Usage
Group Type (Page 1)	Enter “sip”.
Group Name (Page 1)	Assign a name for identification purposes.
TAC (Page 1)	Enter the Trunk Access Code allocated in **Figure 15**.
Service Type (Page 1)	Enter “tie”.
Signaling Group (Page 1)	Enter the number of the signaling group allocated in **Figure 12**.
Number of Members (Page 1)	Enter a number large enough to support the maximum number of anticipated simultaneous calls to be made via the SIP trunk.
Preferred Minimum Session Refresh Interval (Page 2)	Enter “600” seconds, as required for the GSM SIP trunk interface. This should be half of the Session Refresh Interval which is configured for the Belgacom VoIP Access SIP Service.

Table 10: GSM Trunk-Group Parameters

```
add trunk-group 4                                              Page   1 of  21
                                TRUNK GROUP

Group Number: 4                    Group Type: sip            CDR Reports: y
  Group Name: GSM OUT                     COR: 1        TN: 1        TAC: *00
   Direction: two-way        Outgoing Display? n
 Dial Access? n                                  Night Service:
Queue Length: 0
Service Type: tie                   Auth Code? n

                                                   Signaling Group: 4
                                                 Number of Members: 2
```

Figure 13: GSM Trunk-Group Form, p.1

```
add change trunk-group 4                                              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): 600
```

Figure 14: GSM Trunk-Group Form, p.2

4.4. Dial Plan

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

Parameter	Usage
Dialed string: "0"	Use a "0" for the ARS Facility Access Code (FAC) to access external telephone numbers.
Dialed string: "9"	4-digit numbers starting with "9" are for local extensions.
Dialed string: "*"	The dialed string starting with "*" are for Trunk Access Codes.

Table 11: 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
     0            1    fac
     9            4    ext
     *            3    dac
     #            3    dac
```

Figure 15: Dialplan Analysis Form

Use the **change feature-access-codes** command to assign dialed digit strings to feature access codes. Use a “0” to use Automatic Route Selection (ARS) to route PSTN calls over a SIP trunk.

```
change feature-access-codes                                     Page   1 of   9
                               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: *11
                       Answer Back Access Code:
                         Attendant Access Code:
      Auto Alternate Routing (AAR) Access Code: *99
    Auto Route Selection (ARS) - Access Code 1: 0     Access Code 2:
                 Automatic Callback Activation: *05    Deactivation: #05
Call Forwarding Activation Busy/DA: *18    All: *07    Deactivation: #07
   Call Forwarding Enhanced Status:        Act:        Deactivation:
                         Call Park Access Code: *04
                       Call Pickup Access Code: *06
CAS Remote Hold/Answer Hold-Unhold Access Code:
                  CDR Account Code Access Code:
                        Change COR Access Code:
                   Change Coverage Access Code:
            Conditional Call Extend Activation:        Deactivation:
                   Contact Closure   Open Code:          Close Code:
```

Figure 16: Feature-Access-Codes Form

4.5. Configure Codec Sets

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

Parameter	Usage
Audio Codec (Page 1)	Enter “G.711A” or “G.729A” as the codec to be used to communication with the Belgacom SIP trunk.

Table 12: IP-Codec-Set Parameters

```
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.711A            n         2        20
 2: G.729A            n         2        20
 3:
 4:
 5:
 6:
 7:


     Media Encryption
 1: none
 2:
 3:
```

Figure 17: IP-Codec-Set Form – Page 1

4.6. Configure IP Network Regions

Use the **change ip-network-region** command to designate a network region for voice calls to be used for the Belgacom SIP trunk using the parameters shown in the following table.

Parameter	Usage
Region	Enter an unassigned network region number (this must be the same number as was assigned as "Far-end Network Region" in **Figure 6**.
Location	Enter "1".
Authoritative Domain	Enter the domain name assigned in **Figure 32**.
Name	Enter a name to identify the region.
Codec Set	Enter the number of the codec set defined in **Figure 17.**

Table 13: IP-Network-Region Parameters

```
change ip-network-region 2                                      Page   1 of  19
                               IP NETWORK REGION
  Region: 2
Location: 1       Authoritative Domain: voip.belgacom.be
    Name: BGC Voice
MEDIA PARAMETERS                Intra-region IP-IP Direct Audio: yes
      Codec Set: 2              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 18: IP-Network-Region Form – Page 1

Use the **change ip-network-region** command to designate a network region for FAX calls to be used for the Belgacom SIP trunk using the parameters shown in the following table.

Parameter	Usage
Region	Enter an unassigned network region number (this must be the same number as was assigned as "Far-end Network Region" in **Figure 9**.
Authoritative Domain	Enter the domain name assigned in **Figure 32**.
Name	Enter a name to identify the region.
Codec Set	Enter the number of the codec set defined in **Figure 17**

Table 14: IP-Network-Region Parameters

```
change ip-network-region 3                                      Page   1 of  19
                               IP NETWORK REGION
  Region: 3
Location: 1       Authoritative Domain: voip.belgacom.be
    Name: BGC FAX / Modem
MEDIA PARAMETERS                Intra-region IP-IP Direct Audio: yes
      Codec Set: 2              Inter-region IP-IP Direct Audio: yes
   UDP Port Min: 2048                      IP Audio Hairpinning? y
   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 19: IP-Network-Region Form – Page 1

Use the **change ip-network-region** command to designate a network region for GSM calls to be used for the Belgacom SIP trunk using the parameters shown in the following table.

Parameter	Usage
Region	Enter an unassigned network region number (this must be the same number as was assigned as "Far-end Network Region" in **Figure 12**.
Authoritative Domain	Enter the domain name assigned in **Figure 32**.
Name	Enter a name to identify the region.
Codec Set	Enter the number of the codec set defined in **Figure 17.**

Table 15: IP-Network-Region Parameters

```
change ip-network-region 4                                      Page   1 of  19
                               IP NETWORK REGION
  Region: 4
Location: 1       Authoritative Domain: voip.belgacom.be
    Name: GSM
MEDIA PARAMETERS                Intra-region IP-IP Direct Audio: yes
      Codec Set: 2              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 – Page 1

4.7. Configure Class of Restriction

Use the **change cor 1** command to configure the Class of Restriction to be used for voice calls.

Parameter	Usage
COR Description	Enter a descriptive name to identify this COR.
FRL	Enter "1", which is used to route outgoing voice calls (see **Figure 26**).
Calling Party Restriction	Set this parameter to "none".

Table 16: Class of Restriction for Voice Stations

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

               COR Number: 1
          COR Description: Voice Calls

                      FRL: 1                                APLT? y
  Can Be Service Observed? n           Calling Party Restriction: none
Can Be A Service Observer? n            Called Party Restriction: none
        Time of Day Chart: 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 21: Class of Restriction for Voice Stations

Use the **change cor 2** command to configure the Class of Restriction to be used by FAX calls.

Parameter	Usage
COR Description	Enter a descriptive name to identify this COR.
FRL	Enter "2", which is used to route outgoing FAX calls (see **Figure 26**).
Calling Party Restriction	Set this parameter to "none".

Table 17: Class of Restriction for Voice Stations

```
change cor 2                                                Page   1 of  23
                              CLASS OF RESTRICTION

               COR Number: 2
          COR Description: FAX / Modem

                      FRL: 2                                APLT? y
  Can Be Service Observed? n           Calling Party Restriction: none
Can Be A Service Observer? n            Called Party Restriction: none
        Time of Day Chart: 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 22: Class of Restriction for FAX Stations

4.8. Configure Stations

4.8.1. Configure Voice Stations

Create a station for each of the extensions shown in **Table 1**, using the **add station** command using the parameters shown in the following table.

Parameter	Usage
Type	Enter the station type designation.
Name	Enter a descriptive name to identify the station.
Security Code	Enter a security code to be used by the stations.
COR	Enter the COR for voice stations which is defined in **Figure 21.**

Table 18: Voice Station Parameters

```
add station 9681                                              Page   1 of   5
                                     STATION

Extension: 9681                          Lock Messages? n               BCC: 0
     Type: 9630                          Security Code: 1234             TN: 1
     Port: IP                          Coverage Path 1:                 COR: 1
     Name: ext 9681                    Coverage Path 2:                 COS: 1
                                       Hunt-to Station:
STATION OPTIONS
                                           Time of Day Lock Table:
              Loss Group: 19         Personalized Ringing Pattern: 7
                                                 Message Lamp Ext: 9681
            Speakerphone: 2-way               Mute Button Enabled? y
        Display Language: unicode                  Button Modules: 0
 Survivable GK Node Name:
          Survivable COR: internal               Media Complex Ext:
   Survivable Trunk Dest? y                           IP SoftPhone? n




                                              Customizable Labels? y
```

Figure 23: Voice Station Form

4.8.2. Configure FAX Devices

Parameter	Usage
Type	Enter the station type for analog interface.
Port	Enter the analog port number for the FAX.
Name	Enter a descriptive name to identify the station.
COR	Enter the COR for fax stations which is defined in **Figure 22.**

Table 19: FAX Station Parameters

```
add station 9689                                              Page   1 of   4
                                     STATION

Extension: 9689                          Lock Messages? n               BCC: 0
     Type: 2500                          Security Code:                  TN: 1
     Port: 001V702                     Coverage Path 1:                 COR: 2
     Name: FAX                         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
                                              Remote Office Phone? n
```

Figure 24: FAX Station Form

4.9. Outgoing Call Routing

Use the **change ars analysis** command to designate that all numbers beginning with “0”, be routed to the PSTN via route pattern “1”.   Numbers beginning with “047-9” should be routed to the GSM network via routing pattern “2”.

```
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
    0                     9    9     1         pubu         n
    00                    8    28    1         pubu         n
    047                   10   10    2         pubu         n
    048                   10   10    2         pubu         n
    049                   10   10    2         pubu         n
    1                     4    4     1         pubu         n
    10                    3    3     1         pubu         n
    11                    3    3     1         pubu         n
```

Figure 25: ARS Digit Analysis Form

Use the **change route-pattern** command to designate that calls be routed using the FRL for trunk selection. Calls from voice stations use COR 1, as shown in **Figure 23**. COR 1 is shown in **Figure 21**. COR 1 designates FRL 1, thus routing voice calls to trunk group 2, as shown in the following form.

Calls from FAX stations use COR 2, as shown in **Figure 24**. COR 2 shown in **Figure 22**, designates FRL 2, thus routing voice calls to trunk group 3, as shown in the following form.

```
change route-pattern 1                                         Page   1 of   3
                    Pattern Number: 1   Pattern Name: BGC PSTN
                             SCCAN? n     Secure SIP? n
    Grp FRL NPA Pfx Hop Toll No.  Inserted                            DCS/ IXC
    No          Mrk Lmt List Del  Digits                              QSIG
                             Dgts                                     Intw
 1: 3    2                                                             n   user
 2: 2    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                                       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 26: PSTN Route Pattern Form

Use the **change route-pattern** command to designate that GSM calls be routed to trunk group 4.

```
change route-pattern 2                                          Page   1 of   3
                    Pattern Number: 2   Pattern Name: GSM
                             SCCAN? n     Secure SIP? n
    Grp FRL NPA Pfx Hop Toll No.  Inserted                             DCS/ IXC
    No          Mrk Lmt List Del  Digits                               QSIG
                             Dgts                                      Intw
 1: 4    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 27: GSM Route Pattern Form

Use the **change public-unknown-numbering** command to designate that the local FAX and the three locally attached Avaya IP Telephones each be assigned public telephone numbers to be used as the Calling Party Number for outgoing calls, as shown in **Figure 1**.

```
change public-unknown-numbering 0                               Page   1 of   2
                      NUMBERING - PUBLIC/UNKNOWN FORMAT
                                         Total
Ext Ext            Trk       CPN           CPN
Len Code           Grp(s)    Prefix        Len
                                                   Total Administered: 3
 4  9              2         02795         9          Maximum Entries: 240
 4  9              3         02795         9
```

Figure 28: Public-Unknown-Numbering Form

4.10. Incoming Call Routing

Use the **change inc-call-handling-trmt trunk-group** command to map calls arriving on trunk group "2" with a public numbering format to the extensions of the locally attached Avaya IP Telephones in **Figure 1**.

```
change inc-call-handling-trmt trunk-group 2                     Page   1 of   3
                       INCOMING CALL HANDLING TREATMENT
 Service/      Number   Number      Del Insert
 Feature         Len      Digits
 tie              9  0                5
```

Figure 29: Incoming Call Handling Treatment Form for Voice Calls

Use the **change inc-call-handling-trmt trunk-group** command to map calls arriving on trunk group “3” with a public numbering format to the extensions of the local FAX devices shown in **Figure 1**.

```
change inc-call-handling-trmt trunk-group 3                     Page   1 of   3
                       INCOMING CALL HANDLING TREATMENT
 Service/      Number   Number      Del Insert
 Feature        Len      Digits
 tie              9  0               5
```

Figure 30: Incoming Call Handling Treatment Form for FAX Calls

5. Configure Avaya Aura™ SIP Enablement Services

Configure SIP Enablement Services by entering "<SES IP Address>/admin" in a web browser. After entering the administrator name and password, the following screen content is displayed:

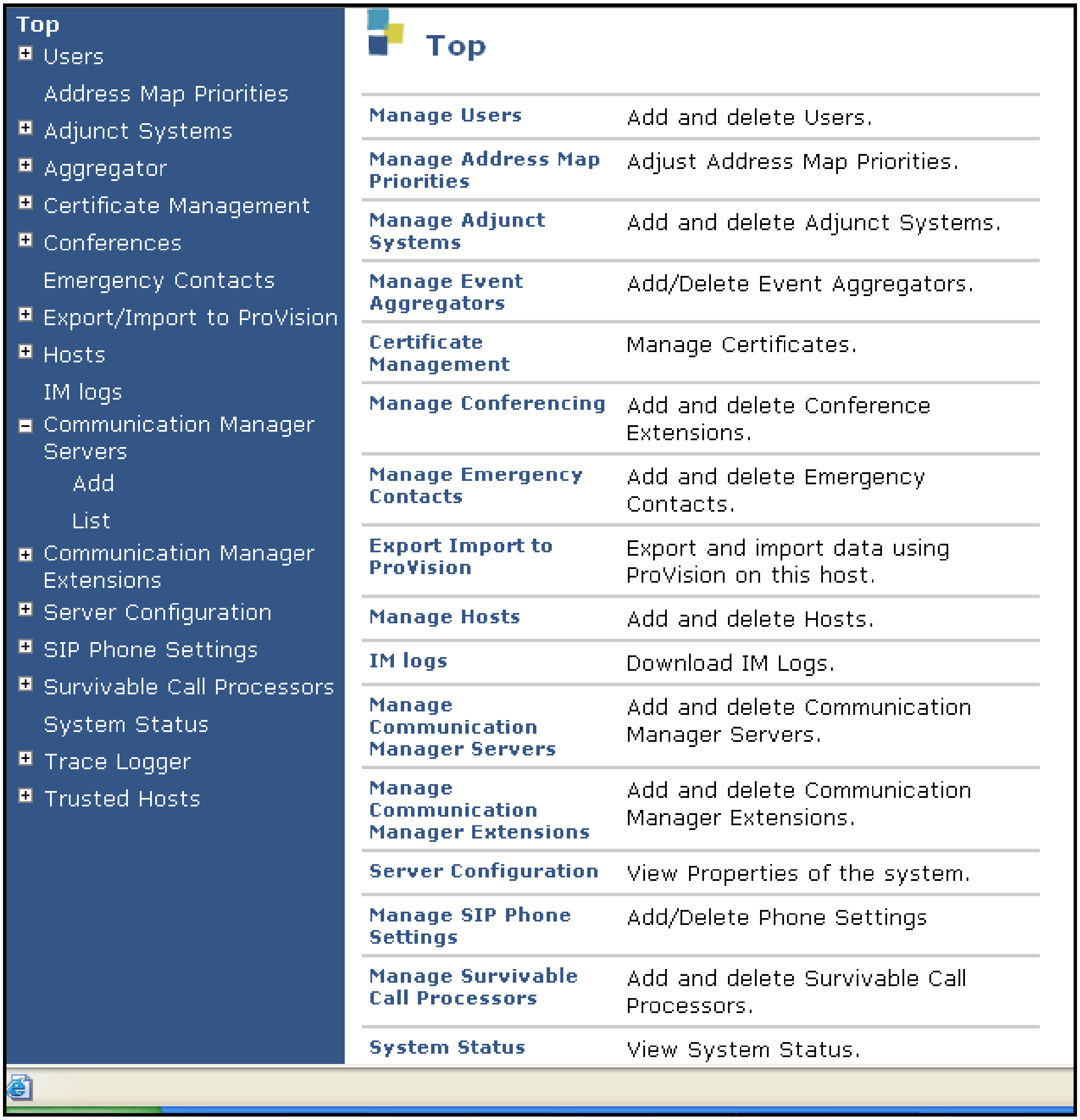

Figure 31: SIP Enablement Services "Top" Configuration Screen

5.1. Server Configuration

Select "System Properties" from the "Server Configuration" menu from the left pane of the screen. Enter values in this screen as shown in the following table:

Parameter	Usage
SIP Domain	Enter SIP domain used by the Belgacom network.
License Host	Enter the IP address of the license host, in this case the IP address of the SES server.

Table 20: Parameters for System Properties

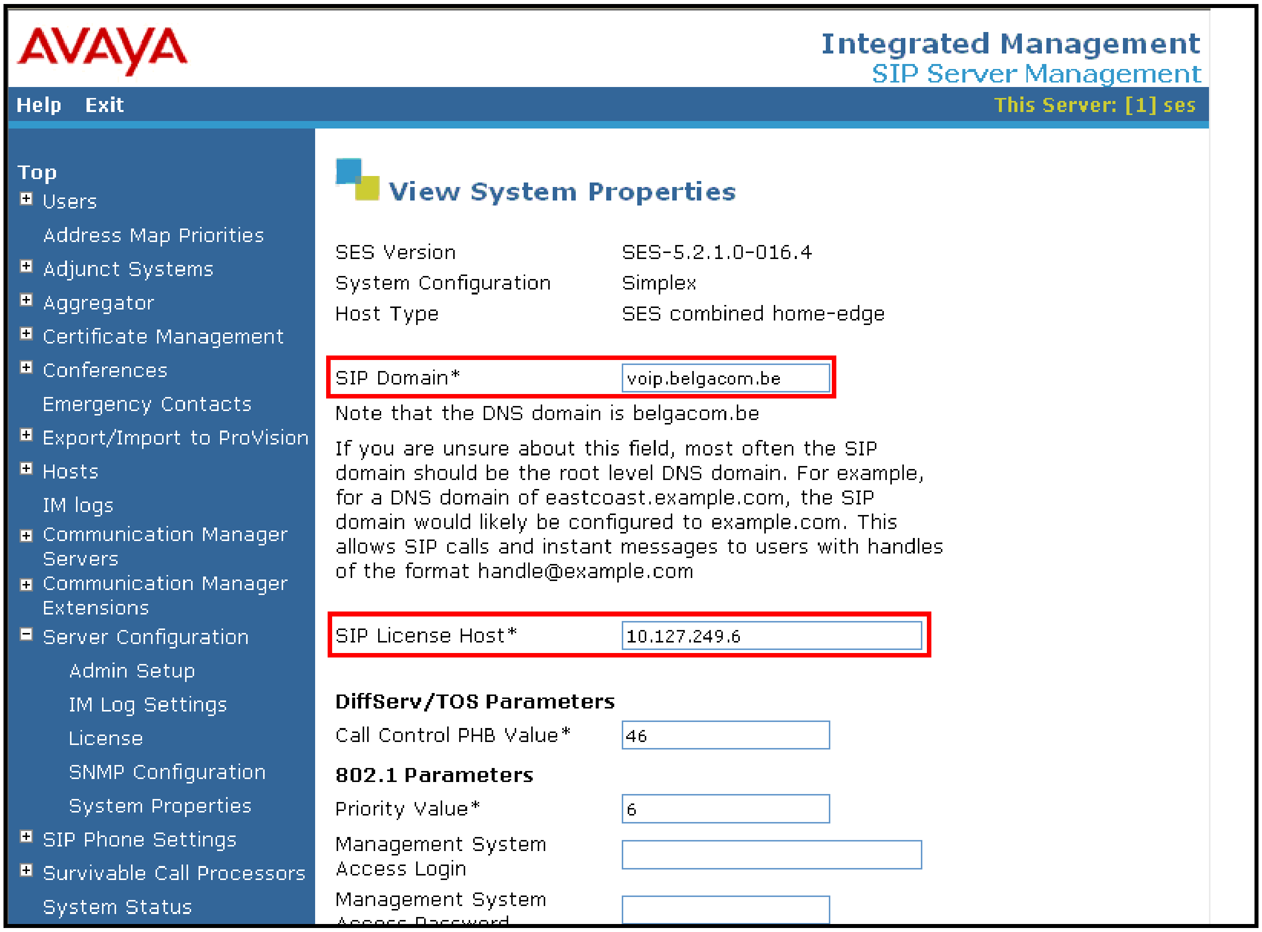

Figure 32: System Properties Screen

5.2. Add Hosts

Select "Hosts" → "Add Host" from the left pane of the top level screen shown in **Figure 31**. Enter values in this screen as shown in the following table, accepting the default values for those parameters which are not listed.

Parameter	Usage
Host IP Address	Enter the IP address of the SES server.
Profile Service Password	Enter the password which was entered from the initial setup script when SES was installed.

Table 21: "Add Host" Parameters

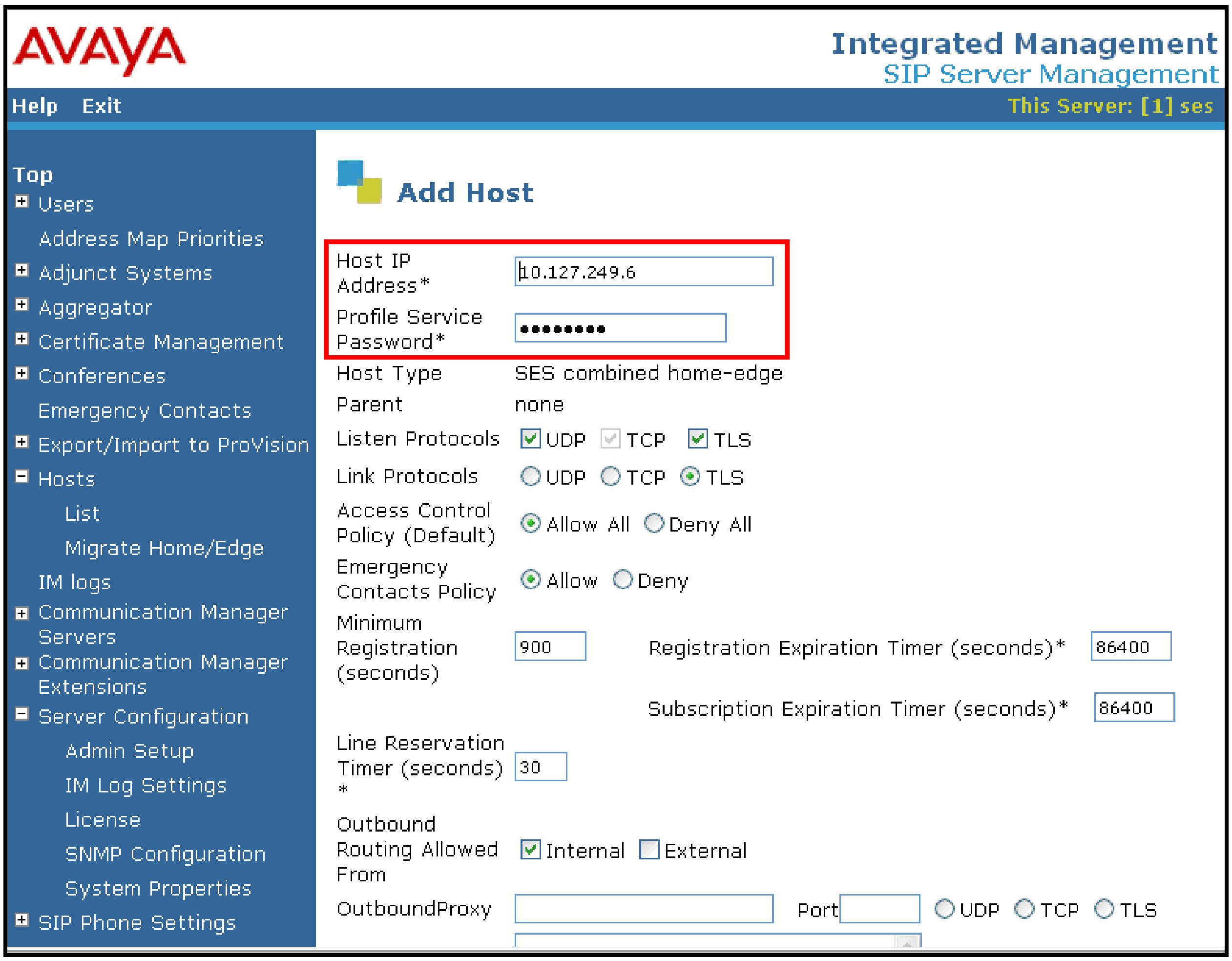

Figure 33: Add Host Screen

Select the "Map" menu point from the "List Hosts" screen.

Figure 34: List Hosts Screen

Click "Add Map In New Group".

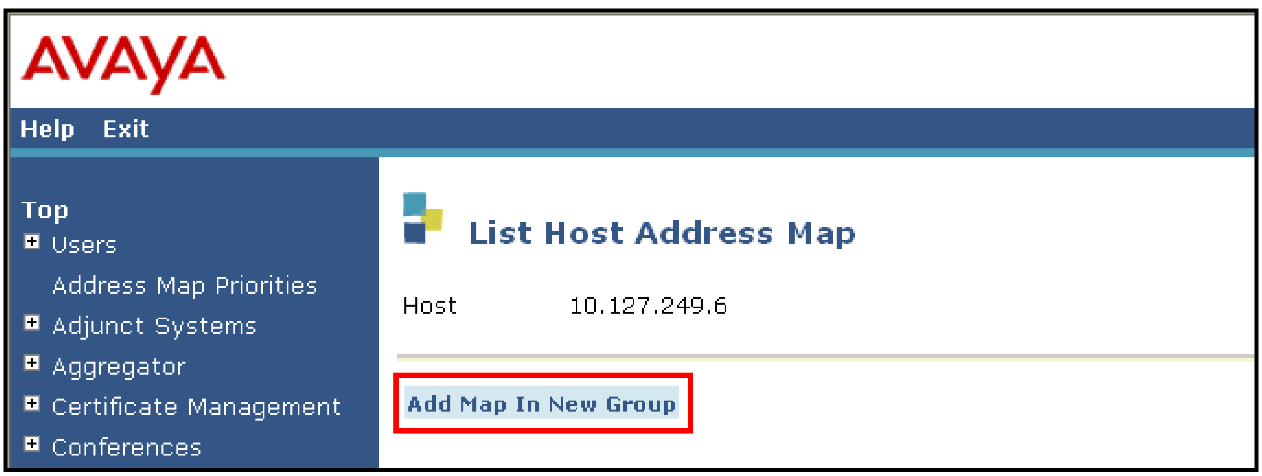

Figure 35: List Host Address Map Screen

Create a map to route outgoing calls to the GSM network. Enter a name to identify the map and the Pattern "^sip:04.*@" to match calls to destinations beginning with "04", the prefix for GSM calls.

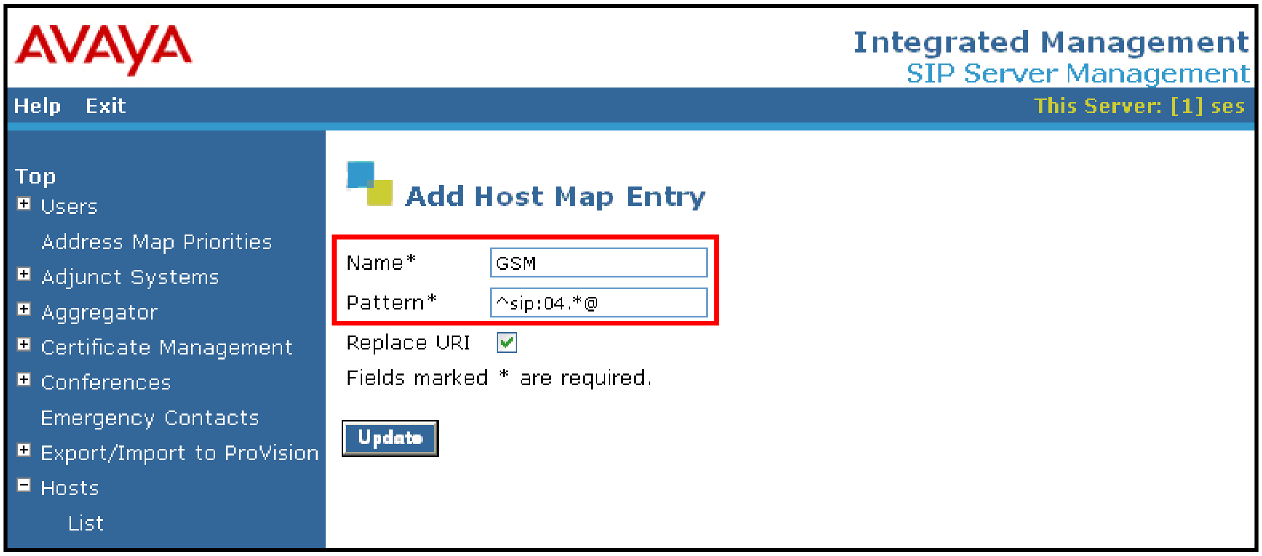

Figure 36: Add GSM Host Map Screen

After the host address maps have been added, select “Add Another Contact” (not shown) and enter the following contact: “sip:$(user)@<GSM SBC IP Address>:5060;transport=udp”. The IP address of the Voice SBC is shown in **Figure 1**.

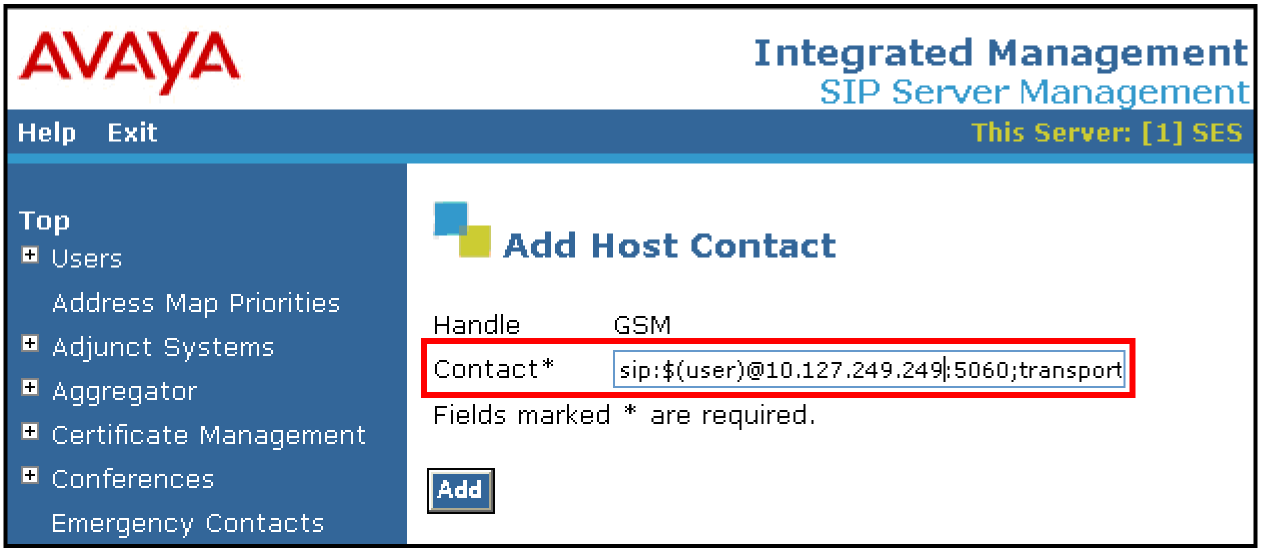

Figure 37: Add Host GSM Contact Screen

From “List Host Address Map” screen, click the “Add Map In New Group”.

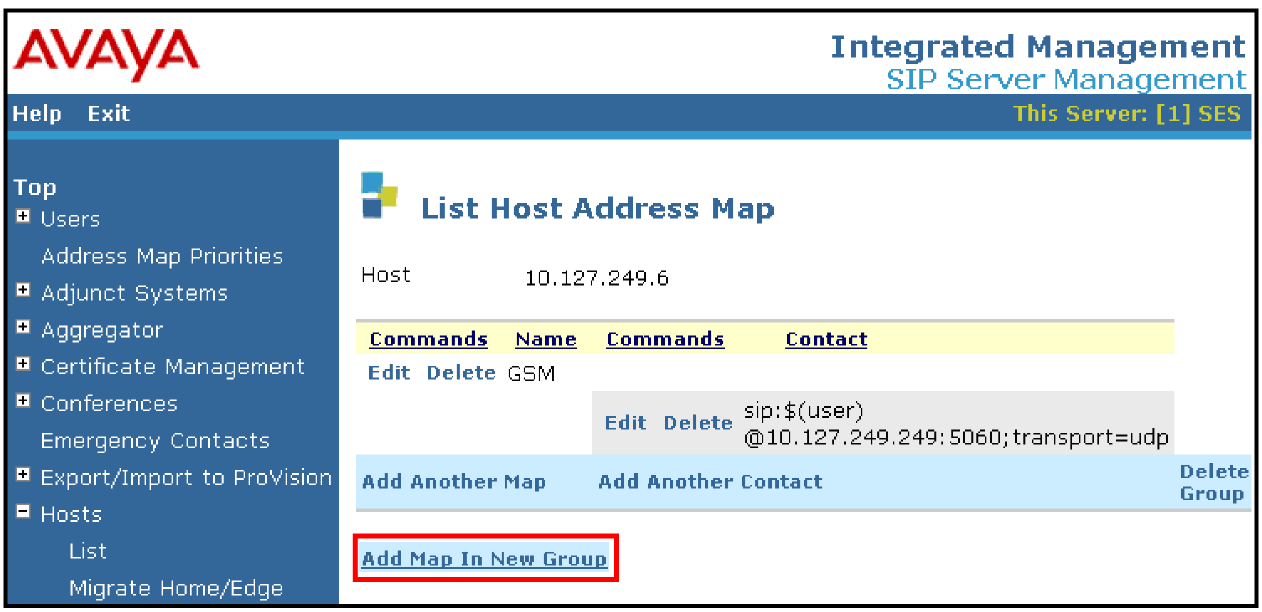

Figure 38: GSM Host Address Map Added Screen

Outbound calls are routed using Host Address Maps to select the destination host. Host Address Maps and Communication Manager Server Address Maps must be unique. This necessitates that the Host Address Maps be defined such that none of them conflict with the Communication Manager Server Address Map. Simply defining a host map with the value of "^sip:0*" to route PSTN calls to the Belgacom SIP network would cause incoming PSTN calls to local extensions to be routed back to the Belgacom SIP network. This can be avoided by defining the Host Address Maps shown in the following table. This causes calls to all numbers, except those which begin with "02795968" to be routed to the Belgacom SIP network. Thus, external calls to local extensions will not be rerouted to the Belgacom SIP network. The "Global-PSTN" map routes calls made to international destinations, i.e. those which have numbers beginning with a "00".

Add a map for each entry in the following table.

Host Address Map Name	Host Address Map Pattern
PSTN-01	^sip:0[^2][0-9]*@
PSTN-02	^sip:02[^7][0-9]*@
PSTN-03	^sip:027[^9][0-9]*@
PSTN-04	^sip:0279[^5][0-9]*@
PSTN-05	^sip:02795[^9][0-9]*@
PSTN-06	^sip:027959[^6][0-9]*@
PSTN-07	^sip:0279596[^8][0-9]@
Global-PSTN	^sip:00.*@

Table 22: "Add Host Map Entry" Parameters

The configuration for the first map in the above table is shown in the following figure. Repeat this procedure for each of the other entries in the table.

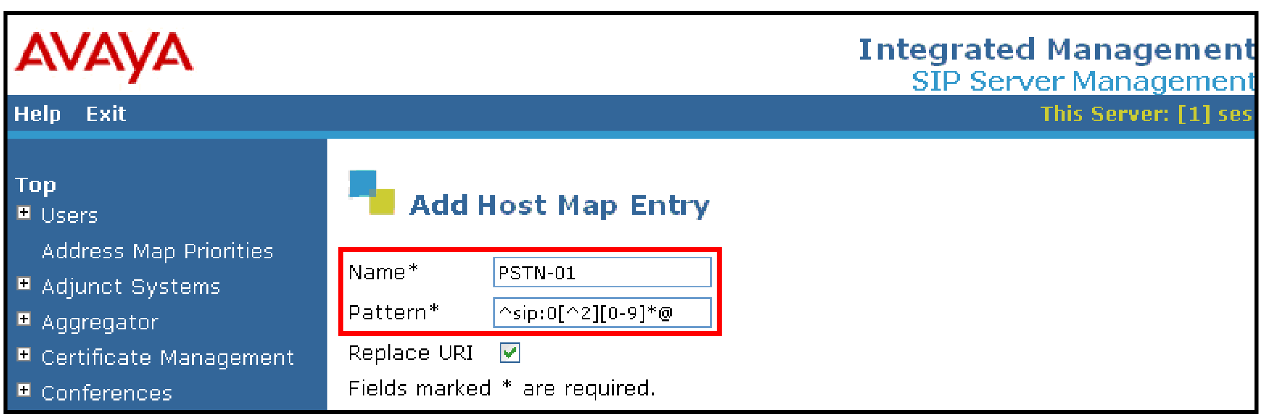

Figure 39: Add Host Map Entry Screen

After the host address maps have been added, select "Add Another Contact" (not shown) and specify the following Contact parameters: "sip:$(user)@<Voice SBC IP Address>:5060;transport=udp". The IP address of the Voice SBC is shown in **Figure 1**.

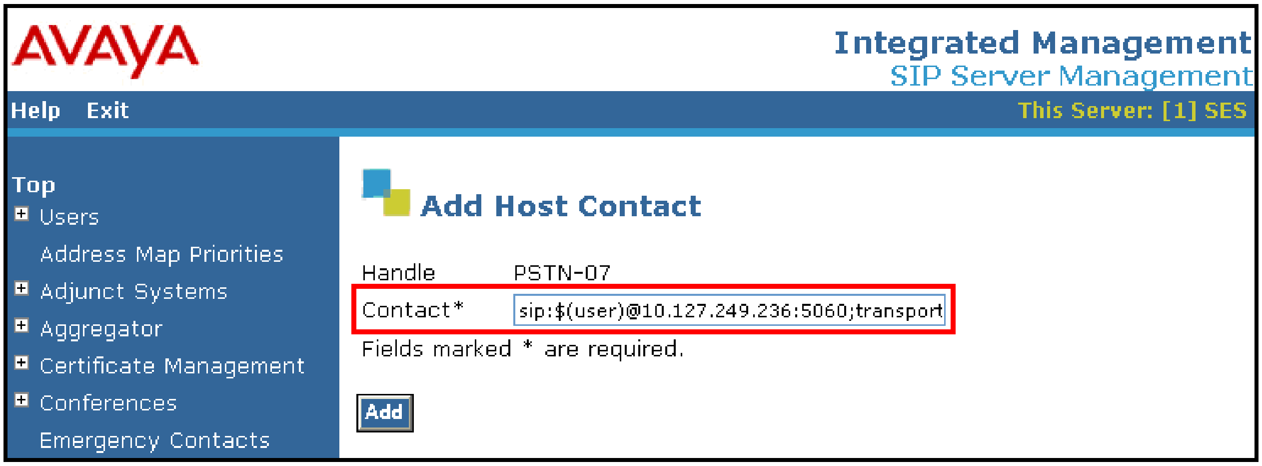

Figure 40: Add Host Contact Screen

The “List Host Address Map” screen now shows the hosts maps for PSTN and GSM routing.

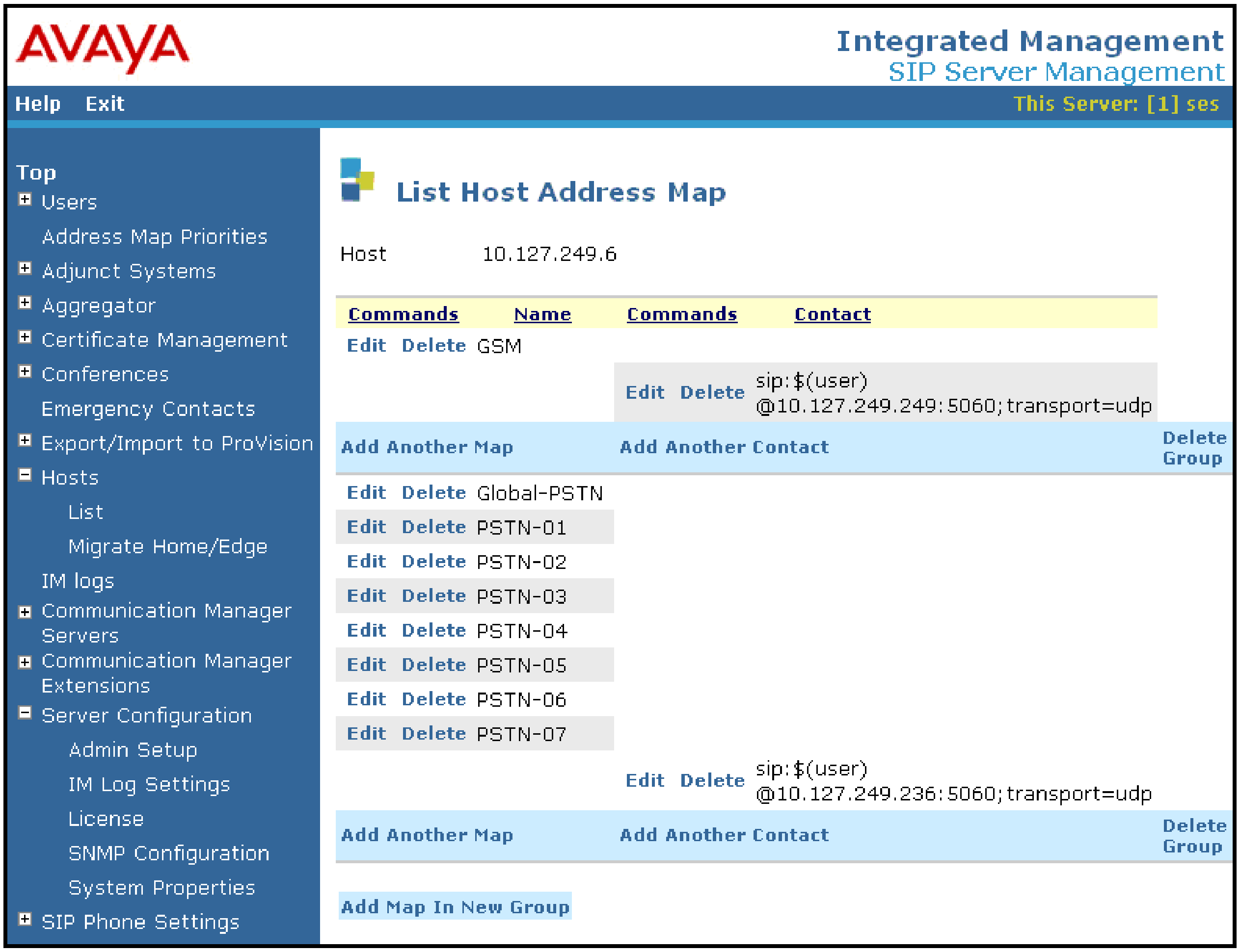

Figure 41: List Completed Host Address Map Screen

5.3. Add Communication Manager Server Interfaces

Select "Communication Manager Servers" → "Add" from the "Top" level menu shown in **Figure 31**, and specify the interface parameters as shown in the following table.

Parameter	Usage
Communication Manager Server Interface Name	Select a suitable name to identify this interface.
Host	Select the IP address of the SES server from the drop-down box.
SIP Trunk IP Address	Enter the IP address of the "procr" interface, as shown in **Figure 5**.
Communication Manager Server Admin Address	Enter the SIP Trunk IP Address.
Communication Manager Server Admin Login	Enter the Communication Manager login name.
Communication Manager Server Admin Password	Enter the Communication Manager login password.

Table 23: Add Communication Manager Server Interface Parameters

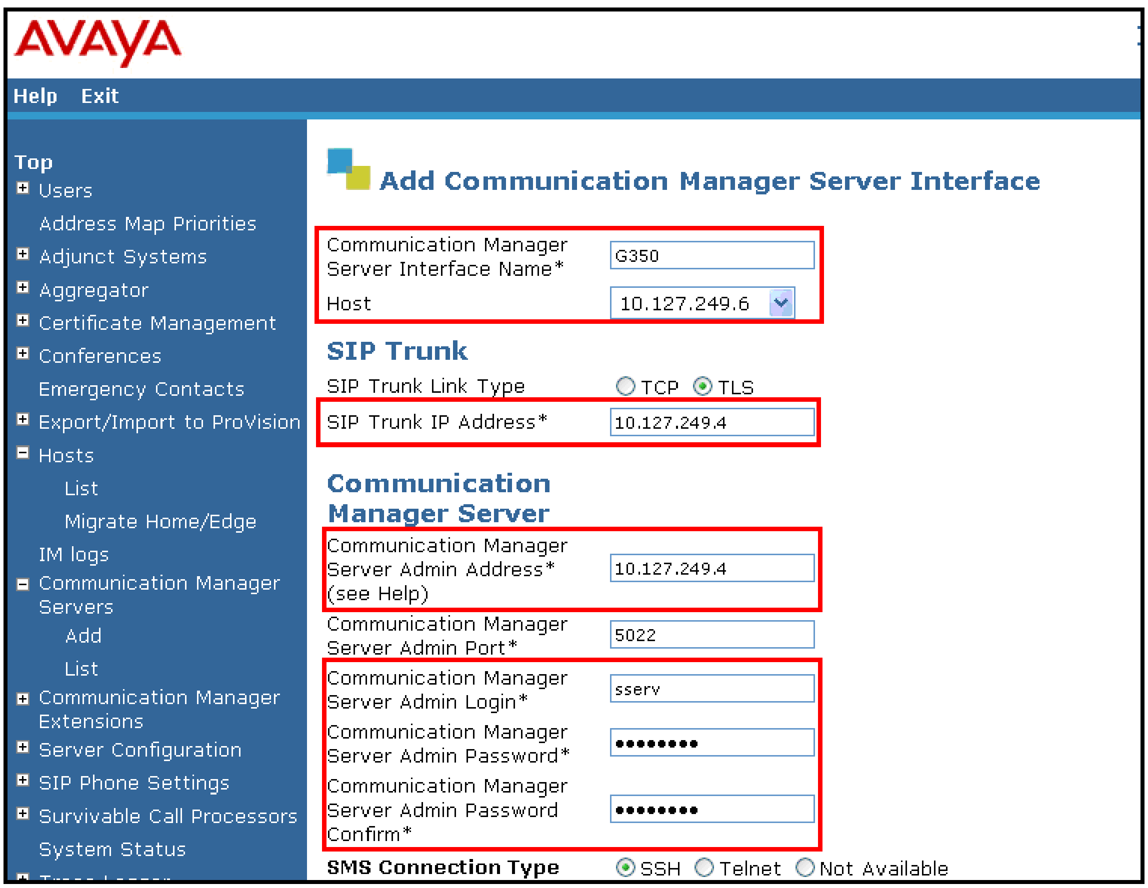

Figure 42: Add Communication Manager Server Interface Screen

5.3.1. Add Communication Manager Server Voice Interface

Select the “Map” menu point from the “List Communication Manager Servers” screen.

Figure 43: List Communication Manager Servers Screen

Click the “Add Map In New Group” control from the following screen.

Figure 44: List Communication Manager Server Address Map Screen

Enter the values shown in the following table in the "Add Communication Manager Server Address Map" screen.

Parameter	Usage
Name	Enter an appropriate name to identify the map.
Pattern	Enter "^sip:02795968[0-7]@ to match that incoming numbers beginning with "02795968" followed by a digit from "0" to "7" for voice stations.

Table 24: Add Communication Manager Server Address Map Parameters

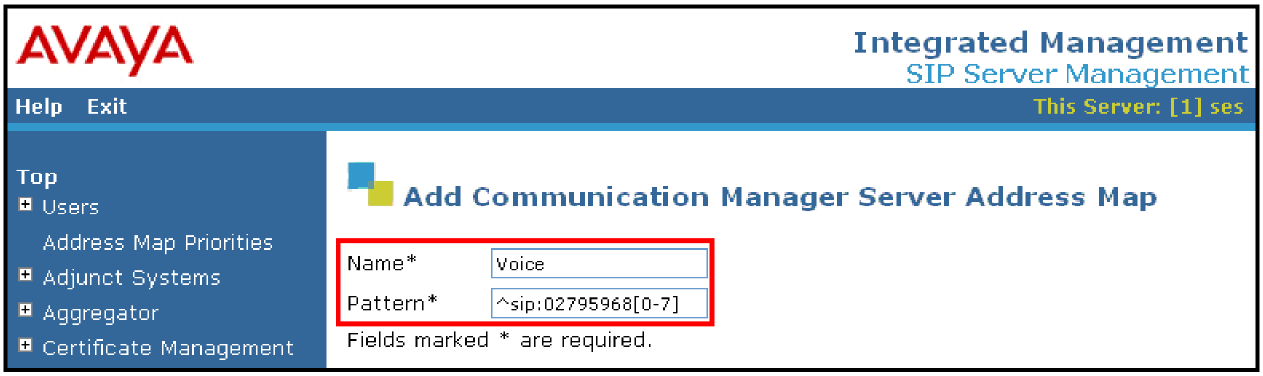

Figure 45: Add Communication Manager Server Address Map

After the map has been added, a Contact with the default port of 5061 is generated automatically. Select the "Edit" contact control.

Figure 46: Add Communication Manager Server Address Map

Specify the following contact: "sip:$(user)@<CM interface IP address>:5062;transport=tls" to specify the port used for Voice calls defined in **Figure 6**.

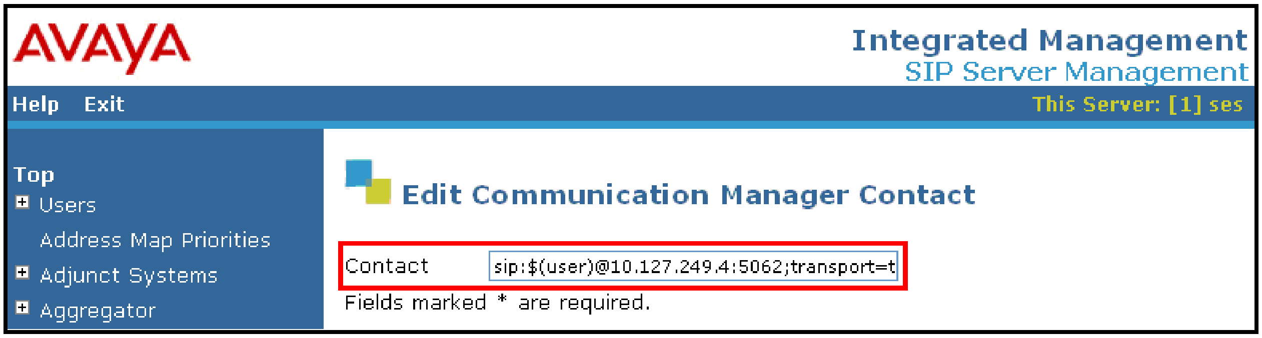

Figure 47: Communication Manager Contact Screen

5.3.2. Add Media Server FAX Interface

Select the "Add Map In New Group".

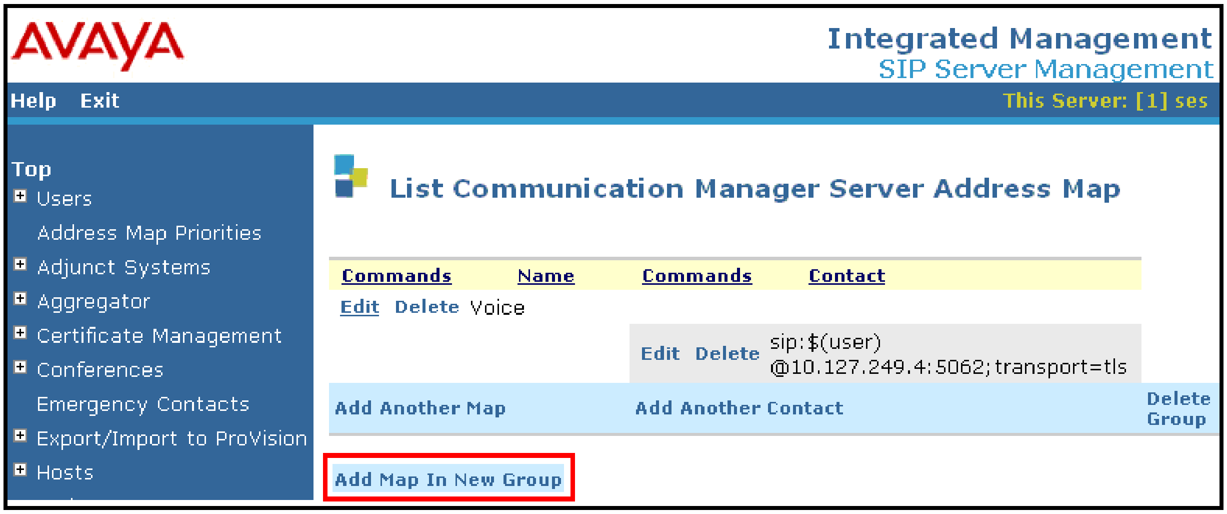

Figure 48: List Communication Manager Server Address Map Screen

Enter the values shown in the following table in the "Add Media Server Address Map" screen.

Parameter	Usage
Name	Enter an appropriate name to identify the map.
Pattern	Enter "^sip:02795968[89]@ to match that incoming numbers beginning with "02795968" followed by either "8" or "9" for FAX devices.

Table 25: Add Communication Manager Server Address Map Parameters

Figure 49: Add Communication Manager Server Address Map

Edit the FAX contact.

Figure 50: Add Communication Manager Server Address Map

Change the FAX contact port number to 5063, to match the port value configured in **Figure 9**.

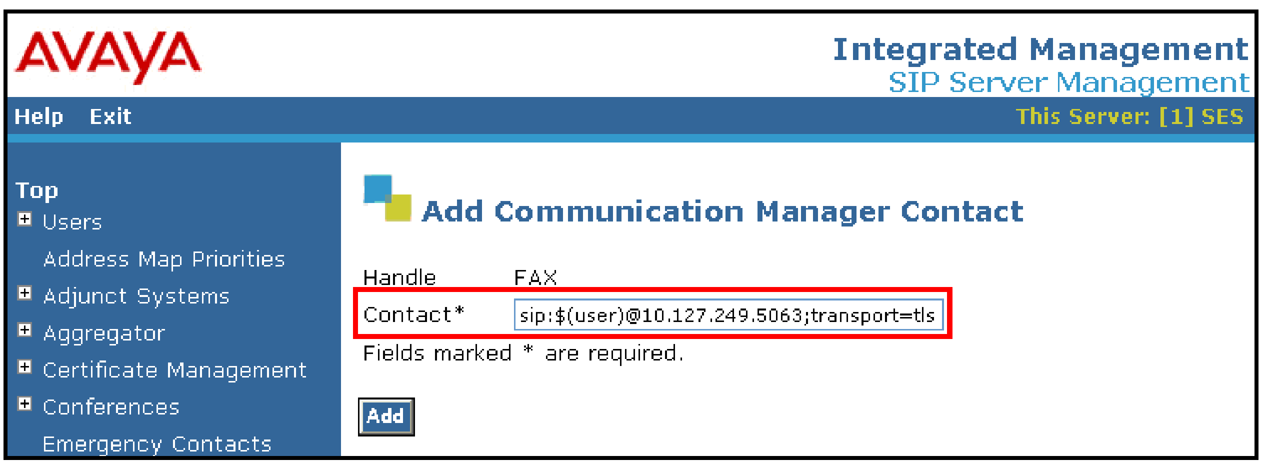

Figure 51: Add Communication Manager Contact Screen

The display now shows the completed Communication Manager Server Address Maps.

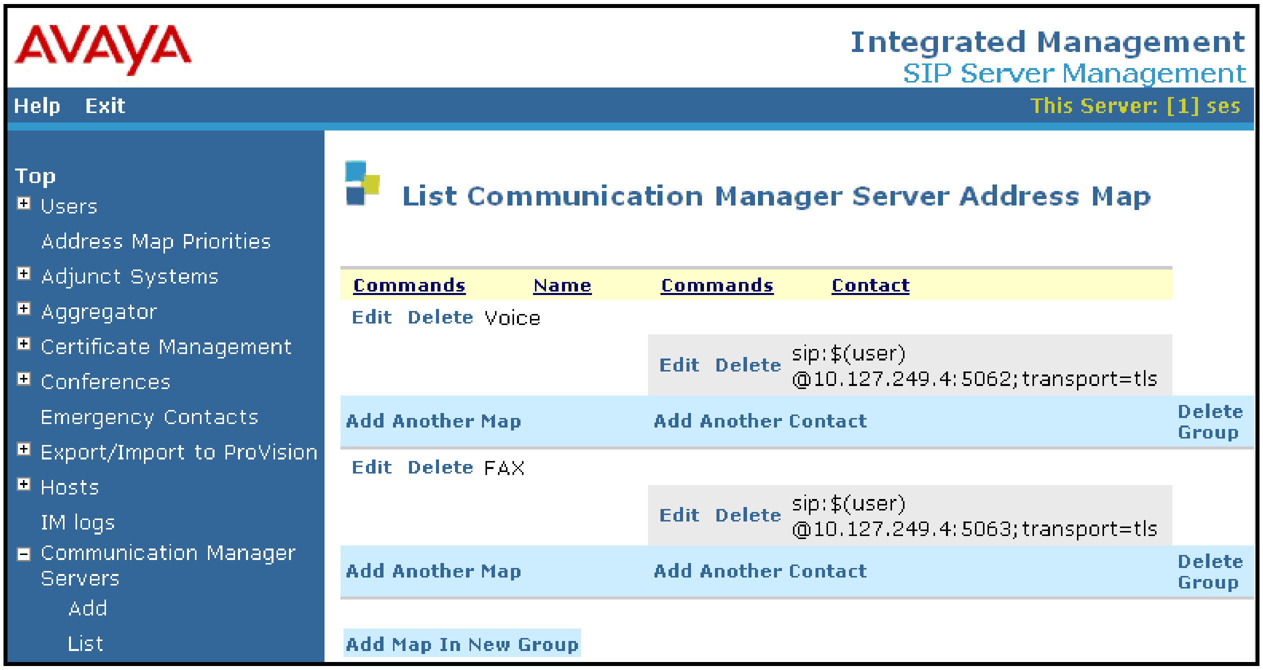

Figure 52: List Communication Manager Server Address Map Screen

5.3.3. Add Media Server FAX Interface

Select the "Add Map In New Group".

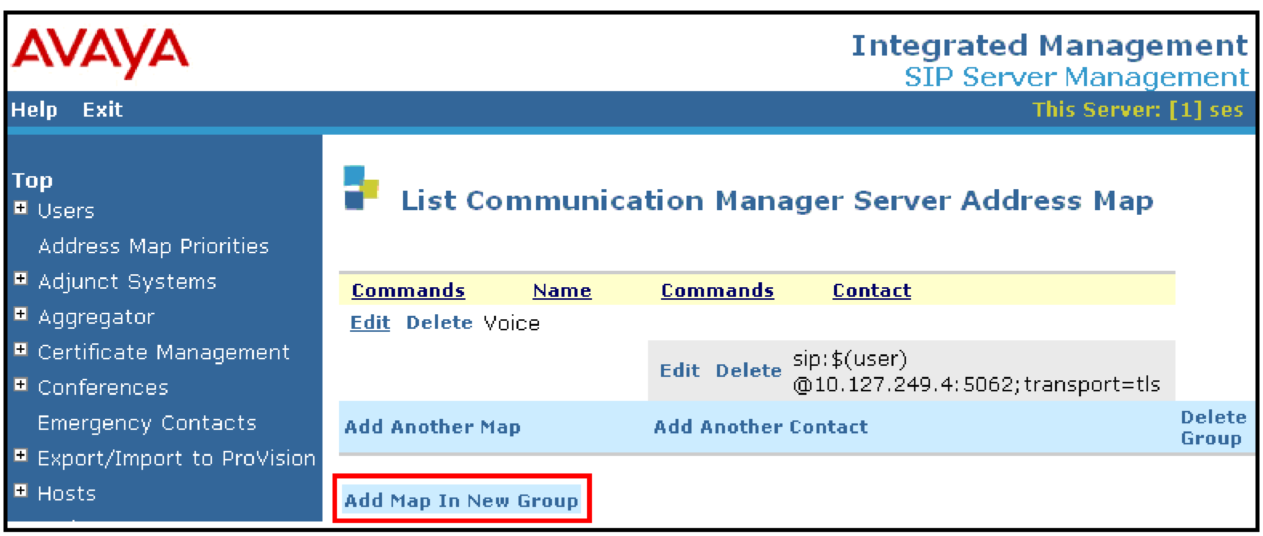

Figure 53: List Communication Manager Server Address Map Screen

Enter the values shown in the following table in the "Add Media Server Address Map" screen.

Parameter	Usage
Name	Enter an appropriate name to identify the map.
Pattern	Enter "^sip:02795968[89]@ to match that incoming numbers beginning with "02795968" followed by either "8" or "9" for FAX devices.

Table 26: "Add Communication Manager Server Address Map" Parameters

Figure 54: Add Communication Manager Server Address Map

Edit the FAX contact.

Figure 55: Add Communication Manager Server Address Map

Change the FAX contact port number to 5063, to match the port value configured in **Figure 9**.

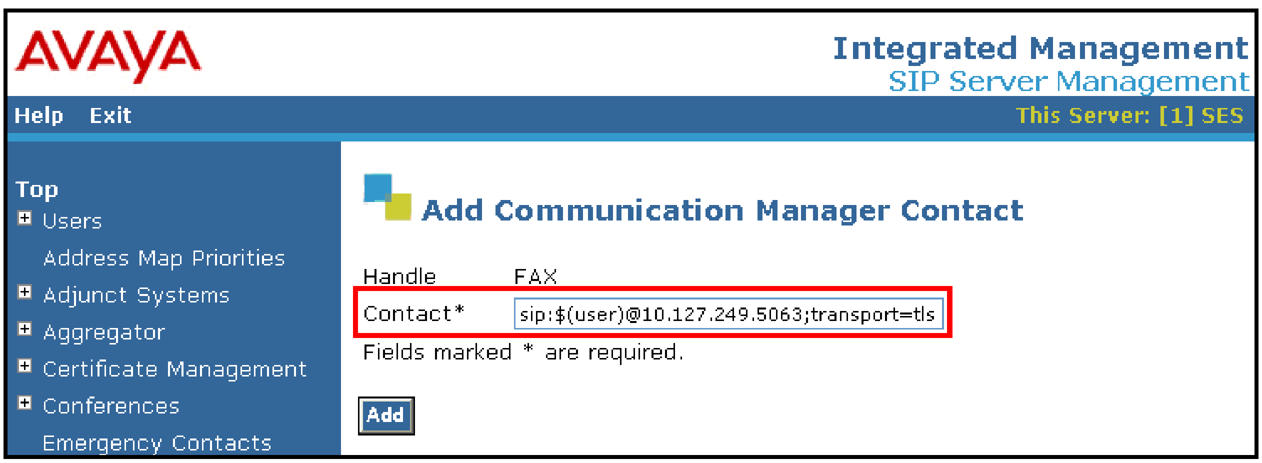

Figure 56: Add Communication Manager Contact Screen

The display now shows the completed Communication Manager Server Address Maps.

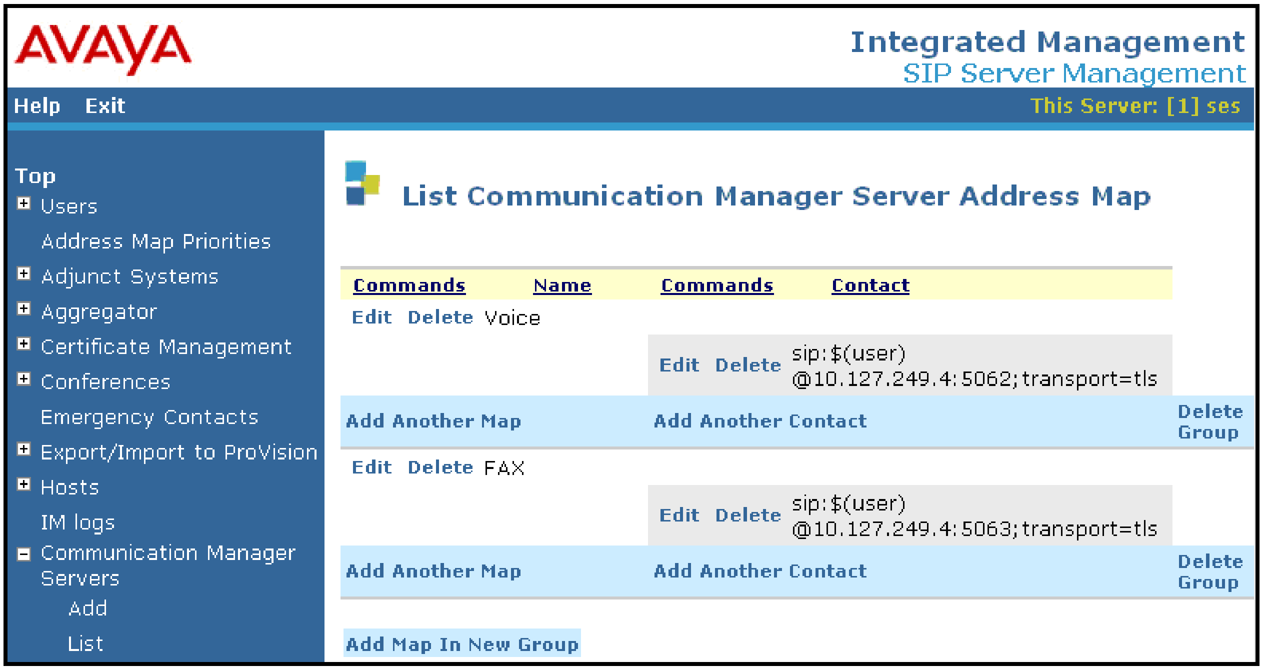

Figure 57: List Communication Manager Server Address Map Screen

5.4. Configure Trusted Host

Select "Trusted Hosts" → "Add" from the "Top" level menu shown in **Figure 31**, and specify the parameters as shown in the following table. Perform this action for both the PSTN and GSM trunks, using the IP addresses shown in **Figure 1**.

Parameter	Usage
IP Address	Enter the IP address on the SBC which is allocated to SIP communications, as shown in **Figure 1**.
Host	Select the IP address of the SES server from the drop-down box.
Comment	Enter an appropriate name to identify the Belgacom VoIP Access SIP Service .

Table 27: Add Trusted Host Parameters

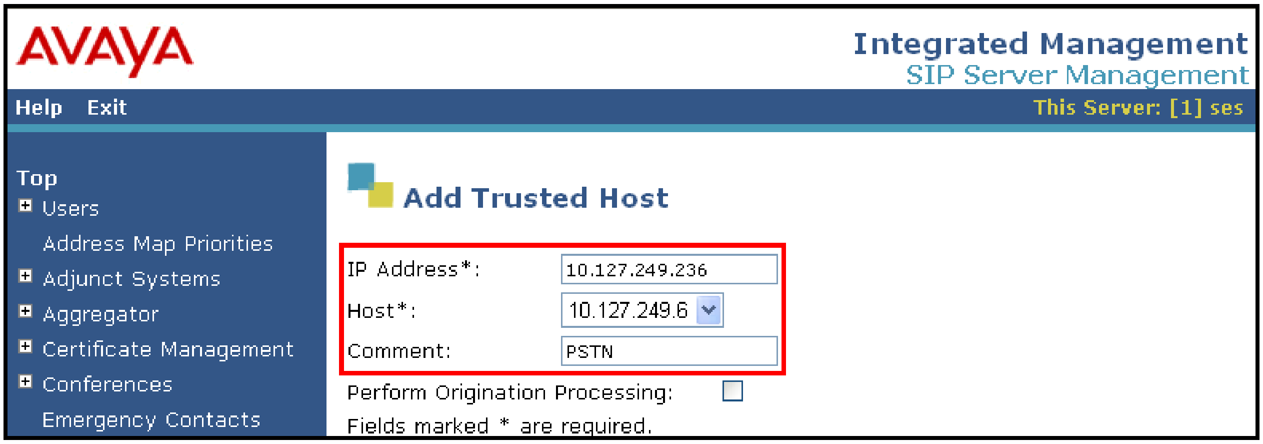

Figure 58: Add PSTN Trusted Host Screen

Repeat this for the GSM trunk:

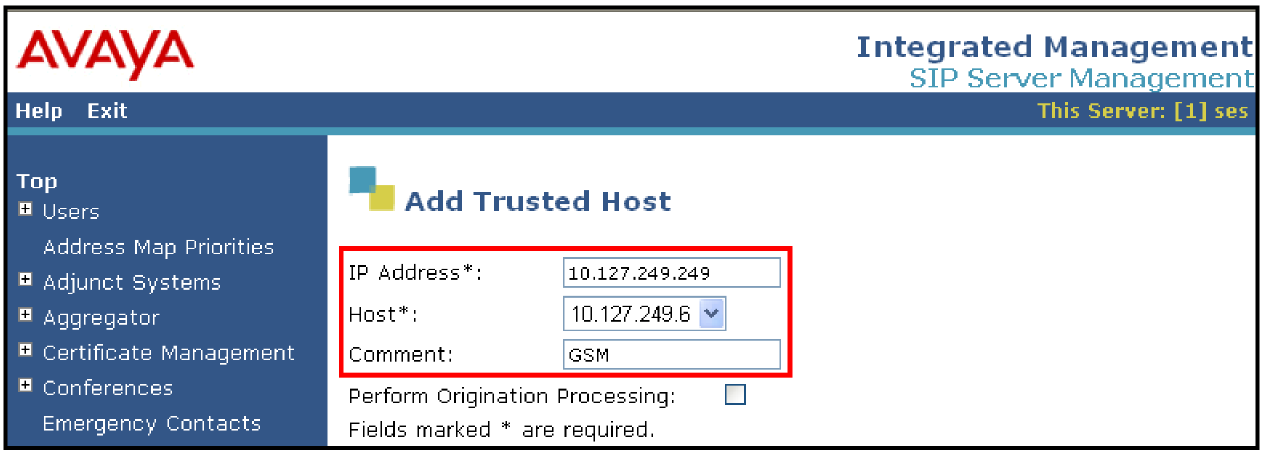

Figure 59: Add GSM Trusted Host Screen

6. General Test Approach and Test Results

The tests listed in **Section 1.1** were performed manually. For each of the tests, correct operation of the endpoints was verified via inspection, and a SIP protocol trace was generated to confirm the expected exchange of SIP protocol messages.

All tests which were conducted were performed successfully. As noted earlier, the operation of GSM SIP trunk was only tested for the operation of incoming and outgoing basic call.

7. Verification Steps

The correct configuration of the system can be verified by performing the following steps:

- Verify that the local Avaya IP Telephones can call each other.
- Verify that the Avaya S8300 Server and SES server can ping each other and the default gateway address of the 2611 router.
- Verify that the Avaya S8300 Server can ping the SBC port allocated to Communication Manager.
- Verify that locally attached Avaya IP Telephones and the telephones attached to the PSTN can call each other.
- Use the “status station” command from the SAT terminal to verify that calls between locally attached telephones and telephones attached to the PSTN are connected with one another without traversing the Avaya G350 Media Gateway.
- Verify that it is possible to send FAX messages between the locally attached FAX device and that which is attached to the PSTN.
- Login to the command line of the SES server and enter “traceSES –no” to view the SIP message traffic between SES and both the Belgacom network and Communication Manager. Make incoming and outgoing calls and verify that the SIP message exchange is correct.

```
--------------------------------------------------------------------------------
       10.127.249.236           10.127.249.4
                        SES
--------------------------------------------------------------------------------
15:20:24:553 |--INVITE-->|           | (1) T:02xxx9682 F:02xxx9333 U:02xxx9682
15:20:24:554 |<--Trying--|           | (1) 100 Trying
15:20:24:563 |           |--INVITE-->| (1) T:02xxx9682 F:02xxx9333 U:02xxx9682
15:20:24:650 |           |<--Trying--| (1) 100 Trying
15:20:24:663 |           |<--Ringing-| (1) 180 Ringing
15:20:24:664 |<--Ringing-|           | (1) 180 Ringing
15:20:27:416 |           |<--200 OK--| (1) 200 OK
15:20:27:418 |<--200 OK--|           | (1) 200 OK
15:20:27:441 |----ACK--->|           | (1) sip:02xxx9682@10.127.249.4
15:20:27:442 |           |----ACK--->| (1) sip:02xxx9682@10.127.249.4
15:20:27:450 |           |<--reINVIT-| (1) T:02xxx9333 F:02xxx9682 U:02xxx9333
15:20:27:451 |           |--Trying-->| (1) 100 Trying
15:20:27:453 |<--reINVIT-|           | (1) T:02xxx9333 F:02xxx9682 U:02xxx9333
15:20:27:508 |--200 OK-->|           | (1) 200 OK
15:20:27:510 |           |--200 OK-->| (1) 200 OK
15:20:27:520 |           |<----ACK---| (1) sip:02xxx9333@10.127.249.236
15:20:27:522 |<----ACK---|           | (1) sip:02xxx9333@10.127.249.236
15:20:31:456 |----BYE--->|           | (1) sip:02xxx9682@10.127.249.4
15:20:31:459 |           |----BYE--->| (1) sip:02xxx9682@10.127.249.4
15:20:31:464 |           |<--200 OK--| (1) 200 OK
15:20:31:465 |<--200 OK--|           | (1) 200 OK
```

Figure 60: Incoming Call traceSES Output Screen

```
--------------------------------------------------------------------------------
        10.127.249.4           10.127.249.236
                        SES
--------------------------------------------------------------------------------
15:17:36:460 |--INVITE-->|           | (3) T:02xxx9040 F:02xxx9682 U:02xxx9040
15:17:36:462 |<--Trying--|           | (3) 100 Trying
15:17:36:478 |           |--INVITE-->| (3) T:02xxx9040 F:02xxx9682 U:02xxx9040
15:17:36:480 |           |<--Trying--| (3) 100 Trying
15:17:39:407 |           |<--Ringing-| (3) 180 Ringing
15:17:39:409 |<--Ringing-|           | (3) 180 Ringing
15:17:43:056 |           |<--200 OK--| (3) 200 OK
15:17:43:057 |<--200 OK--|           | (3) 200 OK
15:17:43:064 |----ACK--->|           | (3) sip:02xxx9040@10.127.249.236
15:17:43:065 |           |----ACK--->| (3) sip:02xxx9040@10.127.249.236
15:17:43:178 |--reINVIT->|           | (3) T:02xxx9040 F:02xxx9682 U:02xxx9040
15:17:43:179 |<--Trying--|           | (3) 100 Trying
15:17:43:181 |           |--reINVIT->| (3) T:02xxx9040 F:02xxx9682 U:02xxx9040
15:17:43:205 |           |<--200 OK--| (3) 200 OK
15:17:43:206 |<--200 OK--|           | (3) 200 OK
15:17:43:229 |----ACK--->|           | (3) sip:02xxx9040@10.127.249.236
15:17:43:230 |           |----ACK--->| (3) sip:02xxx9040@10.127.249.236
15:17:48:088 |----BYE--->|           | (3) sip:02xxx9040@10.127.249.236
15:17:48:091 |           |----BYE--->| (3) sip:02xxx9040@10.127.249.236
15:17:48:099 |           |<--200 OK--| (3) 200 OK
15:17:48:101 |<--200 OK--|           | (3) 200 OK
```

Figure 61: Outgoing Call traceSES Output Screen

8. Conclusion

These Application Notes contain instructions for configuring Avaya Aura™ Communication Manager and Avaya Aura™ SIP Enablement Services to connect to the Belgacom SIP network. A list of instructions is provided to enable the user to verify that the various components have been correctly configured.

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

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

Appendix A: Sample SIP INVITE Messages

These traces were made at a port which mirrored the connection between Avaya Aura™ SIP Enablement Services and the Belgacom network.

Below is a sample SIP INVITE message received from the Belgacom network for an incoming call:

```
Request-Line: INVITE sip:02xxx9681@10.127.249.6:5060 SIP/2.0
Message Header
    Via: SIP/2.0/UDP 10.127.249.236:5060;branch=z9hG4bKrv5dlq20fge07ig0q1o1.1;origin=26.1.251.71
    To: <sip:02xxx9681@voip.belgacom.be;user=phone>
    From: <sip:02xxx9040@voip.belgacom.be;user=phone>;tag=SD01vm501-5c8e796d
    Call-ID: SD01vm501-f924946968208ed2fe5af44f06a6dfb8-vrvvfv3
    CSeq: 1 INVITE
    Max-Forwards: 69
    Contact: <sip:02xxx9040@10.127.249.236:5060;transport=udp>
    Date: Tue, 9 Feb 2010 16:24:51 GMT
    Allow: INVITE, ACK, PRACK, CANCEL, BYE, OPTIONS, MESSAGE, NOTIFY, UPDATE, REGISTER, INFO,
REFER, SUBSCRIBE
    P-Asserted-Identity: <sip:02xxx9040@voip.belgacom.be>
    Accept: application/sdp, application/isup, application/xml, application/dtmf-relay
    Content-Type: application/sdp
    Content-Length: 203
Message Body
    Session Description Protocol
        Session Description Protocol Version (v): 0
        Owner/Creator, Session Id (o): - 0 225088 IN IP4 10.127.249.236
        Session Name (s): IMSS
        Connection Information (c): IN IP4 10.127.249.236
        Time Description, active time (t): 0 0
        Media Description, name and address (m): audio 10378 RTP/AVP 8 101
        Media Attribute (a): rtpmap:101 telephone-event/8000
        Media Attribute (a): fmtp:101 0-15
        Media Attribute (a): X-sqn: 0
        Media Attribute (a): X-cap:  1 image udptl t38
```

Below is a sample SIP INVITE message sent to the Belgacom network for an outgoing call:

```
Request-Line: INVITE sip:02xxx9040@10.127.249.236;user=phone SIP/2.0
Message Header
Call-ID: 022ddbc9c20df1d514b864e3c00
CSeq: 1 INVITE
From: "ext 9682" <sip:02xxx9682@belgacom.be;user=phone>;tag=022ddbc9c20df1d414b864e3c00
Record-Route: <sip:10.127.249.6:5060;lr>,<sip:10.127.249.4:5061;lr;transport=tls>
To: "02xxx9040" <sip:02xxx9040@10.127.249.236;user=phone>
Via: SIP/2.0/UDP 10.127.249.6:5060;branch=z9hG4bK030303232323232323232434d.0,SIP/2.0/TLS
10.127.249.4;psrrposn=2;received=10.127.249.4;branch=z9hG4bK022ddbc9c20df1d614b864e3c00
Content-Length: 189
Content-Type: application/sdp
Contact: "ext 9682" <sip:02xxx9682@10.127.249.4;transport=tls;user=phone>
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@10.127.249.236>;avaya-cm-alert-type=internal
Min-SE: 1800
Session-Expires: 1800;refresher=uac
P-Asserted-Identity: "ext 9682" <sip:02xxx9682@belgacom.be:5061;user=phone>
History-Info: <sip:02xxx9040@10.127.249.236;user=phone>;index=1,"02xxx9040"
<sip:02xxx9040@10.127.249.236;user=phone>;index=1.1
Message Body
Session Description Protocol
Session Description Protocol Version (v): 0
Owner/Creator, Session Id (o): - 1 1 IN IP4 10.127.249.4
Session Name (s): -
Connection Information (c): IN IP4 10.127.249.5
Bandwidth Information (b): AS:64
Time Description, active time (t): 0 0
Media Description, name and address (m): audio 2050 RTP/AVP 8 18 101
Media Attribute (a): rtpmap:8 PCMA/8000
Media Attribute (a): rtpmap:18 G729/8000
Media Attribute (a): rtpmap:101 telephone-event/8000
```

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.