Avaya Solution & Interoperability Test Lab

Application Notes for Avaya Aura® Communication Manager, Avaya Aura® Session Manager and Avaya Aura® Session Border Controller to support Vodafone Netherlands Office Voice and Vodafone Netherlands OneVoice Corporate SIP Trunk Services - Issue 1.0

Abstract

These Application Notes describe the steps to configure Session Initiation Protocol (SIP) trunking between Vodafone Netherlands Office Voice, Vodafone Netherlands OneVoice Corporate SIP Trunk Services and an Avaya SIP enabled enterprise solution. The Vodafone Netherlands Office Voice trunk is used for calls to and from fixed line PSTN locations, Vodafone Netherlands OneVoice Corporate trunk is used for calls to and from mobile telephone numbers as well as providing the ability for enterprise users to reach Vodafone mobile telephone numbers assigned to their account by dialing a four digit short code. The Avaya solution consists of Avaya Aura® Session Border Controller, Avaya Aura® Session Manager and Avaya Aura® Communication Manager Evolution Server. Vodafone Netherlands are a member of the DevConnect Service Provider program.

Information in these Application Notes has been obtained through DevConnect compliance testing and additional technical discussions. Testing was conducted via the DevConnect Program at the Avaya Solution and Interoperability Test Lab.

1. Introduction

These Application Notes describe the steps to configure Session Initiation Protocol (SIP) trunking between Vodafone Netherlands and an Avaya SIP enabled enterprise solution using Vodafone Netherlands Office Voice and Vodafone Netherlands OneVoice Corporate SIP Trunk Services. These services are offerred in conjunction with each other as a total solution, for clarity theses services will be collectivly refrered to in this document as Vodafone Netherlands SIP Trunk Solution. The Avaya solution consists of Avaya Aura® Session Border Controller (AASBC), Avaya Aura® Session Manager and Avaya Aura® Communication Manager Evolution Server. Customers using this Avaya SIP enabled Enterprise solution with Vodafone Netherlands SIP Trunk Solution are able to place and receive calls via standards-based SIP trunks as an alternative to legacy analog or digital trunks.

The Vodafone Netherlands SIP Trunk Solution referenced within these Application Notes is designed for business customers. The solution provides two connections to the enterprise, Vodafone Netherlands Office Voice is a fixed line SIP trunk and Vodafone Netherlands OneVoice Corporate is a mobile SIP trunk. The Vodafone Netherlands Office Voice trunk is used for calls to and from fixed line PSTN locations, Vodafone Netherlands OneVoice Corporate trunk is used for calls to and from mobile telephone numbers as well as providing the ability for enterprise users to reach Vodafone mobile telephone numbers assigned to their account by dialing a four digit short code.

2. General Test Approach and Test Results

The general test approach was to configure a simulated enterprise site using an Avaya SIP telephony solution consisting of Communication Manager, Session Manager and AASBC. The enterprise site was configured to use the SIP Trunk Solution provided by Vodafone Netherlands.

2.1. Interoperability Compliance Testing

The interoperability test included the following:

- Incoming PSTN calls to various phone types. Phone types included SIP, H.323 and digital telephones at the enterprise. All inbound PSTN calls were routed to the enterprise across the SIP trunk from the Service Provider.
- Outgoing calls from the enterprise site were completed via Vodafone Netherlands to PSTN destinations.
- Outgoing calls from the enterprise to the PSTN were made from SIP, H.323 and Digital telephones.
- Inbound and outbound PSTN calls to/from the Avaya one-X® Communicator soft phone.
- Calls to Emergency Services (112)
- Calls using G.729, and G.711A codec's.
- Fax calls to/from a fax machine at the enterprise to a PSTN connected fax machine.
- DTMF transmission using RFC 2833 with successful Voice Mail/Vector navigation for inbound and outbound calls.
- User features such as hold and resume, transfer, conference, call forwarding, etc.
- Caller ID Presentation and Caller ID Restriction.
- Direct IP-to-IP media (also known as "shuffling") with SIP and H.323 telephones.

- Call coverage and call forwarding for endpoints at the enterprise site.
- Transmission and response of SIP OPTIONS messages sent by Vodafone Netherlands requiring Avaya response and sent by Avaya requiring Vodafone Netherlands response.

2.2. Test Results

Interoperability testing of the sample configuration was completed with successful results for the Vodafone Netherlands SIP Trunk Service with the following observations:

- No inbound toll free numbers were tested, however routing of inbound DID numbers and the relevant number translation was successfully tested.
- T.38 Fax calls were unsuccessful when transiting the Netherlands Public Switch Telephone Network (PSTN) due to baud rate alterations occurring mid call. Fax calls using G.711A pass-through were successful.
- Queued ACD calls disconnect if they are not answered within 60 seconds. The Communication Manager sends a SIP 182 message when a call is queued. Vodafone Netherlands do not use the SIP 182 message to cancel any SIP timers but do cancel their SIP timers upon receipt of a SIP 200 message. To workaround this, a null announcement is used at the beginning of the ACD vector to force Communication Manager to answer the call in the vector and send a SIP 200 message.

2.3. Support

For technical support on the Avaya products described in these Application Notes visit http://support.avaya.com.

For technical support on Vodafone Netherlands SIP trunk services, contact Vodafone Netherlands support at http://www.vodafone.nl/zakelijk/totaal_oplossingen/vast_en_mobiel/.

3. Reference Configuration

Figure 1 illustrates the test configuration. The test configuration shows an enterprise site connected to the Vodafone Netherlands SIP Trunk Solution. The Vodafone Netherlands Office Voice connection is represented in **Figure 1** as (Fixed) and the Vodafone Netherlands OneVoice Corporate connection is represented in **Figure 1** as (Mobile). Located at the enterprise site are an AASBC, Session Manager and Communication Manager. Endpoints are Avaya 96x0 series and Avaya 96x1 series IP telephones (with SIP and H.323 firmware), Avaya 46xx series IP telephones (with H.323 firmware), one-X Communicator (SIP and H.323), Avaya Digital telephones and an Analogue fax machine. Also included in the test configuration was an Avaya Desktop Video Device incorporating the Avaya Flare experience.

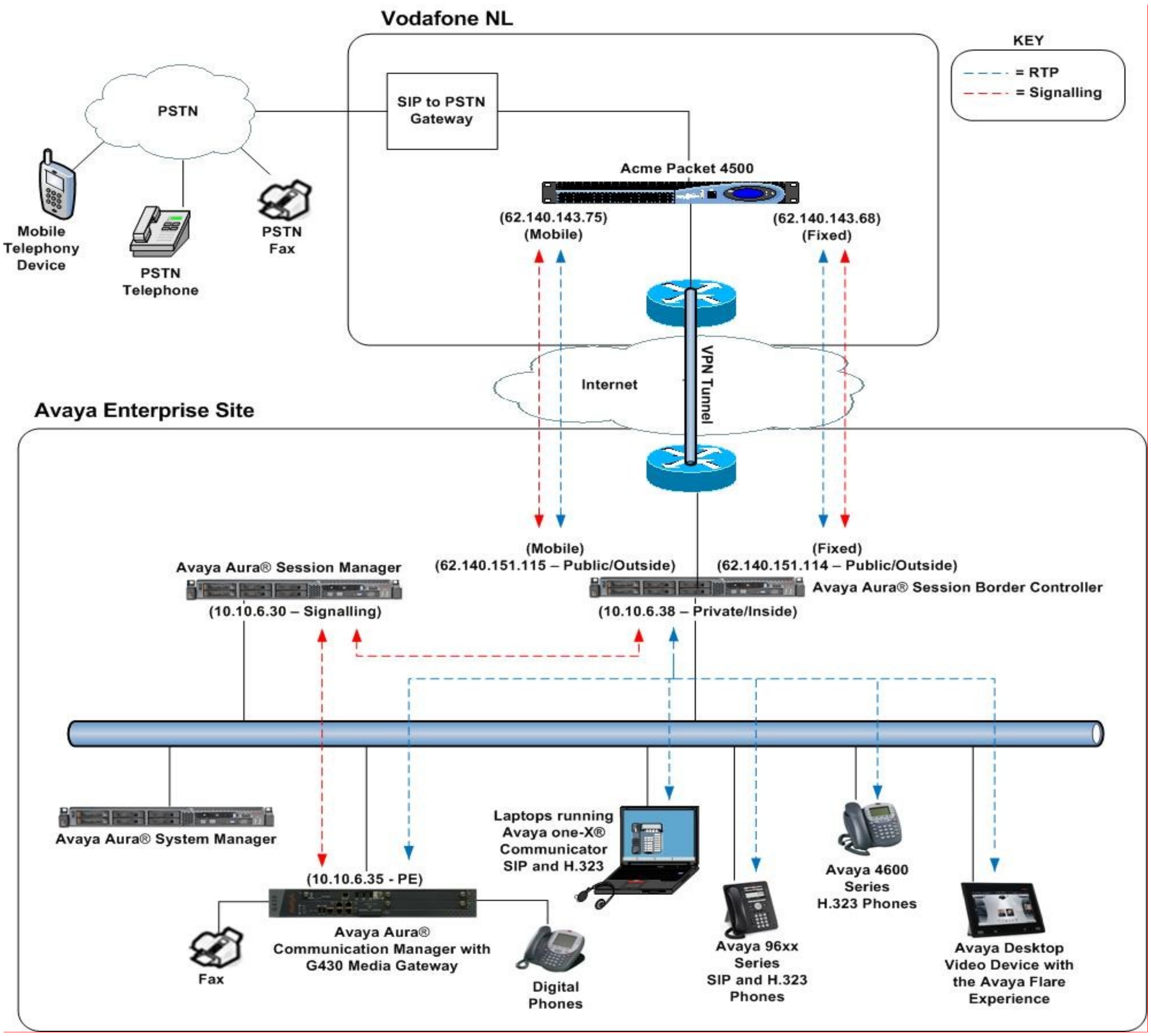


Figure 1: Avaya SIP Telephony Solution using Vodafone Netherlands Office Voice and Vodafone Netherlands OneVoice Corporate services

4. Equipment and Software Validated

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

Avaya	
Equipment	**Software**
Avaya S8300 Media Server	Avaya Aura® Communication Manager R6.0.1 (R016x.00.1.510.1) Service Pack 3 (19009) (System Platform 6.0.3.1.3)
Avaya G430 Media Gateway MM712 MM711	FW 31.19.2 HW07 FW14 HW33 FW95
Avaya S8800 Media Server	Avaya Aura® Session Manager R6.1 Service Pack 3 (6.1.3.0.613006)
Avaya S8800 Media Server	Avaya Aura® System Manager R6.1 Service Pack 3 (6.1.0.0.7345-615.112) (System Platform 6.0.2.0.5)
Avaya S8800 Media Server	Avaya Aura® Session Border ControllerR 6.1 (System Platform 6.0.3.0.3, Template E362P4)
Avaya 9620 Phone (H.323)	S3.102S
Avaya 9641 Phone (H.323)	S6.010f
Avaya 4621 Phone (H.323)	2.9.2
Avaya 9620 Phone (SIP)	2.6.4
Avaya 9641 Phone (SIP)	6.1.3
Avaya Desktop Video Device, A175, Incorporating the Avaya Flare experience	1.0.2
Avaya one–X® Communicator (SIP)	Avaya one–X® Communicator 6.0.1.16-SP1-25226
Digital Phone 2410	N/A
Vodafone Netherlands	
Vodafone Office Voice	1.0
Vodafone OneVoice Corporate	1.0
Acme Packet 4250	SC6.1.0 MR-2 Patch 9 (Build 542)

5. Configure Avaya Aura® Communication Manager

This section describes the steps for configuring Communication Manager for SIP Trunking. SIP trunks are established between Communication Manager and Session Manager. These SIP trunks will carry SIP Signaling associated with Vodafone Netherlands SIP Trunk Solution. For incoming calls, the Session Manager receives SIP messages from the AASBC and directs the incoming SIP messages to Communication Manager. Once the message arrives at Communication Manager, further incoming call treatment, such as incoming digit translations and class of service restrictions may be performed. All outgoing calls to the PSTN are processed within Communication Manager and may be first subject to outbound features such as automatic route selection, digit manipulation and class of service restrictions. Once Communication Manager selects a SIP trunk, the SIP signaling is routed to the Session Manager. The Session Manager directs the outbound SIP messages to the AASBC; the AASBC then sends the SIP messages to the Vodafone Netherlands network. Communication Manager Configuration was performed using the System Access Terminal (SAT). Some screens in this section have been abridged and highlighted for brevity and clarity in presentation. The general installation of the Avaya S8300 Server and Avaya G430 Media Gateway is presumed to have been previously completed and is not discussed here.

5.1. Confirm System Features

The license file installed on the system controls the maximum values for these attributes. If a required feature is not enabled or there is insufficient capacity, contact an authorized Avaya sales representative to add additional capacity. Use the **display system-parameters customer-options** command and on **Page 2**, verify that the **Maximum Administered SIP Trunks** supported by the system is sufficient for the combination of trunks to the Vodafone Netherlands network, and any other SIP trunks used.

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

IP PORT CAPACITIES                                              USED
                     Maximum Administered H.323 Trunks: 12000 10
           Maximum Concurrently Registered IP Stations: 18000 4
             Maximum Administered Remote Office Trunks: 12000 0
Maximum Concurrently Registered Remote Office Stations: 18000 0
              Maximum Concurrently Registered IP eCons: 113   0
  Max Concur Registered Unauthenticated H.323 Stations: 100   0
                        Maximum Video Capable Stations: 18000 0
                   Maximum Video Capable IP Softphones: 0     0
                       Maximum Administered SIP Trunks: 24000 24
  Maximum Administered Ad-hoc Video Conferencing Ports: 24000 0
```

On **Page 4** verify that **IP Trunks** field is set to **y**.

```
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? y
    Enterprise Survivable Server? n                          ISDN-BRI Trunks? y
       Enterprise Wide Licensing? n                                 ISDN-PRI? y
              ESS Administration? y              Local Survivable Processor? n
          Extended Cvg/Fwd Admin? y                     Malicious Call Trace? y
     External Device Alarm Admin? y                 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? 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? y
                       IP Trunks? y

           IP Attendant Consoles? y
```

5.2. Administer IP Node Names

The node names defined here will be used in other configuration screens to define a SIP signaling group between Communication Manager and Session Manager. In the **IP Node Names** form, assign the node **Name** and **IP Address** for the Session Manager. In this case, **SessionMngr** and **10.10.6.30** are the **Name** and **IP Address** for the Session Manager. Also note the **procr** name as this is the processor interface that Communication Manager will use as the SIP signaling interface to Session Manager.

```
display node-names ip
                                  IP NODE NAMES
  Name               IP Address
procr                10.10.6.35
SessionMngr          10.10.6.30
default              0.0.0.0
```

5.3. Administer IP Network Region

Use the **change ip-network-region x** command to set the following values:

- The **Authoritative Domain** field is configured to match the domain name configured on Session Manager. In this configuration, the domain name is **avaya.com**
- By default, **IP-IP Direct Audio** (both **Intra-** and **Inter-Region**) is enabled to allow audio traffic to be sent directly between endpoints without using gateway VoIP resources. When a PSTN call is shuffled the enterprise end point will talk directly to the private interface of the AASBC.
- The **Codec Set** is set to the number of the IP codec set to be used for calls within the IP network region. In this case, codec set **1** is used.

```
change ip-network-region 1                                          Page   1 of  20
                               IP NETWORK REGION
  Region: 1
Location: 1       Authoritative Domain: avaya.com
    Name:Main NR
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
 Call Control PHB Value: 46
        Audio PHB Value: 46
        Video PHB Value: 26
```

5.4. Administer IP Codec Set

Open the **IP Codec Set** form for the codec set specified in the **IP Network Region** form. Enter the list of audio codec's eligible to be used in order of preference. For the interoperability test the codec's supported by Vodafone Netherlands were configured, namely G.729 and G.711A. During compliance testing, other codec set configurations were also verified.

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

On **Page 2** of the codec set form set the **Fax Mode** to **pass-through** as shown below.

```
change ip-codec-set 1                                               Page   2 of   2
                          IP Codec Set
                              Allow Direct-IP Multimedia? n
                Mode                Redundancy
    FAX         pass-through         0
    Modem       off                  0
    TDD/TTY     US                   3
    Clear-channel   n                0
```

5.5. Administer SIP Signaling Groups

This signaling group (and associated trunk group) will be used for inbound and outbound PSTN calls to Vodafone Netherlands SIP Trunk Solution and will be configured using TLS (Transport Layer Security) and the default TLS port of 5061. Configure the **Signaling Group** using the **add signaling-group x** command as follows:

- Set the **Group Type** field to **sip.**
- The **Transport Method** field is set to **tls** (Transport Layer Security).
- The **Peer Detection Enabled** field should be set to **y** allowing the Communication
- Manager to automatically detect if the peer server is a Session Manager
- Set the **Near-end Node Name** to the processor interface (node name **procr**). This value is taken from the **IP Node Names** form shown in **Section 5.2.**
- Set the **Far-end Node Name** to the node name defined for the Session Manager (node name **SessionMngr**), also shown in **Section 5.2**.
- Ensure that the recommended TLS port value of **5061** is configured in the **Near-end Listen Port** and the **Far-end Listen Port** fields.
- In the **Far-end Network Region** field, enter the IP Network Region configured in **Section 5.3.** This field logically establishes the **far-end** for calls using this signaling group as network region 1
- Leave the **Far-end Domain** field blank to accept any domain from the far end.
- The **Direct IP-IP Audio Connections** field is set to **y**.
- The **DTMF over IP** field should remain set to the default value of **rtp-payload**. This value enables Communication Manager to send DTMF transmissions using RFC 2833.
- The default values for the other fields may be used.

```
add signaling-group 2
                                SIGNALING GROUP

 Group Number: 2              Group Type: sip
  IMS Enabled? n        Transport Method: tls
        Q-SIP? n                                                SIP Enabled LSP? n
     IP Video? n                                      Enforce SIPS URI for SRTP? y
  Peer Detection Enabled? y  Peer Server: SM



   Near-end Node Name: procr                  Far-end Node Name: SessionMngr
 Near-end Listen Port: 5061                 Far-end Listen Port: 5061
                                         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? y
Session Establishment Timer(min): 3                      IP Audio Hairpinning? n
         Enable Layer 3 Test? y                     Initial IP-IP Direct Media? n
H.323 Station Outgoing Direct Media? n              Alternate Route Timer(sec): 6
```

5.6. Administer SIP Trunk Group

A trunk group is associated with the signaling group described in **Section 5.5.** Configure the trunk group using the **add trunk-group x** command, where **x** is an available trunk group. On **Page 1** of this form:

- Set the **Group Type** field to **sip**.
- Choose a descriptive **Group Name**.
- Specify a trunk access code (**TAC**) consistent with the dial plan.
- The **Direction** is set to **two-way** to allow incoming and outgoing calls.
- Set the **Service Type** field to **tie**.
- Specify the signaling group associated with this trunk group in the **Signaling Group** field as previously configured in **Section 5.5**.
- Specify the **Number of Members** supported by this SIP trunk group.

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

Group Number: 2                   Group Type: sip          CDR Reports: y
  Group Name: SIP TLS SM                 COR: 1        TN: 1        TAC: 702
   Direction: two-way       Outgoing Display? y
 Dial Access? n                                     Night Service:
Queue Length: 0
Service Type: tie                  Auth Code? n
                                             Member Assignment Method: auto
                                                      Signaling Group: 2
                                                    Number of Members: 10
```

On **Page 2** of the trunk-group form the **Preferred Minimum Session Refresh Interval (sec)** field should be set to a value mutually agreed with Vodafone Netherlands to prevent unnecessary SIP messages during call setup. Also note that the value for **Redirect On OPTIM Failure** was set to **8000** to allow additional set-up time for calls destined for an EC500 destination.

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

TRUNK PARAMETERS
     Unicode Name: auto
                                          Redirect On OPTIM Failure: 8000

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

On **Page 3** set the **Numbering Format** field to **public.**

```
add trunk-group 2                                              Page   3 of  21
TRUNK FEATURES
          ACA Assignment? n            Measured: both
                                                         Maintenance Tests? y

                     Numbering Format: public
                                               UUI Treatment: service-provider
                                                Replace Restricted Numbers? n
                                               Replace Unavailable Numbers? n
```

On **Page 4** set the **Mark Users as Phone** to **y**, this field inserts a parameter to SIP requests indicating to any receiving SIP entity that the user part of the request URI should be treated as a telephone number. Set **Telephone Event Payload Type** to **101** the value preferred by Vodafone Netherlands. As Vodafone Netherlands do not support The P-Asserted Identity header the **Identity for Calling Party Display** field was set to **From** so that the Communication Manager will use the value in the From header for the calling party display on enterprise telephones.

```
add trunk-group 2                                              Page   4 of  21
                              PROTOCOL VARIATIONS

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

       Convert 180 to 183 for Early Media? n
 Always Use re-INVITE for Display Updates? n
       Identity for Calling Party Display: From
                             Enable Q-SIP? n
```

5.7. Administer Calling Party Number Information

Use the **change public-unknown-numbering** command to configure Communication Manager to send the calling party number. In the sample configuration, all stations with a **5-digit** extension beginning with **59** will send the calling party number **+3188003xxxx** to Vodafone Netherlands SIP Trunk Solution, where **x** is the last 4 digits of the 5-digit extension. This calling party number will be sent in the SIP From and Contact headers, and displayed on display-equipped PSTN telephones.

```
change public-unknown-numbering 0                              Page   1 of   2
                      NUMBERING - PUBLIC/UNKNOWN FORMAT
                                         Total
Ext Ext           Trk       CPN           CPN
Len Code          Grp(s)    Prefix        Len
                                                Total Administered: 1
 5  59            2          3188003      11       Maximum Entries: 240
```

5.8. Administer Route Selection for Outbound Calls

In these Application Notes, the Automatic Route Selection (ARS) feature will be used to route outbound calls via the SIP trunk to Vodafone Netherlands SIP Trunk Service. In the sample configuration, the single digit 9 is used as the ARS access code. Avaya telephone users will dial 9 to reach an outside line. Use the **change feature-access-codes** command to configure or observe 9 as the **Auto Route Selection (ARS) - Access Code 1.**

```
change feature-access-codes                                   Page   1 of  10
                               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:
      Auto Alternate Routing (AAR) Access Code: 7
    Auto Route Selection (ARS) - Access Code 1: 9     Access Code 2:
```

Use the **change ars analysis x** command to configure the routing of dialed digits following the first digit 9. A small sample of dial patterns is illustrated here. Further administration of ARS is beyond the scope of these Application Notes. The example entries shown will match outgoing calls to numbers beginning **00** or **210**. Calls are sent to **Route Pattern 2**.

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

   Dialed                 Total     Route    Call   Node  ANI
   String               Min  Max  Pattern    Type   Num   Reqd
    00                    9    11     2        pubu          n
    210                   4    4      2        pubu          n
```

Use the **change route-pattern x** command to add the SIP trunk group to the route pattern that ARS selects. In this configuration, route pattern **2** is used to route calls to trunk group **2**.

```
change route-pattern 2                                         Page   1 of   3
                    Pattern Number: 2   Pattern Name: sip trk to SM6
                             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: 2    0                                                             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
```

5.9. Administer Incoming Digit Translation

This step configures the settings necessary to map incoming DID calls to the proper Communication Manager extension(s). The incoming digits sent in the INVITE message from Vodafone Netherlands can be manipulated as necessary to route calls to the desired extension. In the examples used in the compliance testing, the incoming DID numbers provided by Vodafone Netherlands correlate to the internal extensions assigned within Communication Manager. The entries displayed below translates incoming DID numbers +3188003xxxx to a 5 digit extension by deleting 8 of the incoming digits and inserting a 5 to create a valid extension.

```
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             12 +3188003         8   5
```

5.10. EC500 Configuration

When EC500 is enabled on the Communication Manager station, a call to that station will generate a new outbound call from Communication Manager to the configured EC500 destination, typically a mobile phone. The following screen shows an example EC500 configuration for the user with station extension 59022. Use the command **change off-pbx-telephone station mapping x** where **x** is the Communication Manager station.

- The **Station Extension** field will automatically populate
- For **Application** enter **EC500**
- Enter a **Dial Prefix** (e.g., 9) if required by the routing configuration
- For the **Phone Number** field enter the phone that will also be called (e.g., **00353861111111**)
- Set the **Trunk Selection** to **ars** so that the ARS tables will be used to determine how Communication Manager will route to the Phone Number destination.
- Set the **Config Set** to **1**
- Other parameters can retain default value

```
change off-pbx-telephone station-mapping 59022                  Page   1 of   3
                  STATIONS WITH OFF-PBX TELEPHONE INTEGRATION

 Station       Application Dial   CC  Phone Number    Trunk       Config  Dual
 Extension                 Prefix                     Selection   Set     Mode
 59022            EC500        -      00353861111111   ars        1
                               -
                               -                                -
```

5.11. ACD Call Routing

The general configuration of the Communication Manager ACD including skill groups and ACD agents is outside the scope of these application notes and presumed to have been previously completed and is not discussed here. The focus of this section is the configuration of the call vector used during testing to route ACD voice calls. To add a new ACD call flow use the command **change vector x** where **x** is an available vector. To ensure that calls received by the vector illustrated in the screen below are answered by Communication Manager, the vector use's an announcement at the beginning of the vector to answer the incoming call (see **Section 2.2**) and then queues to **skill 1** where skill 1 is a hunt group staffed by ACD agents.

```
change vector 45                                                  Page   1 of   6
                                  CALL VECTOR

    Number: 45               Name: test queue
Multimedia? n      Attendant Vectoring? n    Meet-me Conf? n           Lock? n
     Basic? y   EAS? y   G3V4 Enhanced? y   ANI/II-Digits? y   ASAI Routing? y
 Prompting? y   LAI? y  G3V4 Adv Route? y   CINFO? y   BSR? y   Holidays? y
 Variables? y   3.0 Enhanced? y
01 announcement 1500
02 queue-to     skill 1    pri m
03 wait-time    60  mins hearing ringback
04 goto step    3             if unconditionally
05 stop
06
```

Save Communication Manager changes by entering the command **save translations** to make them permanent.

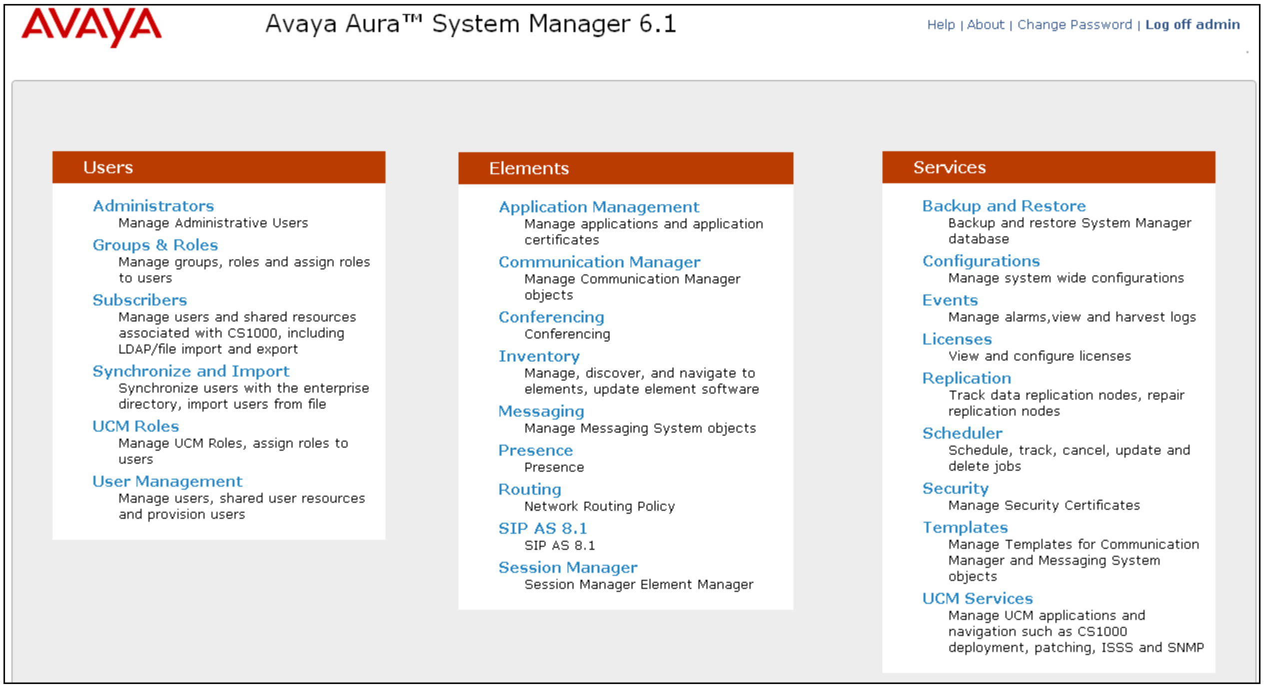
6. Configuring Avaya Aura® Session Manager

This section provides the procedures for configuring Session Manager. The Session Manager is configured via the System Manager. The procedures include the following areas:

- Log in to Avaya Aura® System Manager
- Administer SIP domain
- Administer SIP Entities
- Administer Entity Links
- Administer Routing Policies
- Administer Dial Patterns
- Administer Avaya Aura® Communication Manager as Managed Element
- Administer Application for Avaya Aura® Communication Manager
- Administer Application Sequence for Avaya Aura® Communication Manager
- Administer SIP Extensions

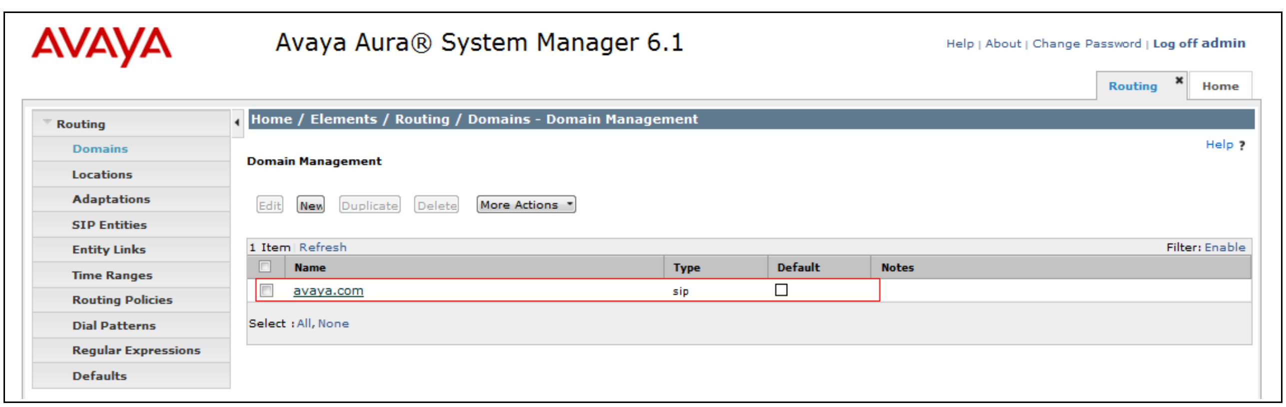
6.1. Log in to Avaya Aura® System Manager

Access the System Manager using a Web Browser by entering **http://<FQDN >/SMGR**, where **<FQDN>** is the fully qualified domain name of System Manager. Log in using appropriate credentials (not shown) and the Home tab will be presented with menu options shown below.



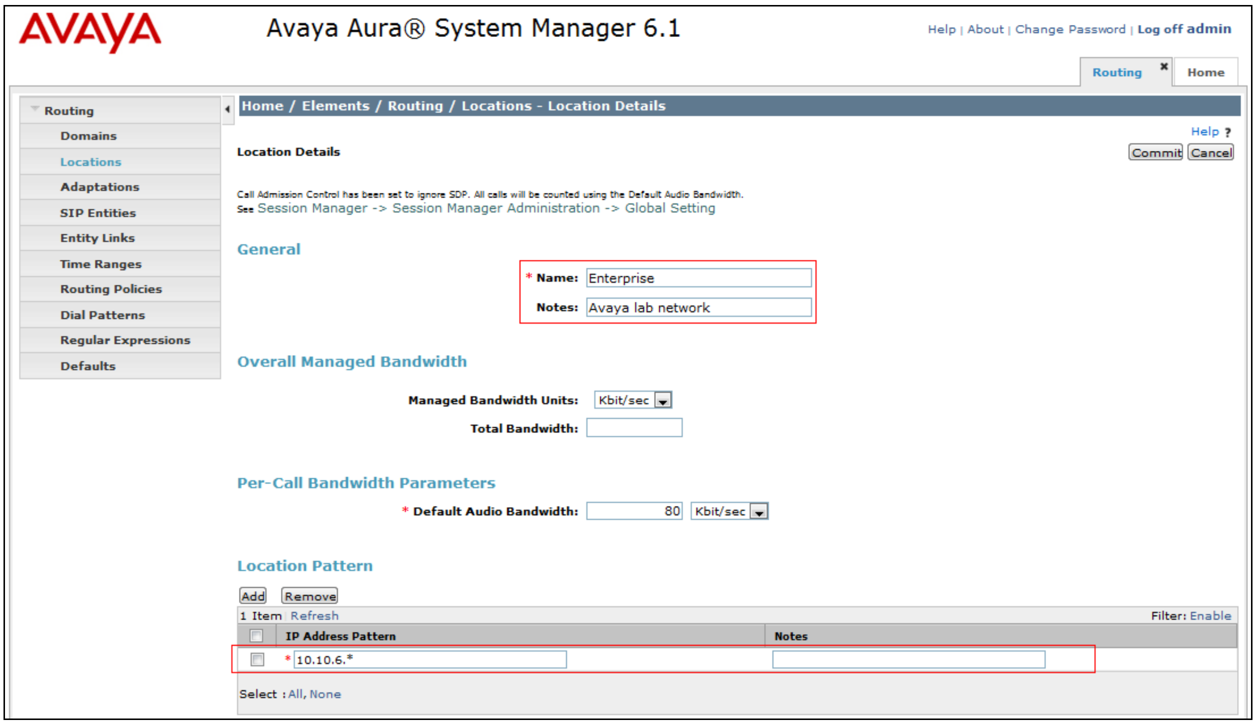
6.2. Administer SIP domain

To add the SIP domain that will be used with Session Manager, select **Routing** from the **Home** tab menu (not shown) and in the resulting tab select **Domains** from left hand menu. Click the **New** button to create a new SIP domain entry. In the **Name** field enter the domain name (e.g., **avaya.com**) and optionally a description for the domain in the **Notes** field. Click **Commit** to save changes (Not shown). The screen below shows the SIP domain that was previously configured.



6.3. Administer Locations

Locations can be used to identify logical and/or physical locations where SIP Entities reside, for the purposes of bandwidth management. One location is added to the sample configuration for all of the enterprise SIP entities. On the **Routing** tab select **Locations** from the left hand menu. Under **General,** in the **Name** field enter an informative name for the location and optionally a description for the location in the **Notes** field. Scroll to the bottom of the page and under **Location Pattern**, click **Add,** then enter an **IP Address Pattern** in the resulting new row, '*' is used to specify any number of allowed characters at the end of the string. Below is the location configuration used for the simulated enterprise.



6.4. Administer SIP Entities

A SIP Entity must be added for each SIP-based telephony system supported by a SIP connection to the Session Manager. To add a SIP Entity, select **SIP Entities** on the left panel menu and then click on the **New** button (not shown). The following fields will need to be populated for each SIP Entity.

Under **General:**

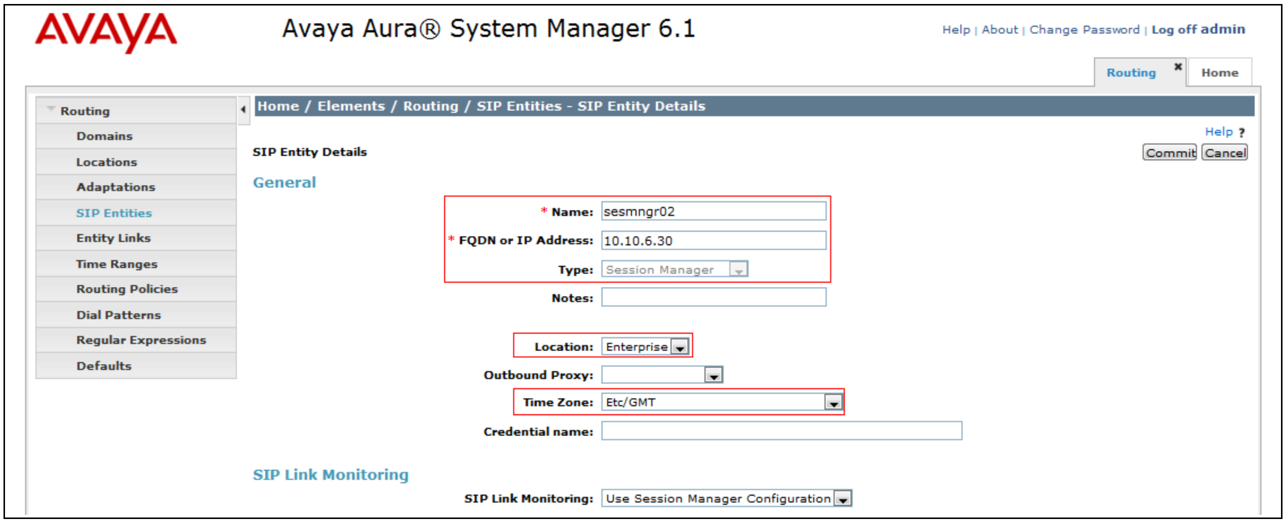
- In the **Name** field enter an informative name.
- In the **FQDN or IP Address** field enter the IP address of Session Manager or the signaling interface on the connecting system.
- In the **Type** field use **Session Manager** for a Session Manager SIP entity, **CM** for a Communication Manager SIP entity and **Gateway** for the AASBC SIP entity.
- In the **Location** field select the appropriate location from the drop down menu.
- In the **Time Zone** field enter the time zone for the SIP Entity.

In this configuration there are three SIP Entities.

- Session Manager SIP Entity
- Communication Manager SIP Entity
- Session Border Controller SIP Entity

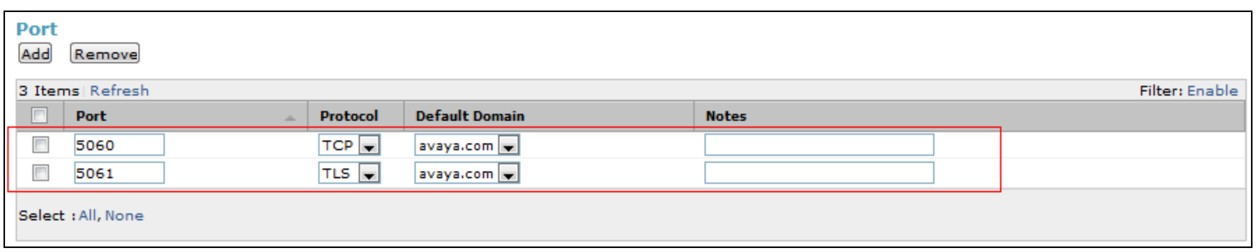
6.4.1. Avaya Aura® Session Manager SIP Entity

The following screens show the SIP entity for Session Manager. The **FQDN or IP Address** field is set to the IP address of the Session Manager SIP signaling interface.



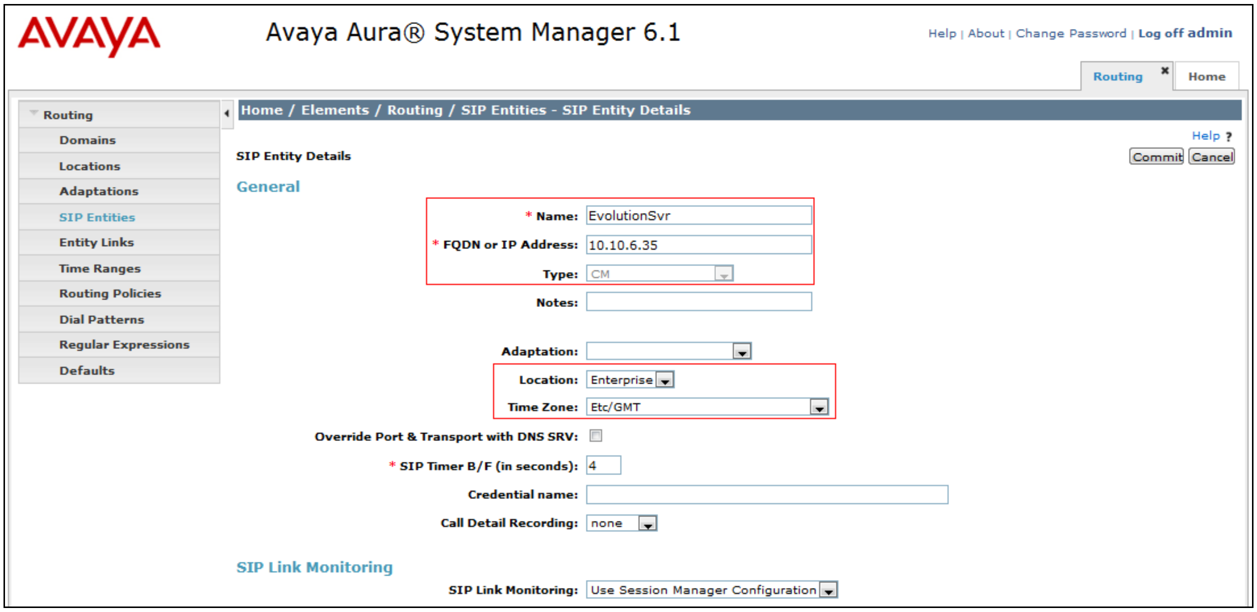
The Session Manager must be configured with the port numbers on the protocols that will be used by the other SIP entities. To configure these scroll to the bottom of the page and under **Port**, click **Add,** then edit the fields in the resulting new row.

- In the **Port** field enter the port number on which the system listens for SIP requests.
- In the **Protocol** field enter the transport protocol to be used for SIP requests.
- In the **Default Domain** field, from the drop down menu select **avaya.com** as the default domain.



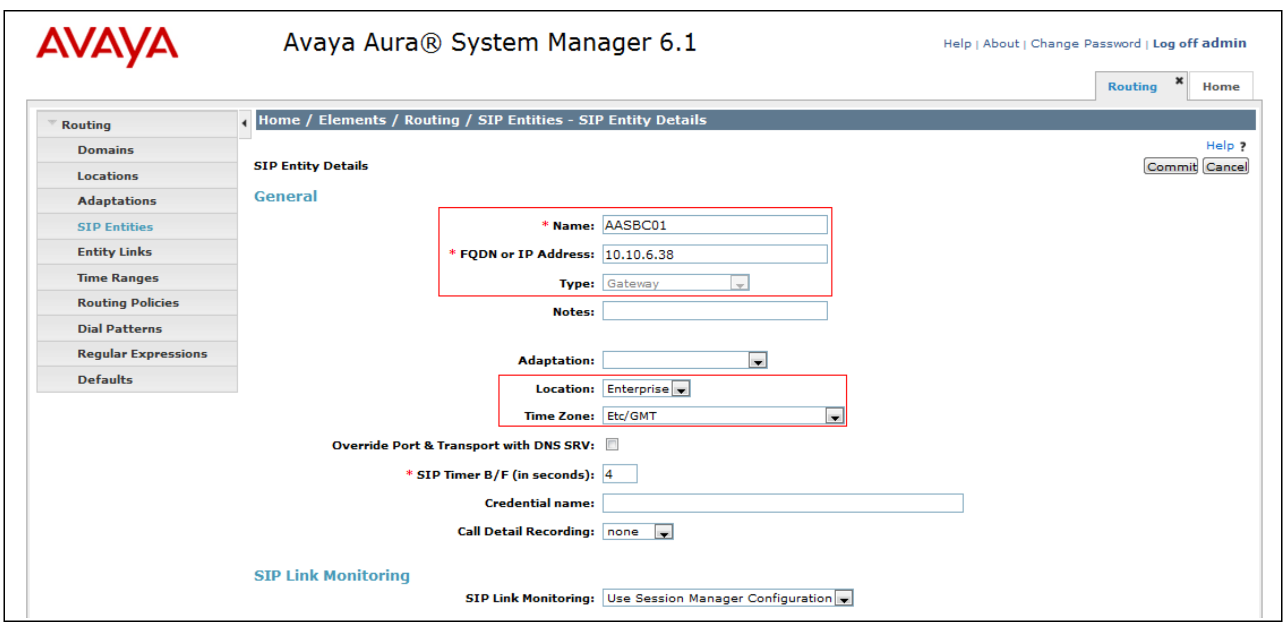
6.4.2. Avaya Aura® Communication Manager SIP Entity

The following screens show the SIP entity for Communication Manager which is configured as an Evolution Server. The **FQDN or IP Address** field is set to the IP address of the interface on Communication Manager that will be providing SIP signaling (see **Figure 1**).



6.4.3. Avaya Aura® Session Border Controller SIP Entity

The following screen shows the SIP Entity for the AASBC. The **FQDN or IP Address** field is set to the IP address of the AASBC private network interface (see **Figure 1**).

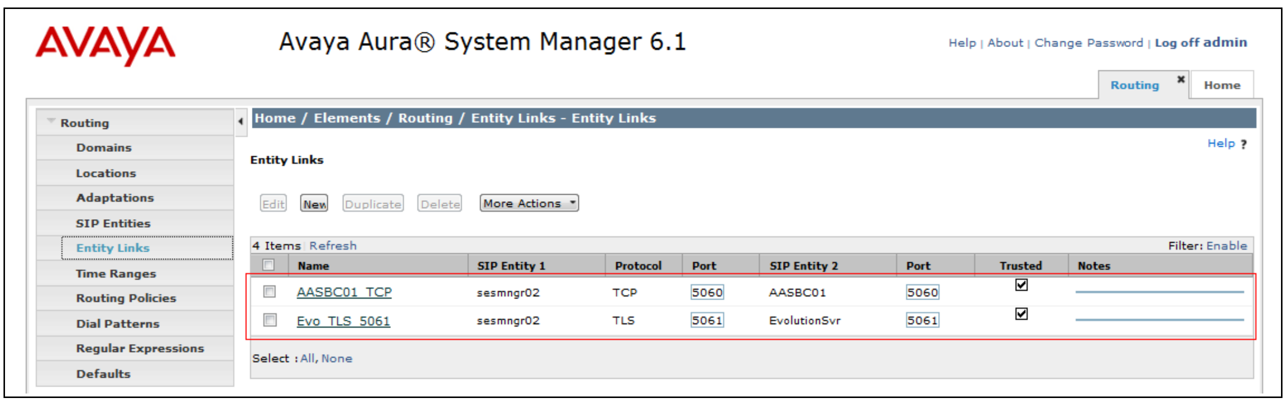


6.5. Administer Entity Links

A SIP trunk between a Session Manager and another system is described by an Entity Link. To add an Entity Link, select **Entity Links** on the left panel menu and click on the **New** button and in the resulting screen fill in the following fields in the new row that is displayed.

- In the **Name** field enter an informative name.
- In the **SIP Entity 1** field select the SIP Entity for SessionManager i.e. **sesmngr02**.
- In the **Port** field enter the port number to which the other system sends its SIP requests.
- In the **SIP Entity 2** field enter the other SIP Entity for this link, created in **Section 6.4**.
- In the **Port** field enter the port number to which the other system expects to receive SIP requests.
- Select the **Trusted** tick box to make the other system trusted.
- In the **Protocol** field enter the transport protocol to be used to send SIP requests.

Click **Commit** to save changes (not shown). The following screen shows the Entity Links used in this configuration.

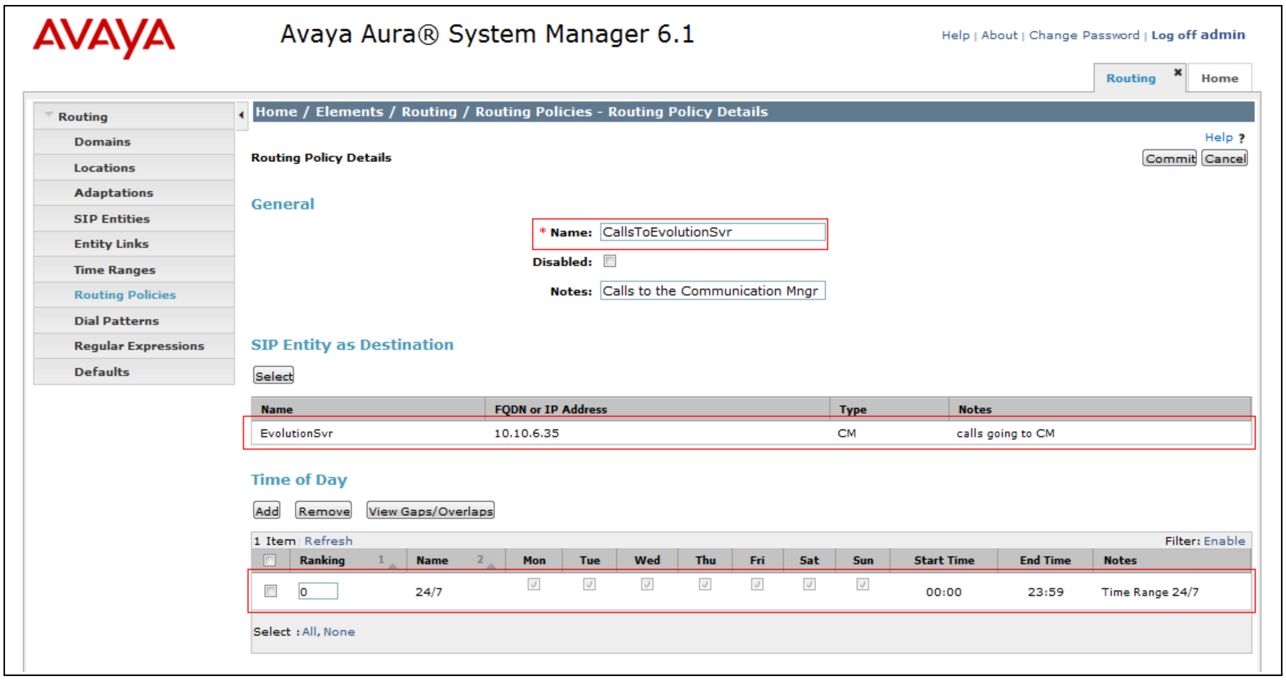


6.6. Administer Routing Policies

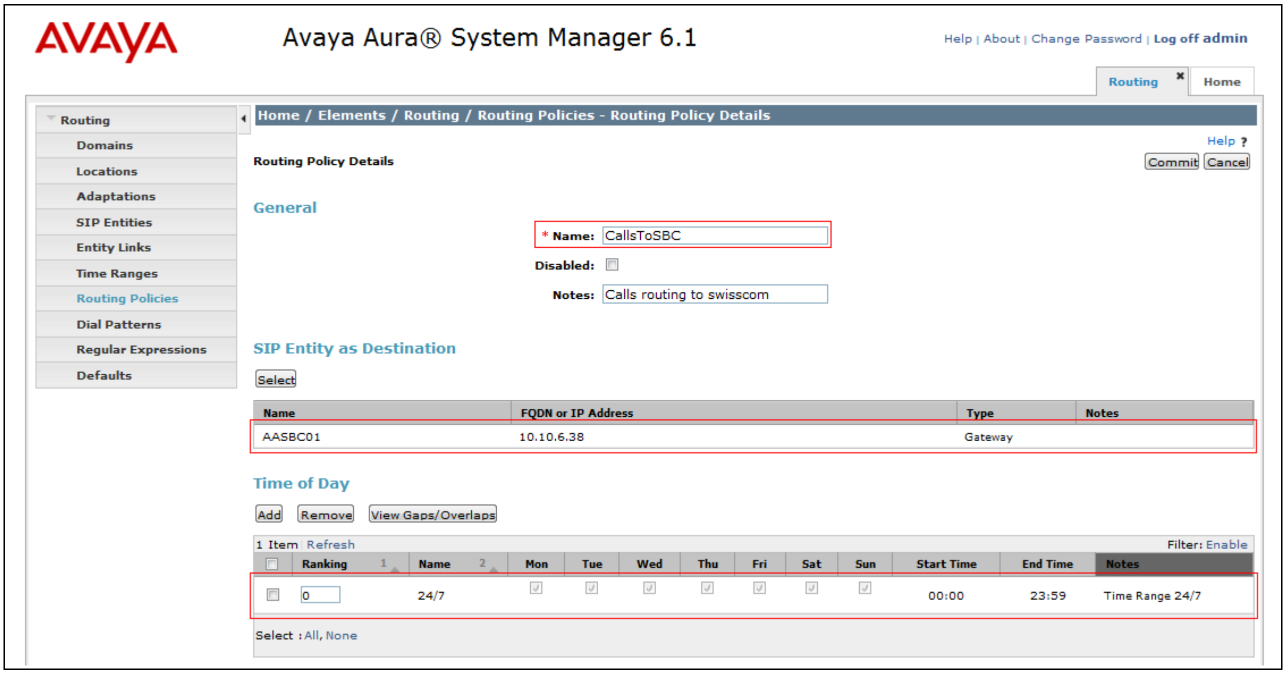
Routing policies must be created to direct how calls will be routed to a system. To add a routing policy, select **Routing Policies** on the left panel menu and then click on the **New** button (not shown).

- Under **General** enter an informative name in the **Name** field.
- Under **SIP Entity as Destination**, click **Select**, and then select the appropriate SIP entity to which this routing policy applies.
- Under **Time of Day**, click **Add**, and then select the time range.

The following screen shows the routing policy for Communication Manager



The following screen shows the routing policy for the AASBC



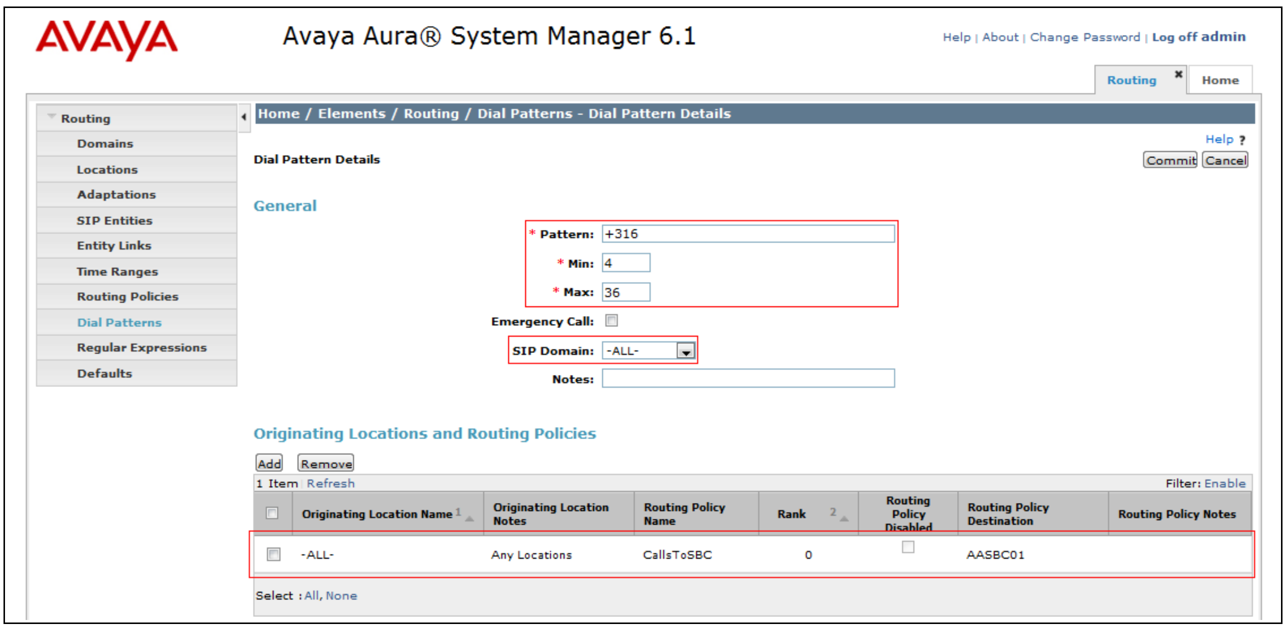
6.7. Administer Dial Patterns

A dial pattern must be defined to direct calls to the appropriate telephony system. To configure a dial pattern select **Dial Patterns** on the left panel menu and then click on the **New** button (not shown).

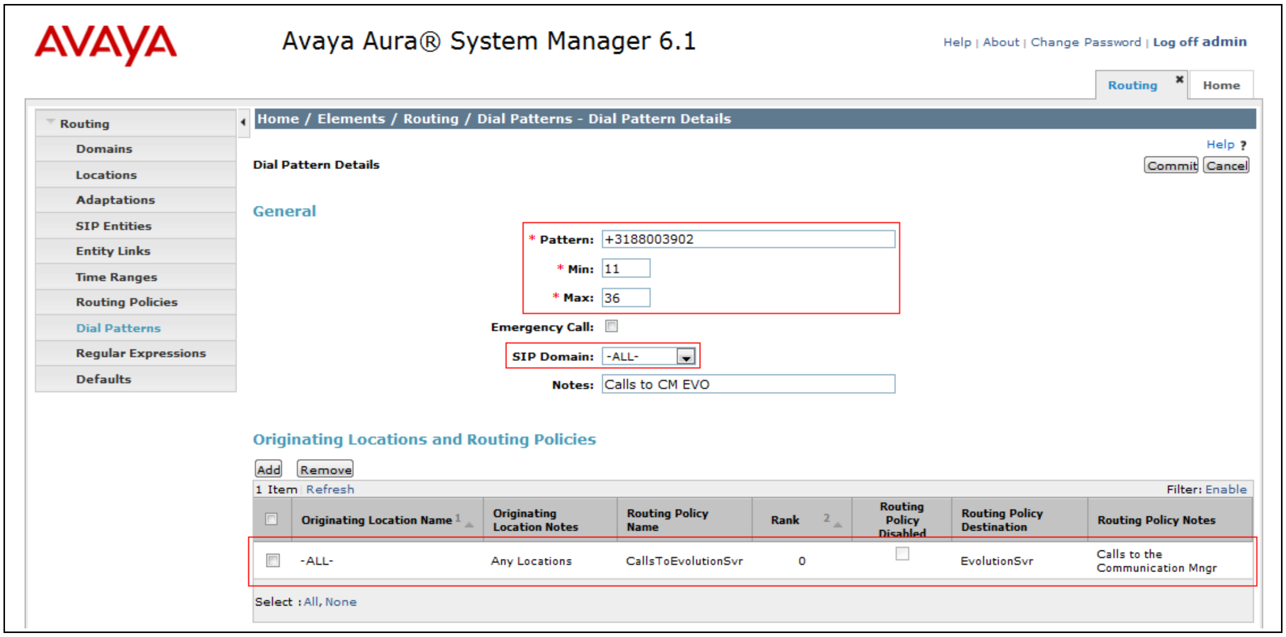
Under **General:**

- In the **Pattern** field enter a dialed number or prefix to be matched
- In the **Min** field enter the minimum length of the dialed number
- In the **Max** field enter the maximum length of the dialed number
- In the **SIP Domain** field select **–ALL-** to allow calls from any domain to match the dial pattern

Under **Originating Locations and Routing Policies.** Click **Add**, in the resulting screen (not shown), under **Originating Location** select **ALL** and under **Routing Policies** select one of the routing policies defined in **Section 6.6.** Click **Select** button to save. The following screen shows an example dial pattern configured for AASBC which will route the calls out to the Vodafone Netherlands SIP Trunk Solution.



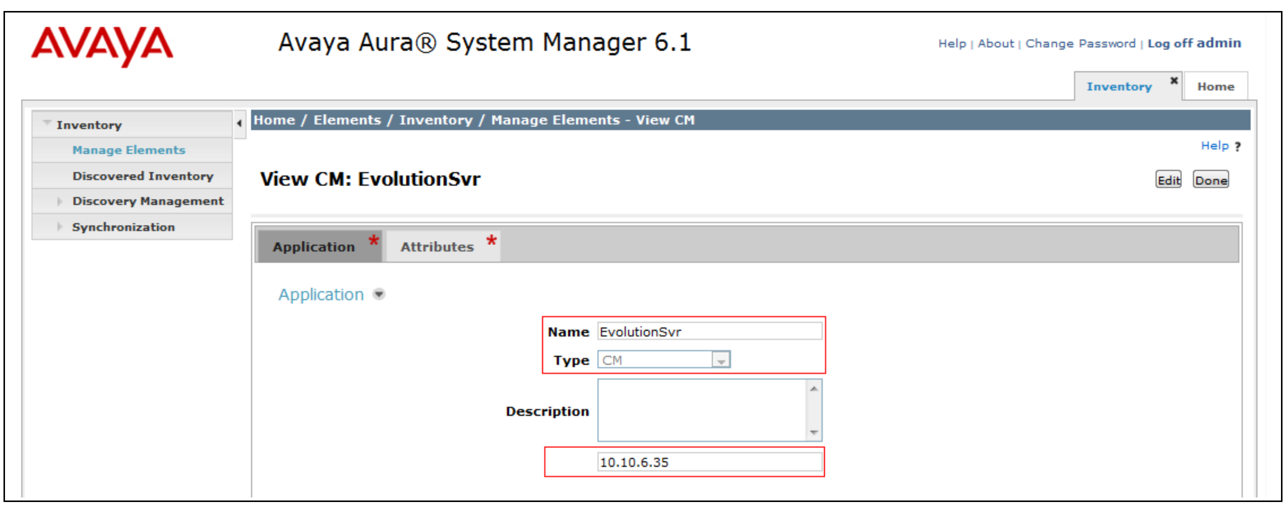
The following screen shows an example dial pattern configured for Communication Manager.



6.8. Administer Avaya Aura® Communication Manager as a Managed Element

From the Home tab select Inventory from the menu. In the resulting tab from the left panel menu select **Manage Elements** and click **New** (not shown)**.** On the **Application** tab, enter values in the following fields and use defaults for the remaining fields:

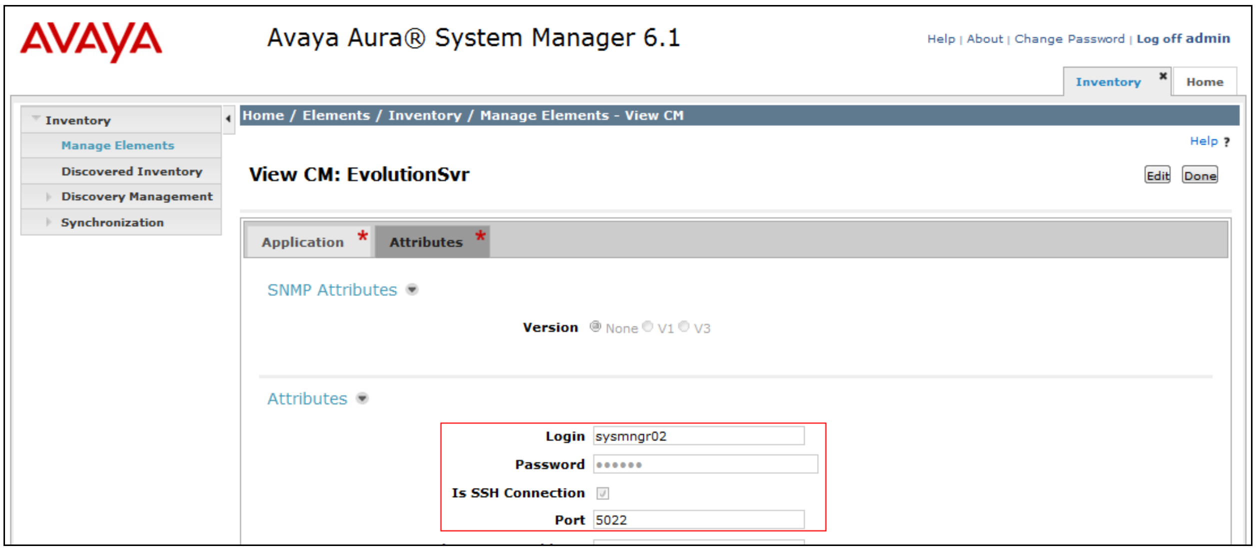
- In the **Name** field enter a descriptive name i.e. **EvolutionSvr**
- In the **Type** field select **CM** from the drop-down menu
- In the **Node** enter the IP address of the Communication Manager



On the **Attributes** tab, under the **Attributes** heading, enter values in the following fields and use defaults for the remaining fields:

- In the **Login** field enter a login name for Communication Manager (SAT SSH login)
- In the **Password** field enter Password for Communication Manager (SAT SSH password)
- Select the **Is SSH Connection** check box if SSH is to be used
- In the **Port** field enter the port number to use for SAT access

Select **Commit**, this causes System Manager to synchronize with the Communication Manager in the background.

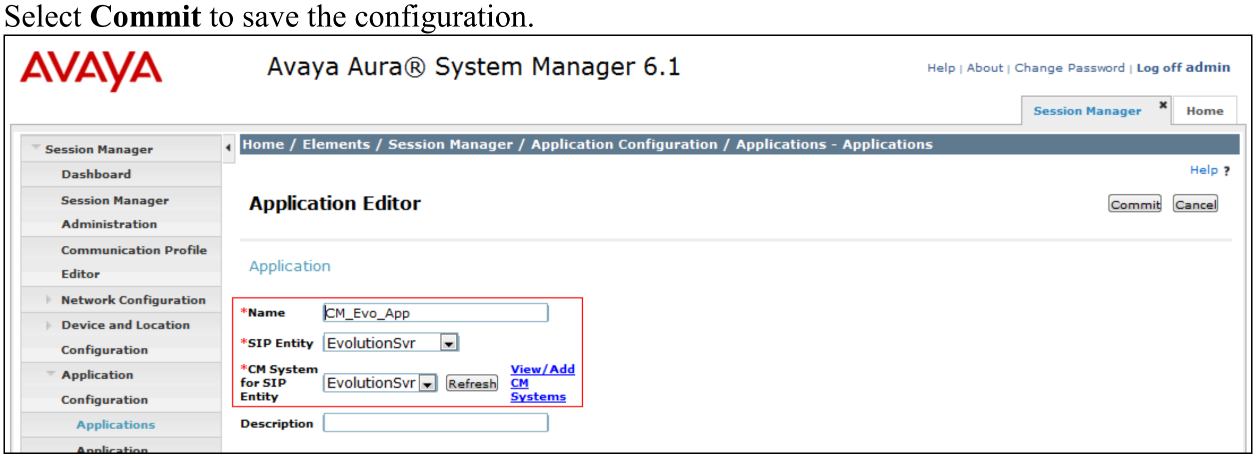


6.9. Administer Application for Avaya Aura® Communication Manager

From the Home tab select Session Manager from the menu. In the resulting tab from the left panel menu select **Application Configuration → Applications** and click **New**

- In the **Name** field enter a name for the application
- In the **SIP Entity** field select the SIP entity for the Communication Manager i.e. **EvolutionSvr**
- In the **CM System for SIP Entity** field select the SIP entity for the Communication Manager

Select **Commit** to save the configuration.

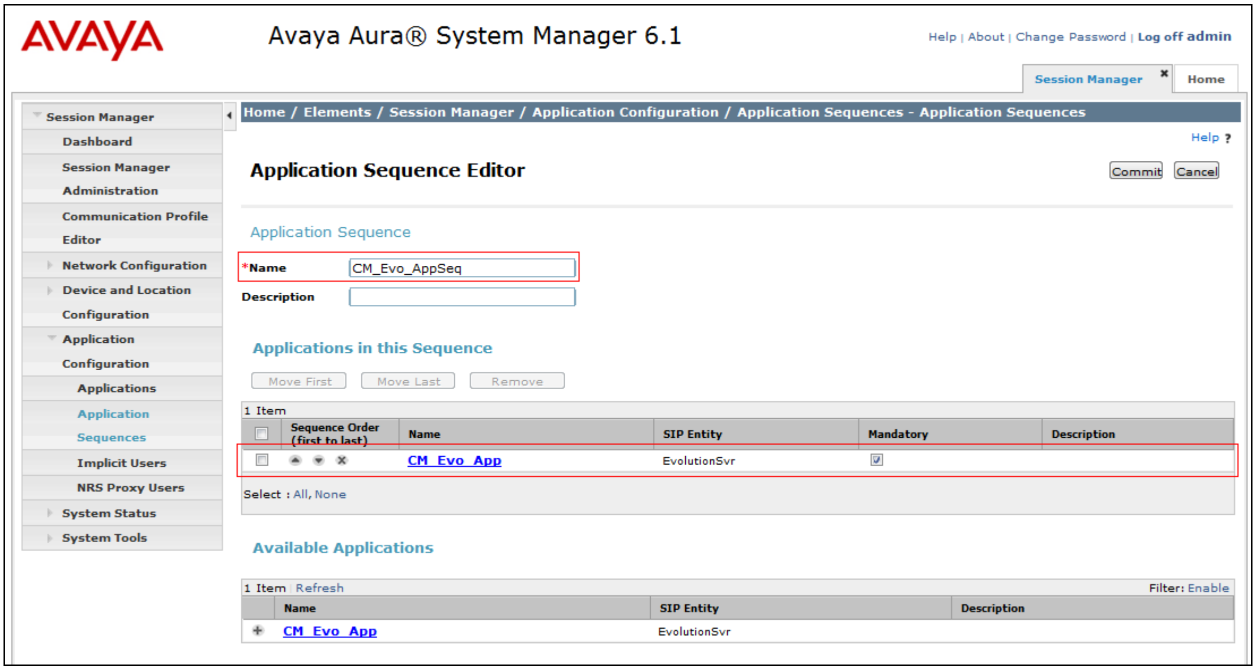


6.10. Administer Application Sequence for Communication Manager

From the left panel navigate to **Session Manager → Application Configuration → Application Sequences** and click on **New.**

- In the **Name** field enter a descriptive name
- Under **Available Applications**, click the + sign in front of the appropriate application instance. When the screen refreshes the application should be displayed under the **Applications in this Sequence** heading.

Select **Commit** to save the configuration.

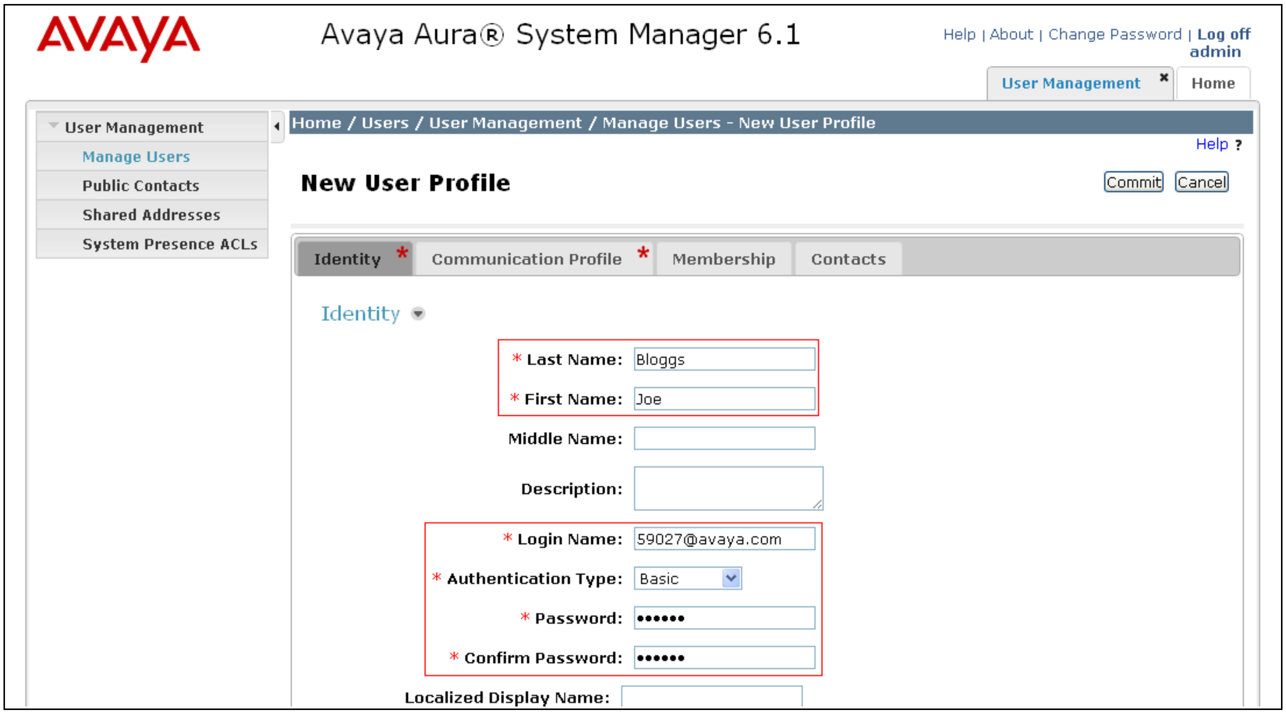


6.11. Administer SIP Extensions

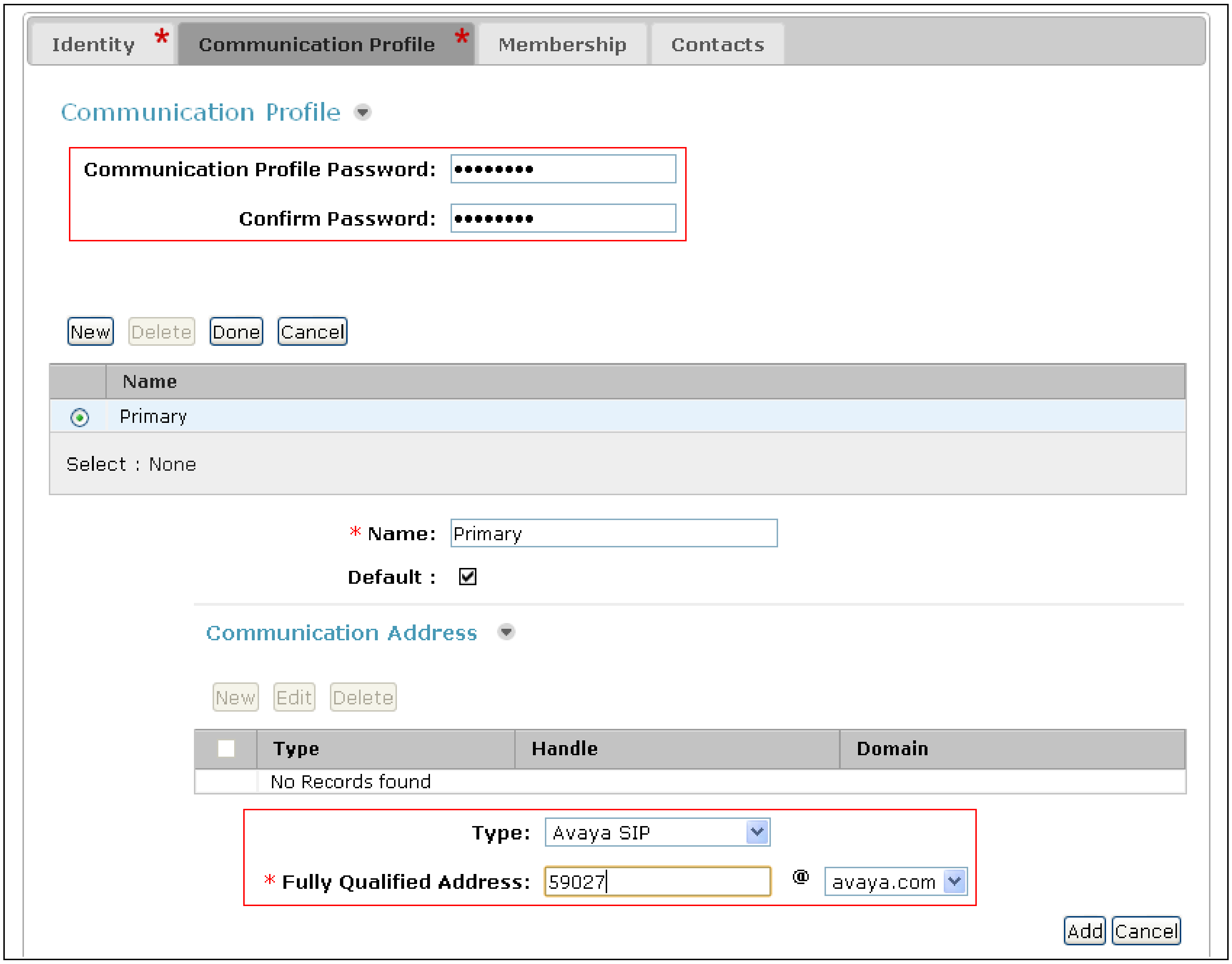
SIP extensions are registered with the Session Manager and use Communication Manager for their feature and configuration settings. From the Home tab select **User Management** from the menu. Then select **Manage Users** and click **New** (not shown).

On the **Identity** tab:

- Enter the user's name in the **Last Name** and **First Name** fields
- In the **Login Name** field enter a unique system login name in the form of user@domain (e.g. **59027@avaya.com**) which is used to create the user's primary handle
- The **Authentication Type** should be **Basic**
- In the **Password/Confirm Password** fields enter an alphanumeric password

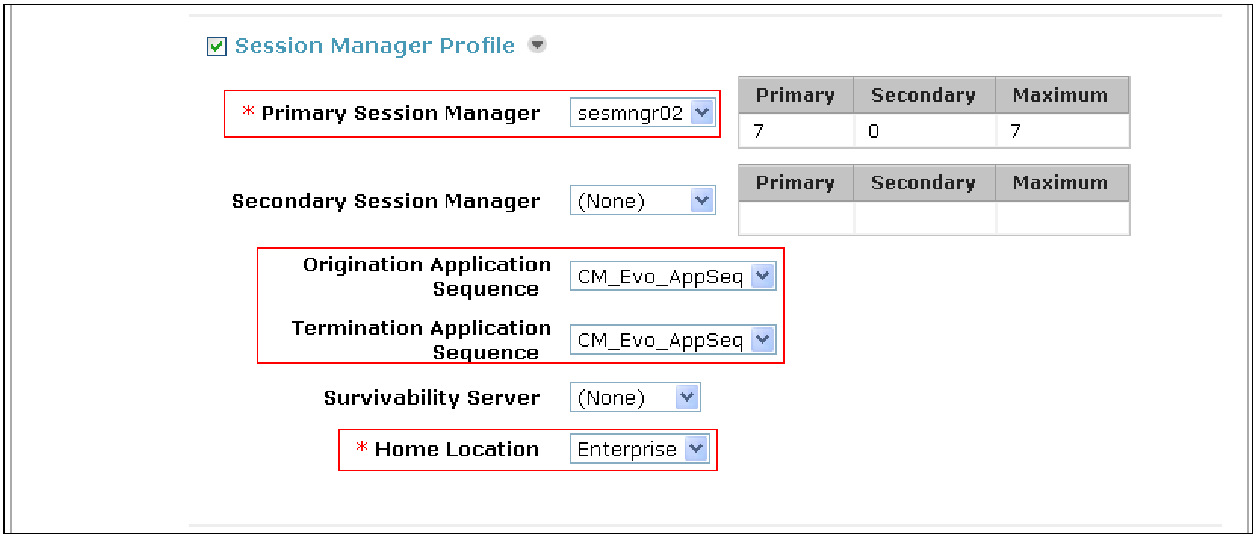


On the **Communication Profile** tab enter a numeric **Communication Profile Password** and confirm it, then click on the show/hide button for **Communication Address** and click **New.** For the **Type** field select **sip** from the drop-down menu. In the **Fully Qualified Address** field, enter an extension number and select the relevant domain from the drop-down menu. Click the **Add** button.



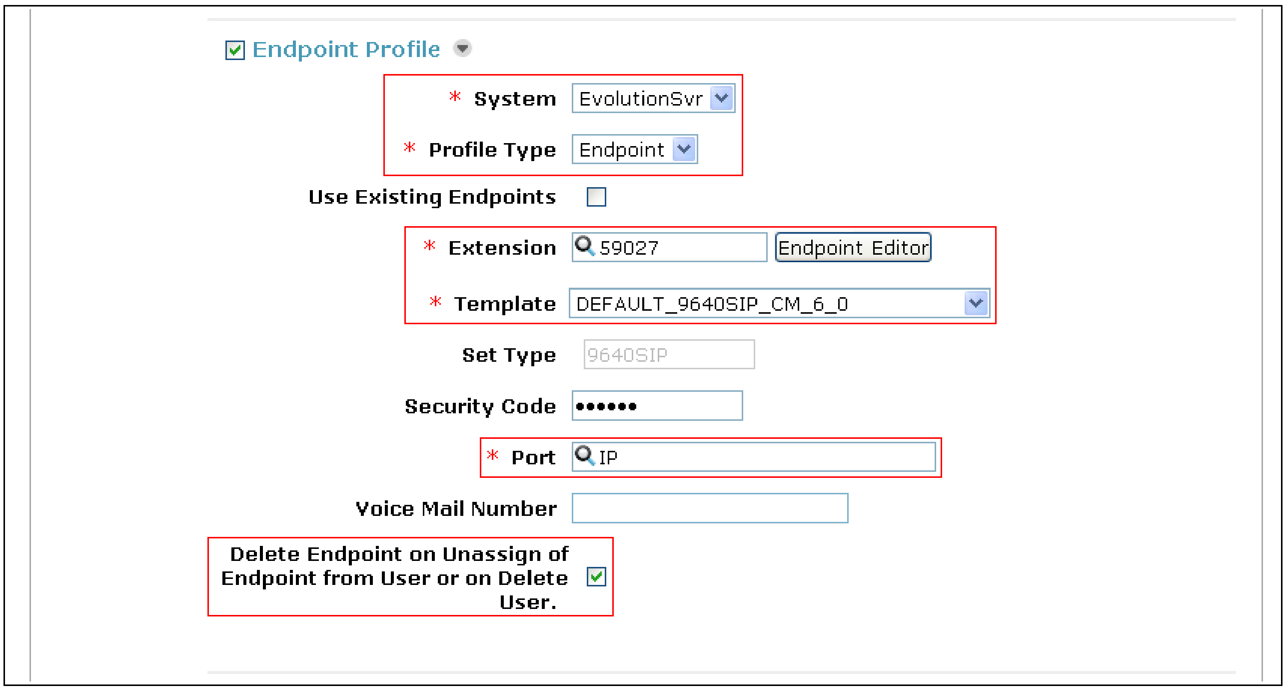
Click the show/hide button next to **Session Manager Profile:**

- Make sure the **Session Manager** check box is checked
- Select the appropriate Session Manager instance from the drop-down menu in the **Primary Session Manager** field
- Select the appropriate application sequence from the drop-down menu in the **Origination Application Sequence** field configured in **Section 6.10**
- Select the appropriate application sequence from the drop-down menu in the **Termination Application Sequence** field configured in **Section 6.10**
- Select the appropriate location from the drop-down menu in the **Home Location** field



Click the show/hide button next to **Endpoint Profile.**

- Select the Communication Manager SIP Entity from the **System** drop-down menu
- Select **Endpoint** from the drop-down menu for **Profile Type**
- Enter the extension in the **Extension** field
- Select the desired template from the **Template** drop-down menu
- For the **Port** field select **IP**
- Select the **Delete Endpoint on Unassign of Endpoint from User or on Delete User** check box
- Click **Commit** (not shown) to save changes and the System Manager will add the Communication Manager user configuration automatically



7. Configure Avaya Aura® Session Border Controller

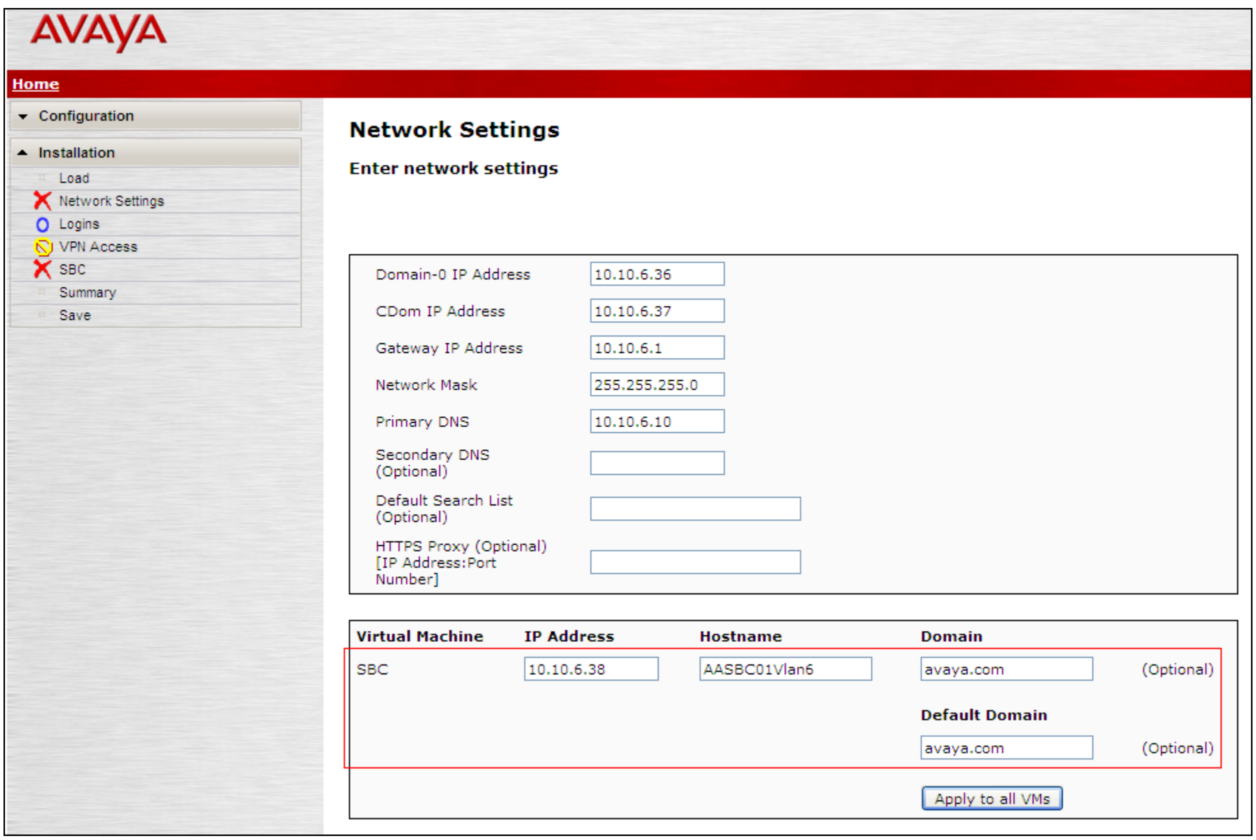
This section describes the configuration of the AASBC. This configuration is done in two parts. The first part is done during the AASBC installation via the installation wizard. These Application Notes will not cover the AASBC installation in its entirety but will include the use of the installation wizard. For information on installing the System Platform and the loading of the AASBC template see **[1]** & **[2]**.The second part of the configuration is done after the installation is complete using the AASBC web interface.

7.1. Installation Wizard

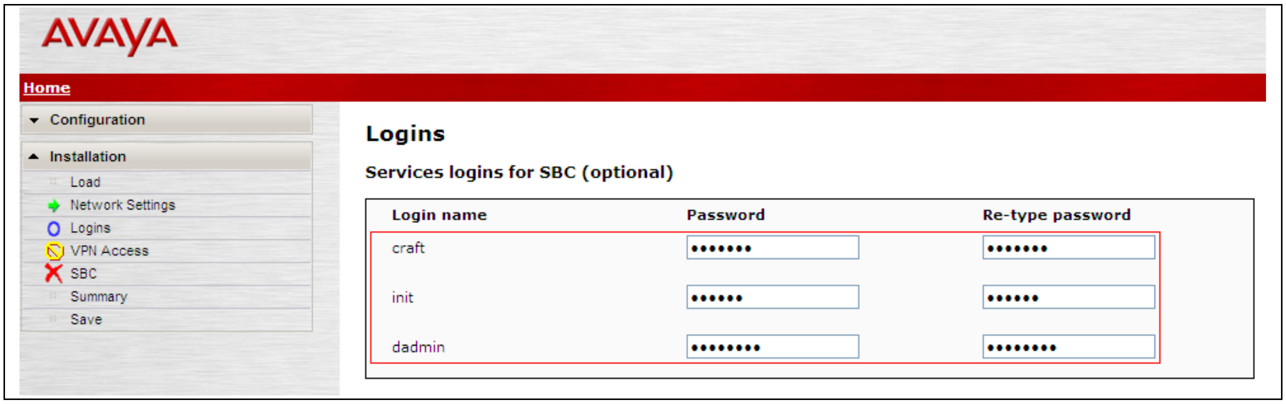
During the installation of the AASBC template, the installation wizard will prompt the installer for information that will be used to create the initial configuration of the AASBC. The first screen of the installation wizard is the Network Settings screen. Fill in the fields as described below and shown in the following screen:

- In the **IP Address** field enter the IP address of the private side of the AASBC
- In the **Hostname** field enter a host name for the AASBC
- Specify a domain in the **Domain** and **Default Domain** fields

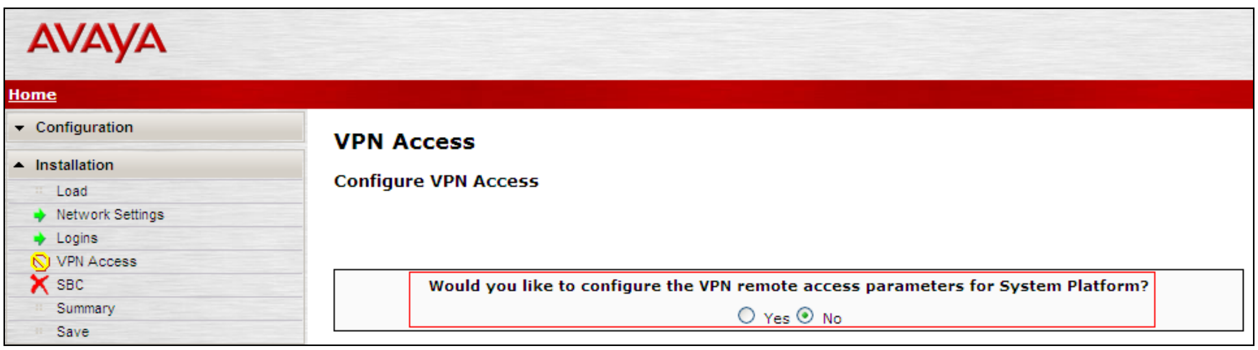
Click **Next Step** (not shown) to continue



From the **Logins** screen specify passwords for the services logins to the AASBC

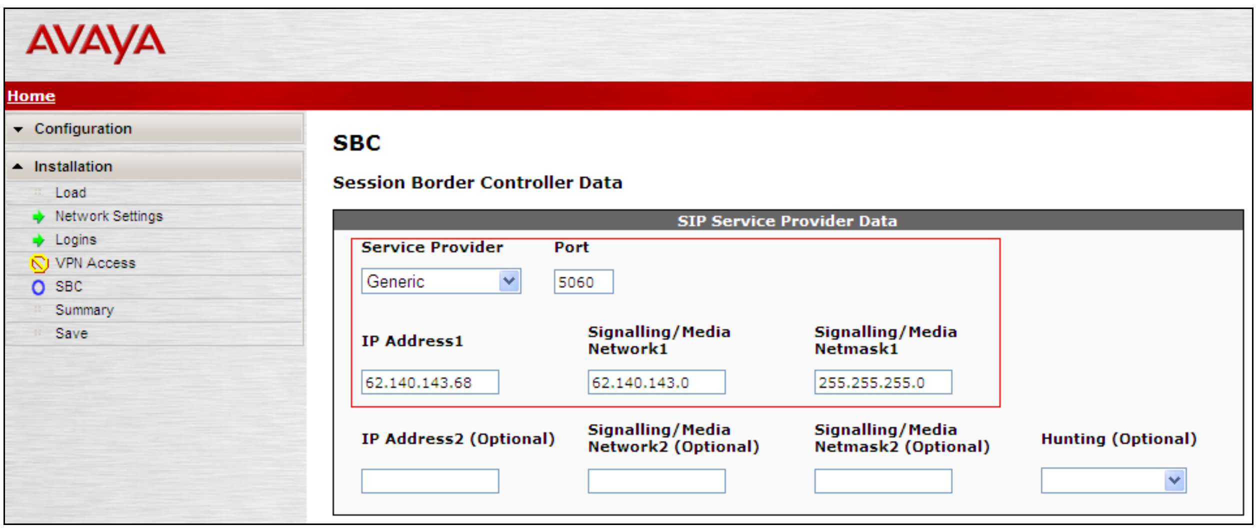


VPN remote access to the AASBC was not part of the compliance test. Thus, on the VPN Access screen, select **No** to the question, **Would you like to configure the VPN remote access parameters for System Platform?**



On the **SBC** screen, in the **SIP Service Provider Data** section fill in the fields as described below and shown in the following screen:

- In the **Service Provider** select the name of the Service Provider to which the AASBC will connect. This will allow the wizard to select a configuration file customized for this Service Provider. At the time of the compliance test, a customized configuration file did not exist for Vodafone Netherlands. Thus, **Generic** was chosen
- In the **Port** field enter the port number that Vodafone Netherlands uses to listen for SIP traffic
- In the **IP Address1** field enter the IP addresses provided by Vodafone Netherlands for the Vodafone Office Voice SIP Trunk Service (fixed). The IP address for the Vodafone OneVoice Corperate SIP Trunk Service (mobile) used during testing will be added after the AASBC template is installed **(Section 7.3)**
- In the **Signaling/Media Network1** field enter the Vodafone Netherlands provided subnet where media traffic will originate. An additional subnet can be provided for **Signaling/Media Network2**
- In the **Media Netmask** field enter the netmask corresponding to the Media Network
- Scroll down to continue



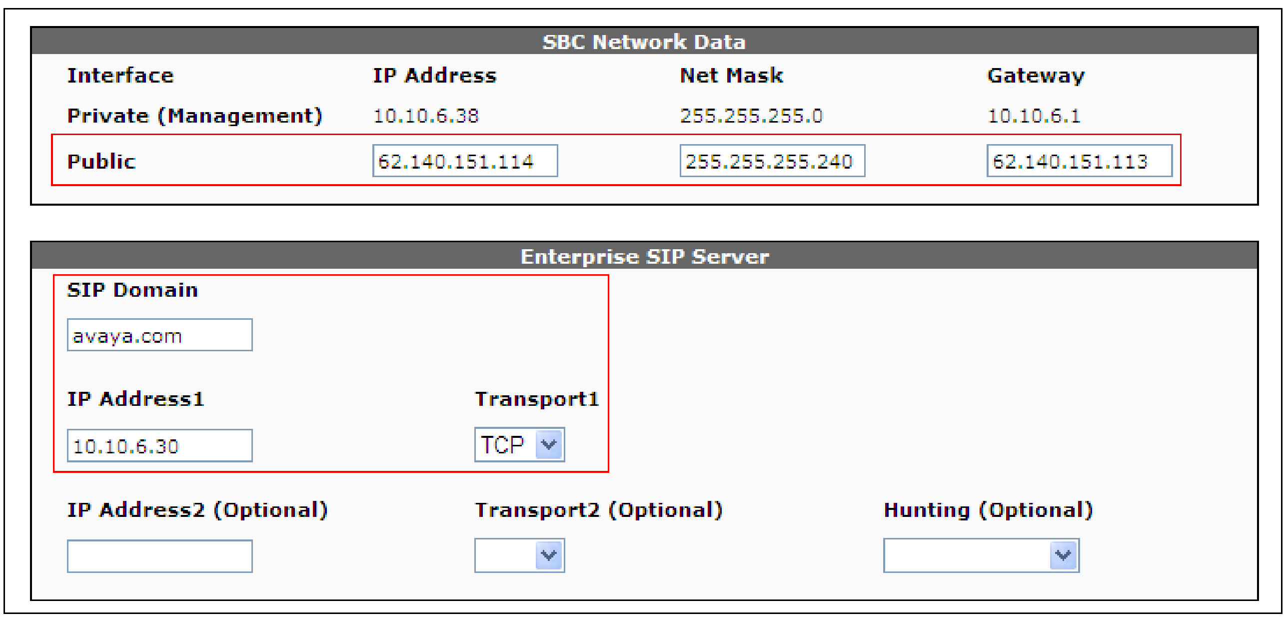
Further down on the same **SBC** screen, in the **SBC Network Data** section fill in the fields as described below:

- In the **Public IP Address** field enter the enterprsieIP address that will be used for the Vodafone Netherlands Office Voice SIP Trunk Service on the public side of the AASBC
- In the **Public Net Mask** field enter the netmask associated with the public network to which the AASBC connects
- In the **Public Gateway** field enter the default gateway of the public network

In the **Enterprise SIP Server** section fills in the fields as described below:

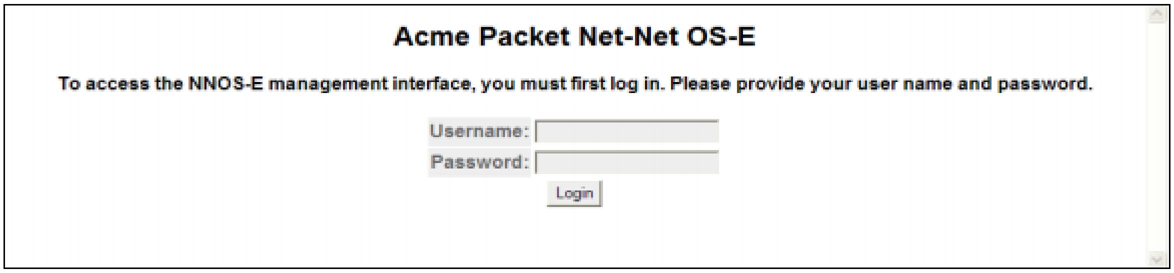
- In the **SIP Domain** field enter the enterprise SIP domain
- In the **IP Address** field enter the IP address of the Enterprise SIP Server to which the AASBC will connect. In the case of the compliance test, this is the IP address of the Session Manager SIP signaling interface
- In the **Transport1** field select the transport protocol to be used for SIP traffic between the AASBC and Session Manager

Click **Next Step** to continue. A summary screen will be displayed (not shown). Check the displayed values and click **Next Step** again to install the template with the values entered.



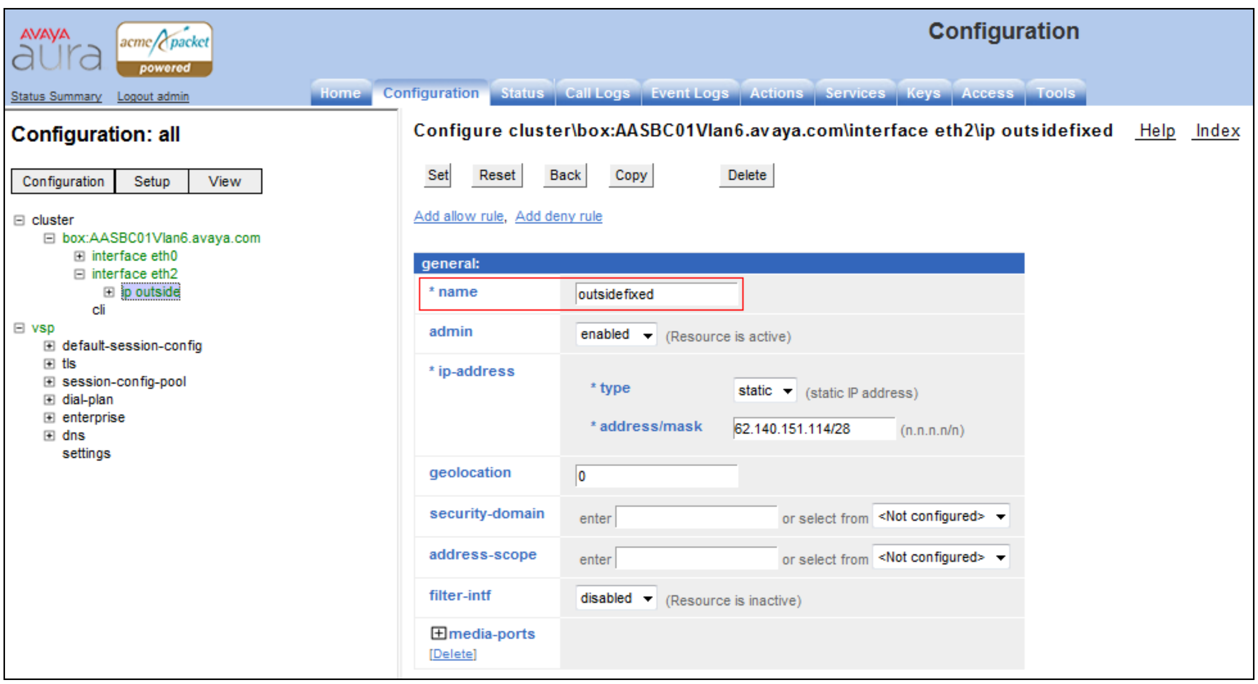
7.2. Access Avaya Aura® Session Border Controller

Access the AASBC using a web browser by entering the URL **https://<ip-address>**, where **<ip-address>** is the private IP address configured in **Section 7.1**. Log in with the appropriate credentials.

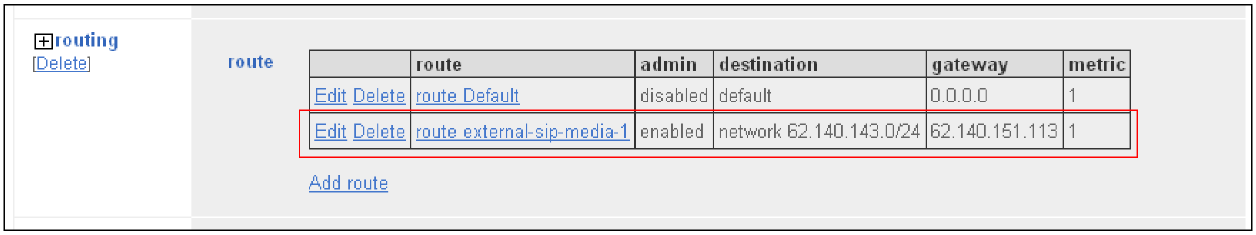


7.3. Configure Outside Interfaces

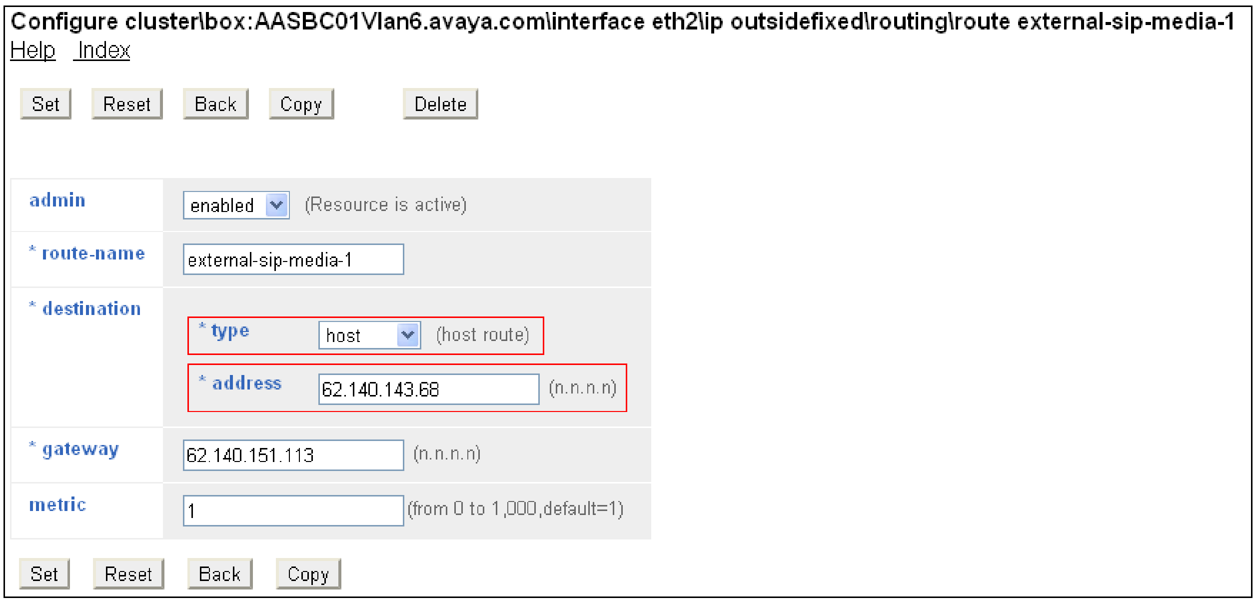
To allow two logical connections to be created between the enterprise and Vodafone Netherlands an additional IP address is created on the outside interface of the AASBC. Rename the IP address configuration created in **Section 7.1** by expanding **cluster → box:AASBC01Vlan6.avaya.com → interface eth2 → ip outside** and enter a descriptive name in the **name** field. The name **outsidefixed** is used as this is the IP address that will be used for the Vodafone Office Voice SIP Trunk Service. Scroll down to continue.



Further down on the same screen in the **routing** section click the edit link relating to the **route external-sip-media-1** route.



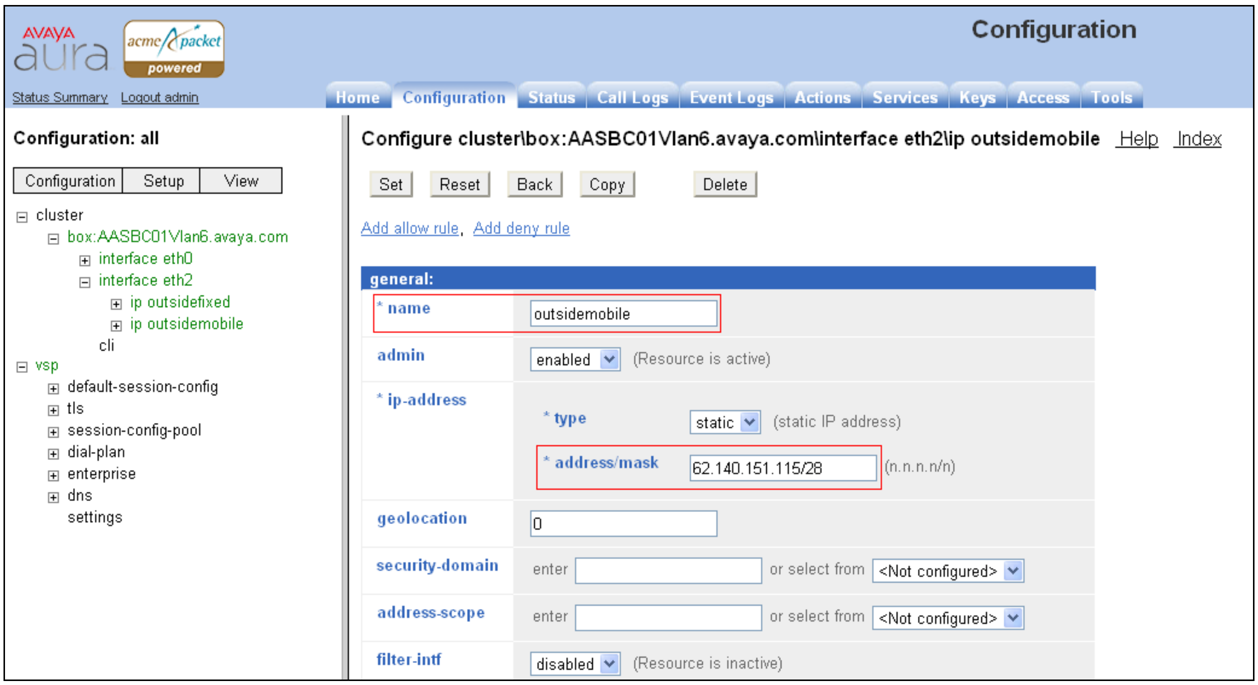
In the resulting screen in the **destination** section, select **host** from the **type** drop down menu. In the **address** field enter the IP address of the Vodafone Netherlands Office Voice SIP trunk service.



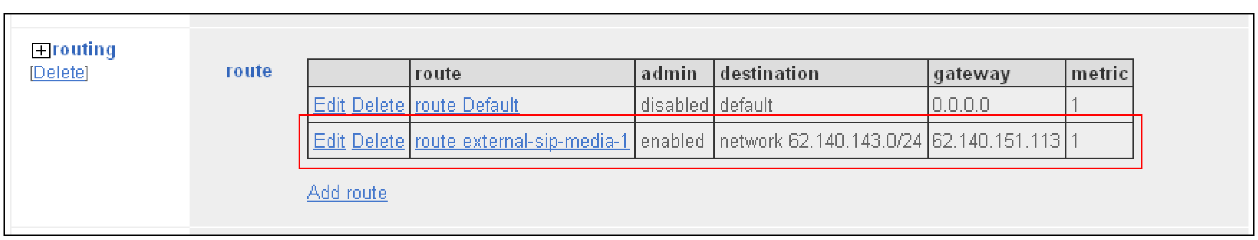
To create another IP address configuration navigate to **box:AASBC01Vlan6.avaya.com → interface eth2 → ip outsidefixed** and click **copy** (Not shown). In the resulting screen update the fields as shown below:

- In the **name** field enter a descriptive name. The name **outsidemobile** is used as this is the IP address that will be used for the Vodafone OneVoice Corporate SIP Trunk Service. Scroll down to continue.
- In the **address/mask** field enter the IP address that will be used on the public side of the AASBC for the Vodafone OneVoice Corporate SIP Trunk Service.

Scroll down to continue.



Further down on the same screen in the **routing** section click the edit link relating to the **route external-sip-media-1** route.



In the resulting screen in the **destination** section, select **host** from the drop down menu for **type**. In the **address** field enter the IP address of the Vodafone Netherlands OneVoice Corperate SIP trunk service.

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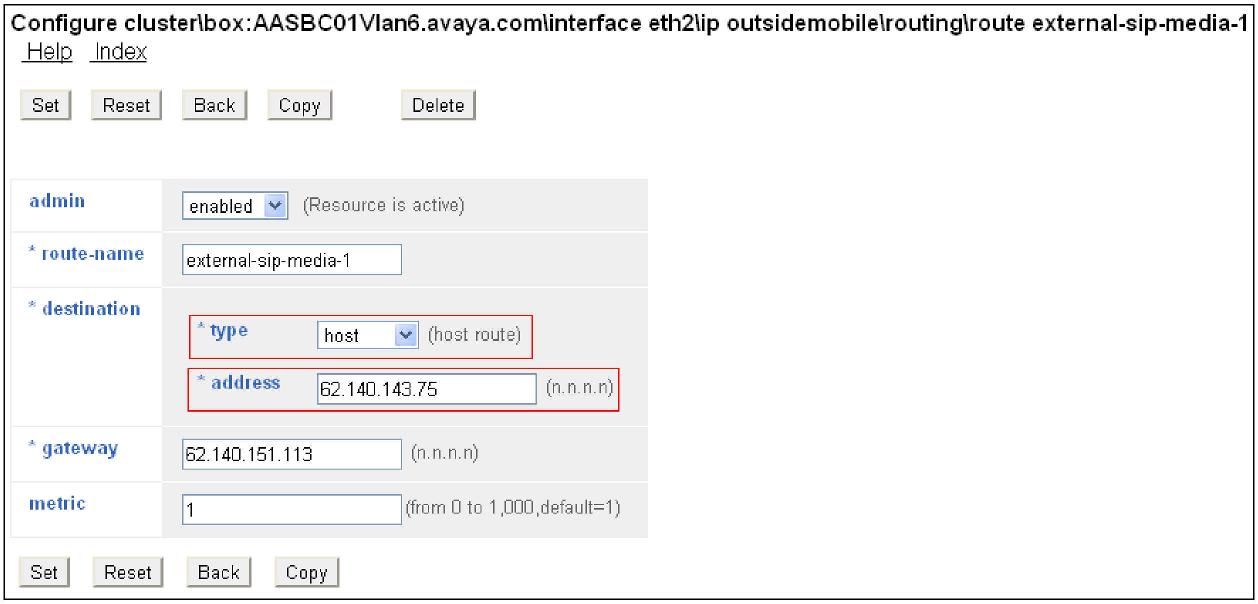
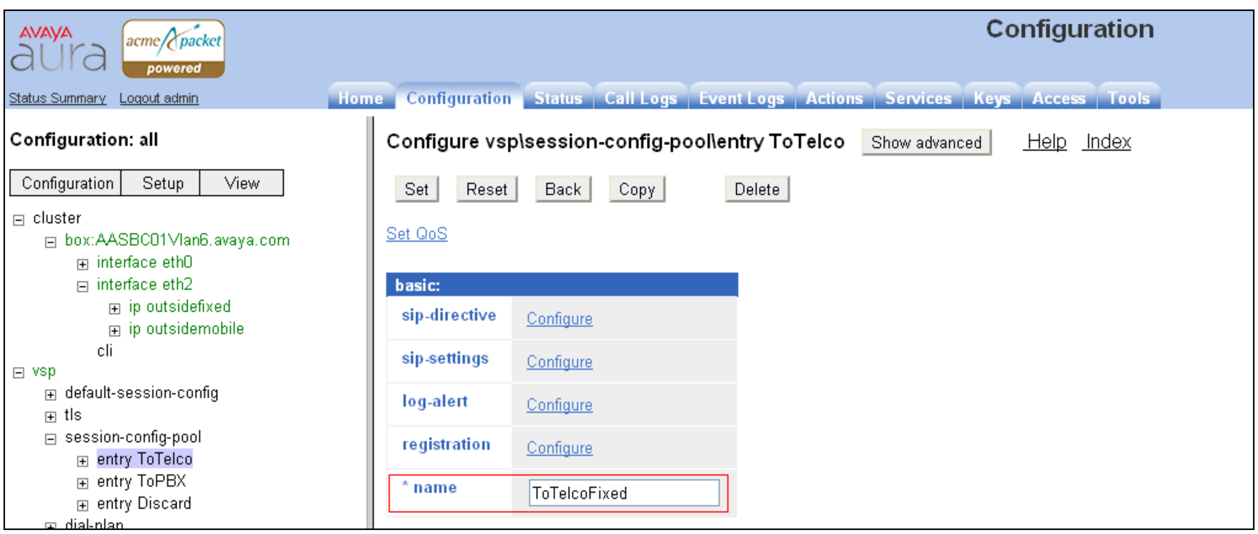


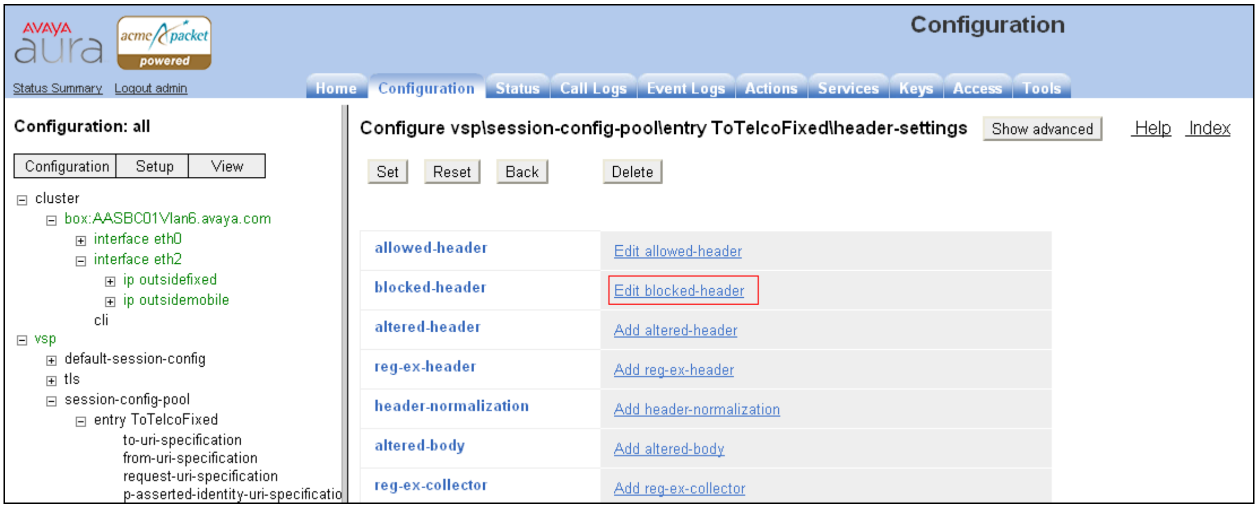
7.4. Session Config Pool

Navigate to **vsp** → **session-config-pool** → **entry ToTelco** and extend the entry in the **name** field to **ToTelcofixed**.

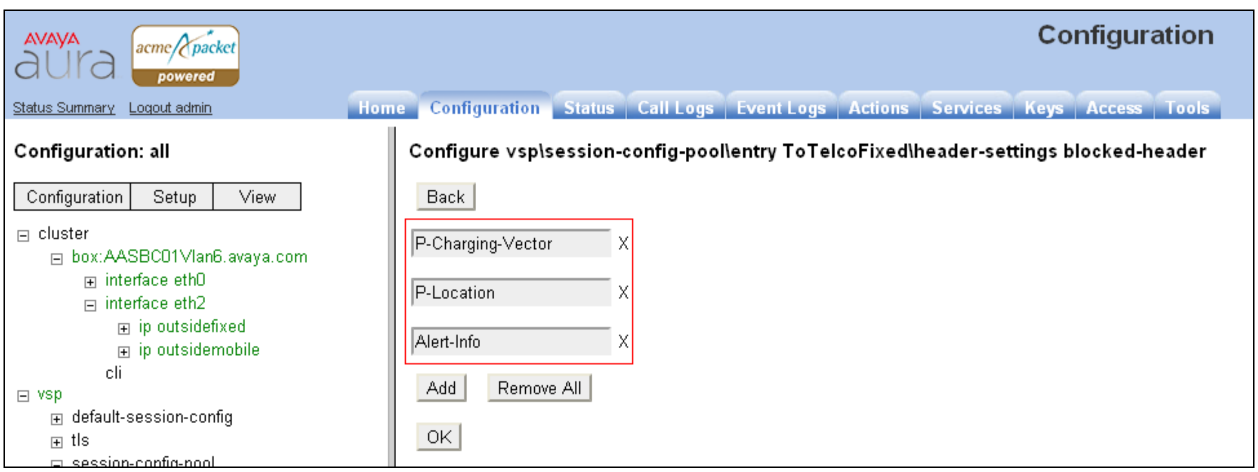


7.4.1. Stripping SIP Headers

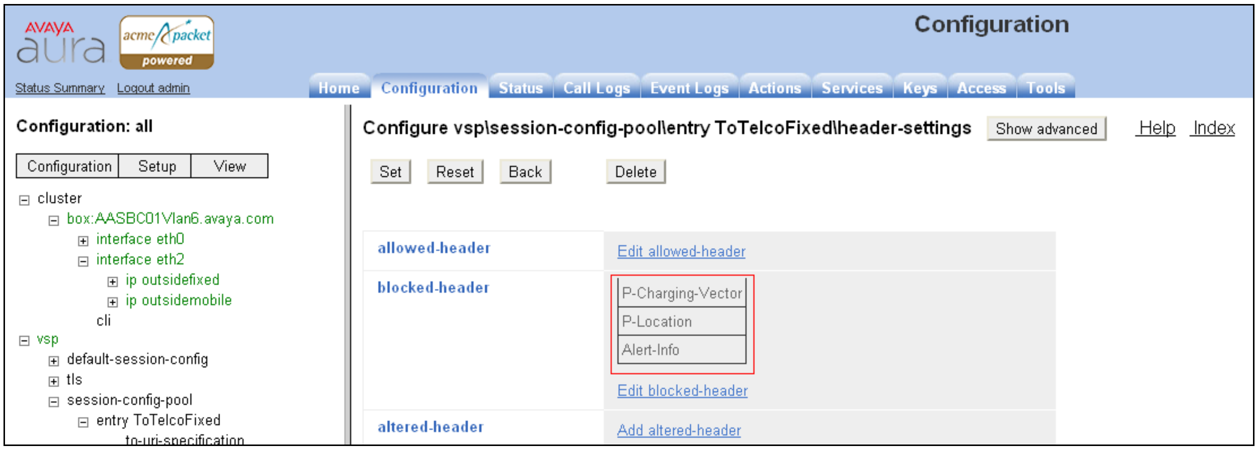
The AASBC can be used to strip SIP headers to prevent the header from being sent to the public SIP Service Provider. To strip a SIP header navigate to **vsp → session-config-pool → entry ToTelco → header-settings** and click on the **Edit blocked-header** link.



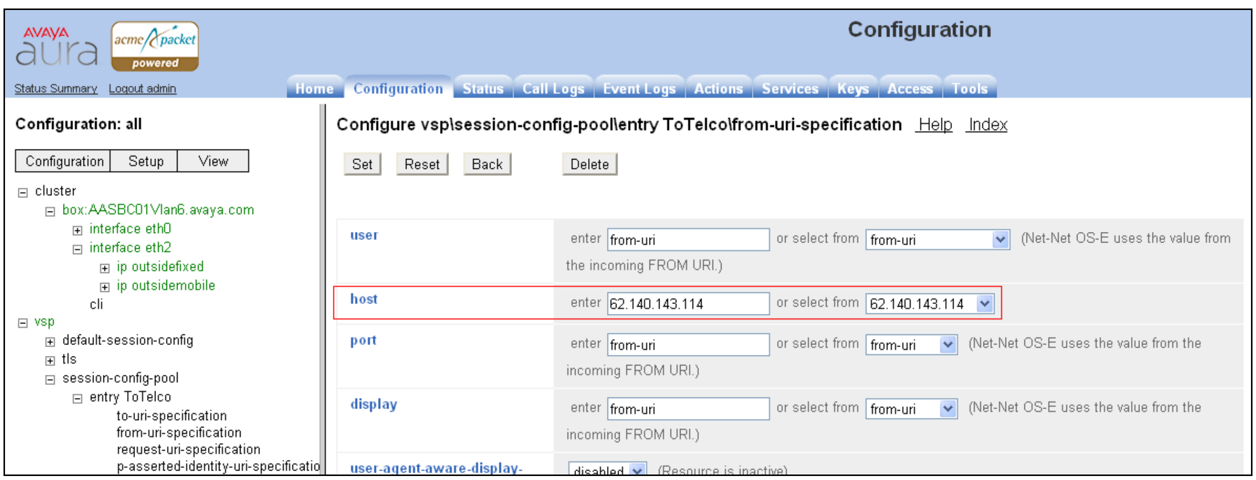
In the resulting page click the **Add** button to open a new entry field and enter the name of the header to be removed, repeat this action for all the headers to be removed. Click the **OK** button when finished.



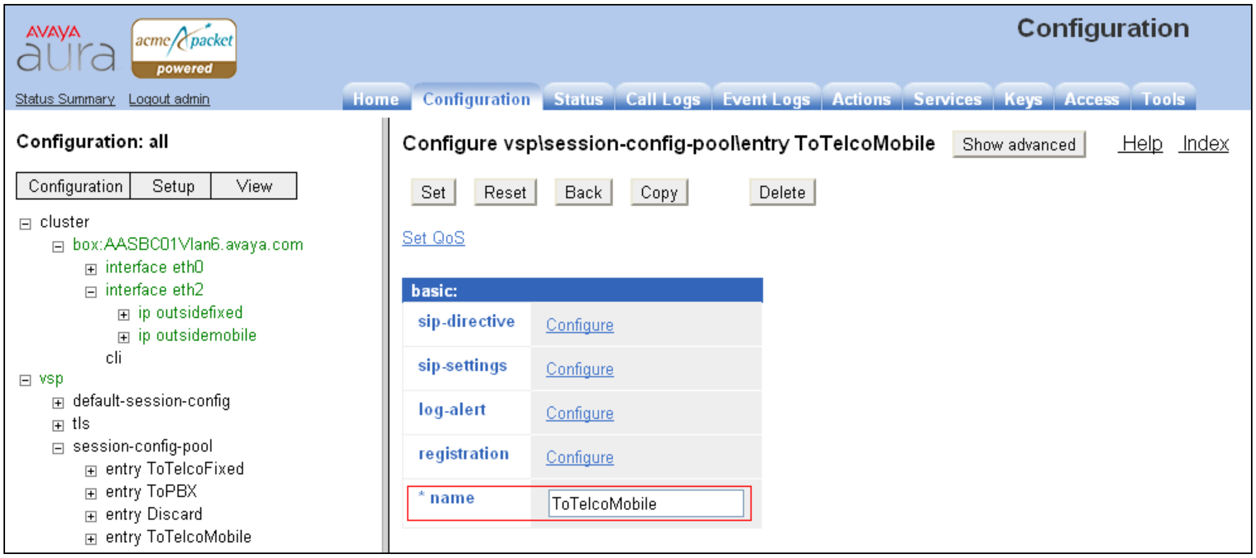
The following screen shows the headers being stripped during testing.



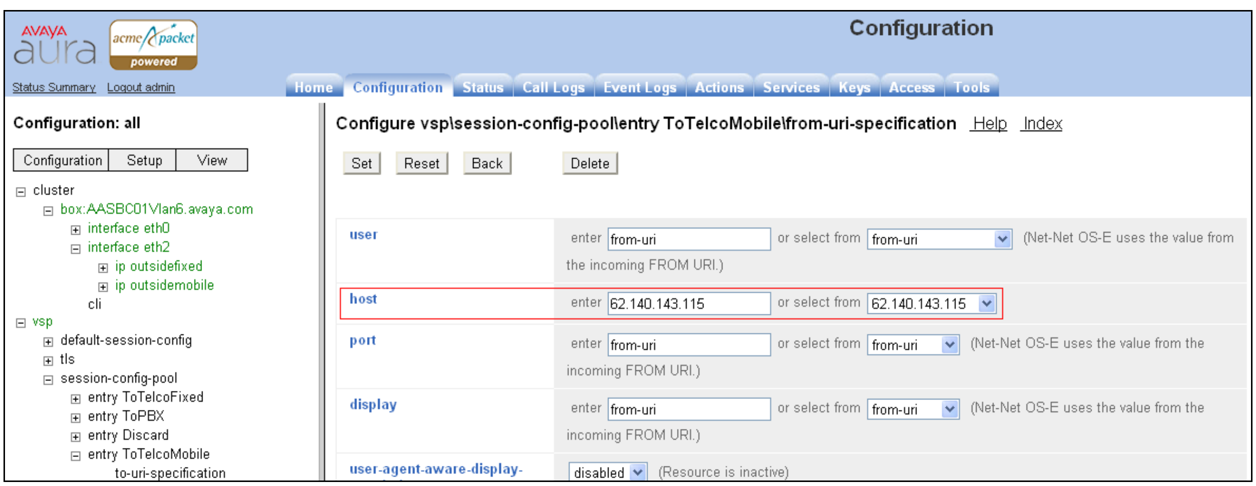
Navigate to **vsp → session-config-pool → entry ToTelco → from-uri-specification** and enter the IP address used on the public side of the AASBC for the Vodafone Netherlands Office Voice SIP trunk into the first host field. This will ensure that the host part of the From header is always set as the entered IP address. Click **Set** to save changes.



Navigate to **vsp → session-config-pool → entry ToTelco** and click **Copy** (not shown). This will produce an exact copy of the session config including the stripped SIP headers. In the resulting screen alter the entry in the **name** field to **ToTelcomobile**.

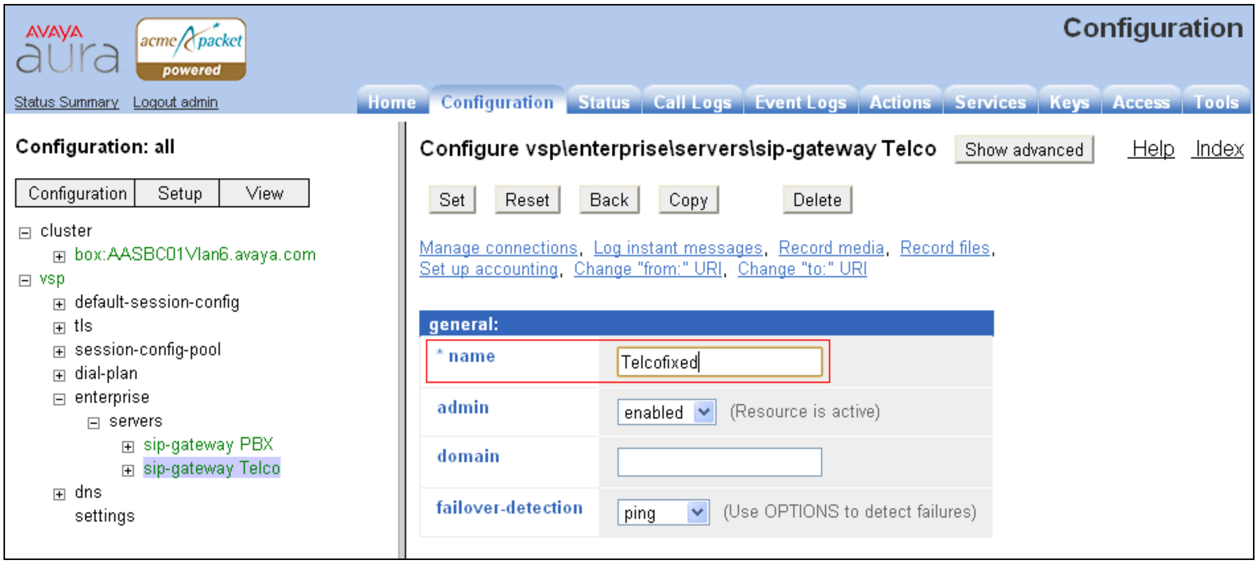


Navigate to **vsp → session-config-pool → entry ToTelcomobile → from-uri-specification** and enter the IP address used on the public side of the AASBC for the Vodafone Netherlands OneVoice Coporate SIP trunk into the first **host** field. This will ensure that the host part of the From header is always set as the entered IP address. Click **Set** to save changes.

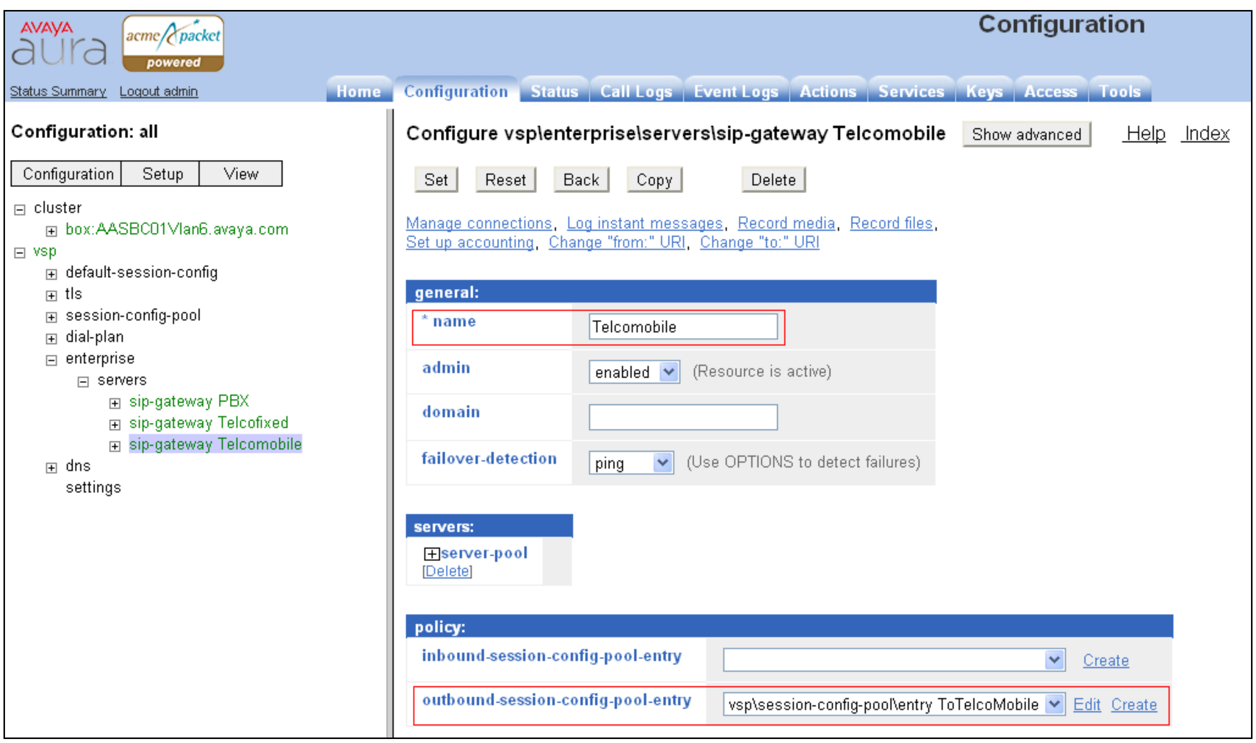


7.5. SIP Servers

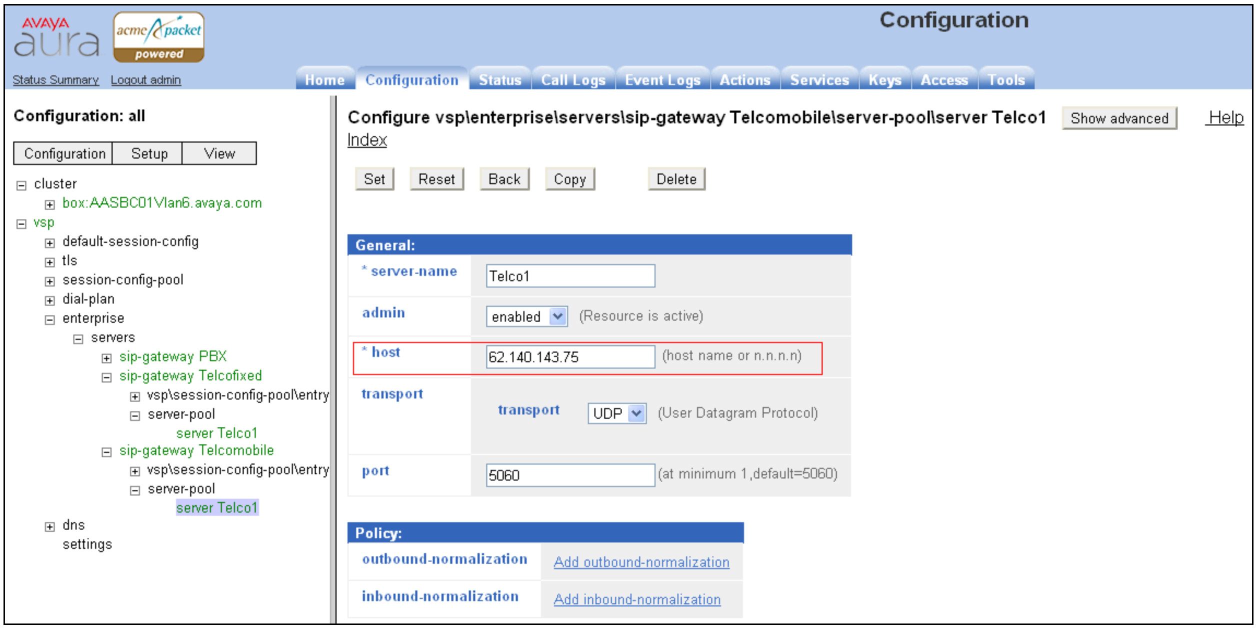
Navigate to **vsp → enterprise → servers → sip-gateway Telco** and alter the entry in the **name** field to **Telcofixed.** Click **Set** to save changes.



Navigate to **vsp → enterprise → servers → sip-gateway Telcofixed** and click **Copy** (Not shown). In the resulting screen alter the entry in the **name** field to **Telcomobile**. In the **outbound-session-config-pool-entry** field select the **ToTelcomobile** session config created in **Section 7.4** from the drop down menu. Click **Set** to save changes.



Navigate to **vsp → enterprise → servers → sip-gateway Telcomobile → server-pool → server Telco1** and enter the IP address provided by Vodafone Netherlands for the Vodafone Netherlands OneVoice Corporate SIP trunk connection in to the **host** field. Click **Set** to save changes.

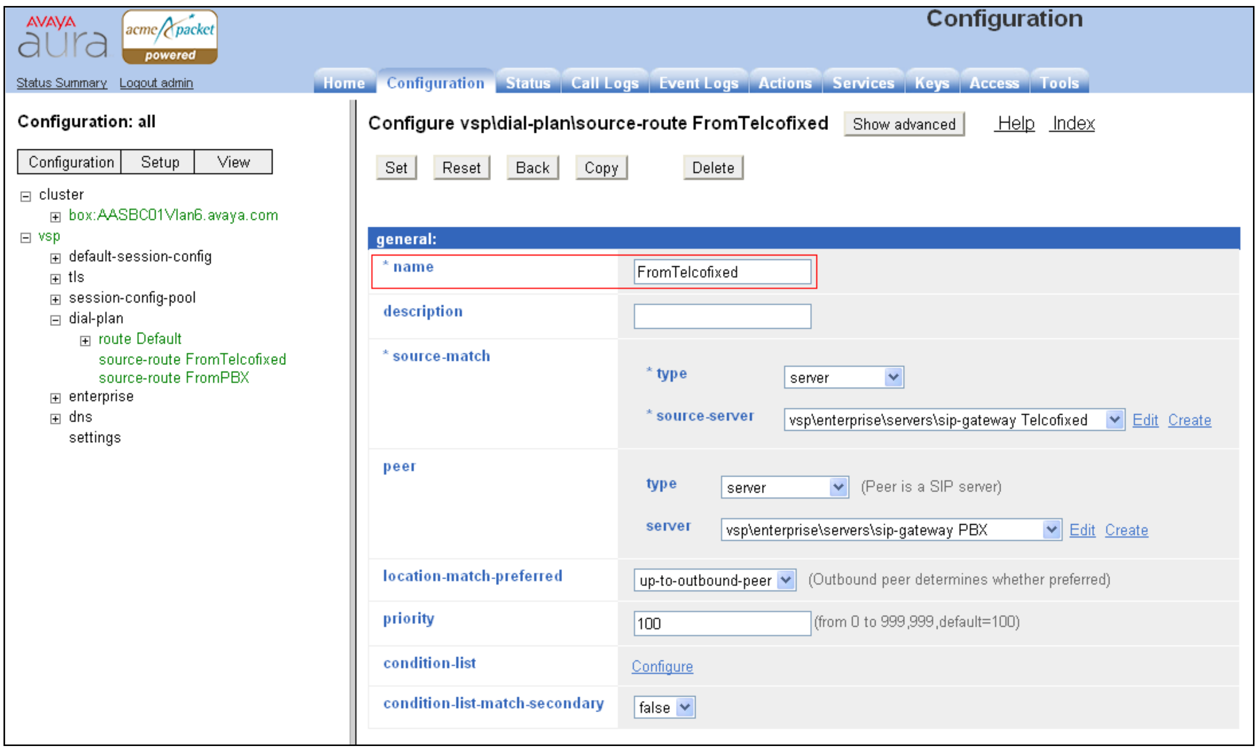


7.6. Dial Plan Configuration

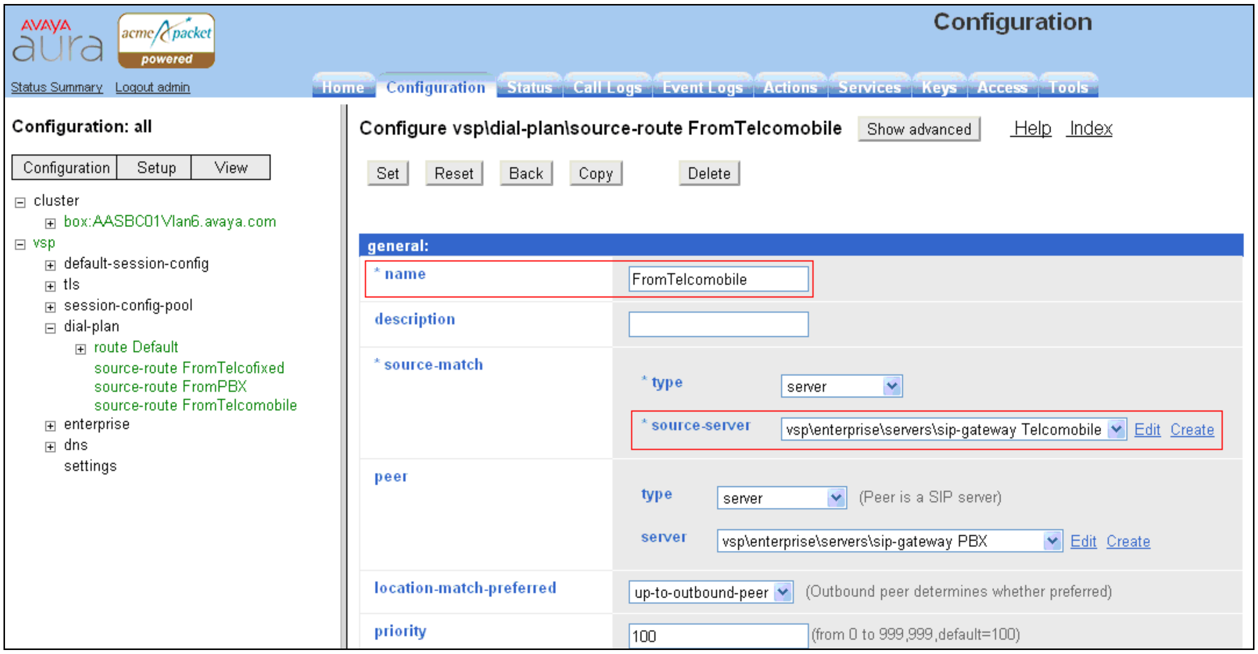
The dial plan is used to define how calls route between SIP entities. For the compliance test four routes are required.

- The route **FromTelcofixed** will be used to route fixed calls from Vodafone Netherlands to the Session Manager.
- The route **FromPBXfixed** will be used to route fixed calls from the Session Manager to Vodafone Netherlands.
- The route **FromTelcomobile** will be used to route mobile calls from Vodafone Netherlands to the Session Manager.
- The route **FromPBXmobile** will be used to route mobile calls from the Session Manager to Vodafone Netherlands.

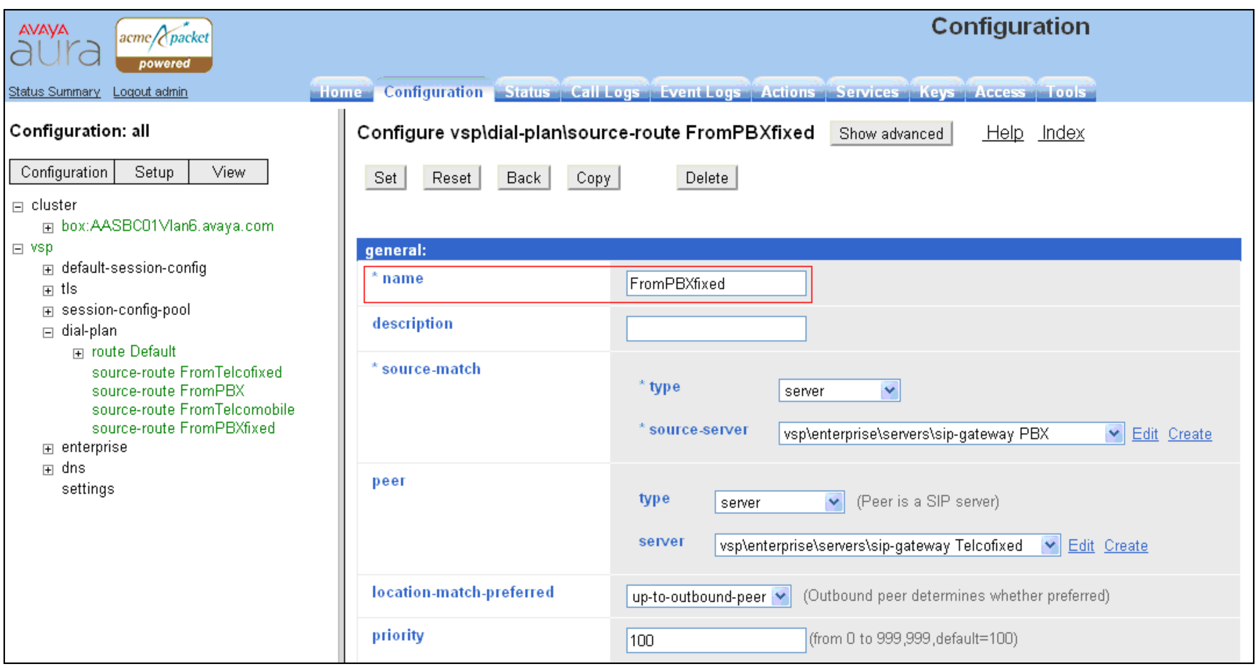
Navigate to **vsp → dial-plan → source-route FromTelco** and alter the entry in the **name** field to **FromTelcofixed.** Click **Set** to save changes.



Navigate to **vsp → dial-plan → source-route FromTelco** and click **Copy** (not shown). In the resulting screen alter the entry in the **name** field to **FromTelcomobile.** In the **source-server** field select the **Telcomobile** SIP server created in **Section 7.5.** Click **Set** to save changes.



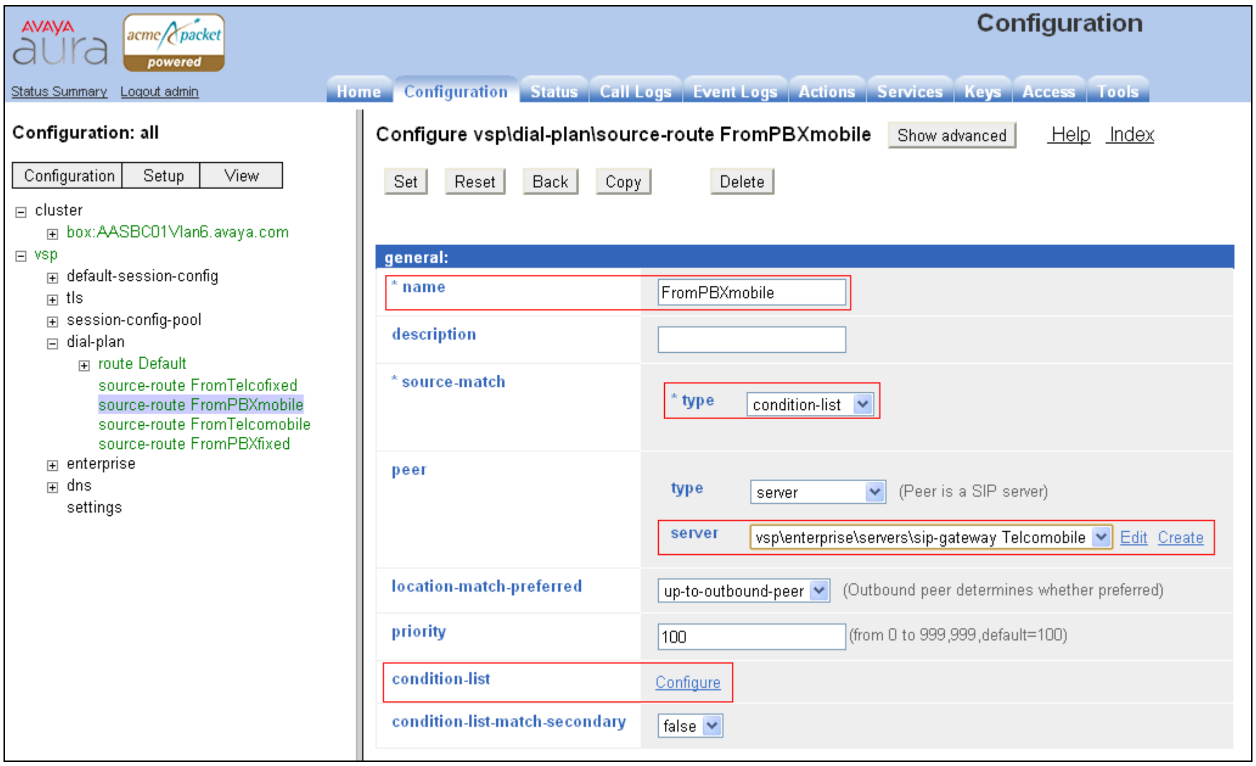
Navigate to **vsp → dial-plan → source-route FromTelco** and alter the entry in the **name** field to **FromPBXfixed.** Click **Set** to save changes.



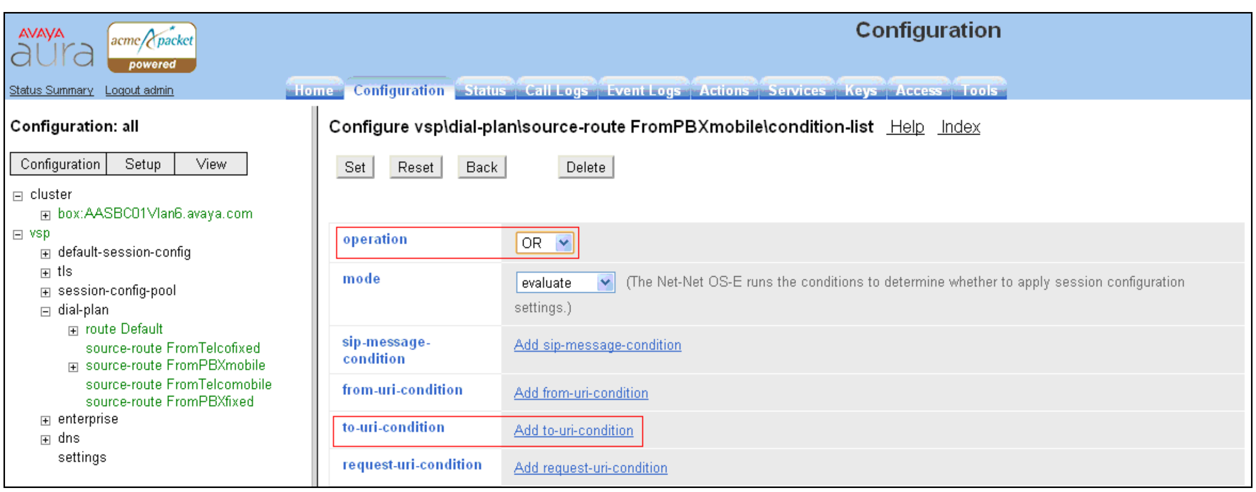
Navigate to **vsp → dial-plan → source-route FromPBXfixed** and click **Copy** (not shown). In the resulting screen update the fields as shown below:

- Alter the entry in the **name** field to **FromPBXmobile.**
- Under the **source-match** section, select **condition-list** from the drop down box in the **type** field.
- Under the peer section, in the **server** field select the **Telcomobile** SIP server created in **Section 7.5**

Click **Set** to save changes and then click the **configure** link under the **condition-list** section.



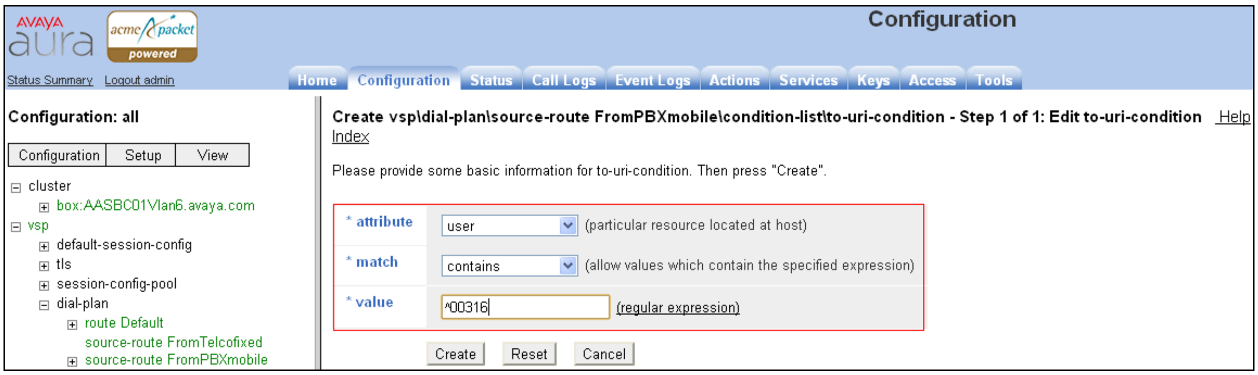
In the resulting screen select the **operation OR** from the drop down menu and click the **Add-to-uri-condition** link under the **to-uri–condition** section.



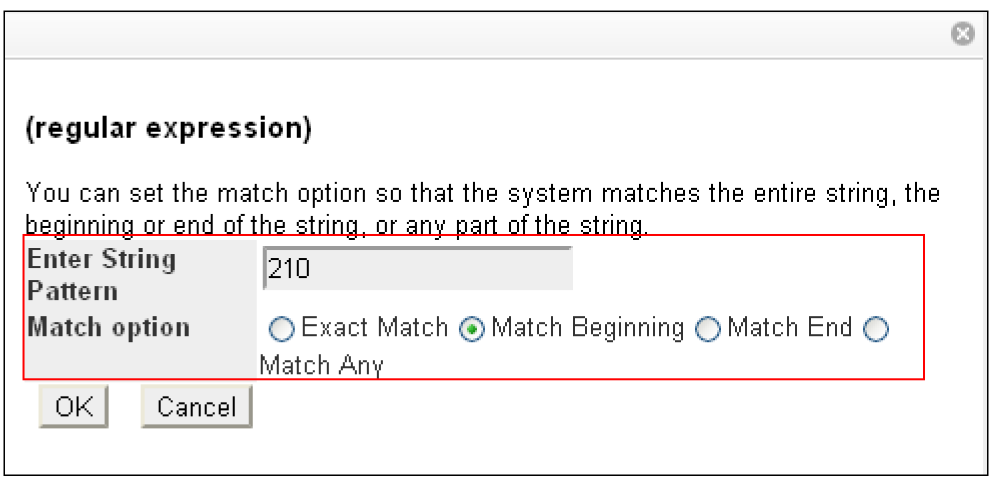
In the resulting screen define the dial patterns that the condition list should match by updating the fields as shown below:

- For **attribute** select **user** from the drop down menu. This means that the condition will try to match the user part of the uri.
- For **match** select **contains** from the drop down menu. This means that the condition list will match anything that contains the entry in the value field.
- In the **value** field enter the digits to match using regular expression.

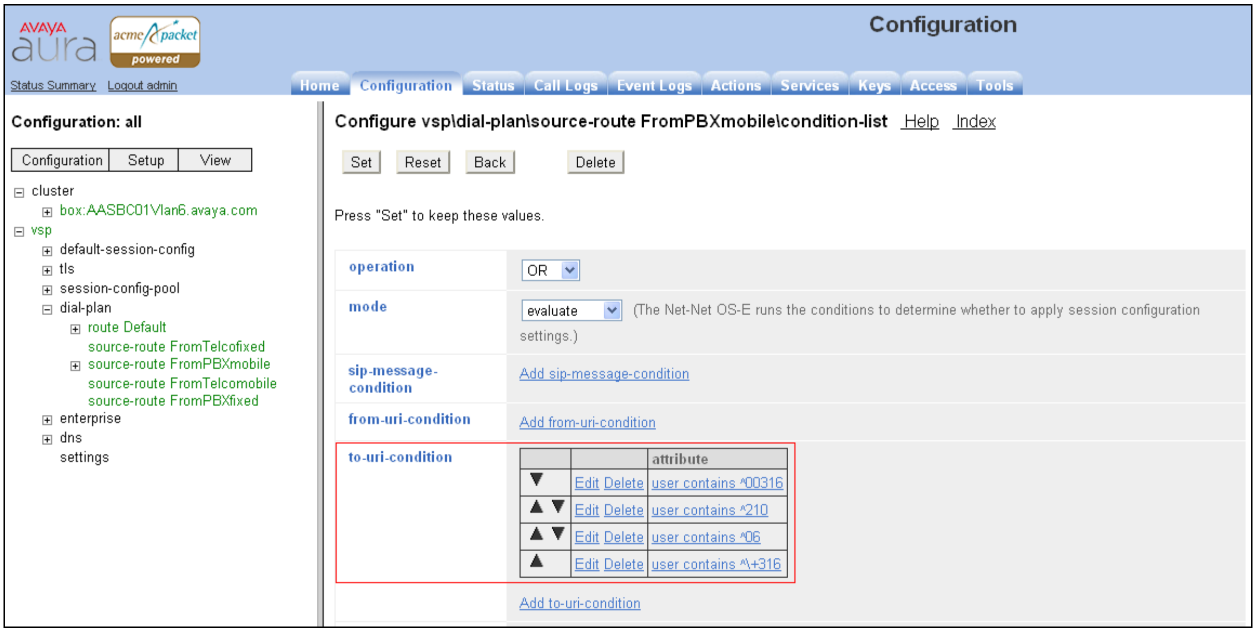
Click **Create** to save the condition.



The AASBC can be used to create the regular expression for the **value** field. Click the **(regular expression)** link next to the **value** field as seen in the previous screen. The following pop up box is displayed. Enter the digits to be matched and select the appropriate radio button for the type of match. The example below will match any digits beginning with 210, this will produce a regular expression of **^210**.

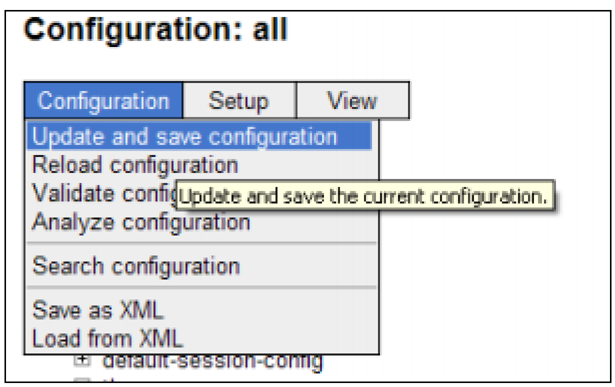


The following screen shows the to-uri-conditions used during the compliance test.



7.7. Save the Configuration

To save the configuration, click on **Configuration** in the left pane to display the configuration menu. Next, select **Update and save configuration**.



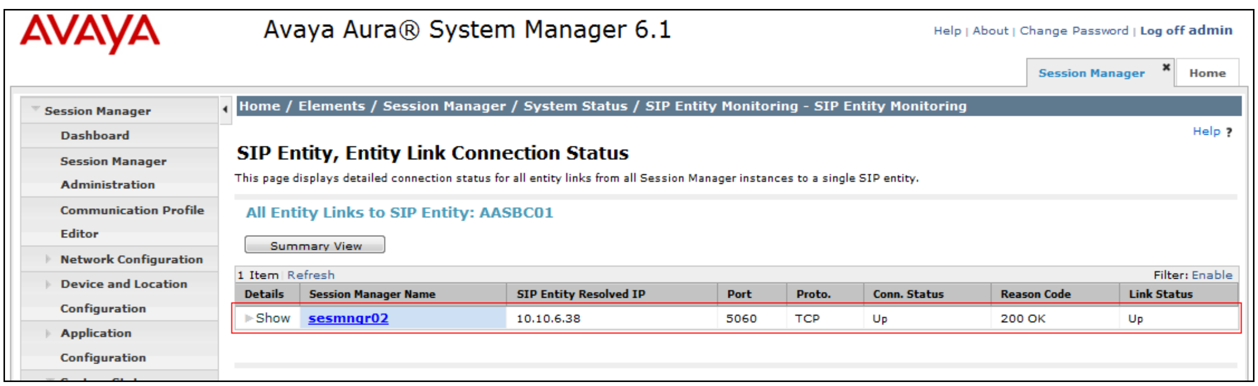
8. Verification Steps

This section provides steps that may be performed to verify that the solution is configured correctly.

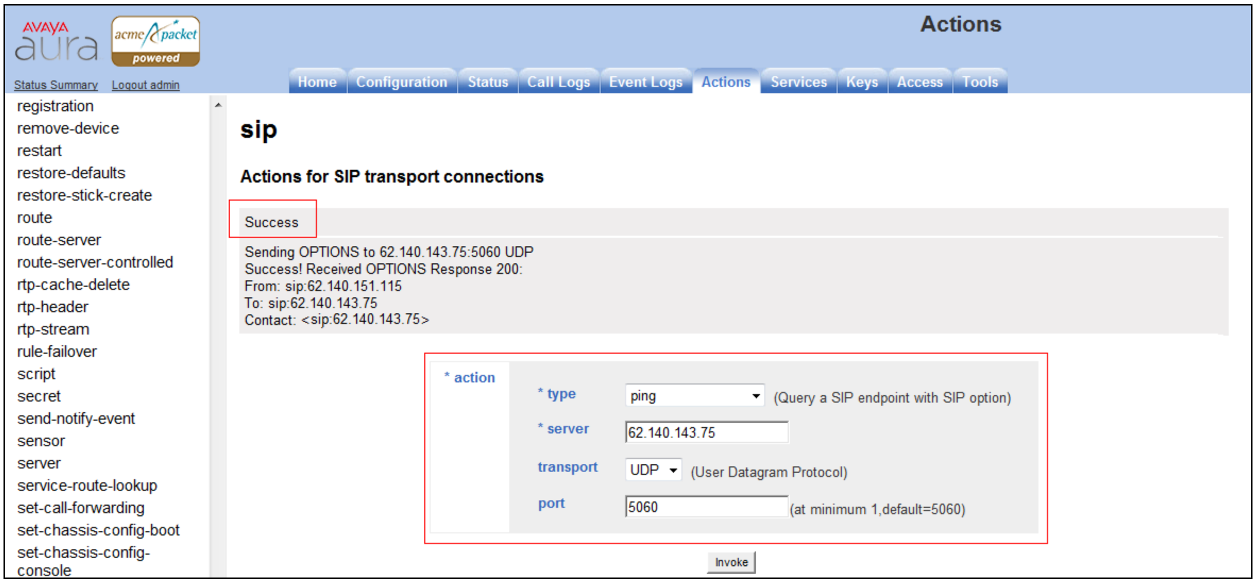
1. From the Communication Manager SAT interface run the command **status trunk n** where **n** is a previously configured SIP trunk. Observe if all channels on the trunk group display **In service/ idle.**

```
status trunk 2

                              TRUNK GROUP STATUS

Member   Port     Service State       Mtce Connected Ports
                                      Busy

0002/001 T00001   in-service/idle     no
0002/002 T00007   in-service/idle     no
0002/003 T00008   in-service/idle     no
0002/004 T00009   in-service/idle     no
0002/005 T00010   in-service/idle     no
```

2. From System Manager **Home** Tab click on **Session Manager** and navigate to **Session Manager → System Status → SIP Entity Monitoring**. Select the relevant SIP Entity from the list and observe if the **Conn Status** and **Link Status** are showing as **up.**



3. From the AASBC **Actions** tab it is possible to send a SIP OPTIONS message to a specified IP address to confirm the correct response. Select **sip** from the left hand menu and select **ping** from the drop down menu in the **type** field. Enter the required IP address in the **server** field and specify the appropriate **transport** type and **port**. Click **Invoke** and the result of the test are shown towards the top of the page.



4. Verify that endpoints at the enterprise site can place calls to the PSTN and that the call remains active.
5. Verify that endpoints at the enterprise site can receive calls from the PSTN and that the call can remain active.
6. Verify that the user on the PSTN can end an active call by hanging up.
7. Verify that an endpoint at the enterprise site can end an active call by hanging up.

9. Conclusion

These Application Notes describe the configuration necessary to connect Avaya Aura® Communication Manager, Avaya Aura® Session Manager and Avaya Aura® Session Border Controller to Vodafone Netherlands SIP Trunk Solution comprising of Vodafone Office Voice and Vodafone OneVoice Corporate. Vodafone Netherlands SIP Trunk Solution is a SIP-based Voice over IP solution providing businesses a flexible, cost-saving alternative to traditional hardwired telephony trunks. Vodafone Vodafone Netherlands SIP Trunk Solution comprising of Vodafone Office Voice and Vodafone OneVoice Corporate passed compliance testing. Please refer to **Section 2.2** for any observations or workarounds relating the testing covered by these Application Notes.

10. References

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

[1] *Installing and Configuring Avaya Aura® System Platform*, Release 6, June 2010.

[2] *Administering Avaya Aura® System Platform*, Release 6, June 2010.

[3] *Administering Avaya Aura® Communication Manager*, May 2009, Document Number 03-300509.

[4] *Avaya Aura® Communication Manager Feature Description and Implementation,* May 2009, Document Number 555-245-205.

[5] *Installing and Upgrading Avaya Aura® System ManagerRelease6.1*, November 2010.

[6] *Installing and Configuring Avaya Aura® Session Manager*, January 2011, Document Number 03-603473

[7] *Administering Avaya Aura® Session Manager,* March 2011, Document Number 03-603324.

[8] *Avaya Aura® Session Border Controller System Administration*, September 2010

[9] *Installing and Configuring Avaya Aura Session Border Controller*, May 2011

[8] RFC 3261 *SIP: Session Initiation Protocol,* http://www.ietf.org/

Additional Vodafone product documentation is available at http://www.vodafone.nl/zakelijk/totaal_oplossingen/vast_en_mobiel/

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.