Avaya Solution & Interoperability Test Lab

Application Notes for Configuring Avaya Aura® Communication Manager R6.0.1 as an Evolution Server, Avaya Aura® Session Manager R6.1 and Avaya Aura® Session Border Controller to support Vodafone DE SIP Trunking Service for IP-PBX- Issue 1.0

Abstract

These Application Notes describe the steps used to configure Session Initiation Protocol (SIP) trunking between the Vodafone DE SIP Trunking service and an Avaya SIP enabled Enterprise Solution. The Avaya solution consists of Avaya Aura® Session Border Controller, Avaya Aura® Session Manager and Avaya Aura® Communication Manager as an Evolution Server. Vodafone DE is 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 used to configure Session Initiation Protocol (SIP) trunking between Vodafone DE SIP Trunking service for IP-PBX and an Avaya SIP-enabled Enterprise Solution. The Avaya solution consists of Avaya Aura® Session Border Controller Avaya Aura® Session Manager and Avaya Aura® Communication Manager Evolution Server. Customers using this Avaya SIP-enabled enterprise solution with Vodafone DE SIP trunking service are able to place and receive PSTN calls via a dedicated Internet connection and the SIP protocol. This converged network solution is an alternative to traditional PSTN trunks. This approach generally results in lower cost for the Enterprise customer.

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 Session Border Controller. The enterprise site was configured to use the SIP Trunk Service provided by Vodafone DE.

2.1. Interoperability Compliance Testing

The interoperability test included the following:

- Incoming calls to the enterprise site from the PSTN routed to the DDI numbers assigned by Vodafone DE
- Incoming PSTN calls made to SIP, H.323 and Digital telephones at the enterprise
- Outgoing calls from the enterprise site completed via Vodafone DE to PSTN destinations
- Outgoing calls from the enterprise to the PSTN made from SIP, H.323 and Analogue telephones
- Calls using G.729, and G.711A codecs
- Fax calls to/from a group 3 fax machine to a PSTN connected fax machine using pass-through mode
- 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 DE requiring Avaya response and sent by Avaya requiring Vodafone DE response

2.2. Test Results

Interoperability testing of the sample configuration was completed with successful results for the Vodafone DE SIP trunking service with the following observations:

- No inbound toll free numbers were tested as none were available from the Service Provider
- No Emergency services numbers tested as test calls to these numbers should be pre-arranged with the Operator
- G.711mu is not offered by Vodafone DE SIP trunking service and thus incoming calls were not tested
- Outgoing calls succeed where G.711mu is offered by the Avaya equipment
- G.711 pass-through was tested for fax as T.38 is not supported by Vodafone DE SIP trunking

2.3. Support

For technical support on Vodafone DE products please visit the website at www.vodafone.de or contact an authorized Vodafone DE representative.

3. Reference Configuration

Figure 1 illustrates the test configuration. The test configuration shows an Enterprise site connected to the Vodafone DE SIP Trunking Service. Located at the Enterprise site is a Session Border Controller, 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), Avaya 16xx series IP telephones (with SIP firmware) Avaya analogue telephones and an analogue fax machine. Also included in the test configuration was a one-X® Communicator soft phone running on a laptop PC configured for H.323.

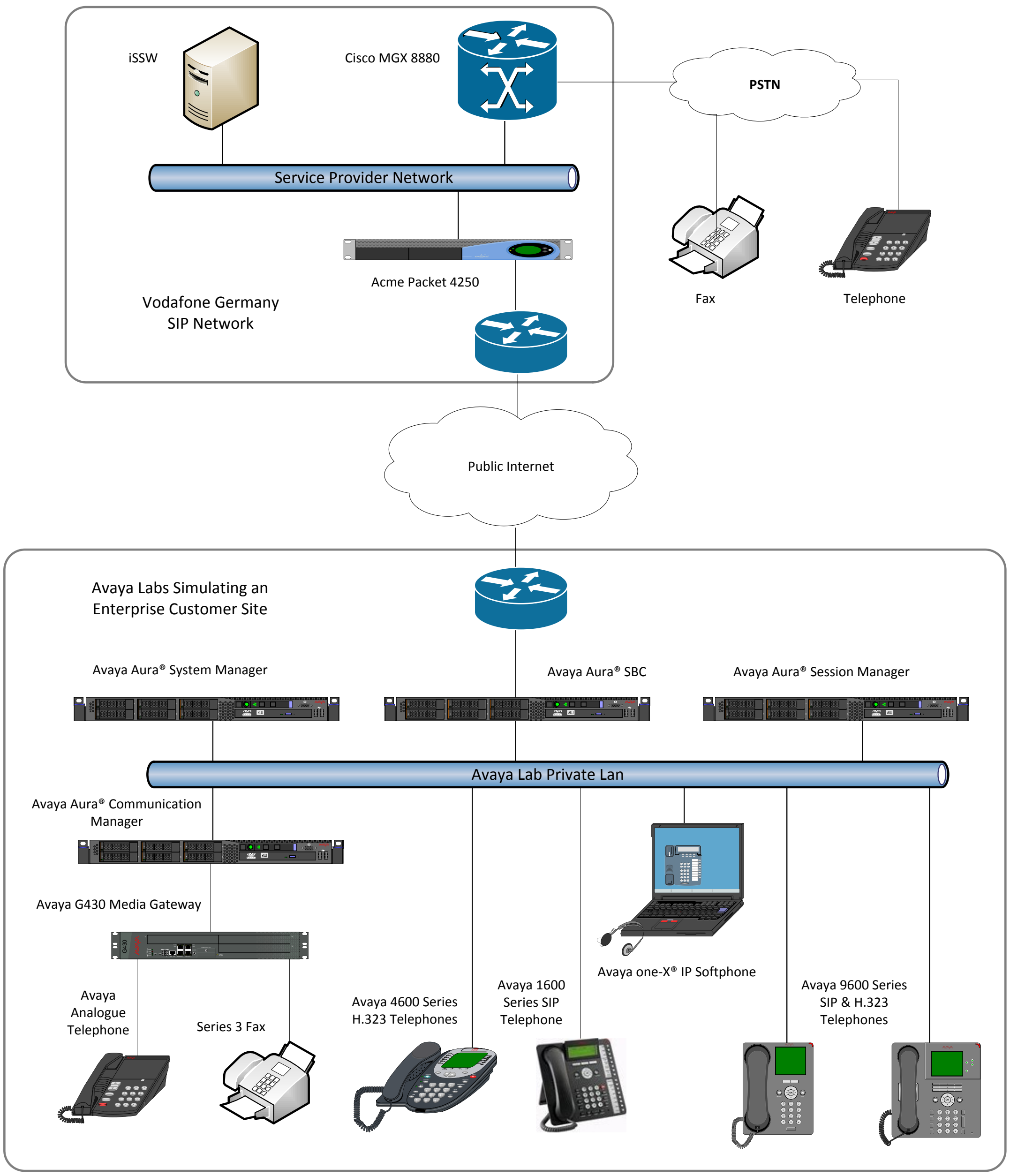


Figure 1: Test Setup Vodafone DE SIP Trunking to Avaya Enterprise

4. Equipment and Software Validated

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

4.1. Avaya Enterprise

Equipment	Software
Avaya S8800 Server	Avaya Aura® Communication Manager R6.0.1 (R016x.00.1.510.1) Service Pack 19009 (System Platform 6.0.3.1.3)
Avaya G430 Media Gateway	FW 30.12.1
Avaya S8800 Server	Avaya Aura® Session Manager R6.1 (6.1.0.0.610023)
Avaya S8800 Server	Avaya Aura® System Manager R6.1 (System Platform 6.0.3.1.3, Template 6.1.5.0)
Avaya S8800 Server	Avaya Aura® Session Border Controller 6.1 (System Platform 6.0.3.0.3, Template E362P4)
Avaya 1616 Phone (H.323)	1.22
Avaya 4621 Phone (H.323)	2.901
Avaya 9670 Phone (H.323)	2.0
Avaya 9601 Phone (SIP)	1.0.11.3
Avaya one–X® Communicator (H.323)	Avaya one–X® Communicator 6.0.1.16-SP1-25226
Analogue Phone	N/A

4.2. Vodafone DE SIP Trunking Service:

Equipment	Software
Softswitch: iSSW	20.50.34-olxi
SBC: Acme Packet 4250	6.1.0 MR5
MG: Cisco MGX 8880	5.5(10.204)P4

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 the Vodafone DE SIP Trunking Service. For incoming calls, the Session Manager receives SIP messages from the SBC 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 Session Border Controller at the enterprise site that then sends the SIP messages to the Vodafone DE 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 S8800 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 DE 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 0
           Maximum Concurrently Registered IP Stations: 18000 3
             Maximum Administered Remote Office Trunks: 12000 0
Maximum Concurrently Registered Remote Office Stations: 18000 0
              Maximum Concurrently Registered IP eCons: 414   0
  Max Concur Registered Unauthenticated H.323 Stations: 100   0
                        Maximum Video Capable Stations: 18000 0
                   Maximum Video Capable IP Softphones: 18000 0
                       Maximum Administered SIP Trunks: 24000 10
  Maximum Administered Ad-hoc Video Conferencing Ports: 24000 0
   Maximum Number of DS1 Boards with Echo Cancellation: 522   0
                             Maximum TN2501 VAL Boards: 128   0
                     Maximum Media Gateway VAL Sources: 250   1
           Maximum TN2602 Boards with 80 VoIP Channels: 128   0
          Maximum TN2602 Boards with 320 VoIP Channels: 128   0
   Maximum Number of Expanded Meet-me Conference Ports: 300   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? n
                  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, **SM100** and **10.10.9.61** are the **Name** and **IP Address** for the Session Manager SIP interface. 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
SM100               10.10.9.61
default             0.0.0.0
procr               10.10.9.52
procr6              ::
```

5.3. Administer IP Network Region

Use the **change ip-network-region 1** 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 (**yes**) to allow audio traffic to be sent directly between endpoints without using gateway VoIP resources. When a PSTN call is shuffled the enterprise endpoint will talk directly to the public interface of the Vodafone DE Session Border Controller.
- 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: default
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
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
```

5.4. Administer IP Codec Set

Open the **IP Codec Set** form for the codec set specified in the **IP Network Region** form, **Section 5.3**. 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 DE were configured, namely **G.711A**, **G.726A-32K** and **G.729A**. 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.711A            n         2        20
 2: G.726A-32K        n         2        20
 3: G.729A            n         2        20
```

Vodafone DE SIP Trunk Service does not support the T.38 fax protocol. Fax pass-through was tested using G.711. Navigate to **Page 2** to configure fax pass-through by setting 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 trunk group) will be used for inbound and outbound PSTN calls to Vodafone DE SIP Trunking Service. During test, this was configured to use TCP and port 5060 to facilitate tracing and fault analysis. It is recommended however, to use TLS (Transport Layer Security) and the default TLS port of 5061 for security. Configure the **Signaling Group** using the **add signaling-group x** command as follows:

- Set **Group Type** to **sip**
- Set **Transport Method** to **tcp**
- Set **Peer Detection Enabled** to **y** allowing the Communication Manager to automatically detect if the peer server is a Session Manager
- Set **Near-end Node Name** to the processor interface (node name **procr** as defined in the **IP Node Names** form shown in **Section 5.2**)
- Set **Far-end Node Name** to the Session Manager (node name **SM100**as defined in the **IP Node Names** form shown in **Section 5.2**)
- Set **Near-end Listen Port** and **Far-end Listen Port** to 5060 (recommended TCP port value)
- Set **Far-end Network Region** to the IP Network Region configured in **Section 6.2.** (logically establishes the **far-end** for calls using this signaling group as network region **1**)
- Leave **Far-end Domain** blank (removes the analysis of the far end domain name and subsequent handling of multiple signaling groups where it is not required)
- Set **Direct IP-IP Audio Connections** to **y**
- Leave **DTMF over IP** at default value of **rtp-payload**(Enables **RFC2833**for DTMF transmission from the Communication Manager)

The default values for the other fields may be used.

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

 Group Number: 1              Group Type: sip
  IMS Enabled? n        Transport Method: tcp
        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: SM100
 Near-end Listen Port: 5060                 Far-end Listen Port: 5060
                                         Far-end Network Region: 1

Far-end Domain:
                                             Bypass If IP Threshold Exceeded? n
Incoming Dialog Loopbacks: eliminate                  RFC 3389 Comfort Noise? n
         DTMF over IP: rtp-payload             Direct IP-IP Audio Connections? 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 1                                              Page   1 of  21
                                TRUNK GROUP

Group Number: 1                   Group Type: sip          CDR Reports: y
  Group Name: Group 1                    COR: 1        TN: 1        TAC: 101
   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: 1
                                                     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 DE to prevent unnecessary SIP messages during call setup. Also note that the value for **Redirect On OPTIM Failure** can be increased to allow additional set-up time for calls destined for an EC500 destination.

```
add trunk-group 1                                              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
```

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

```
add trunk-group 1                                                    Page   3 of  21
TRUNK FEATURES
          ACA Assignment? n            Measured: none
                                                          Maintenance Tests? y

                     Numbering Format: public
                                                UUI Treatment: service-provider

                                                 Replace Restricted Numbers? n
                                                Replace Unavailable Numbers? n
```

On **Page 4,** set the **Convert 180 to 183 for Early Media** to **n.** If the 183 Session Progress message is received by Vodafone DE SIP Trunking, ring tone is expected by the terminating equipment. The Avaya G430 Media gateway does not play ring tone, so none is heard by the caller. The default value for **Network Call Redirection** is **n**, but the setting is changed on the test system to facilitate testing of User to User Information.

```
add trunk-group 1                                                    Page   4 of  21
                              PROTOCOL VARIATIONS

                      Mark Users as Phone? n
            Prepend '+' to Calling Number? n
      Send Transferring Party Information? n
                 Network Call Redirection? y
                    Send Diversion Header? n
                  Support Request History? y
             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: P-Asserted-Identity
                             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, individual stations are mapped to send numbers allocated from the Vodafone DE DDI range supplied. This calling party number will be sent in the SIP From, Contact and PAI 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: 5
 4  2291          1         49691382698104  14       Maximum Entries: 9999
 4  2296          1         49691382698103  14
 4  2316          1         49691382698105  14    Note: If an entry applies to
 4  2346          1         49691382698102  14    a SIP connection to Avaya
 4  2396          1         49691382698101  14    Aura(tm) Session Manager,
                                                  the resulting number must
```

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 DE SIP Trunking 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: *69
                       Answer Back Access Code:
                         Attendant 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** 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 **0 or 00**. Calls are sent to route pattern **1**.

```
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
    0                     9    14     1         pubu          n
    00                    10   17     1         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 **1** is used to route calls to trunk group **1**.

```
change route-pattern 1                                          Page   1 of   3
                    Pattern Number: 1   Pattern Name: all calls
                             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: 1    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                              intl-pub  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 DDI calls to the proper Communication Manager extension(s). The incoming digits sent in the INVITE message from Vodafone DE can be manipulated as necessary to route calls to the desired extension. In the examples used in the compliance testing, the incoming DDI numbers provided by Vodafone DE correlate to the internal extensions assigned within Communication Manager. The entries displayed below translate incoming DDI numbers 0691382698101 - 0691382698110 to a 4 digit extension by deleting all of the incoming digits and inserting an extension.

```
change inc-call-handling-trmt trunk-group 1                     Page   1 of  30
                      INCOMING CALL HANDLING TREATMENT
 Service/      Number   Number      Del Insert
 Feature        Len      Digits
 tie             13 0691382698101    all 2396
 tie             13 0691382698102    all 2346
 tie             13 0691382698103    all 2296
 tie             13 0691382698104    all 2291
 tie             13 0691382698105    all 2316
 tie             13 0691382698106    all 6101
 tie             13 0691382698107    all 2000
 tie             13 0691382698108    all 2400
 tie             13 0691382698109    all 6102
 tie             13 0691382698110    all 2501
```

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 2396. Use the command **change off-pbx-telephone station mapping x** where **x** is the Communication Manager station.

- The **Station Extension** field will automatically populate with station extension
- For **Application** enter **EC500**
- Enter a **Dial Prefix** (e.g., 9) if required by the routing configuration
- For the **Phone Number** enter the phone that will also be called (e.g. **01723456789**)
- Set the **Trunk Selection** to **1** so that Trunk Group 1 will be used for routing
- Set the **Config Set** to **1**
- Other parameters can retain default value

```
change off-pbx-telephone station-mapping 2396                   Page   1 of   3
                 STATIONS WITH OFF-PBX TELEPHONE INTEGRATION

 Station       Application Dial   CC  Phone Number    Trunk        Config  Dual
 Extension                 Prefix                     Selection    Set     Mode
 2396             EC500         -     01723456789      1           1
                                -
```

Save Communication Manager changes by entering **save translation** to make them permanent.

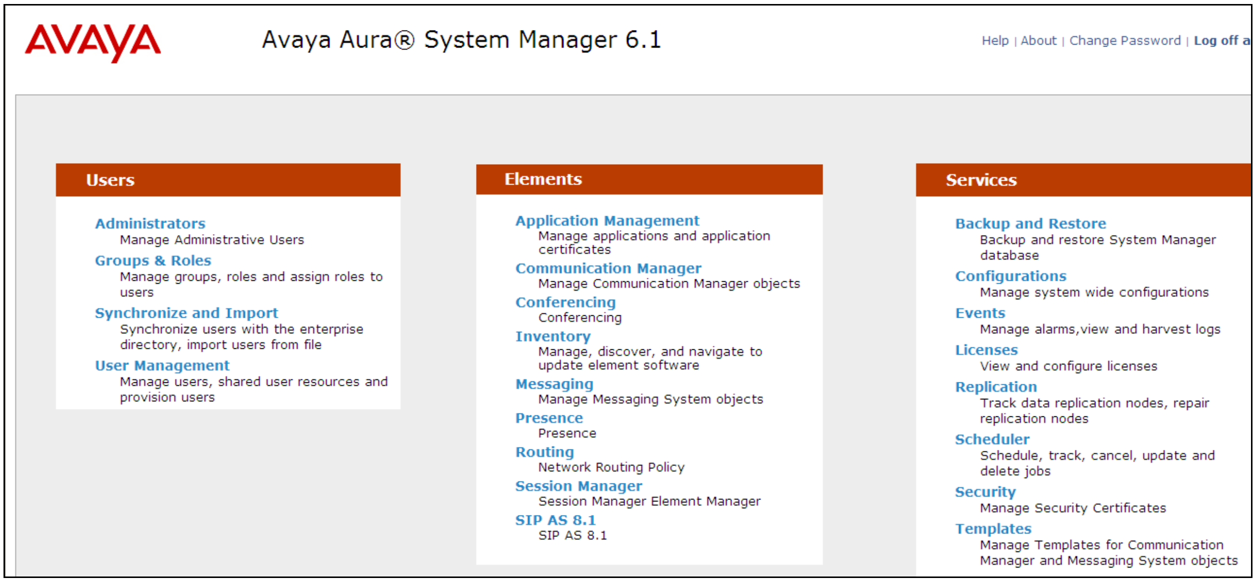
6. Configuring Avaya Aura® Session Manager

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

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

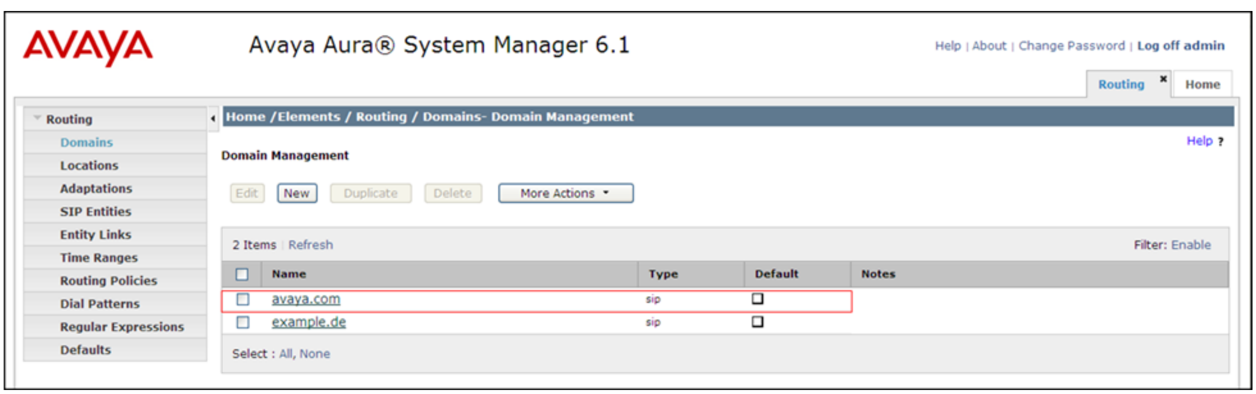
6.1. Log in to Avaya Aura® System Manager

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.



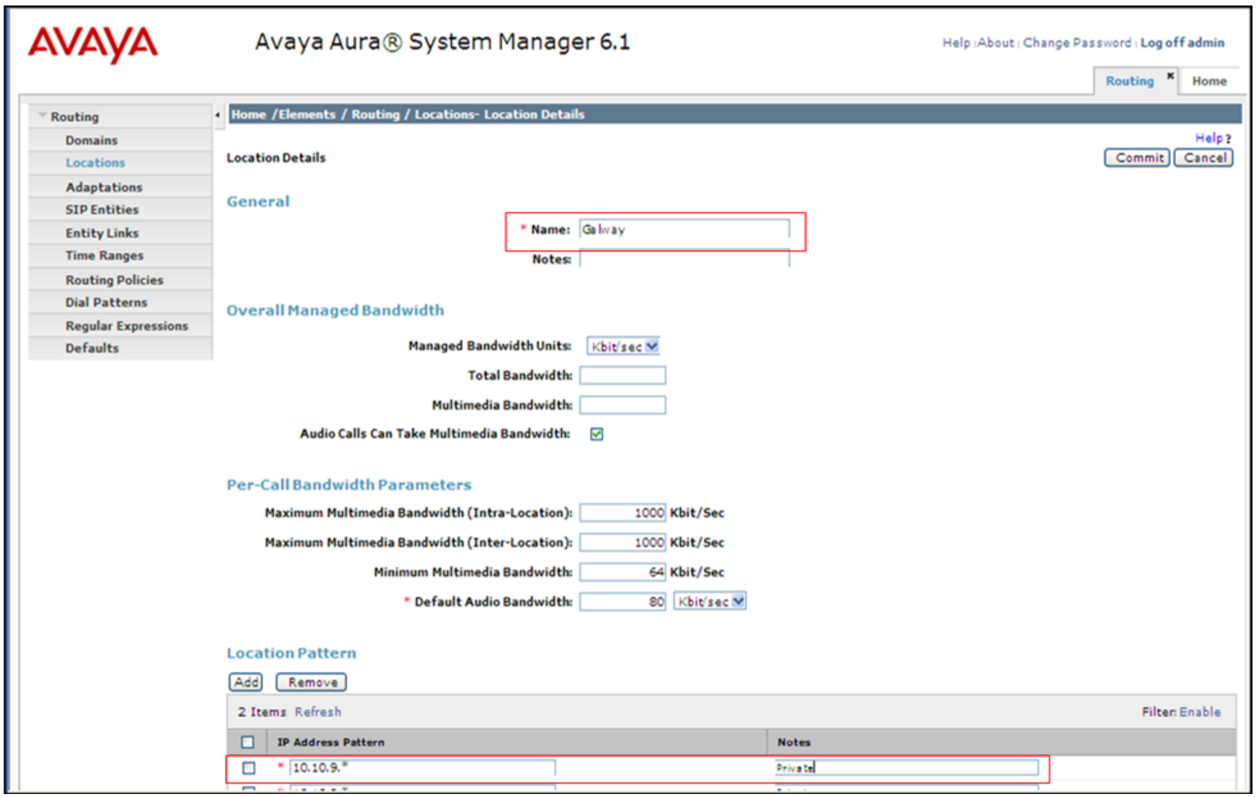
6.2. Administer SIP Domain

To add the SIP domain that will be used with Session Manager, select **Routing** from the **Home** tab menu 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.



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. 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 will need to be entered for each SIP Entity. Under **General:**

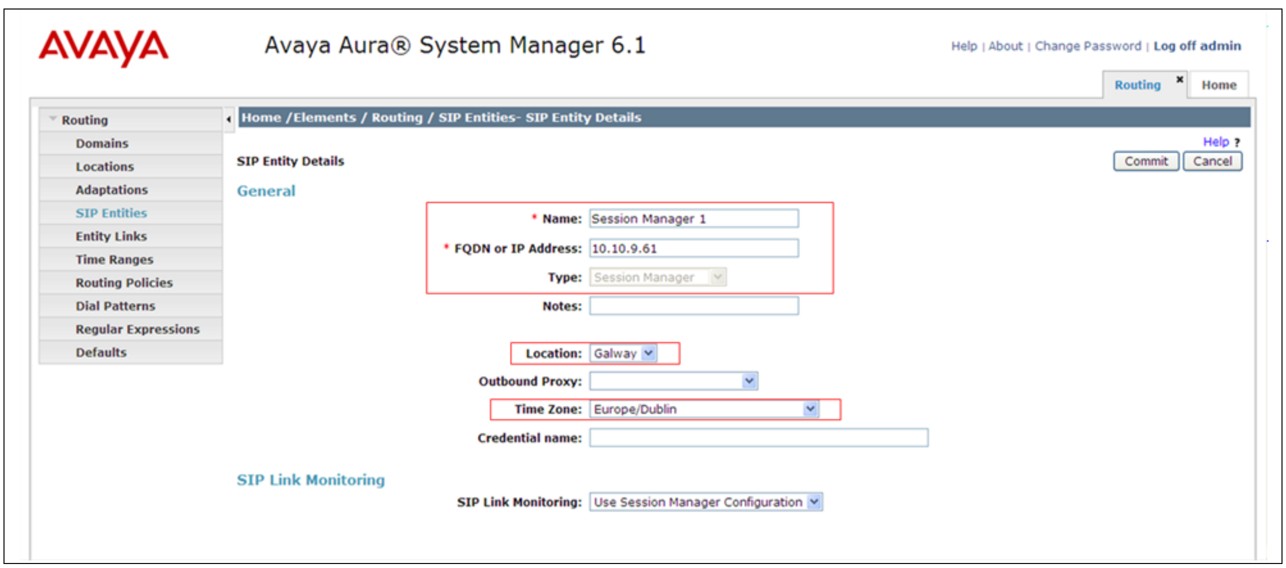
- In the **Name** field enter an informative name
- In the **FQDN or IP Address** field enter the IP address of Session Manager or the signaling interface on the connecting system
- In the **Type** field use **Session Manager** for a Session Manager SIP entity, **CM** for a Communication Manager SIP entity and **Gateway** for the Session Border Controller 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:

- Avaya Aura® Session Manager SIP Entity
- Avaya Aura® Communication Manager SIP Entity
- Avaya Aura® Session Border Controller SIP Entity

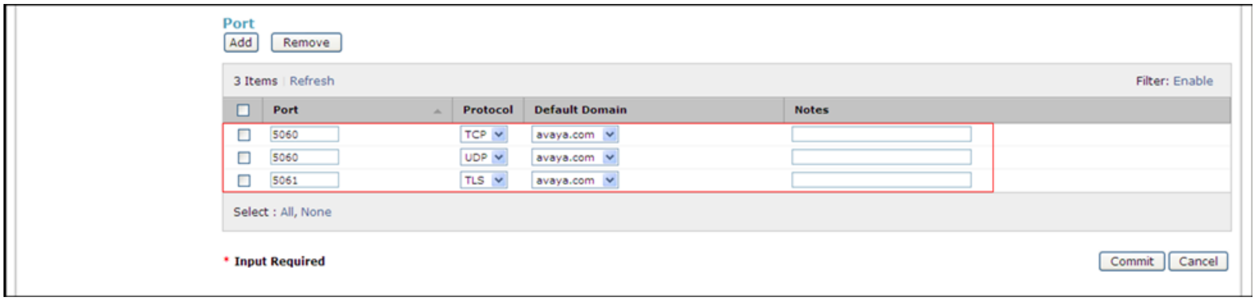
6.4.1. Avaya Aura® Session Manager SIP Entity

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.



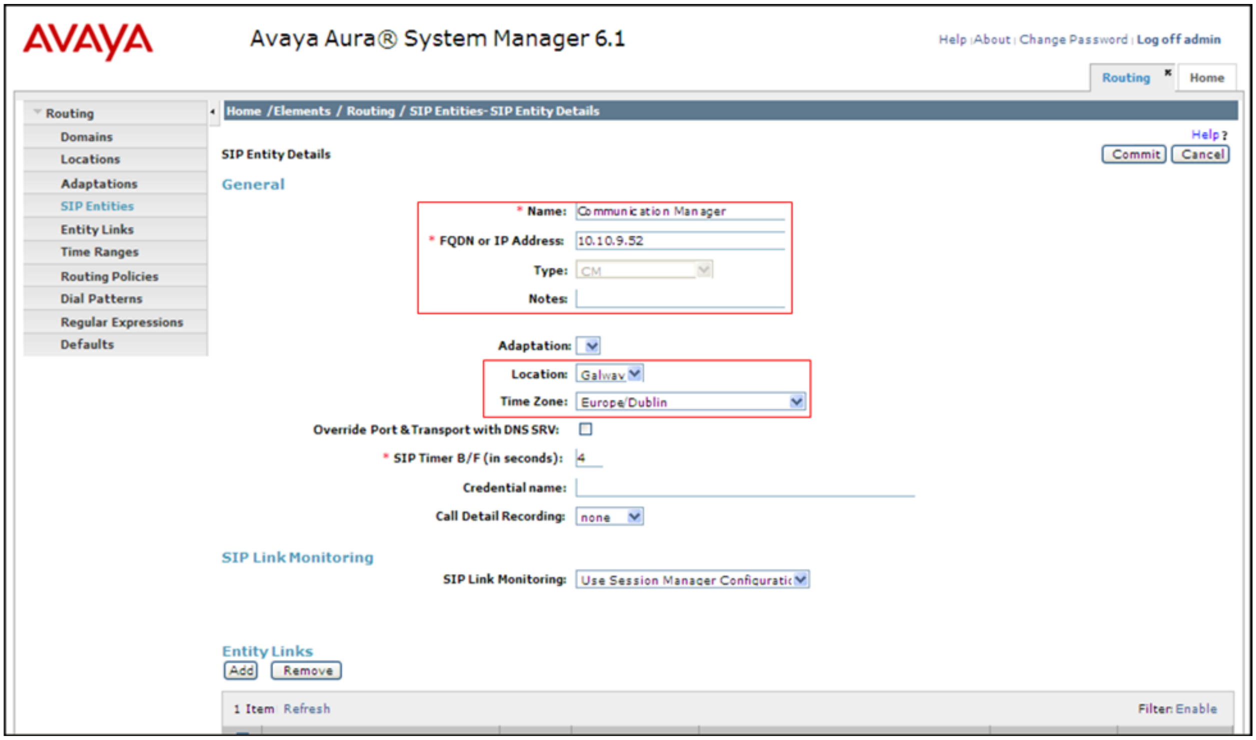
The Session Manager must be configured with the port numbers on the protocols that will be used by the other SIP entities. To configure these scroll to the bottom of the page and under **Port**, click **Add,** then edit the fields in the resulting new row.

- 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



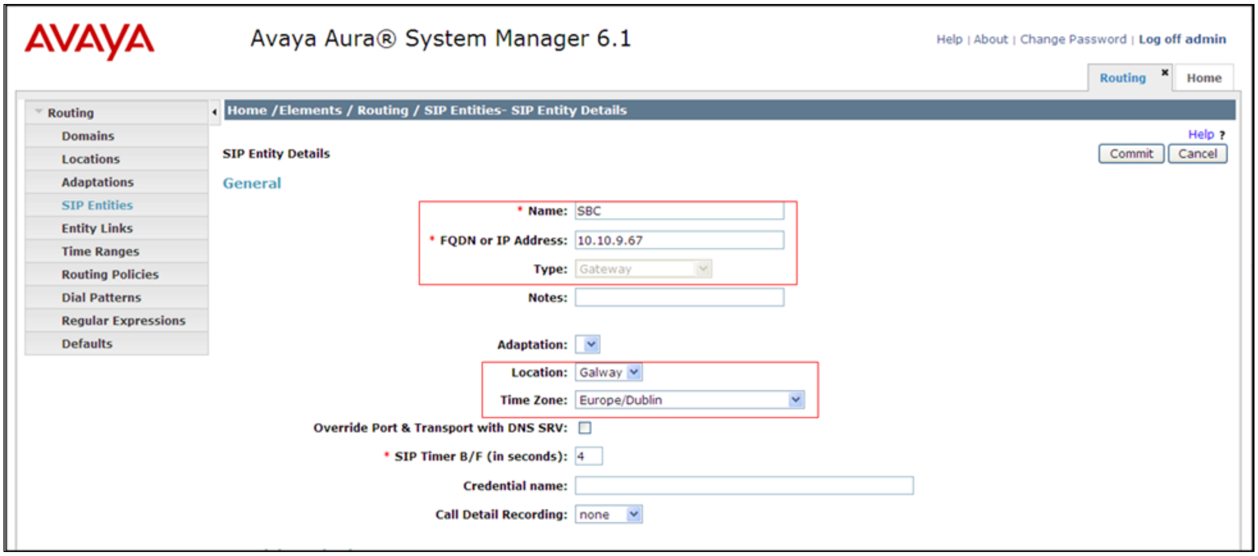
6.4.2. Avaya Aura® Communication Manager SIP Entity

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.



6.4.3. Avaya Aura® Session Border Controller SIP Entity

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

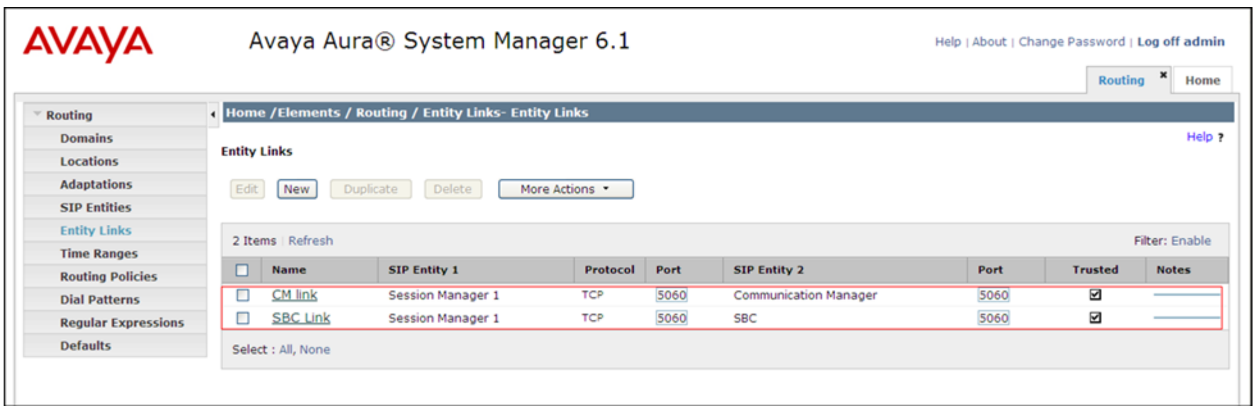


6.5. Administer Entity Links

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 (not shown). 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 **Session Manager 1**
- 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. The following screen shows the Entity Links used in this configuration.



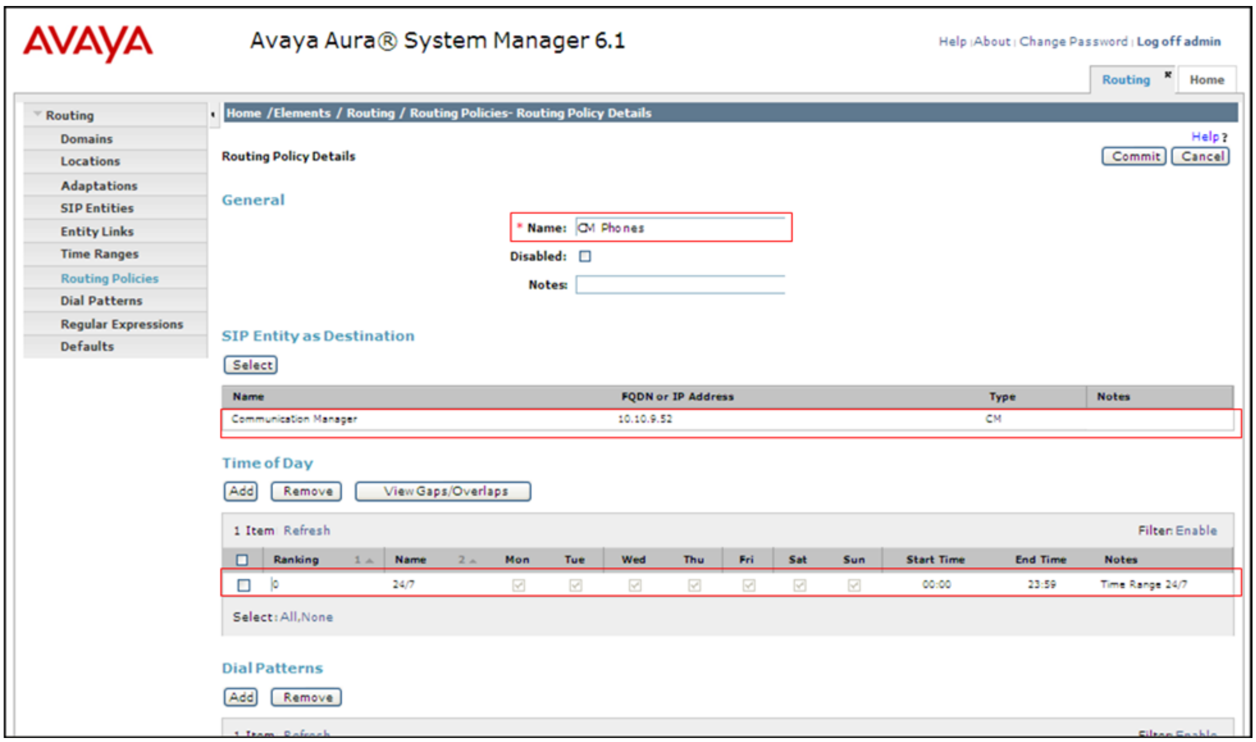
6.6. Administer Routing Policies

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).

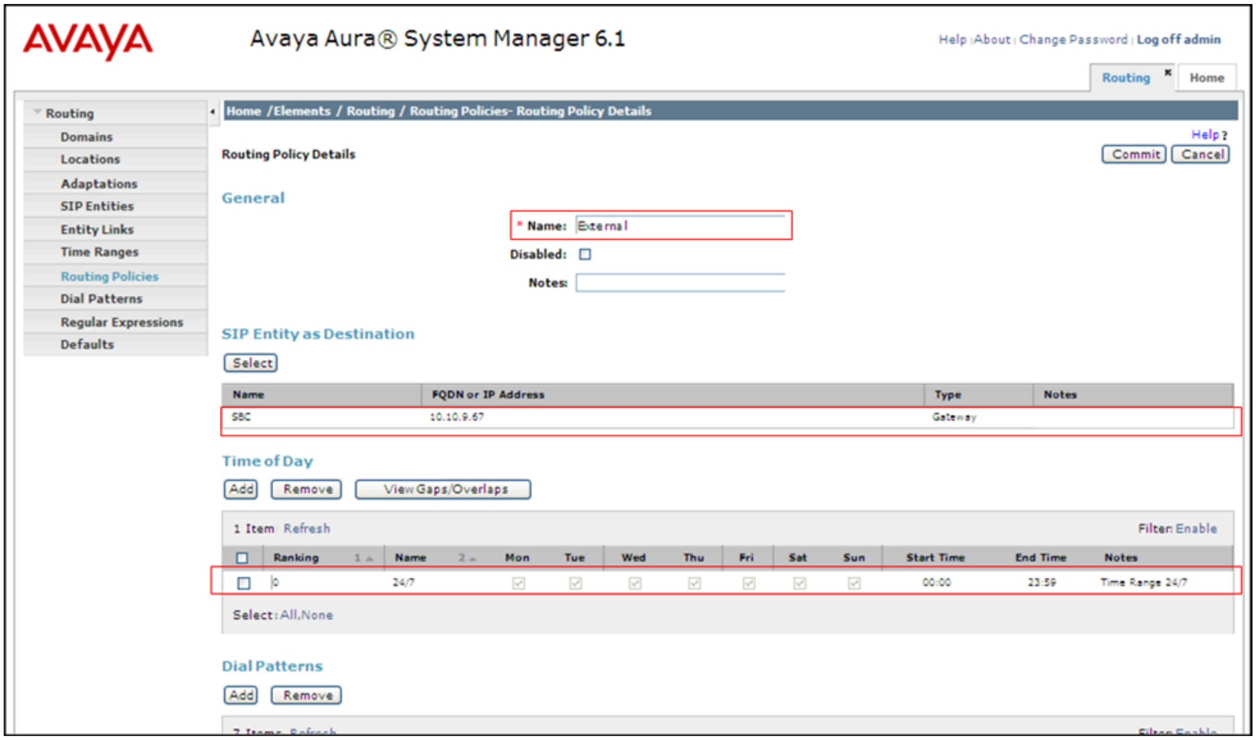
Under **General:**

- 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 Session Border Controller.



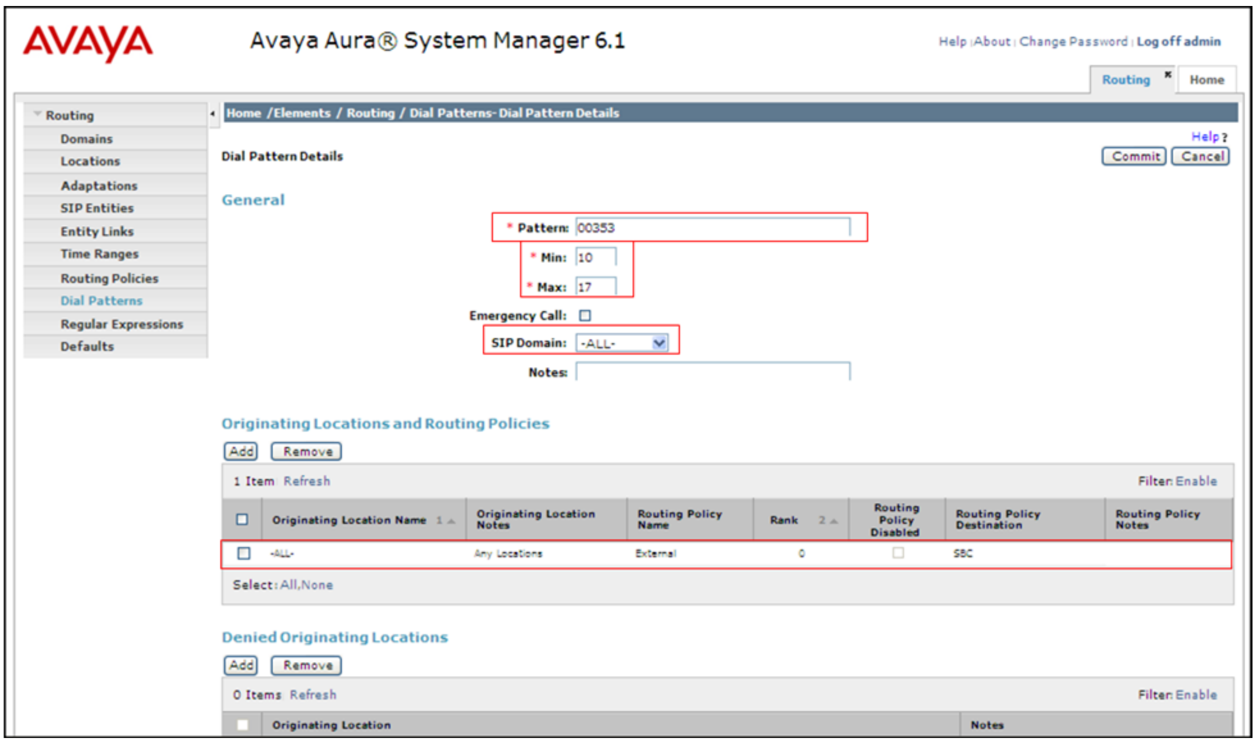
6.7. Administer Dial Patterns

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).

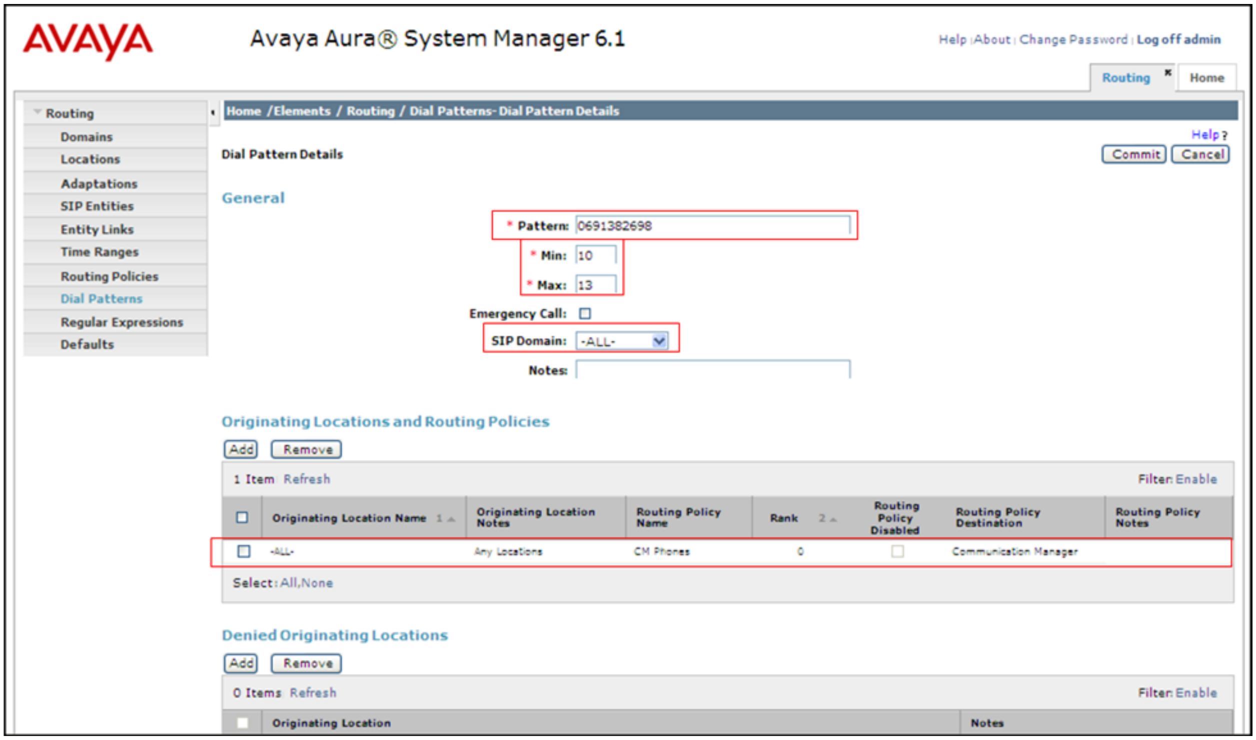
Under **General:**

- 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 the domain configured in **Section 6.2**

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 the Session Border Controller which will route the calls out to the Vodafone DE SIP Trunking Service.



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

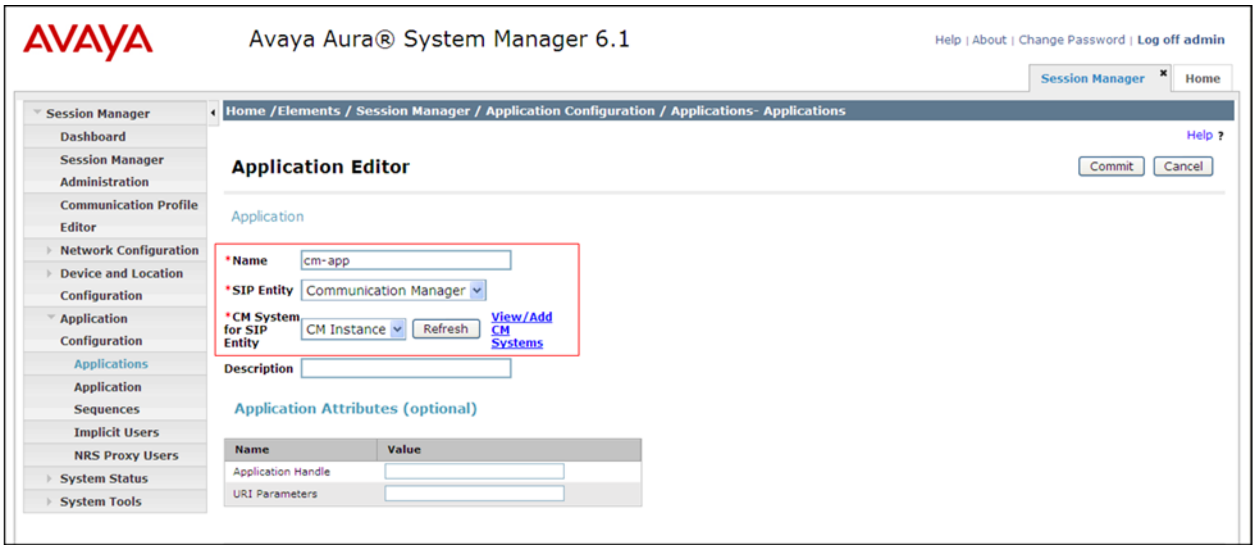


6.8. Administer Application for Avaya Aura® Communication Manager

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
- In the **CM System for SIP Entity** field select the SIP entity for the Communication Manager

Select **Commit** to save the configuration.

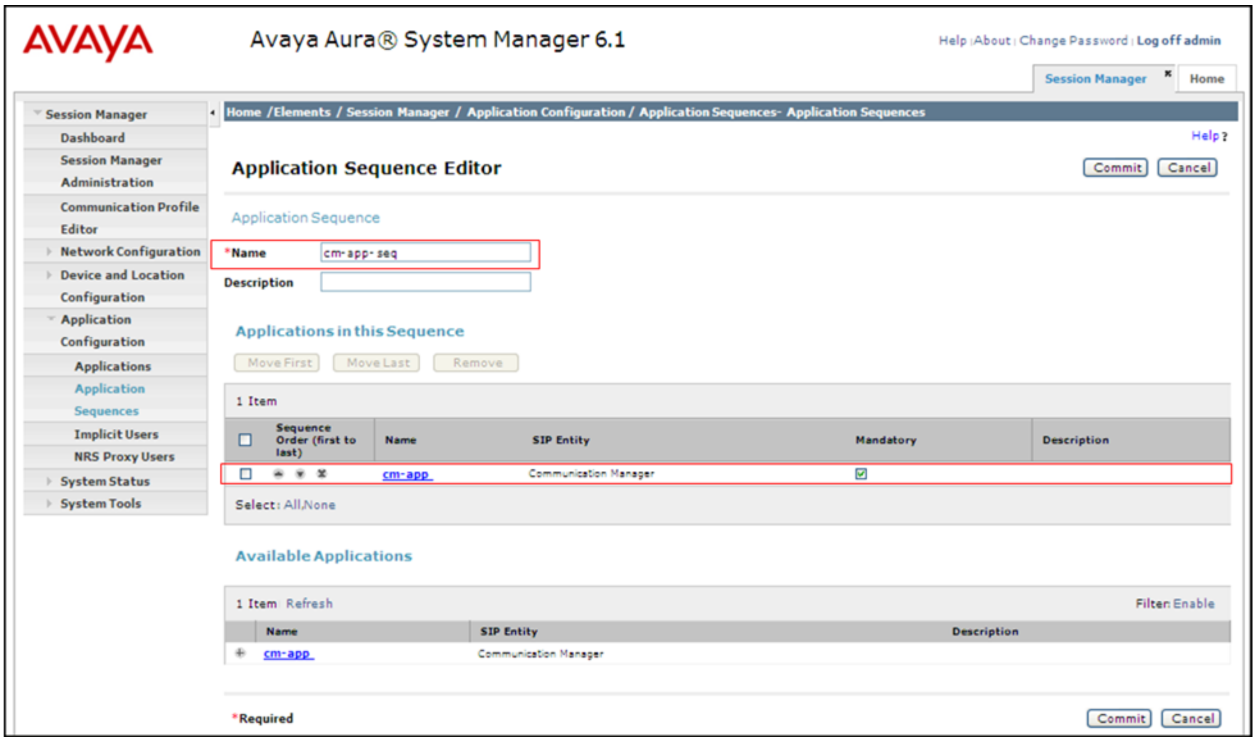


6.9. Administer Application Sequence for Communication Manager

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.**

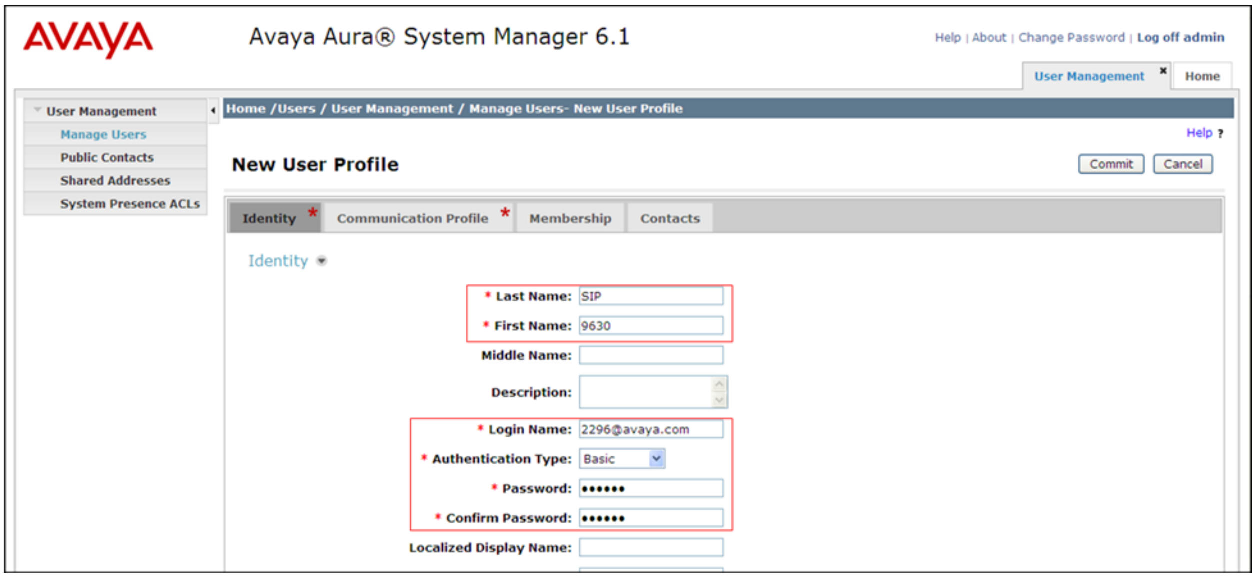


6.10. Administer SIP Extensions

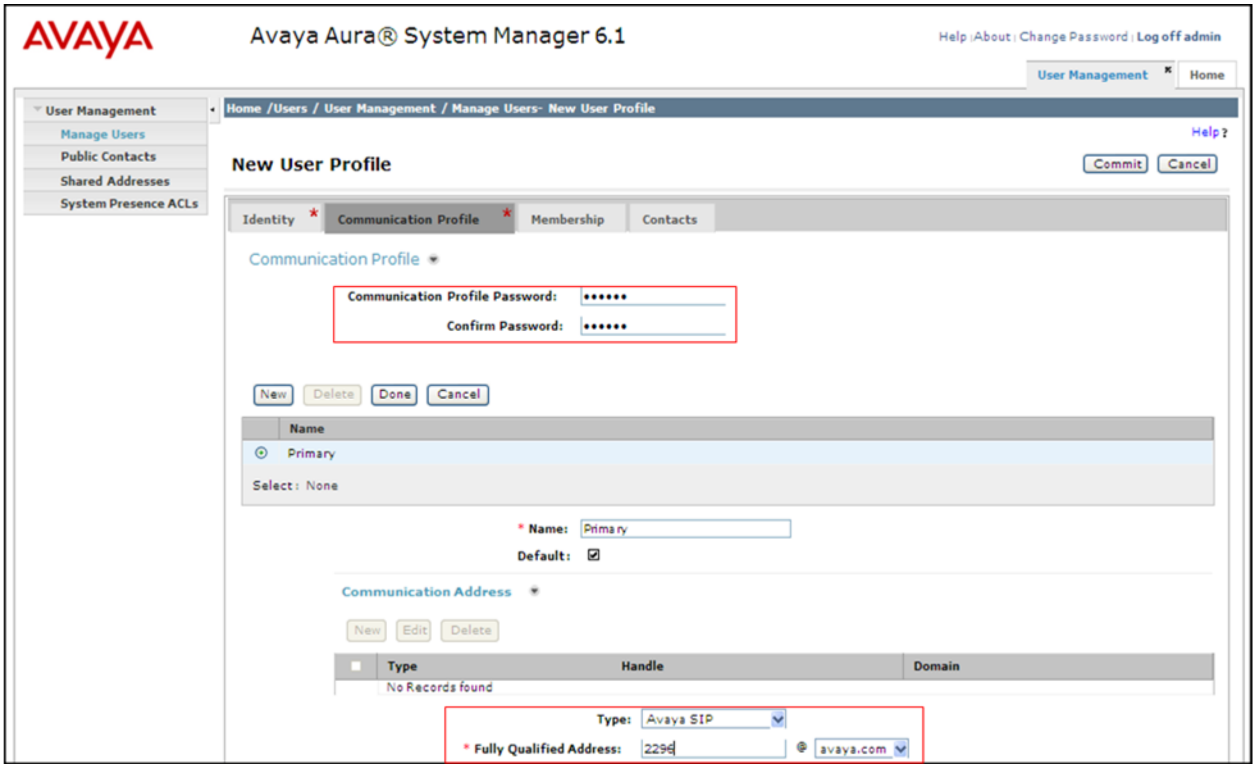
SIP extensions are registered with the Session Manager and use Communication Manager for their feature and configuration settings. From the Home tab select **User Management** from the menu. Then select **Manage Users** and click **New** (not shown).

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. **2296@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

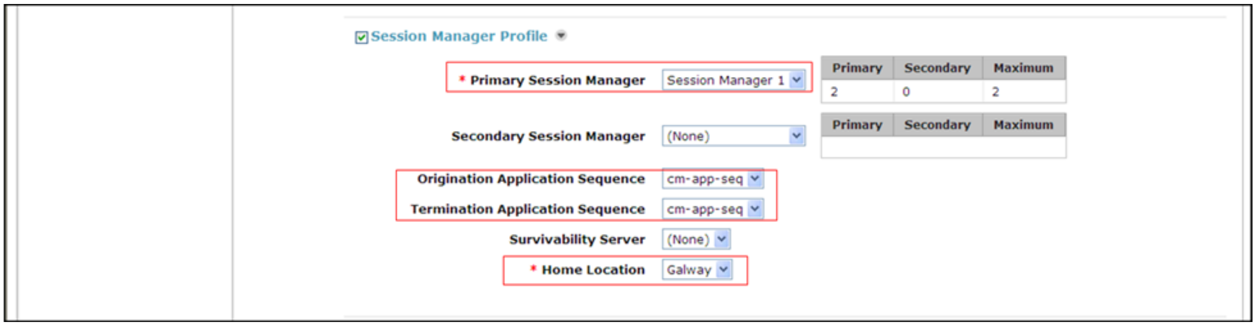


On the **Communication Profile** tab enter a numeric **Communication Profile Password** and confirm it, then expand the **Communication Address** section and click **New.** For the **Type** field select **sip** from the drop-down menu. In the **Fully Qualified Address** field, enter an extension number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



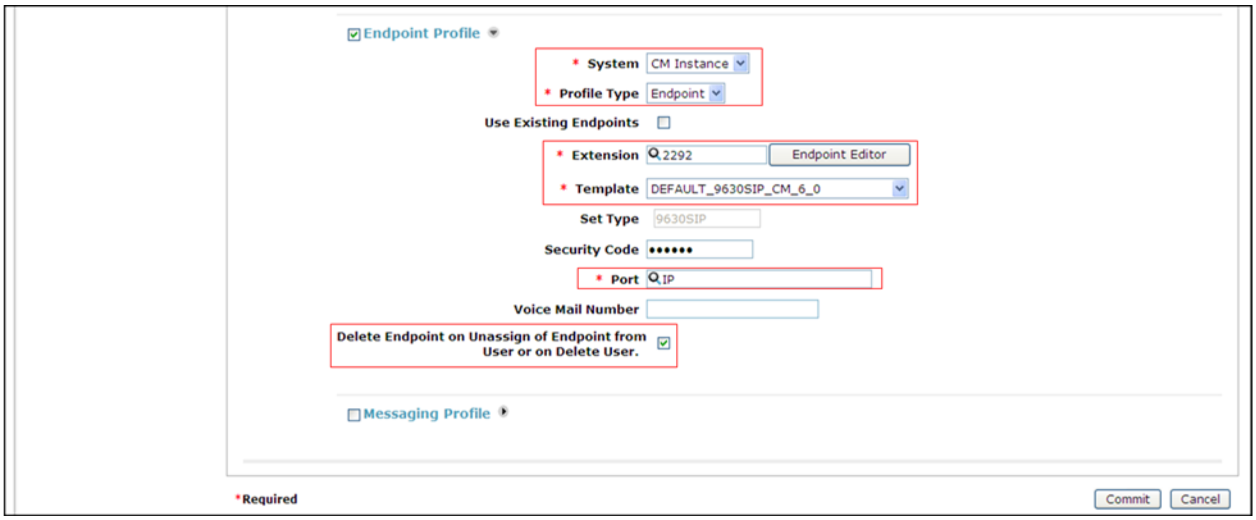
Expand the **Session Manager Profile** section.

- 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 6.9**
- Select the appropriate application sequence from the drop-down menu in the **Termination Application Sequence** field configured in **Section 6.9**
- Select the appropriate location from the drop-down menu in the **Home Location** field



Expand the **Endpoint Profile** section**.**

- 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
- Select **Commit** to save changes and the System Manager will add the Communication Manager user configuration automatically

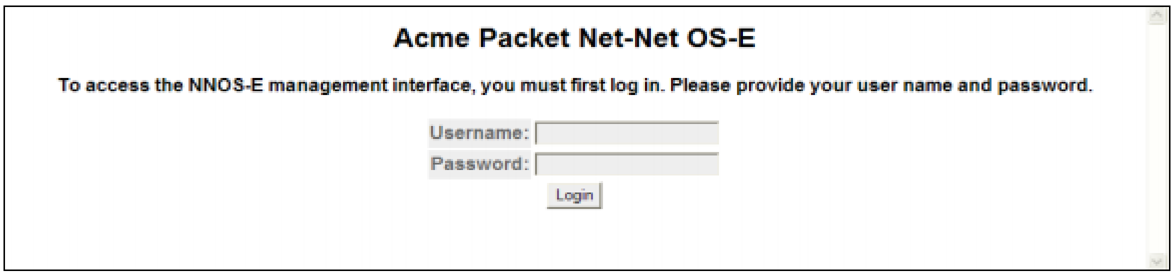


7. Configure Avaya Aura® Session Border Controller

This section describes the configuration of the Session Border Controller The configuration is done using the Session Border Controller web interface.

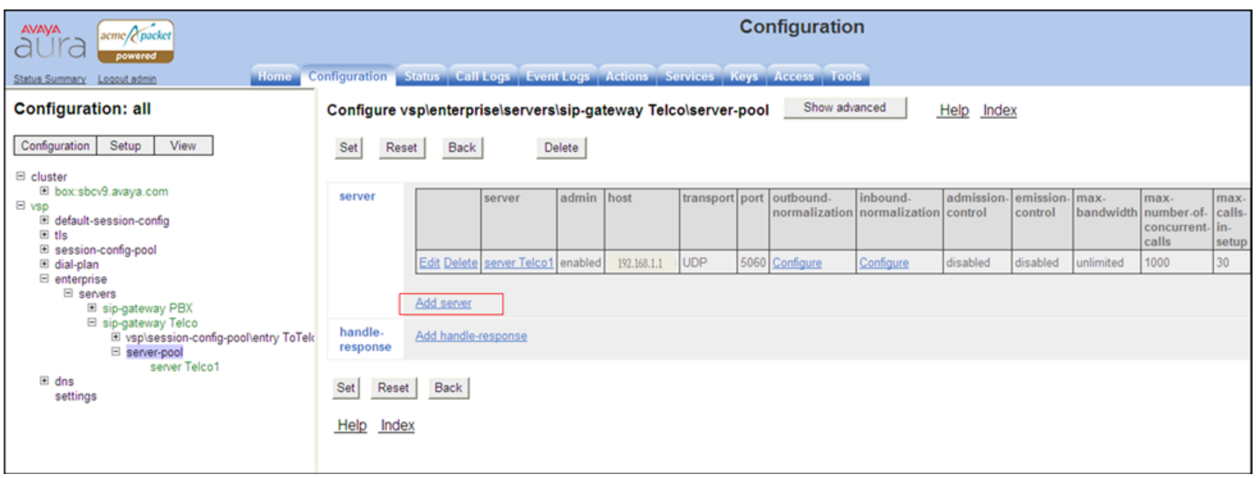
7.1. Access Avaya Aura® Session Border Controller

Access the Session Border Controller using a web browser by entering the URL **https://<ip-address>**, where **<ip-address>** is the private IP address configured at installation. Log in with the appropriate credentials.

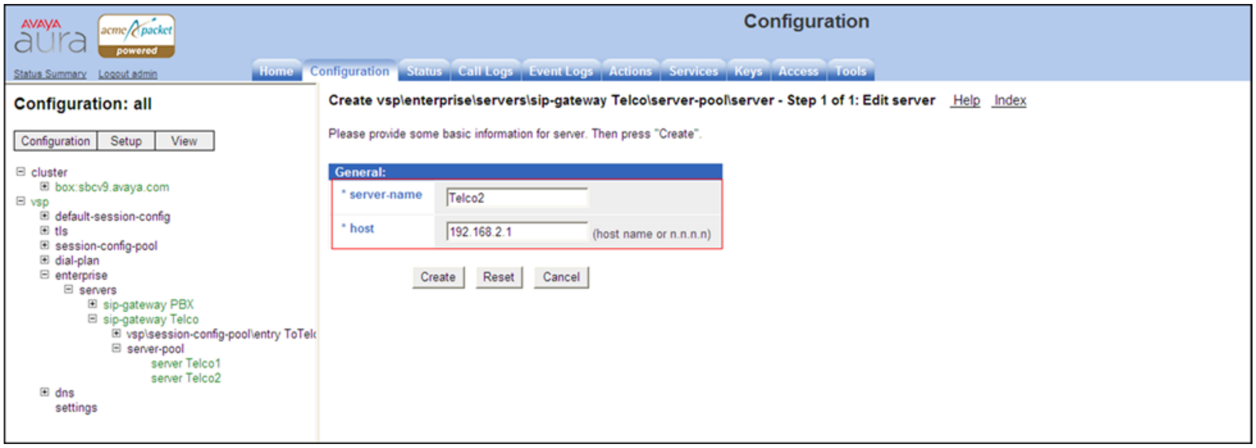


7.2. Add Additional Service Provider IP Addresses

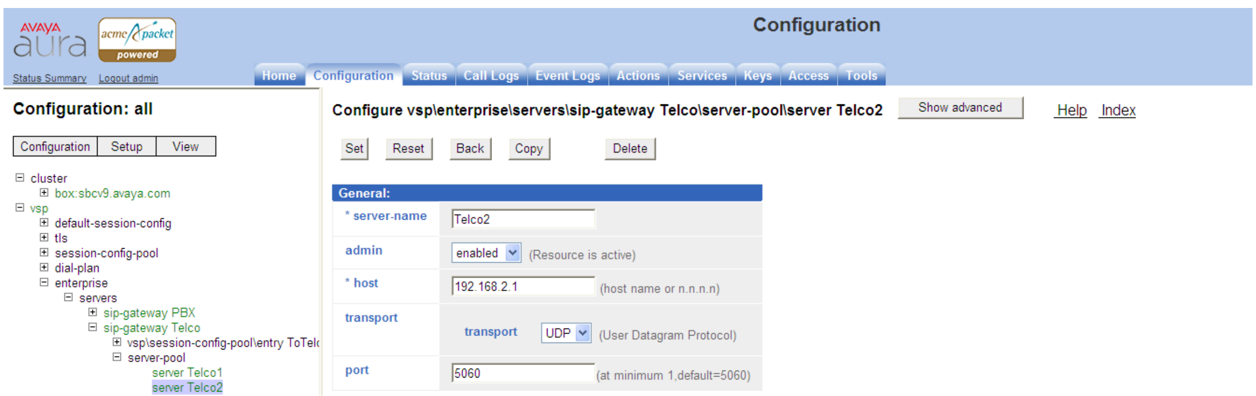
To add the additional IP addresses for the Vodafone DE SBCs click on the **Configuration** tab and browse to **vsp → enterprise → servers → sip-gateway Telco → server-pool.** A list of the IP addresses already configured in the server pool is displayed in the right hand pane. Click the **Add server** link**.**



In the resulting page enter a name for the server in the **server-name** field and an IP address in the **host** field. Click **Create** to continue.



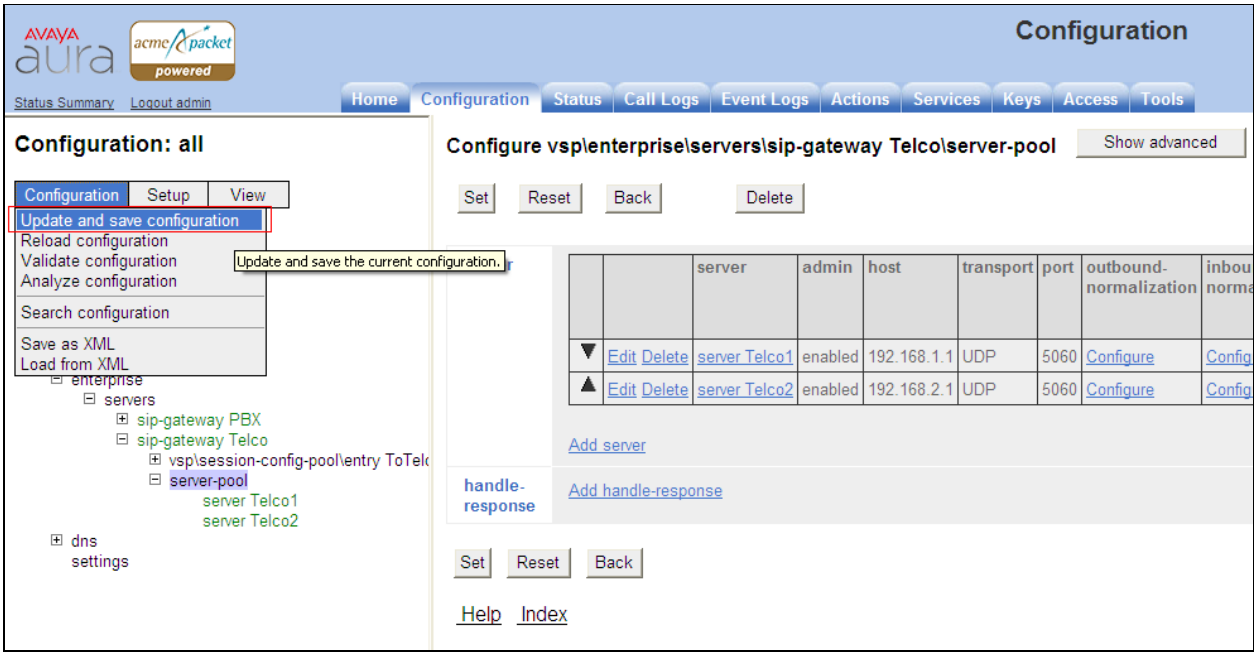
In the resulting page verify the details entered and click the **Set** button.



Repeat these steps for each additional IP address that needs to be added to the Session Border Controller server pool.

7.3. 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**.



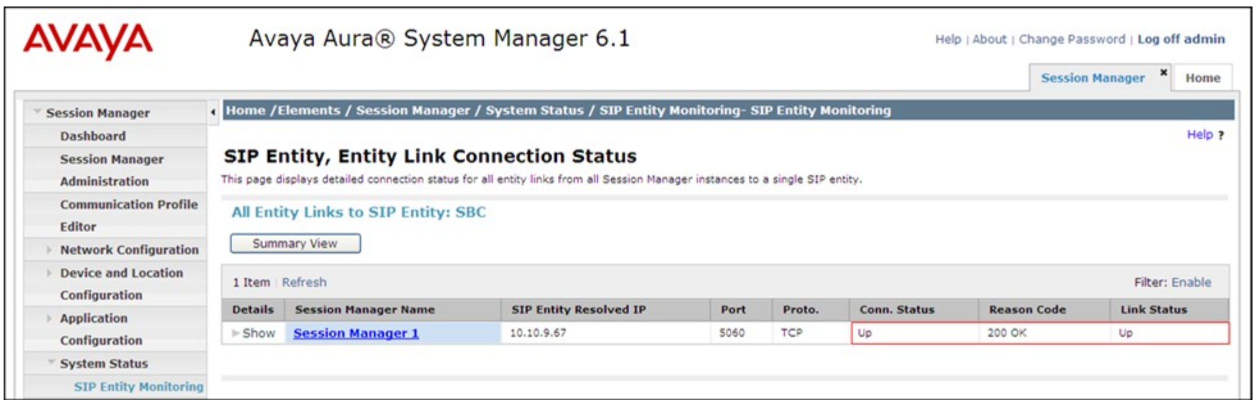
8. Service Provider Configuration

The configuration of the Vodafone DE equipment used to support the Vodafone DE SIP Trunking service is outside of the scope of these Application Notes and will not be covered. To obtain further information on Vodafone DE equipment and system configuration please contact an authorised Vodafone representative.

9. Verification Steps

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

1. 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.**



2. 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 1

                                    TRUNK GROUP STATUS

Member    Port     Service State      Mtce Connected Ports
                                      Busy

0001/001 T00001   in-service/idle    no
0001/002 T00002   in-service/idle    no
0001/003 T00003   in-service/idle    no
0001/004 T00004   in-service/idle    no
0001/005 T00005   in-service/idle    no
0001/006 T00006   in-service/idle    no
0001/007 T00007   in-service/idle    no
0001/008 T00008   in-service/idle    no
0001/009 T00009   in-service/idle    no
0001/010 T00010   in-service/idle    no
```

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

10. 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 DE SIP Trunking Service. Vodafone DE SIP Trunking Service is a SIP-based Voice over IP solution providing businesses a flexible, cost-saving alternative to traditional hardwired telephony trunks. The service was successfully tested with a number of observations listed in **Section 2.2.**

11. 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 6.0.3, February 2011.

[2] *Administering Avaya Aura® System Platform*, Release 6.0.3, February 2011.

[3] *Administering Avaya Aura® Communication Manager*, Release 6.0.1, April 2011.

[4] *Avaya Aura® Communication Manager Feature Description and Implementation,* August 2010, *D*ocument Number 555-245-205.

[5] *Installing and Upgrading Avaya Aura® System Manager Release 6.1*, November 2010.

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

[7] *Administering Avaya Aura® Session Manager,* May 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/

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.