Avaya Solution & Interoperability Test Lab

Application Notes for Configuring Avaya Aura® Communication Manager R6.2 as an Evolution Server, Avaya Aura® Session Manager R6.3 and Avaya Aura® Session Border Controller to support Updata SIP Trunking service - Issue 1.0

Abstract

These Application Notes describe the steps used to configure Session Initiation Protocol (SIP) trunking between Updata 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. Updata 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 Updata SIP Trunking service and an Avaya SIP-enabled enterprise solution. The Avaya solution consists of Avaya Aura® Session Border Controller (SBC), Avaya Aura® Session Manager and Avaya Aura® Communication Manager Evolution Server. Customers using this Avaya SIP-enabled enterprise solution with the Updata 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 SBC. The enterprise site was configured to use the SIP Trunking service provided by Updata.

DevConnect Compliance Testing is conducted jointly by Avaya and DevConnect members. The jointly-defined test plan focuses on exercising APIs and/or standards-based interfaces pertinent to the interoperability of the tested products and their functionalities. DevConnect Compliance Testing is not intended to substitute full product performance or feature testing performed by DevConnect members, nor is it to be construed as an endorsement by Avaya of the suitability or completeness of a DevConnect member's solution.

2.1. Interoperability Compliance Testing

The interoperability test included the following:

- Incoming calls to the enterprise site from mobile phones using the SIP Trunk provided by Updata, calls made to SIP and H.323 telephones at the enterprise
- Outgoing fixed and mobile calls from the enterprise site completed via Updata to PSTN and mobile destinations, calls made from SIP and H.323 telephones
- Calls using the G.711A, G.711MU and G.729 codecs
- Fax calls to/from a group 3 fax machine to a PSTN connected fax machine using T.38
- 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 with SIP and H.323 telephones
- No initial direct IP to IP media so that media is established to the G430 Media Gateway initially then converted to a direct connection after call set-up (shuffling)
- Call coverage and call forwarding for endpoints at the enterprise site
- Transmission and response of SIP OPTIONS messages sent by Updata requiring Avaya response and sent by Avaya requiring Updata response

2.2. Test Results

Interoperability testing of the sample configuration was completed with successful results for the Updata SIP Trunking service with the following observations:

- Shuffling was delayed on outgoing calls from the 9620 SIP phone and media was routed via the Media Gateway – this was end-point specific as shuffling worked effectively from the Flare and SIP soft client
- Although codec G.729 is supported, it was observed during test that annex B is not supported
- When making incoming calls with no matching codec available, CM sends a 488 "Not Acceptable Here". The network responds to this with several re-attempts resulting in a delay before a tone is heard by the caller.
- When testing DTMF voicemail menu navigation from the Flare, an upgrade was required to the latest Flare firmware for navigation to work effectively.
- Incoming Toll-Free access was not available for test
- No test call was booked with Emergency Services Operator
- When incoming faxes are received, the network attempts to change back to the original codec once the fax is completed. The re-INVITE is rejected by CM with a 488 "Not Acceptable Here". During test the fax machine indicated an intermittent failure even though the faxes were sent successfully.
- When all trunks are busy and the Communication Manager sends 500 "Service Unavailable", the network re-attempts the call and there is a delay before the caller hears the call failure treatment.
- When the signalling is unavailable and the Session Manager sends 500 "Server Link Monitor Status Down", the network re-attempts the call and there is a delay before the caller hears the call failure treatment.

2.3. Support

For technical support on Updata products please visit the website at www.updata.net or contact an authorized Updata representative.

3. Reference Configuration

Figure 1 illustrates the test configuration. The test configuration shows an Enterprise site connected to the Updata SIP Trunking service. Located at the Enterprise site is an Avaya Aura® 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 H.323 firmware), Avaya A175 Desktop Video Device running Flare Experience (audio only), Avaya analogue telephones and an analogue fax machine. Also included in the test configuration was an Avaya one-X® Communicator soft phone running on a laptop PC configured for SIP.

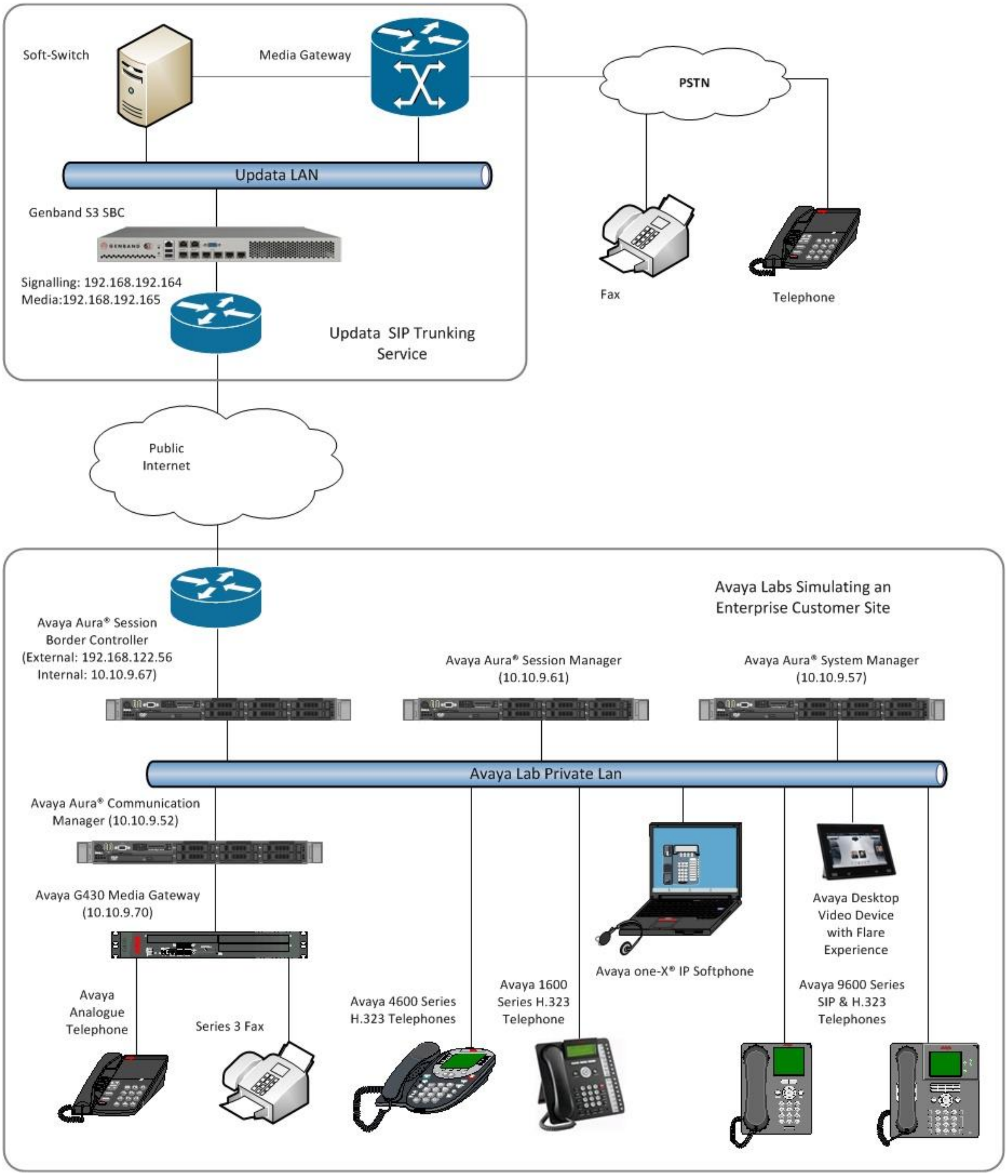


Figure 1: Test Setup Updata SIP Trunking to Avaya Enterprise

4. Equipment and Software Validated

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

Equipment/Software	Release/Version
Avaya	
Avaya S8800 Server running Session Manager	R6.3 - 6.3.0.0.630039
Avaya S8800 Server running System Manager	R6.3 - Build No. - 6.3.0.8.5682-6.3.8.818 Software Update Revision No: 6.3.0.8.923
Avaya S8800 Server running Communication Manager	R016x.02.0.823.0 Patch 20396 (SP5)
Avaya S8800 Server running Session Border Controller	Version E3.6.2.M1P2 – Build No. 50293
Avaya 1616 Phone (H.323)	1.301
Avaya 4621 Phone (H.323)	2.902
Avaya 9630 Phone (H.323)	3.103
Avaya A175 Desktop Video Device (SIP)	Flare Experience Release 1.1.2
Avaya 9620 Phone (SIP)	R2.6 SP9
Avaya one–X® Communicator (H.323) on Lenovo T510 Laptop PC	6.1.7.04-SP7-39506
Analogue Handset	NA
Analogue Fax	NA
Updata	
Genband S3 THN (Annapolis)	7.1
Genband S3 LD5 (Annapolis)	7.1

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 signalling associated with the Updata SIP Trunking service. For incoming calls, the Session Manager receives SIP messages from the Avaya Aura® Session Border Controller (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 signalling is routed to the Session Manager. The Session Manager directs the outbound SIP messages to the SBC at the enterprise site that then sends the SIP messages to the Updata 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 Servers 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 Updata 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 20
  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 signalling 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 signalling interface to Session Manager.

```
display node-names ip
                                  IP NODE NAMES
    Name              IP Address
SM100               10.10.9.61
Sipera-SBC          10.10.9.71
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 media stream is established directly between the enterprise end-point and the internal media interface of the SBC.
- 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 in **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 Updata were configured, namely **G.711A**, **G.729** and **G.711MU**.

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

Note: G.729A is shown here. In fact, Updata supports G.729 and both annexes A and B were tested. While annex A was tested successfully, annex B was not fully supported.

The Updata SIP Trunking service supports T.38 for transmission of fax. Navigate to **Page 2** to configure T.38 by setting the **FAX - Mode** to **t.38-standard** as shown below

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

                          IP Codec Set

                              Allow Direct-IP Multimedia? n



                    Mode                Redundancy
    FAX             t.38-standard        0
    Modem           off                  0
    TDD/TTY         US                   3
    Clear-channel   n                    0
```

5.5. Administer SIP Signaling Groups

This signalling group (and trunk group) will be used for inbound and outbound PSTN calls to the Updata 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** (Commonly used TCP port value)
- Set **Far-end Network Region** to the IP Network Region configured in **Section 5.3**. (logically establishes the far-end for calls using this signalling group as network region **1**)
- Leave **Far-end Domain** blank (allows the CM to accept calls from any SIP domain on the associated trunk)
- 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.

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

 Group Number: 1              Group Type: sip
  IMS Enabled? n        Transport Method: tcp
        Q-SIP? n
     IP Video? n                                 Enforce SIPS URI for SRTP? n
  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 **public-netwrk**
- Specify the signalling 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: public-ntwrk          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 Updata to prevent unnecessary SIP messages during call setup.

```
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): 300

 Disconnect Supervision - In? y  Out? y
```

On **Page 3**, set the **Numbering Format** field to **private**. This allows delivery of CLI with leading zeros.

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



                     Numbering Format: private
                                                UUI Treatment: service-provider

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

Note: The **Numbering Format** setting of **private** is only effective with non-international numbering formats specified in the route pattern.

On **Page 4** of this form:

- Set **Support Request History** to **y**
- Set the **Telephone Event Payload Type** to **101** to match the value preferred by Updata (this Payload Type is not applied to calls from SIP end-points)
- Set **Always Use re-INVITE for Display Updates** to **y**
- Set the **Identity for Calling Party Display** to **From** to ensure that where CLI for incoming calls is withheld, it is not displayed on the Communication Manager extension

```
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? n
                         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? y
            Identity for Calling Party Display: From
Block Sending Calling Party Location in INVITE? n
                                  Enable Q-SIP? n
```

5.7. Administer Calling Party Number Information

Use the **change private-unknown-numbering** command to configure Communication Manager to send the calling party number with leading zeros. As with the public numbering table, individual stations were mapped to send numbers allocated from the Updata DDI range supplied.

```
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: 8
 4  2000            1          01737nnnnn3      11    Total Administered: 9
 4  2291            1          01737nnnnn5      11       Maximum Entries: 540
 4  2296            1          01737nnnnn6      11
 4  2316            1          01737nnnnn3      11
 4  2346            1          01737nnnnn5      11
 4  2396            1          01737nnnnn4      11
 4  2400            1          01737nnnnn4      11
 4  2401            1          01737nnnnn7      11
 4  2601            1          01737nnnnn7      11
```

Note: The private numbering table is used when **private** is specified in the trunk settings described in **Section 5.6** and when a non-international number format is specified in the route pattern settings specified in **Section 5.8** for example **unk-unk**.

5.8. Administer Route Selection for Outbound Calls

In the test environment, the Automatic Route Selection (ARS) feature was used to route outbound calls via the SIP trunk to the Updata SIP Trunking service. The single digit **9** was used as the ARS access code providing a facility for telephone users to dial 9 to reach an outside line. Use the **change feature-access-codes** command to configure a digit 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 dialled digits following the first digit 9. A small sample of dial patterns are shown here as an example. Further administration of ARS is beyond the scope of this document. The example entries shown will match outgoing calls to numbers beginning 0 or 00. Note that exact maximum number lengths should be used where possible to reduce post-dial delay. Calls are sent to **Route Pattern 1**.

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

          Dialed            Total      Route     Call   Node  ANI
          String           Min  Max   Pattern    Type   Num   Reqd
    0                      8    14     1         pubu         n
    00                     13   17     1         pubu         n
    00353                  10   14     1         pubu         n
    0044                   12   14     1         pubu         n
    0800                   11   11     1         pubu         n
    118                    5    6      1         pubu         n
```

Use the **change route-pattern x** command, where **x** is an available route pattern, 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**. Set the **Numbering Format** to **unk-unk** to allow sending of national format CLI.

```
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                               unk-unk   none
 2: y y y y y n  n            rest                                         none
 3: y y y y y n  n            rest                                         none
 4: y y y y y n  n            rest                                         none
 5: y y y y y n  n            rest                                         none
 6: y y y y y n  n            rest                                         none
```

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 Updata can be manipulated as necessary to route calls to the desired extension. In the example, the incoming DDI numbers provided by Updata for testing are assigned to the internal extensions of the test equipment configured within the Communication Manager. The **change inc-call-handling-trmt trunk-group x** command is used to translate numbers **01737nnnnn3** to **01737nnnnn7** to the 4 digit extension by deleting all (**11**) of the incoming digits and inserting the extension number. Note that the significant digits beyond the area code have been obscured.

```
change inc-call-handling-trmt trunk-group 1                     Page   1 of  30
                       INCOMING CALL HANDLING TREATMENT
 Service/      Number   Number      Del Insert
 Feature        Len      Digits
 public-ntwrk    11 01737nnnnn3      11  2000
 public-ntwrk    11 01737nnnnn4      11  2396
 public-ntwrk    11 01737nnnnn5      11  2346
 public-ntwrk    11 01737nnnnn6      11  2296
 public-ntwrk    11 01737nnnnn7      11  2401
```

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. **0035386nnnnnnn**)
- Set the **Trunk Selection** to **1** so that Trunk Group 1 will be used for routing
- Set the **Config Set** to **1**

```
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        -     0035386nnnnnnn   1           1
                               -
```

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

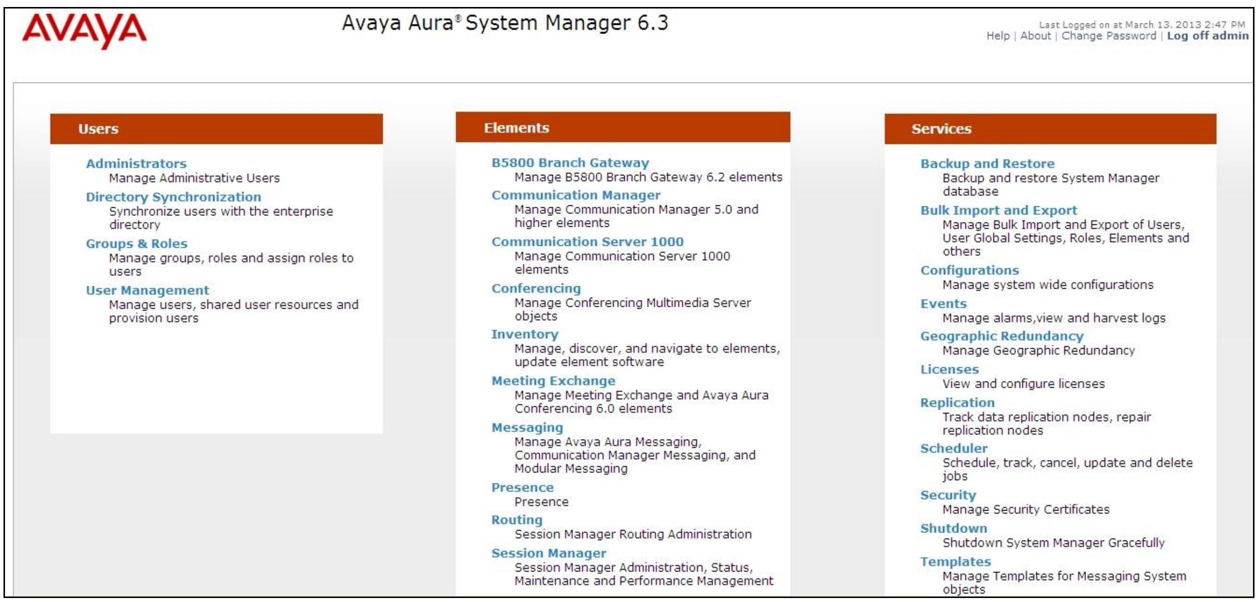
6. Configuring Avaya Aura® Session Manager

This section provides the procedures for configuring Session Manager. The Session Manager is configured via the 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

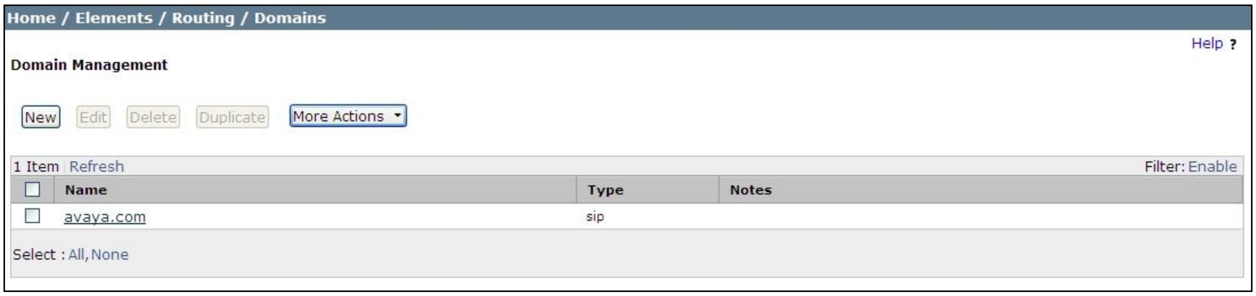
6.1. Log in to Avaya Aura® System Manager

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.



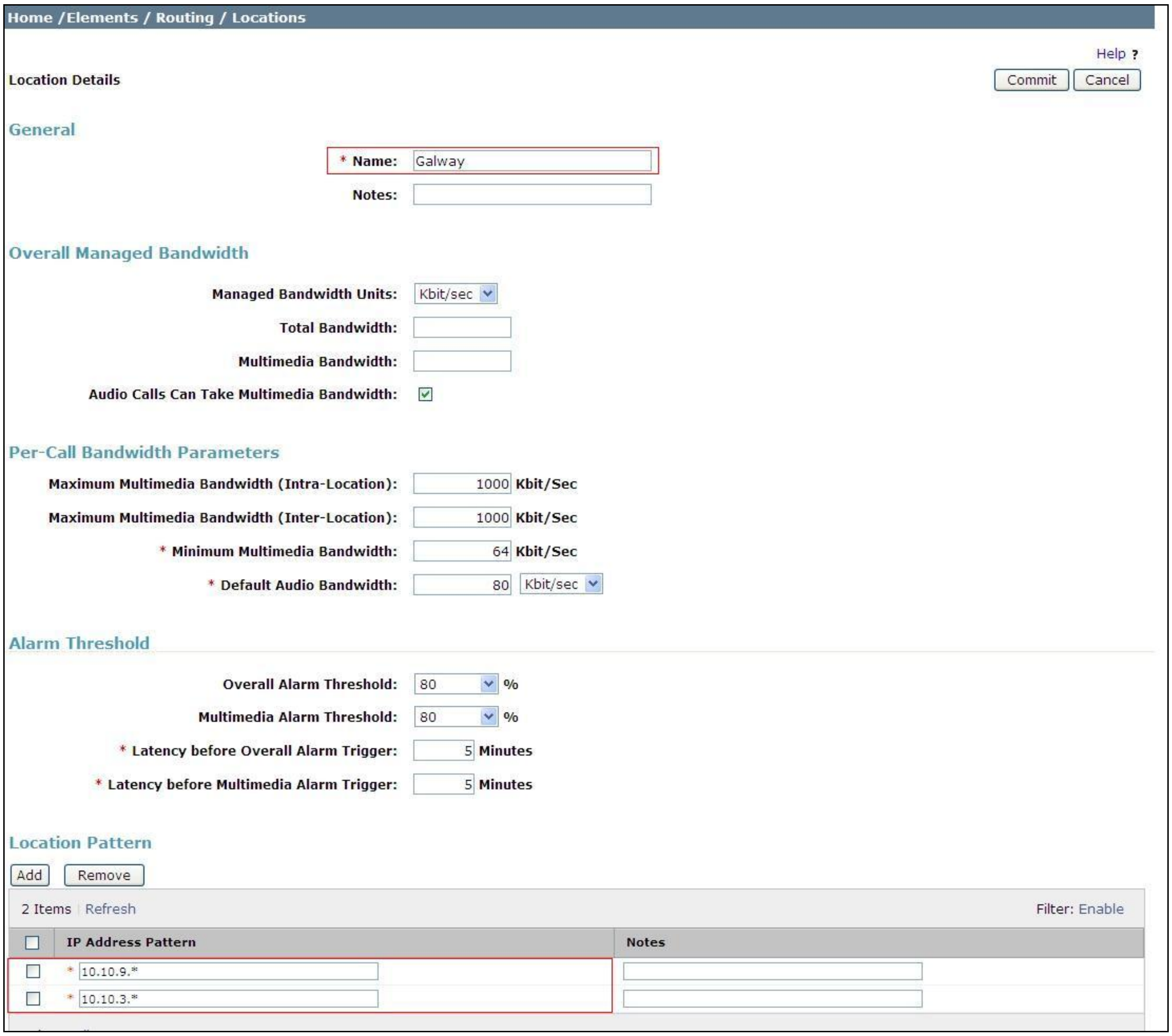
6.2. Administer SIP Domain

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 agreed with Updata; this will be the same as specified in the Authoritative Domain specified in the IP Network Region on the Communication Manager. Refer to **Section 5.3** for details. In test, **avaya.com** was used. Optionally, a description for the domain can be entered 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 test 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**:

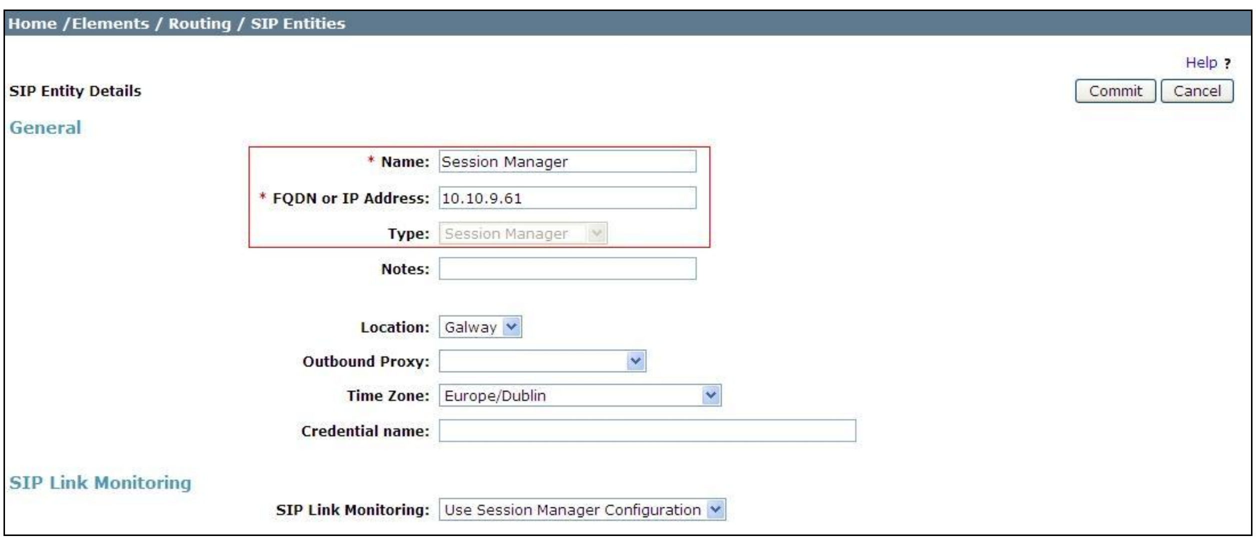
- In the **Name** field enter an informative name
- In the **FQDN or IP Address** field enter the IP address of the Session Manager or the signalling interface on the connecting system
- In the **Type** field use **Session Manager** for a Session Manager SIP entity, **Other** for a Communication Manager SIP entity and **SIP Trunk** for the SBC 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 (SBC) SIP Entity

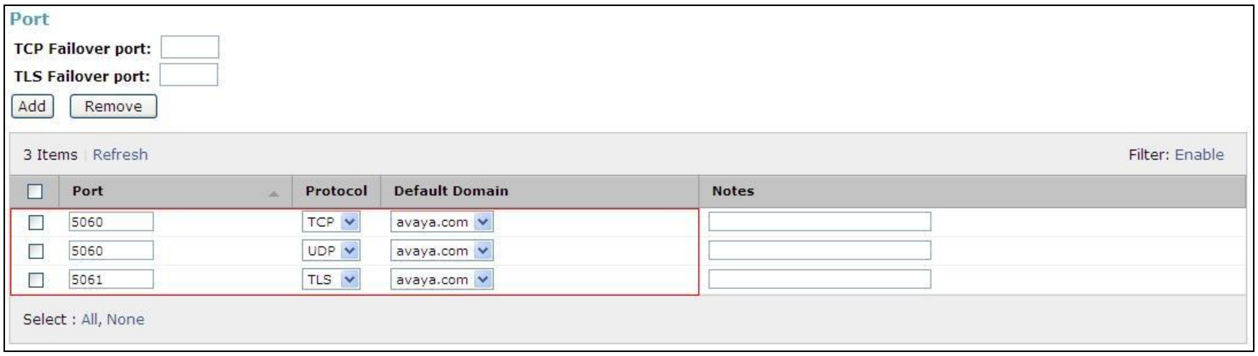
6.4.1. Avaya Aura® Session Manager SIP Entity

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 signalling interface.



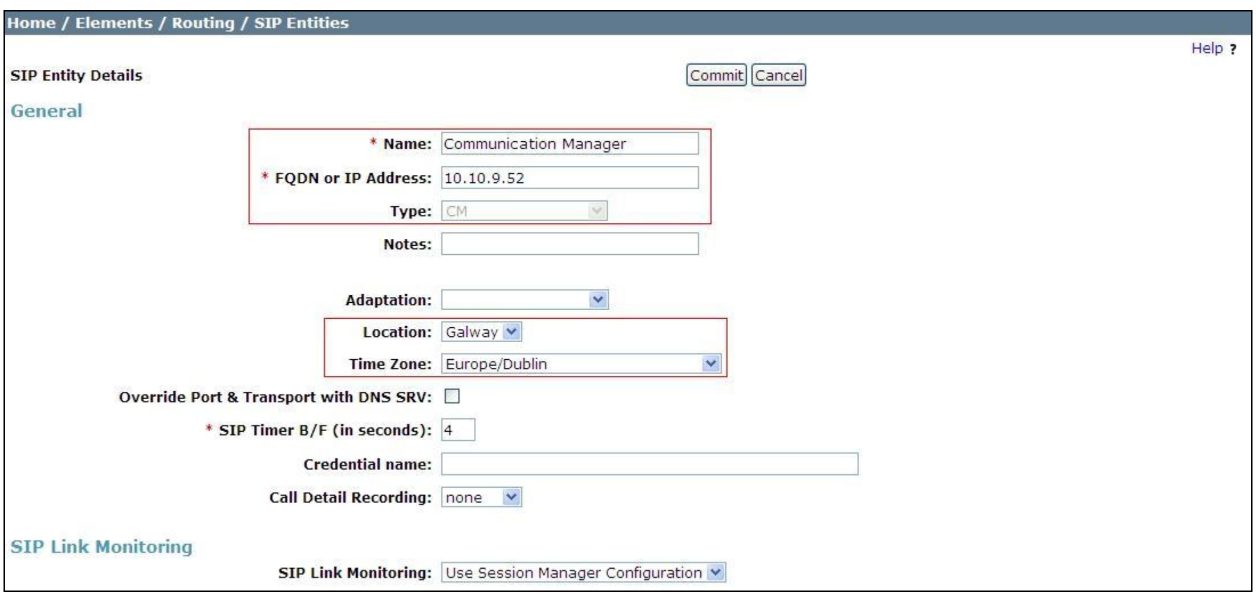
The Session Manager must be configured with the port numbers on the protocols that will be used by the other SIP entities. To configure these scroll to the bottom of the page and under **Port**, click **Add**, then edit the fields in the resulting new row.

- 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 the domain added in **Section 6.2** as the default domain



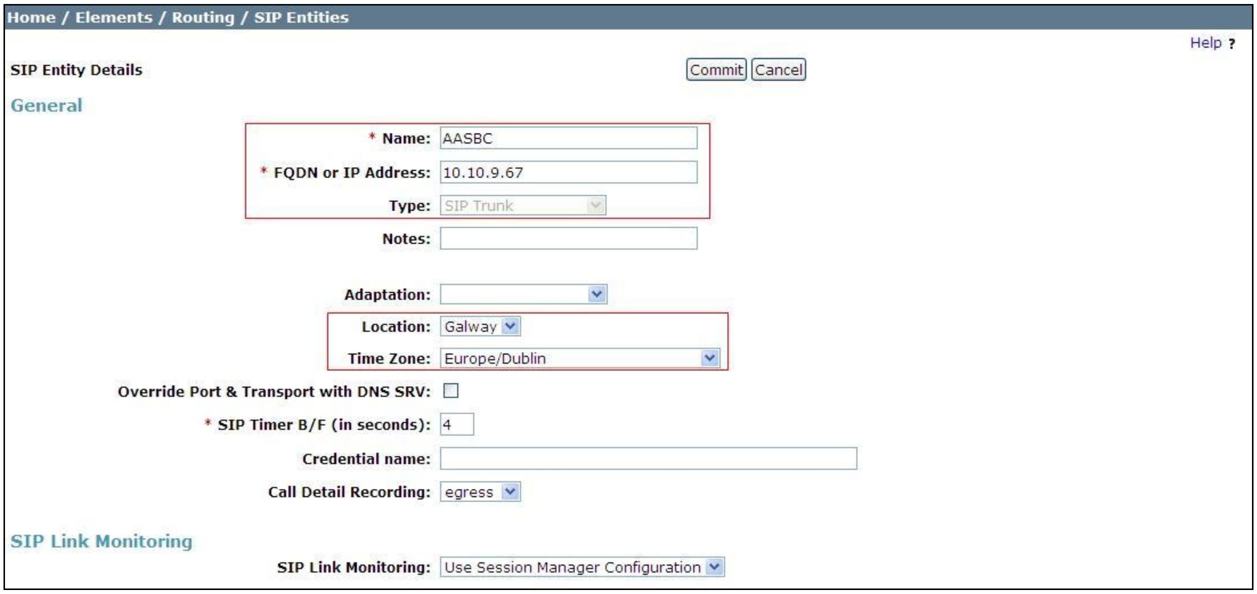
6.4.2. Avaya Aura® Communication Manager SIP Entity

The following screen shows 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 signalling. Set the location to that defined in **Section 6.3** and the **Time Zone** to the appropriate time zone.



6.4.3. Avaya Aura® Session Border Controller (SBC) SIP Entity

The following screen shows the SIP Entity for the SBC. The **FQDN or IP Address** field is set to the IP address of the SBC private network interface (see **Figure 1**). Set the location to that defined in **Section 6.3** and the **Time Zone** to the appropriate time zone.

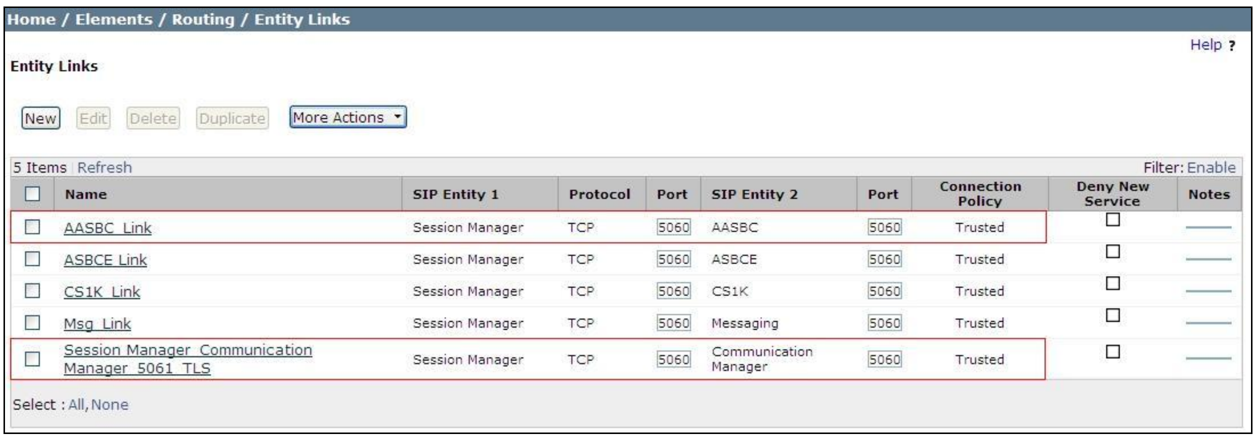


6.5. Administer Entity Links

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**
- 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.


Home / Elements / Routing / Entity Links

Help ?

Entity Links

New Edit Delete Duplicate More Actions

5 Items | Refresh Filter: Enable

	Name	SIP Entity 1	Protocol	Port	SIP Entity 2	Port	Connection Policy	Deny New Service	Notes
☐	AASBC_Link	Session Manager	TCP	5060	AASBC	5060	Trusted	☐	
☐	ASBCE Link	Session Manager	TCP	5060	ASBCE	5060	Trusted	☐	
☐	CS1K_Link	Session Manager	TCP	5060	CS1K	5060	Trusted	☐	
☐	Msg_Link	Session Manager	TCP	5060	Messaging	5060	Trusted	☐	
☐	Session Manager_Communication Manager_5061_TLS	Session Manager	TCP	5060	Communication Manager	5060	Trusted	☐	

Select : All, None

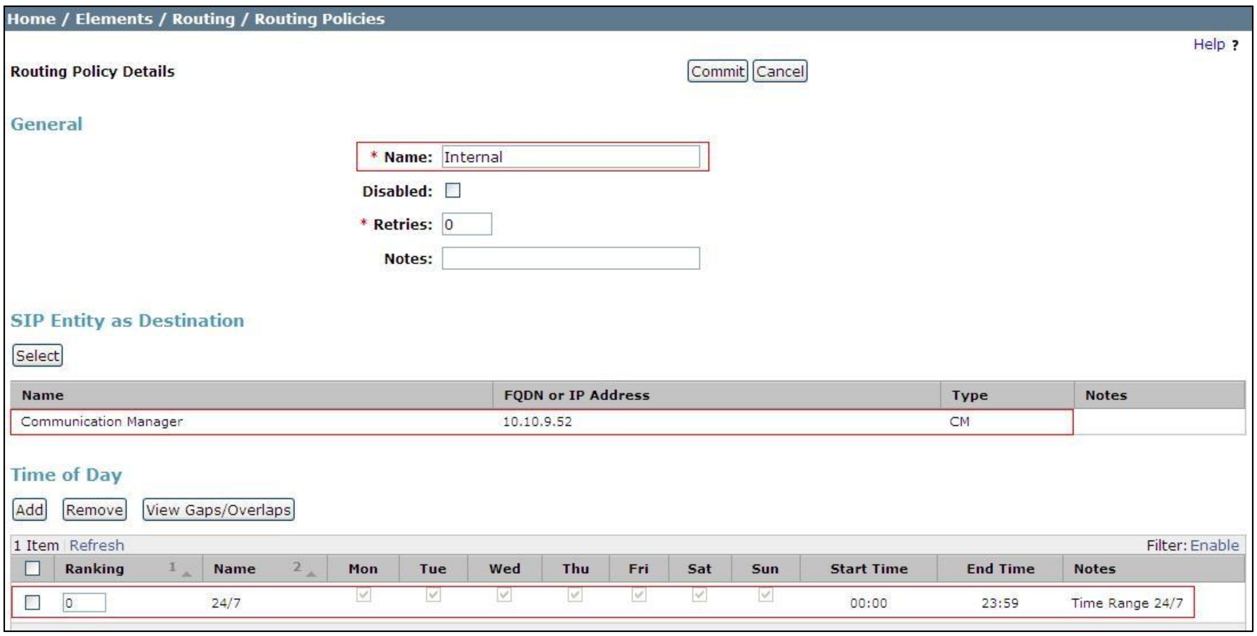
6.6. Administer Routing Policies

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).

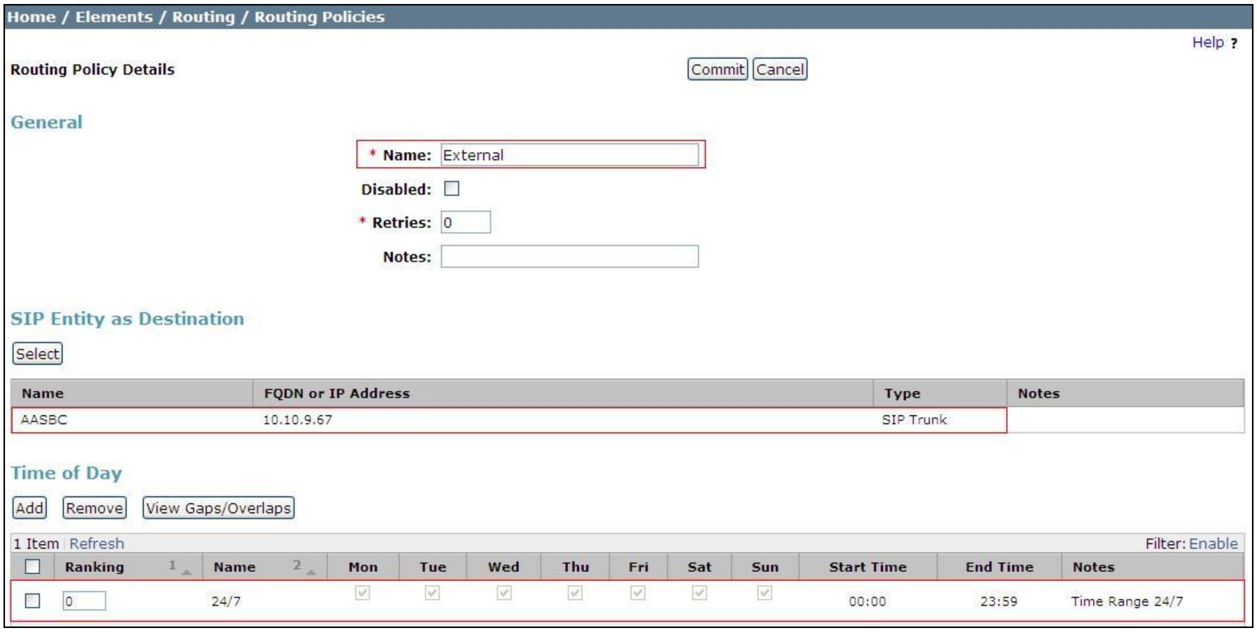
Under **General**:

- 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 SBC.



6.7. Administer Dial Patterns

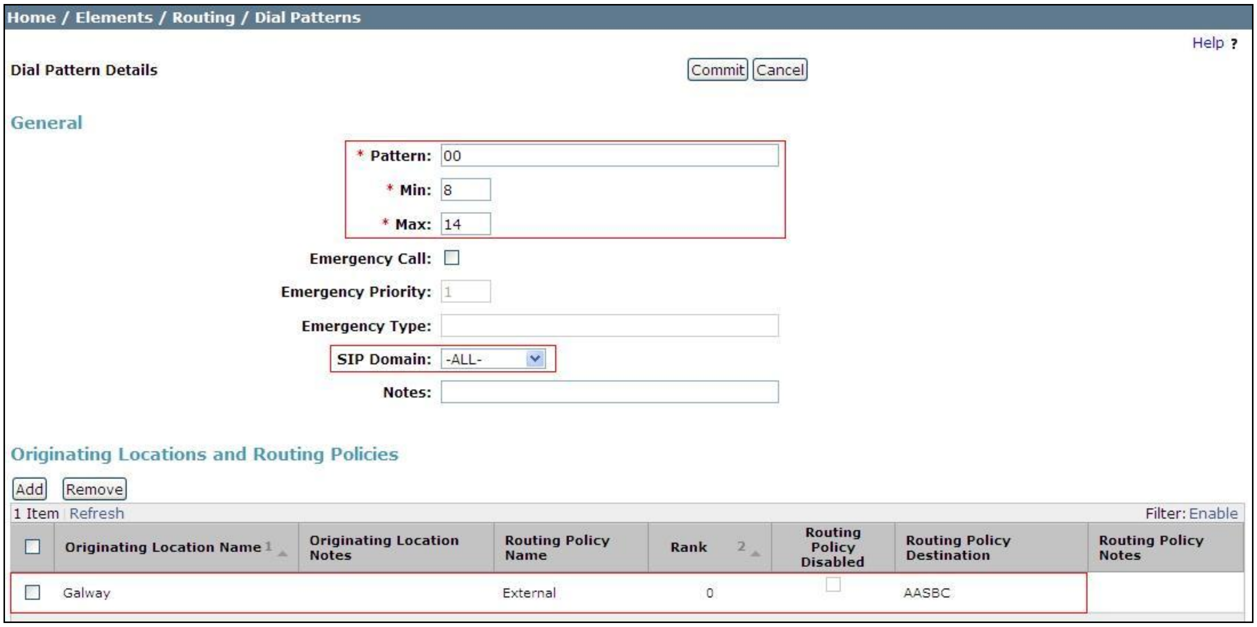
A dial pattern must be defined to direct calls to the appropriate telephony system. To configure a dial pattern select **Dial Patterns** on the left panel menu and then click on the **New** button (not shown).

Under **General**:

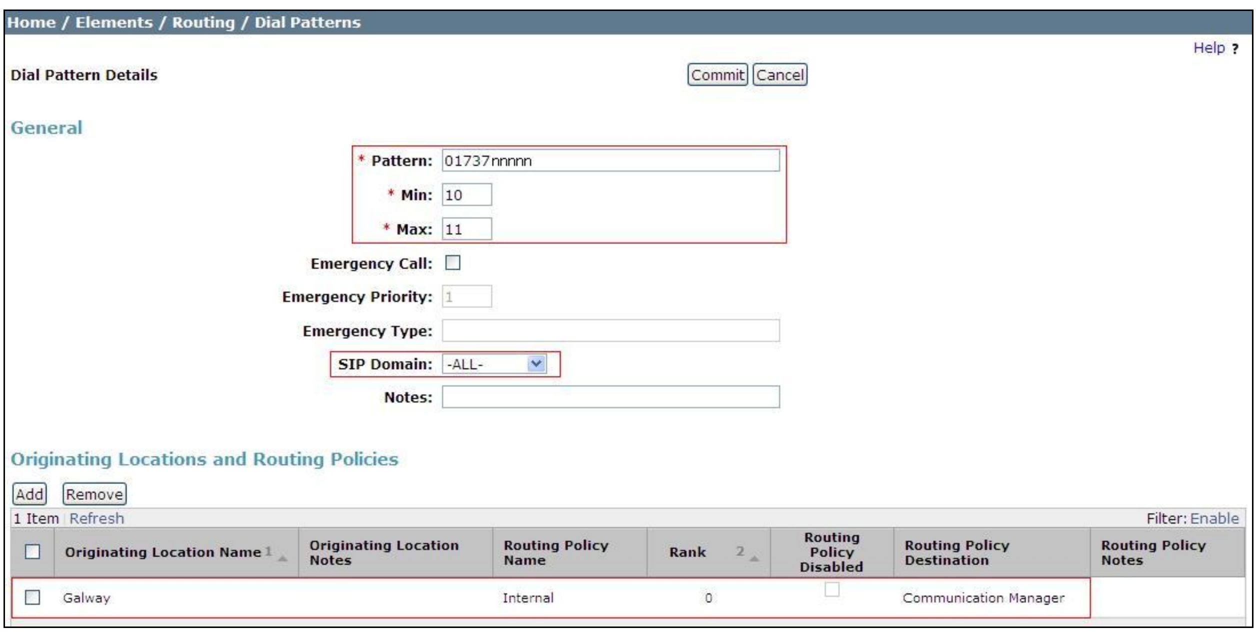
- In the **Pattern** field enter a dialled number or prefix to be matched
- In the **Min** field enter the minimum length of the dialled number
- In the **Max** field enter the maximum length of the dialled number
- In the **SIP Domain** field select **ALL** or alternatively one of those configured in **Section 6.2**

Configuration is continued on the next page.

Under **Originating Locations and Routing Policies**. Click **Add**, in the resulting screen (not shown), under **Originating Location** select the location defined in **Section 6.3** or **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 SBC which will route the calls out to the Updata Business Trunk service.



The following screen shows the test dial pattern configured for Communication Manager.

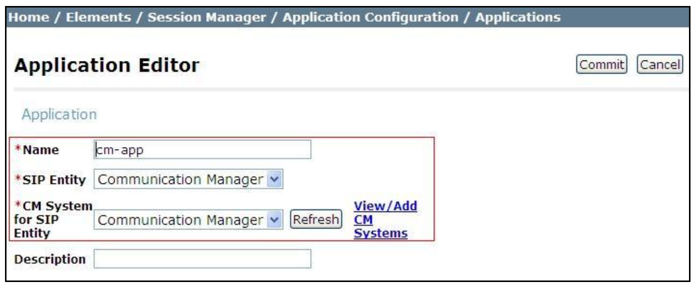


Note: The pattern to be matched has been obscured.

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**.

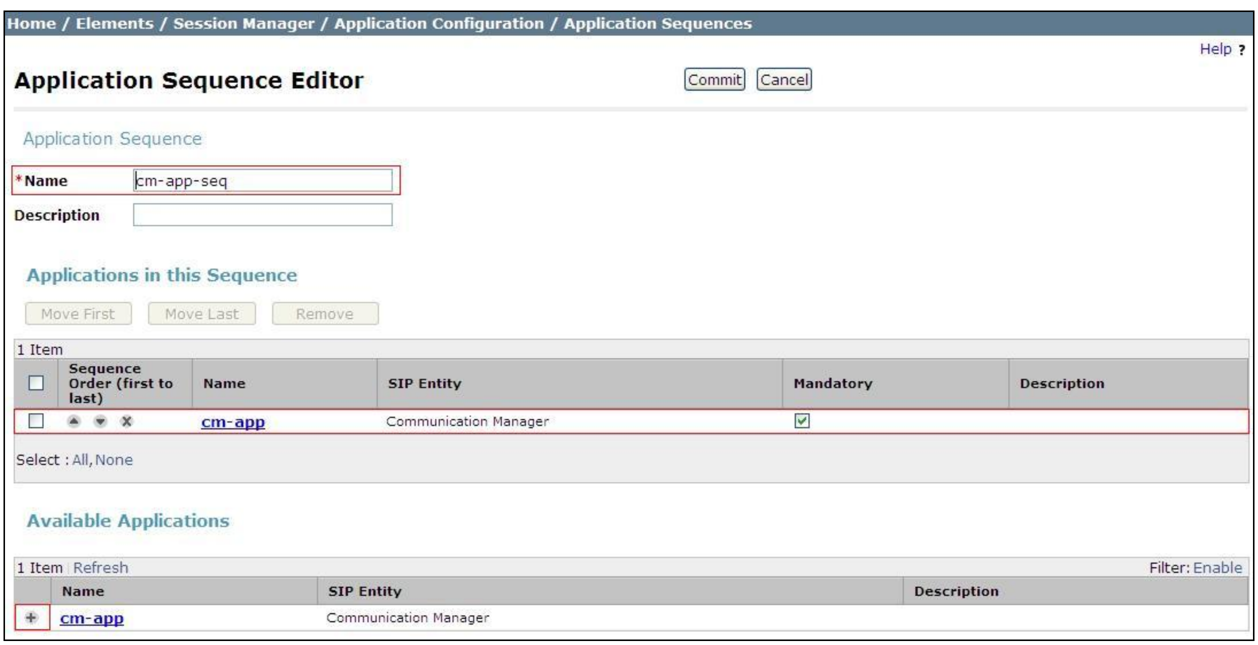
- In the **Name** field enter a name for the application
- In the **SIP Entity** field select the SIP entity for the Communication Manager
- In the **CM System for SIP Entity** field select the SIP entity for the Communication Manager and select **Commit** to save the configuration.



6.9. Administer Application Sequence for Avaya Aura® 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**.

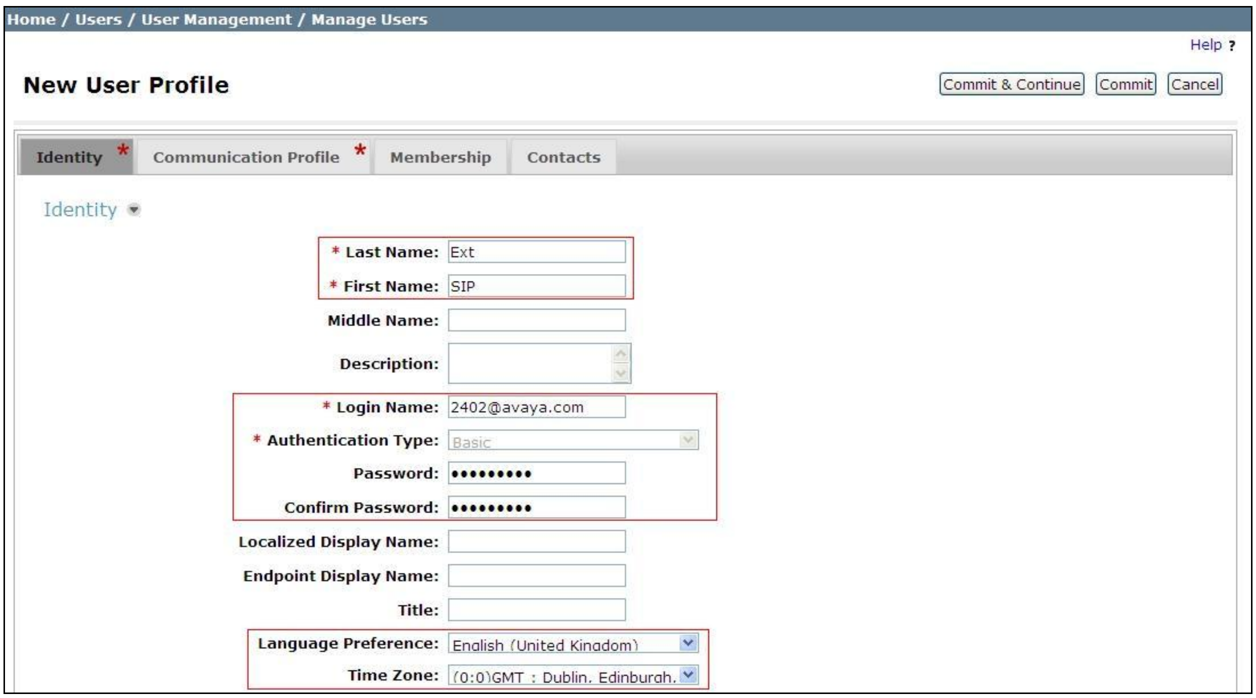


6.10. Administer SIP Extensions

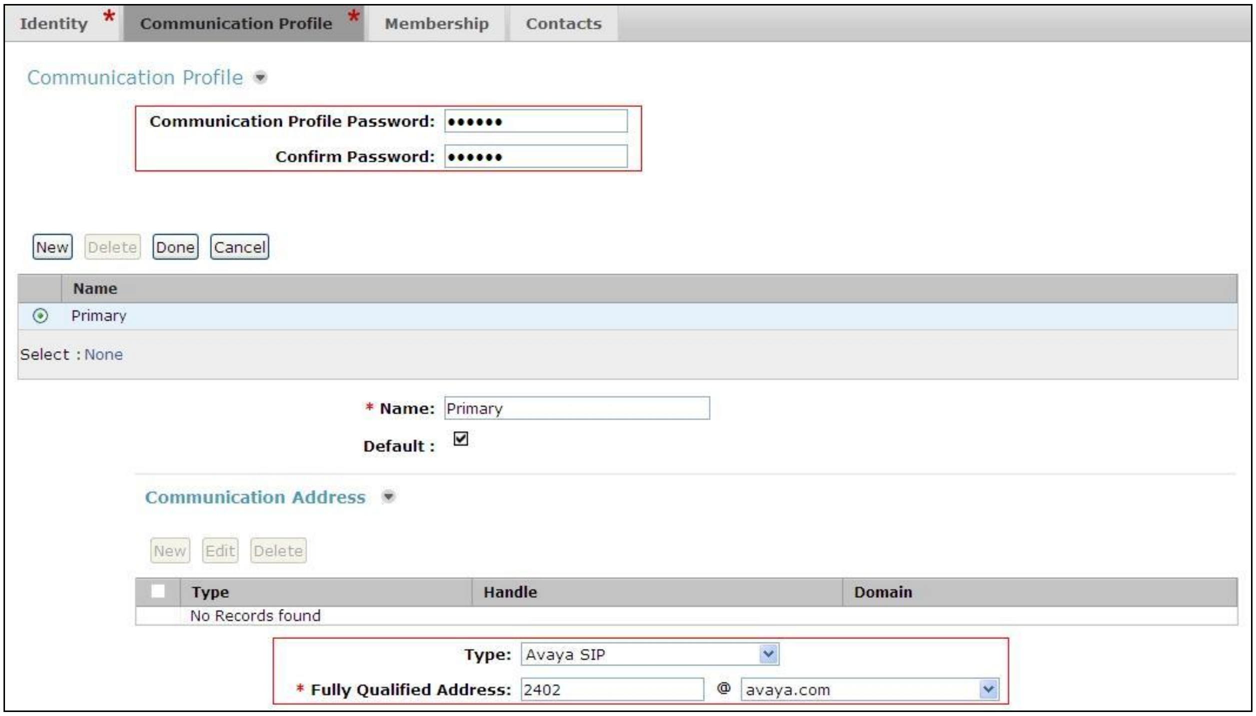
SIP extensions are registered with the Session Manager and use Communication Manager for their feature and configuration settings. From the **Home** tab select **User Management** from the menu. Then select **Manage Users** and click **New** (not shown).

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. **2402@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
- Set the **Language Preference** and **Time Zone** as required



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 **Avaya 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 (not shown).

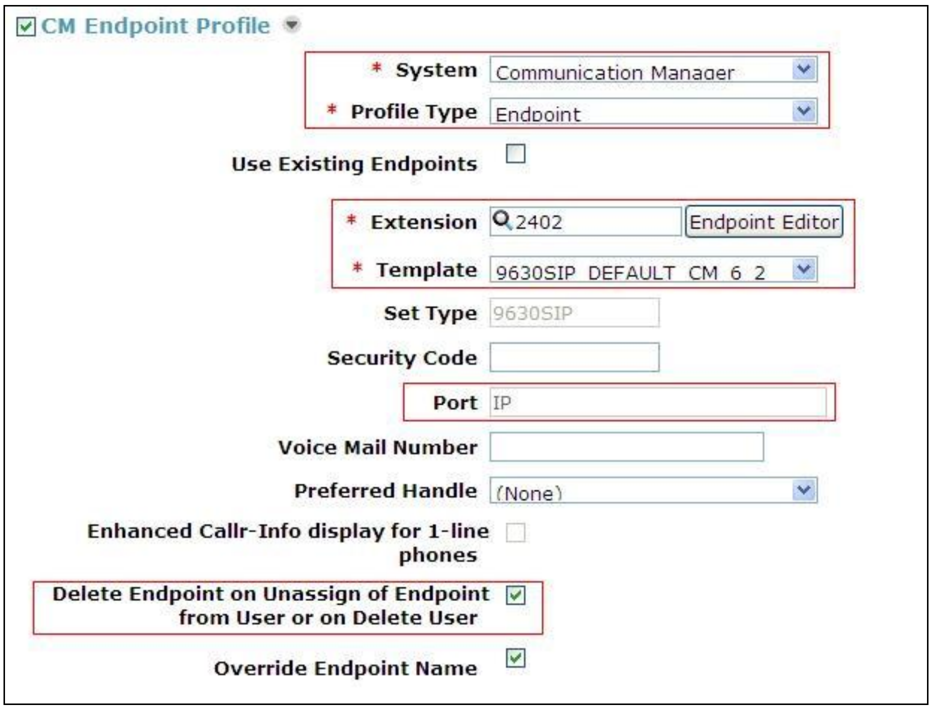


Expand the **Session Manager Profile** section.

- Make sure the **Session Manager Profile** 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** (Not Shown) to save changes and the System Manager will add the Communication Manager user configuration automatically



7. Configure Avaya Aura® Session Border Controller

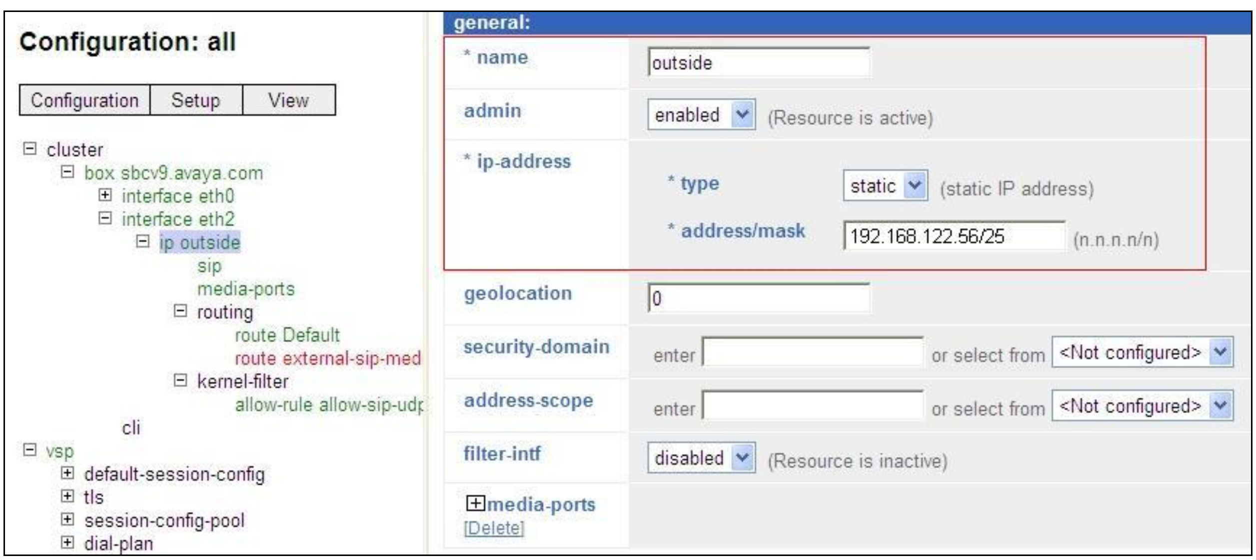
This section provides the procedures for configuring the SBC to receive and route calls over the SIP trunk between Communication Manager and Updata SIP Trunking. These instructions assume other administration activities have already been completed such as the default configuration. This section will cover the configuration that was put in place specifically for Updata.

7.1. Access Avaya Aura® Session Border Controller

Access the SBC using a web browser by entering the URL **https://**<**ip-address**>, where <**ip-address**> is the private IP address configured.

7.2. Verify Outside Interface was configured at installation

An IP address was given to the outside interface that is on the public internet. The IP address is modified in the screenshot below for security purposes. Click on the **Configuratio**n tab and browse to **cluster → interface eth2 → ip outside**.

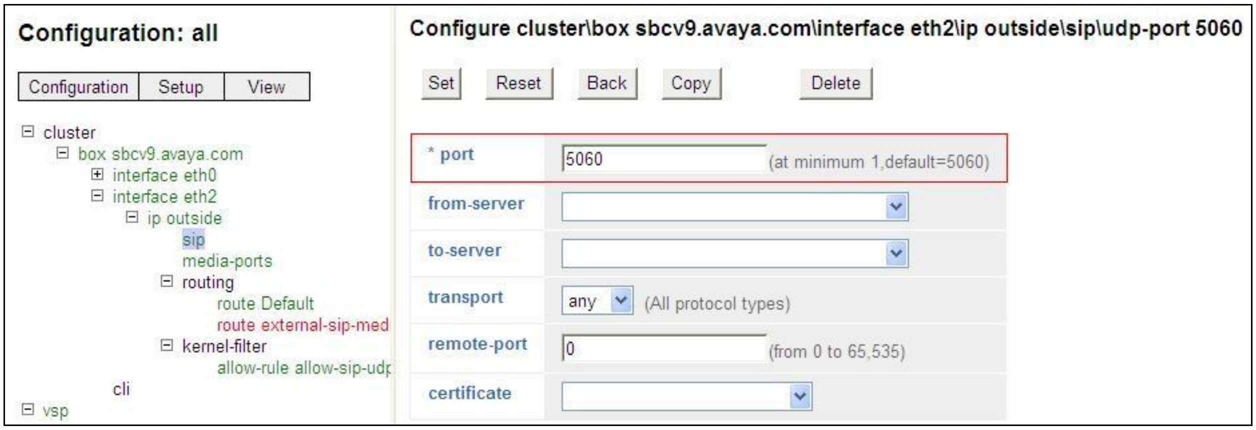


7.3. Configure External Interface

7.3.1. Configure SIP Port

For the outside interface a transport protocol needs to be configured. In the compliance testing UDP was used for the SIP messaging. Click on the Configuration tab and browse to **cluster → interface eth2 → ip outside → sip**.

- **Port** Port number to be used for SIP messaging, default is **5060**



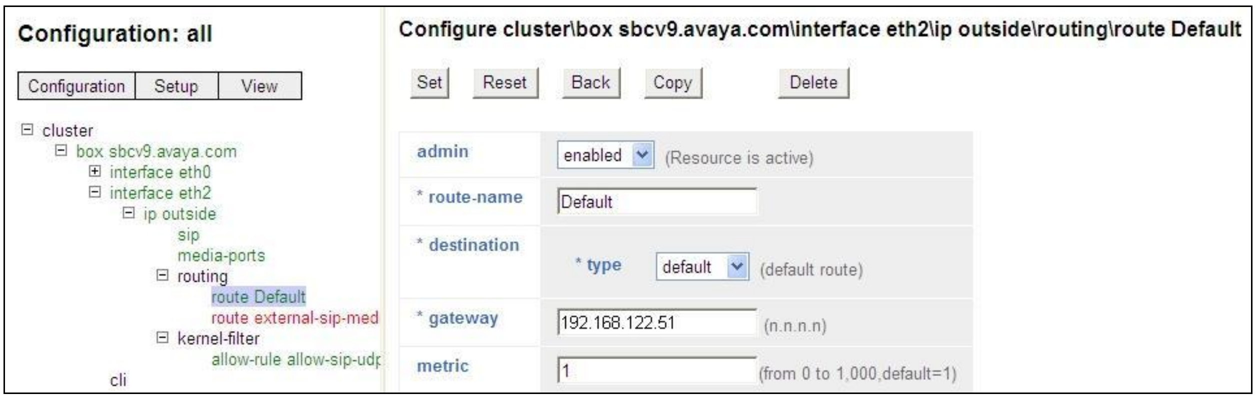
The newly created UDP port is shown below

7.3.2. Configure Routing

Configure routing on the outside interface to correctly route the SIP traffic from the interface to Updata’s network. The IP address is modified in the screenshot below for security purposes. Click on the Configuration tab and browse to **cluster → interface eth2 → ip outside → routing → add route**.

The following values need to be added for the new route that is being created:

- **Admin** Enables or disables this route configuration
- **route name** Enter a name for the route
- **destination type** Use **default** as the network route
- **destination address/mask** The destination address is the subnet used by the service provider and mask
- **gateway** Sets the gateway or next hop IP address for the packet
- **metric** Associates a cost for the route, default is 1



Note: Default routing is used so that all IP packets not recognised as being destined for the enterprise will be routed to the network via the external interface. This will apply to both signalling and media. The gateway address is the next hop router, the address has been modified for security purposes. Note also that a route is configured for external media on the test equipment. This route isn’t used, however, and media is routed by the **Default** route along with signalling.

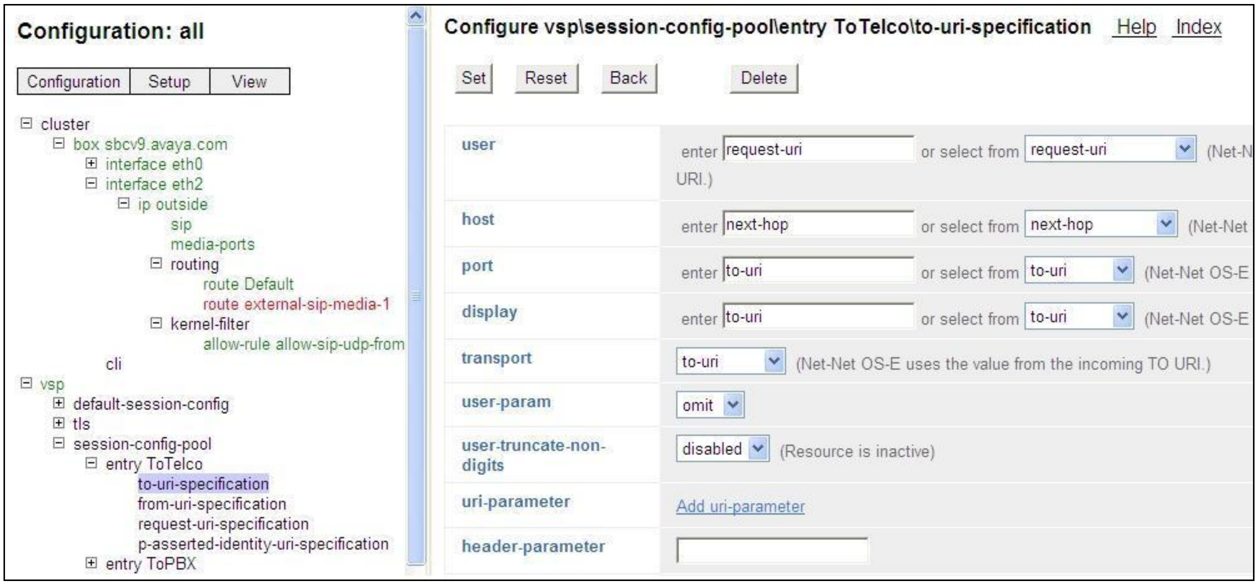
7.3.3. Kernel Filter

The kernel filter was not used during test and configuration is not covered in this document. It is important to check, however, as previous configuration may block IP traffic from the new SIP trunk. To check, browse to **cluster → interface eth2 → ip outside → kernel-filter**. Ensure that no filter is configured and enabled that would disallow traffic from the network.

7.4. Configure VSP

7.4.1. Configure Session-Config-Pool Entry ToTelco

During test, the default action with regard to uri modification for signalling going out to the network was used was used. This replaces domain names and internal IP addresses with external IP addresses only. If modification of the "To" uri is required, expand **vsp → session-config pool → entry ToTelco → to-uri-specification**. The screenshot below shows the default settings.

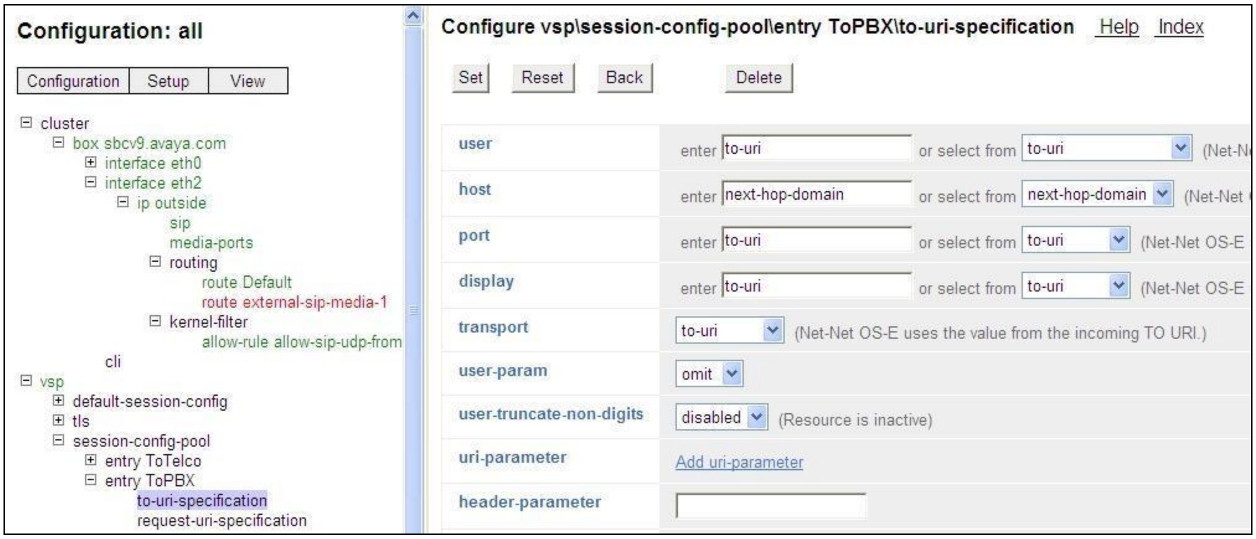


If modification of the "From" uri is required, expand **vsp → session-config pool → entry ToTelco → from-uri-specification**. If modification of the "P-Asserted-ID" uri is required, expand **vsp → session-config pool → entry ToTelco → p-asserted-identity-uri-specification**. Screenshots are not shown for the above settings as default values were used..

7.4.2. Configure Session-Config-Pool Entry ToPBX

During test, the default action with regard to uri modification for signalling going in to the PBX was used. This replaces domain names and external IP addresses with internal IP addresses only If modification is required, expand **vsp → session-config pool → entry ToPBX → to-uri-specification**.

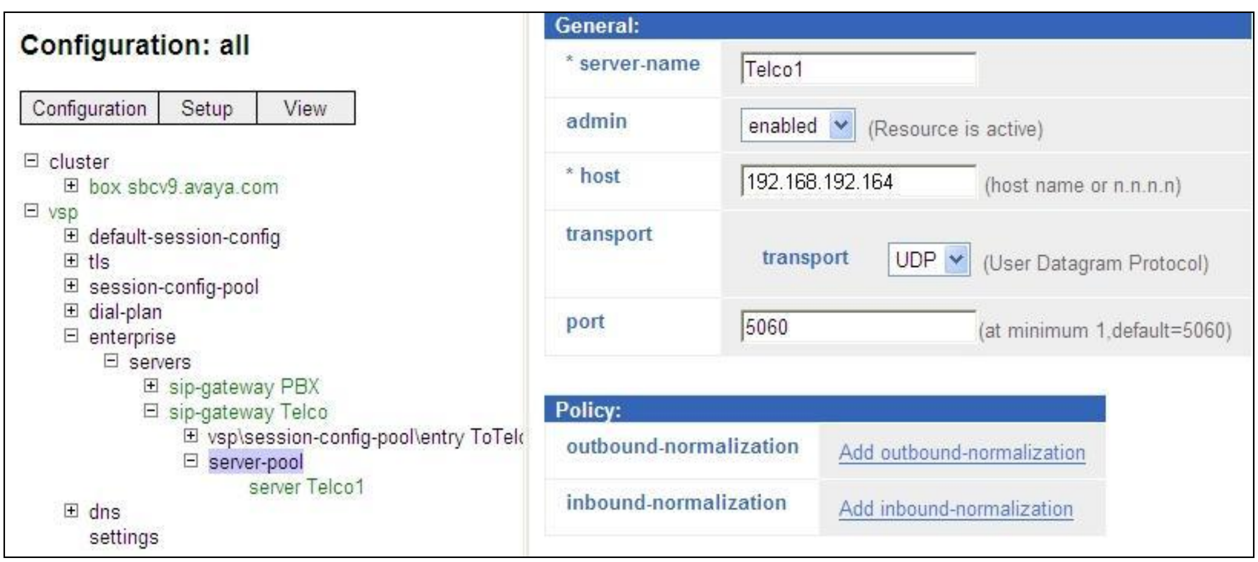
The screenshot below shows the default settings for modification of the “To” uri for signalling going in to the PBX.



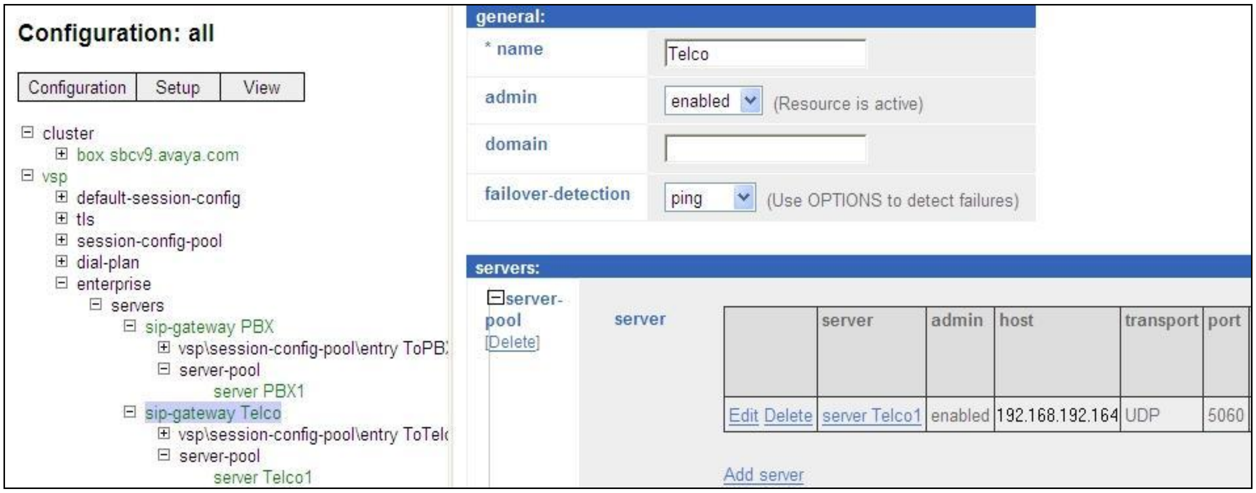
7.4.3. Configure Enterprise

To add the additional IP addresses for the Updata Network SBC 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. Either edit an existing server (not shown) or click the **Add server** link (not shown) as required.

In the resulting page enter a name for the server in the **server-name** field and an IP address in the **host** field. Ensure the **transport** and **port** fields are configured as required. The IP address in the screenshot below has been modified for security purposes. Click the **Set** button (not shown).



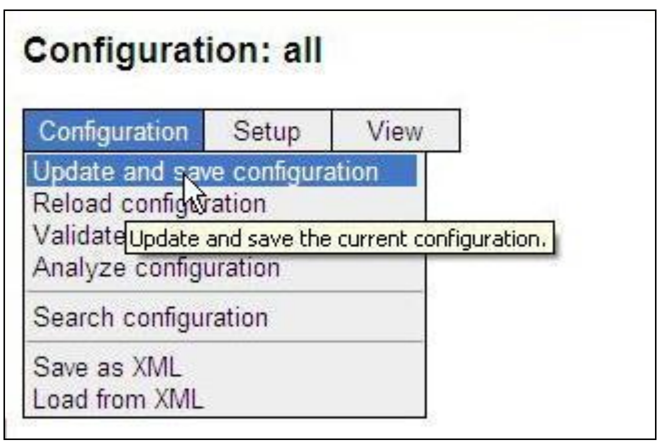
In the resulting page verify the details entered and edit if necessary.



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

7.5. 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**.



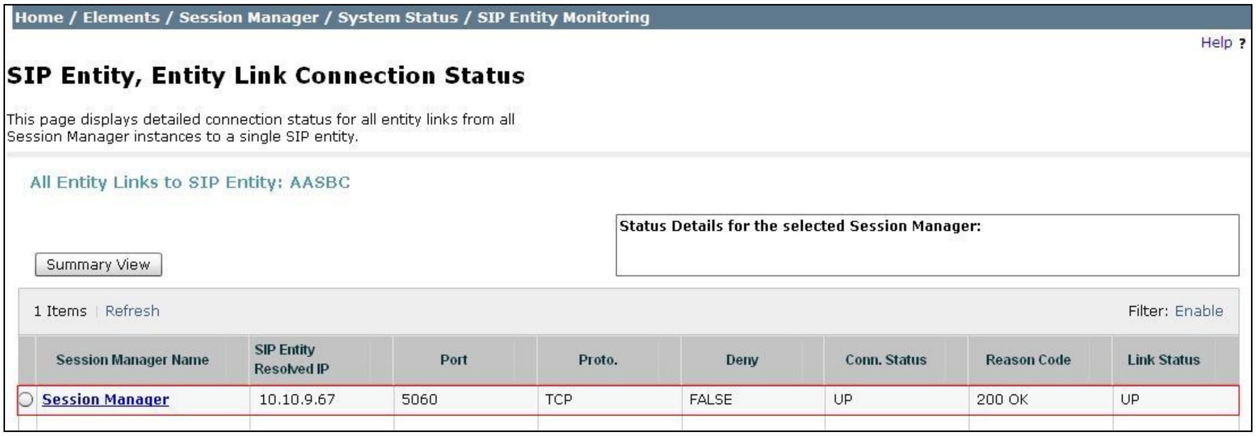
8. Configure Updata Equipment

The configuration of the Updata equipment used to support the Updata SIP Trunking service is outside of the scope of these Application Notes and will not be covered. To obtain further information on Updata equipment and system configuration please contact an authorised Updata 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 R6.2 as an Evolution Server, Avaya Aura® Session Manager R6.3 and Avaya Aura® Session Border Controller to the Updata SIP Trunking service. Updata SIP Trunking 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. Additional 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 6.2.2, December 2012.

[2] *Administering Avaya Aura® System Platform*, Release 6.2.1, July 2012.

[3] *Administering Avaya Aura® Communication Manager*, Release 6.2, December 2012.

[4] *Avaya Aura® Communication Manager Feature Description and Implementation*, December 2012, Document Number 555-245-205.

[5] *Implementing Avaya Aura® System Manager* Release 6.3, December 2012

[6] *Upgrading Avaya Aura® System Manager to 6.3*, January 2013.

[7] *Administering Avaya Aura® System Manager* Release 6.3, December 2012

[8] *Implementing Avaya Aura® Session Manager* Release 6.3, December 2012

[9] *Upgrading Avaya Aura® Session Manager* Release 6.3, December 2012

[10] *Administering Avaya Aura® Session Manager* Release 6.3, December 2012,

[11] *Installing and Configurinng Avaya Aura® Sesion Border Controller* Release 6.0.1, November 2010

[12] *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.