Nexmo SIP Trunking Configuration Guide

Avaya Aura 6.3.18.0.631804 With Avaya SBCe 6.3.7-01-12611

June 2017

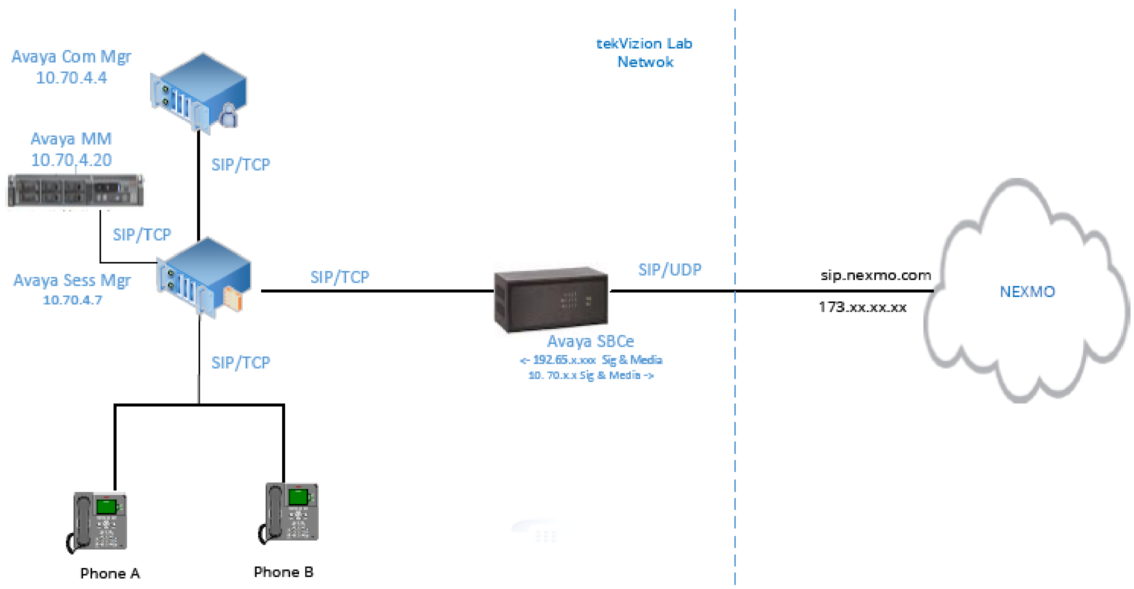
Contents

1 Introduction

This document is intended for the SIP trunk customer's technical staff and Value Added Retailer (VAR) having installation and operational responsibilities. This configuration guide provides steps for configuring Avaya Aura 6.3.18.0.631804 and Avaya SBCe 6.3.7-01-12611 to Nexmo SIP Trunking services.

2 SIP Trunking Network Components

The network for the SIP trunk reference configuration is illustrated below and is representative of an Avaya Aura and Avaya SBCe configuration to Nexmo SIP trunking.



2.1 Network Components

Component	Version	Comments
Avaya Aura	6.3.18.0.631804	
Avaya SBCe	6.3.7-01-12611	
Avaya MM	5.2-11.0	Avaya Voicemail

Avaya 9630G	Version: SIP96xx_2_6_14.5.bin	Avaya Phone
Cisco IP Phone	Model: CP-7965	This Cisco IP Phone is the PSTN test device
	App Load ID: jar45sccp.9-4-2TH1-1.sbn	
	Boot Load ID: tnp65.9-3-1-CR17.bin	

3 Features

3.1.1 Features Supported

- Incoming and outgoing off-net calls using G711ULAW & G711ALAW voice codecs
- Calling Line (number) Identification Presentation
- Calling Line (number) Identification Restriction
- Call hold and resume
- Call transfer (unattended and attended)
- Call Conference
- Call forward (all, no answer)
- DTMF relay both directions (RFC2833)
- Media flow-through on Avaya SBCe

3.1.2 Features Not Supported by PBX

- None

3.1.3 Caveats and Limitations

- Session refresh is always done by Avaya Aura. The issue does not impact the calls.

4 Configuration

4.1 IP Address Worksheet

The specific values listed in the table below and in subsequent sections are used in the lab configuration described in this document, and are for **illustrative purposes only**. The customer must obtain and use the values for your deployment.

Table 1 – IP Addresses

Component	Lab Value	Customer Value
Avaya SBCe		
LAN IP Address	10.70.4.13	
LAN Subnet Mask	255.255.255.0	
WAN IP Address	192.xx.xx.XXX	
WAN Subnet Mask	255.255.255.128	
Avaya Aura		
System IP Address	10.70.4.3	

4.2 Configuring Avaya Aura Communication Manager

This section describes the Avaya Aura Communication Manager configuration necessary to support connectivity to Avaya SBCe. A SIP trunk is established between Communication Manager and Session Manager for use by signaling traffic to and from Nexmo via Avaya SBCe. It is assumed that the general installation of Communication Manager, the Avaya G430 Media Gateway and Session Manager has been previously completed.

The Avaya Aura Communication Manager configuration was performed using System Access Terminal (SAT) via Putty.

```
This system is restricted to authorized users for legitimate business purposes.
          Unauthorized access is a criminal violation of the law.
           Copyright 1992 - 2013 Avaya Inc. All Rights Reserved.
Except where expressly stated otherwise, this Product is protected by copyright
and other laws respecting proprietary rights. Certain software programs or
portions thereof included in this Product may contain software distributed
under third party agreements, which may contain terms that expand or limit
rights to use certain portions of the Product. Information identifying third
party components and terms that apply to them are available on Avaya's web
           site at: http://support.avaya.com/ThirdPartyLicense/.
```

4.2.1 Licenses

In order to connect to Nexmo, Avaya Aura Communication Manager needs to have enough SIP trunk licenses. Use the display system-parameters customer-options command to verify the available SIP Trunk licenses

```
display system-parameters customer-options                    Page   2 of  12
                                OPTIONAL FEATURES

IP PORT CAPACITIES                                              USED
                     Maximum Administered H.323 Trunks: 4000  0
           Maximum Concurrently Registered IP Stations: 2400  1
             Maximum Administered Remote Office Trunks: 4000  0
Maximum Concurrently Registered Remote Office Stations: 2400  0
              Maximum Concurrently Registered IP eCons: 28    0
  Max Concur Registered Unauthenticated H.323 Stations: 100   0
                        Maximum Video Capable Stations: 2400  0
                   Maximum Video Capable IP Softphones: 20    0
                       Maximum Administered SIP Trunks: 4000  35
  Maximum Administered Ad-hoc Video Conferencing Ports: 4000  0
   Maximum Number of DS1 Boards with Echo Cancellation: 80    0
```

Figure 1: Cisco UCM Version

4.2.2 System Features

Use the change system-parameters features command and ensure Trunk to Trunk Transfer is set to all

```
change system-parameters features                               Page   1 of  19
                            FEATURE-RELATED SYSTEM PARAMETERS
                               Self Station Display Enabled? n
                                    Trunk-to-Trunk Transfer: all
               Automatic Callback with Called Party Queuing? n
    Automatic Callback - No Answer Timeout Interval (rings): 3
                       Call Park Timeout Interval (minutes): 10
        Off-Premises Tone Detect Timeout Interval (seconds): 20
                                 AAR/ARS Dial Tone Required? n

              Music (or Silence) on Transferred Trunk Calls? no
              DID/Tie/ISDN/SIP Intercept Treatment: attendant
    Internal Auto-Answer of Attd-Extended/Transferred Calls: transferred
                  Automatic Circuit Assurance (ACA) Enabled? n



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

4.2.3 IP Node Names

Use the display node-names ip command to verify that node names have been properly defined for Communication Manager (procr) and Session Manager (AASM in this test). These node names will be needed for configuring the Signaling Group later.

```
display node-names ip
                                  IP NODE NAMES
    Name                  IP Address
AASM                    10.70.4.7
AASMHA                  10.70.4.24
default                 0.0.0.0
procr                   10.70.4.4
procr6                  ::
```

4.2.4 IP Codecs

The change ip-codec-set command is used for assigning the proper codecs. For this setup, ip-codec-set 1 is used.

```
display 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.711MU            n          2        20
 2: G.711A             n          2        20
 3: G.729              n          2        20
 4: G.722-64K                     2        20
 5:
 6:
 7:


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

4.2.5 IP Network Region

For this test, IP Network region 3 was created using the change ip-network-region 1 command

```
display ip-network-region 1                                     Page   1 of  20
                               IP NETWORK REGION
  Region: 1
Location: 1       Authoritative Domain: lab.tekvizion.com
    Name: SIP                    Stub Network Region: n
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
```

4.2.6 Signaling Group

Use the add signaling-group x command to create a signaling group 2 between Communication Manager and Session Manager for SIP trunk calls.

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

 Group Number: 2                Group Type: sip
  IMS Enabled? n          Transport Method: tcp
        Q-SIP? n
     IP Video? y            Priority Video? n        Enforce SIPS URI for SRTP? n
  Peer Detection Enabled? y  Peer Server: SM
 Prepend '+' to Outgoing Calling/Alerting/Diverting/Connected Public Numbers? y
Remove '+' from Incoming Called/Calling/Alerting/Diverting/Connected Numbers? n
Alert Incoming SIP Crisis Calls? n
   Near-end Node Name: procr                    Far-end Node Name: AASM
 Near-end Listen Port: 5060                   Far-end Listen Port: 5060
                                           Far-end Network Region: 1

Far-end Domain: lab.tekvizion.com
                                             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
```

4.2.7 Trunk Group

Use the add trunk-group x command to create trunk groups for the associated signaling group, trunk group 2 is associated with Signaling group 2 for SIP trunk between CM and SM.

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

Group Number: 2                    Group Type: sip           CDR Reports: y
  Group Name: Trunk to PSTN               COR: 1        TN: 1        TAC: #002
   Direction: two-way        Outgoing Display? n
 Dial Access? n                                  Night Service:
Queue Length: 0
Service Type: public-ntwrk          Auth Code? n
                                              Member Assignment Method: auto
                                                       Signaling Group: 2
                                                     Number of Members: 6
```

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

TRUNK PARAMETERS

     Unicode Name: auto

                                            Redirect On OPTIM Failure: 5000

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

 Disconnect Supervision - In? y  Out? y


             XOIP Treatment: auto    Delay Call Setup When Accessed Via IGAR? n




 Caller ID for Service Link Call to H.323 1xC: station-extension
```

```
change trunk-group 2                                            Page   3 of  21
TRUNK FEATURES
          ACA Assignment? n            Measured: none
                                                    Maintenance Tests? y



   Suppress # Outpulsing? n  Numbering Format: public
                                              UUI Treatment: service-provider

                                               Replace Restricted Numbers? n
                                              Replace Unavailable Numbers? n


                                Modify Tandem Calling Number: no




 Show ANSWERED BY on Display? y
```

- The Numbering Format is set to Public. Outbound calls to Nexmo uses this trunk and uses the Public Numbering table to send the calling party number.

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

                                       Mark Users as Phone? n
Prepend '+' to Calling/Alerting/Diverting/Connected Number? n
                       Send Transferring Party Information? n
                                  Network Call Redirection? n

                                     Send Diversion Header? y
                                   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: From
            Block Sending Calling Party Location in INVITE? n
                 Accept Redirect to Blank User Destination? n
                                              Enable Q-SIP? n

          Interworking of ISDN Clearing with In-Band Tones: keep-channel-active
```

- Send Diversion Header is enabled to send the diversion information for voice mail.

4.2.8 Route Pattern

Use the change route-pattern 2 command to add routing preference for SIP trunk to Session Manager.

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

     BCC VALUE  TSC  CA-TSC     ITC  BCIE  Service/Feature  PARM  No.  Numbering  LAR
    0 1 2 M 4 W      Request                                     Dgts  Format
                                                              Subaddress
 1: y y y y y n  n              rest                                   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
```

- This route is associated with trunk group 2

4.2.9 Dialing Pattern and Feature Code

Use the change dialplan analysis and change feature-access-codes commands

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

    Dialed      Total  Call        Dialed     Total  Call        Dialed     Total  Call
    String      Length Type        String     Length Type        String     Length Type
   0              1    attd
   2              4    ext
   4              4    ext
   8              1    fac
   9              1    fac
   *              3    fac
   #              4    dac
```

- ARS access code is set to 9

```
change feature-access-codes                                      Page   1 of  10
                               FEATURE ACCESS CODE (FAC)
         Abbreviated Dialing List1 Access Code:
         Abbreviated Dialing List2 Access Code:
         Abbreviated Dialing List3 Access Code:
Abbreviated Dial - Prgm Group List Access Code:
                      Announcement Access Code:
                       Answer Back Access Code: *72

      Auto Alternate Routing (AAR) Access Code: 8
    Auto Route Selection (ARS) - Access Code 1: 9      Access Code 2:
                 Automatic Callback Activation:         Deactivation:
Call Forwarding Activation Busy/DA:       All:          Deactivation:
   Call Forwarding Enhanced Status:       Act:          Deactivation:
                         Call Park Access Code: *70
                       Call Pickup Access Code: *71
CAS Remote Hold/Answer Hold-Unhold Access Code:
                CDR Account Code Access Code:
                      Change COR Access Code:
                 Change Coverage Access Code:
          Conditional Call Extend Activation:           Deactivation:
                   Contact Closure  Open Code:           Close Code:
```

4.2.10 Call Routing

4.2.10.1 Outbound Calls

The change ars analysis command is used for outbound PSTN call routing. 121 is shown as an example setup for this test.

```
change ars analysis 1                                          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
    101xxxx0             8    8    deny     op           n
    101xxxx0             18   18   deny     op           n
    101xxxx01            16   24   deny     iop          n
    101xxxx011           17   25   deny     intl         n
    101xxxx1             18   18   deny     fnpa         n
    10xxx0               6    6    deny     op           n
    10xxx0               16   16   deny     op           n
    10xxx01              14   22   deny     iop          n
    10xxx011             15   23   deny     intl         n
    10xxx1               16   16   deny     fnpa         n
    120                  11   11   deny     fnpa         n
    1200                 11   11   deny     fnpa         n
    121                  11   11   2        natl         n
    122                  11   11   deny     fnpa         n
    123                  3    3    1        svcl         n
```

- Route pattern 2 is used for PSTN call routing.

4.2.11 Caller ID

The change private-numbering 2 command is used to assign the Caller ID for 4 digit Avaya Aura extensions

```
change public-unknown-numbering 2                                Page   1 of   2
                        NUMBERING - PUBLIC/UNKNOWN FORMAT
                                             Total
Ext Ext            Trk       CPN              CPN
Len Code           Grp(s)    Prefix           Len
                                                    Total Administered: 9
 4  2035           2         12014[redacted]  11       Maximum Entries: 9999
 4  2101           2         [redacted]       10
 4  2105           2         [redacted]       10    Note: If an entry applies to
 4  2140           2         12063[redacted]  11    a SIP connection to Avaya
 4  2620           2         [redacted]       10    Aura(R) Session Manager,
 4  2621           1         [redacted]       12    the resulting number must
 4  2622           1         [redacted]       12    be a complete E.164 number.
 4  2623           1         [redacted]       12
 4  2624           1         [redacted]       12    Communication Manager
                                                    automatically inserts
                                                    a '+' digit in this case.
```

- Trunk group number 2 is used.

4.2.12 Avaya Aura Extensions

Create a SIP extension as shown below.

```
change off-pbx-telephone station-mapping 2140                   Page   1 of   3
                 STATIONS WITH OFF-PBX TELEPHONE INTEGRATION

 Station       Application Dial   CC  Phone Number    Trunk       Config  Dual
 Extension                 Prefix                     Selection   Set     Mode
 2140              OPS      ____ - ___ 2140_______________ aar    1_      ___
```

- Enter station extension.
- Application: Type OPS
- Trunk Selection: Type aar

```
change station 2140                                             Page   1 of   6
                                    STATION

Extension: 2140                          Lock Messages? n               BCC: 0
     Type: 9630SIP                       Security Code: *                TN: 1
     Port: S00008                      Coverage Path 1: 1               COR: 1
     Name: Nexmo, U1                   Coverage Path 2:                 COS: 1
                                       Hunt-to Station:

STATION OPTIONS
                                           Time of Day Lock Table: _
              Loss Group: 19
                                                 Message Lamp Ext: 2140

        Display Language: english                  Button Modules: 0

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

                                                         IP Video? n
```

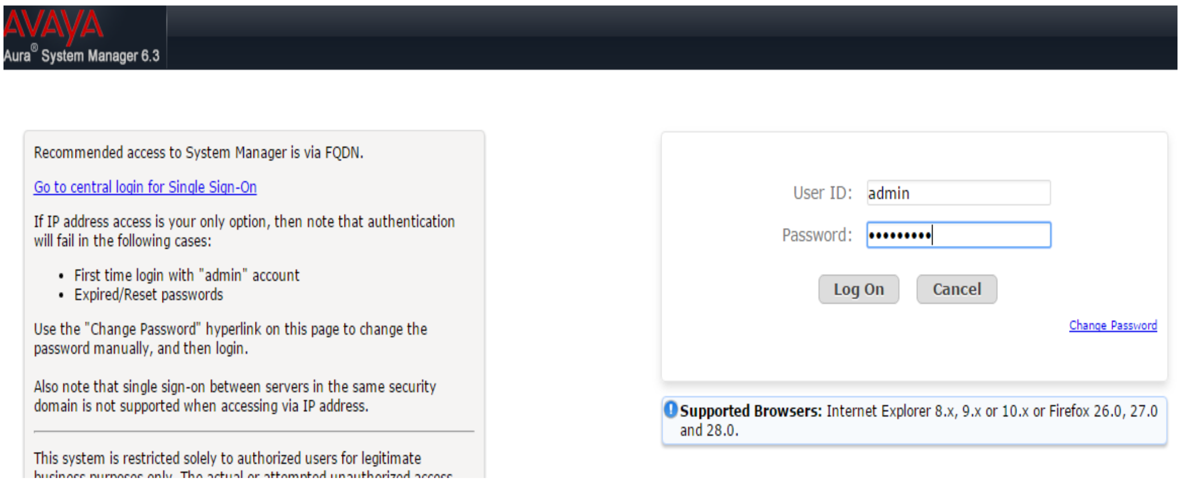
```
change station 2140                                            Page   2 of   6
                                     STATION
FEATURE OPTIONS
           LWC Reception: spe
          LWC Activation? y                        Coverage Msg Retrieval? y
                                                              Auto Answer: none
             CDR Privacy? n                              Data Restriction? n
                                                   Idle Appearance Preference? n
 Per Button Ring Control? n                      Bridged Idle Line Preference? n
   Bridged Call Alerting? n                        Restrict Last Appearance? y
  Active Station Ringing: single

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

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

                                              Direct IP-IP Audio Connections? y
  Emergency Location Ext: 2140         Always Use? n IP Audio Hairpinning? n
```

```
change station 2140                                            Page   3 of   6
                                     STATION


   Bridged Appearance Origination Restriction? n



                         IP Phone Group ID: ___


                           ENHANCED CALL FORWARDING
                                    Forwarded Destination          Active
 Unconditional For Internal Calls To: 2301                           y
                   External Calls To: 2301                           y
          Busy For Internal Calls To:                                n
                   External Calls To:                                n
      No Reply For Internal Calls To:                                n
                   External Calls To: 187[redacted]                  y
```

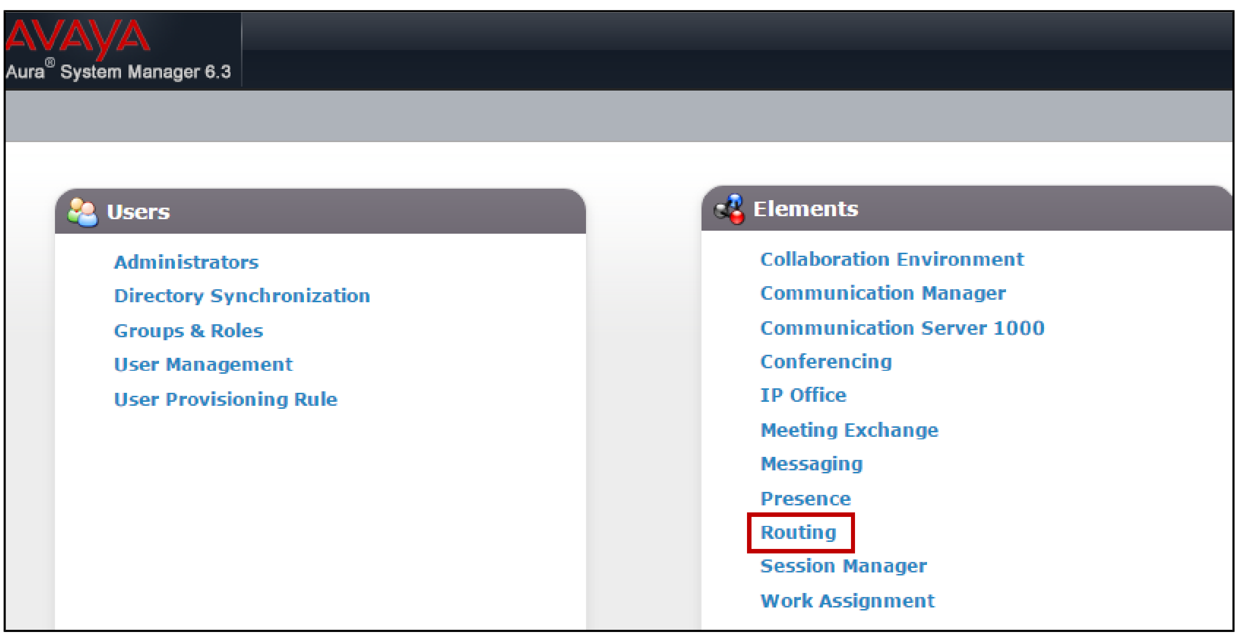
```
change station 2140                                            Page   6 of   6
                                     STATION
SIP FEATURE OPTIONS

         Type of 3PCC Enabled: None
                    SIP Trunk: aar
```

4.3 Configuring Avaya Aura Session Manager

The Avaya Aura Session Manager configuration utilizes Avaya Aura System Manager. The Avaya Aura System Manager Web login screen is accessed via https://<IP Address/FQDN>. Use admin as User ID and input associated password, and then click Log on. It is assumed that the Domain, Location and Endpoint for Session Manager have been previously configured.



Navigate to **Home > Elements > Routing**



4.3.1 Add Adaptations

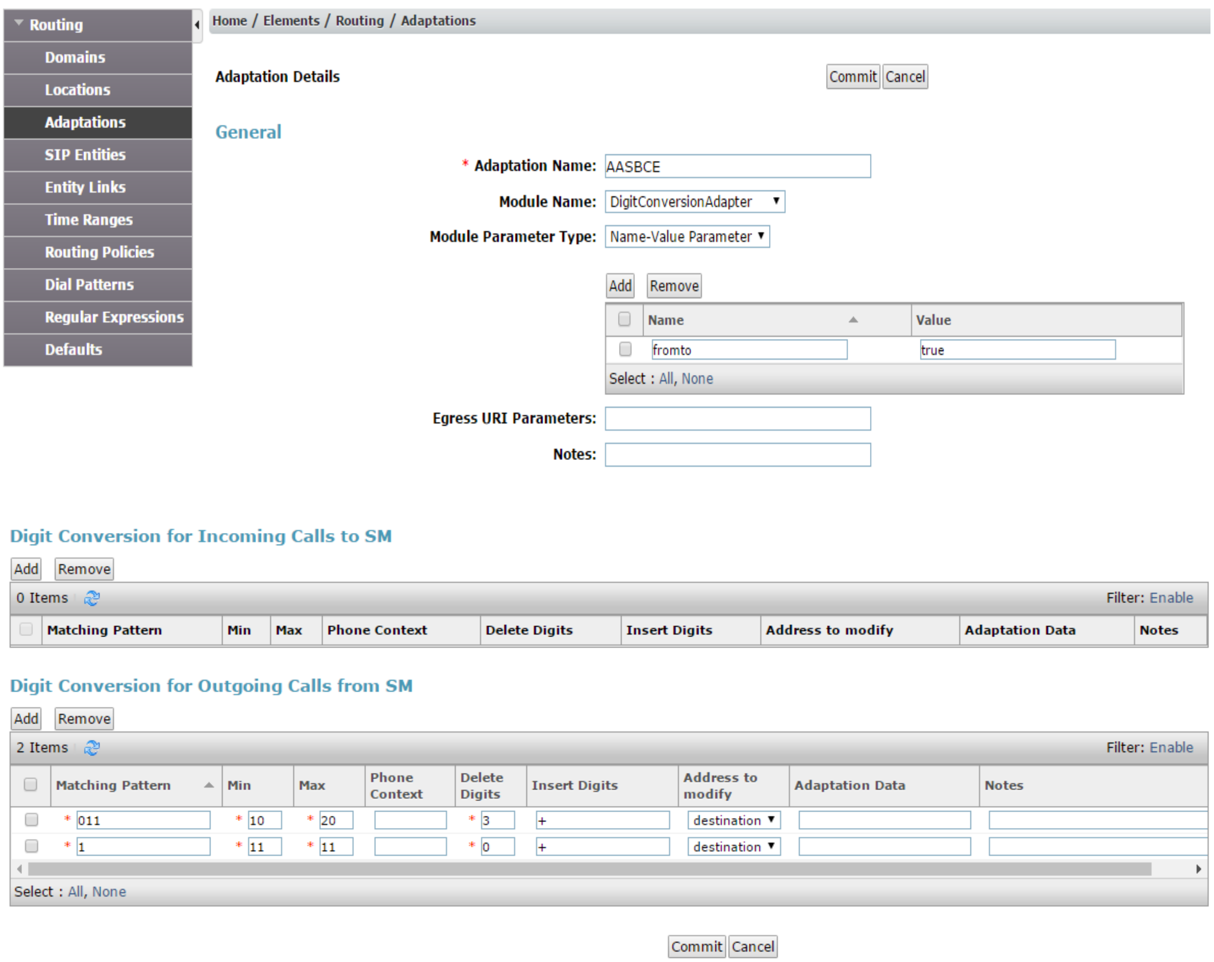
Modifications to the SIP messaging within the Session Manager can be made in the Adaptions module. The idea here is to create an adaptation entity, identified by its Name, and then assign it to a SIP Entity.

Navigate to **Routing > Adaptations > New**

4.3.1.1 Adaptation for Avaya SBCe

The following adaption rules are provisioned in the “Module parameter” field:

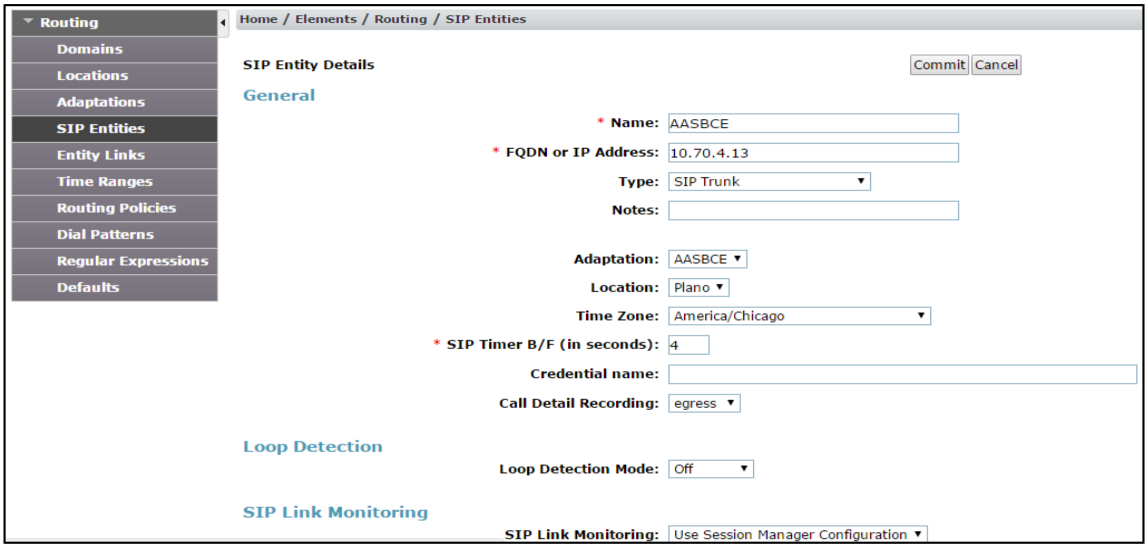
fromto =true: If set to true, then adaptation modifies From and To headers of the message.

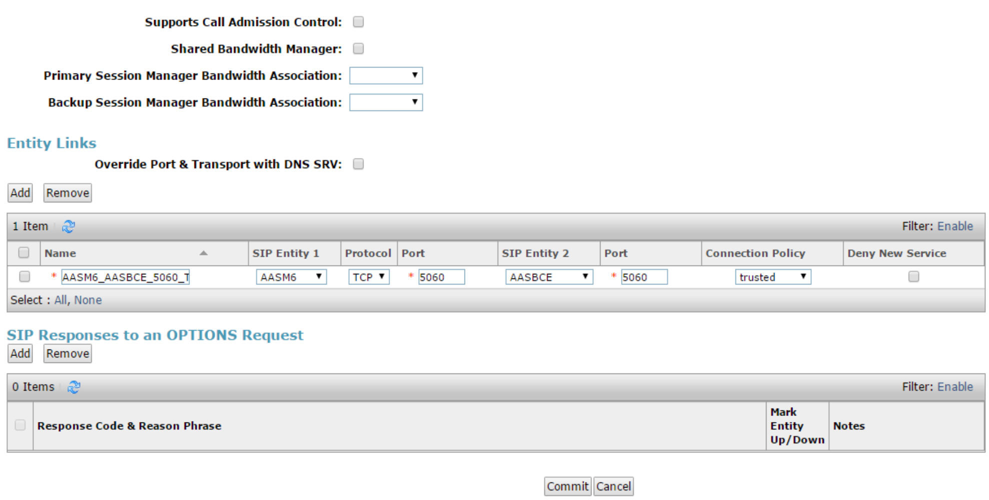


- An adaptation is created under the Digit Conversion for Outgoing Calls from SM to cause SM to insert the + sign in the From and To headers on SM-originated calls routed to Avaya SBCe.

4.3.2 SIP Entities

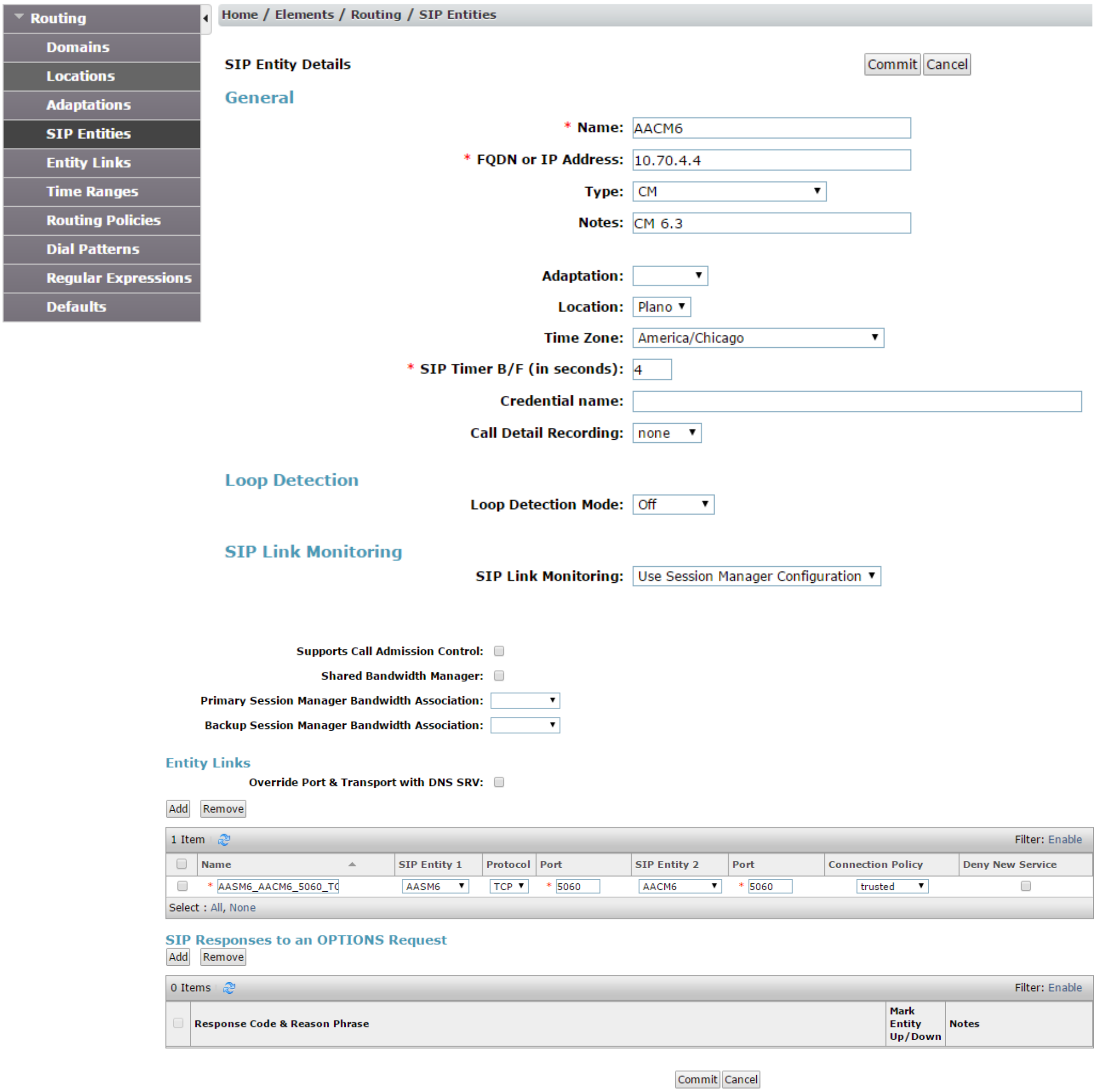
4.3.2.1 SIP Entity for Avaya SBCe





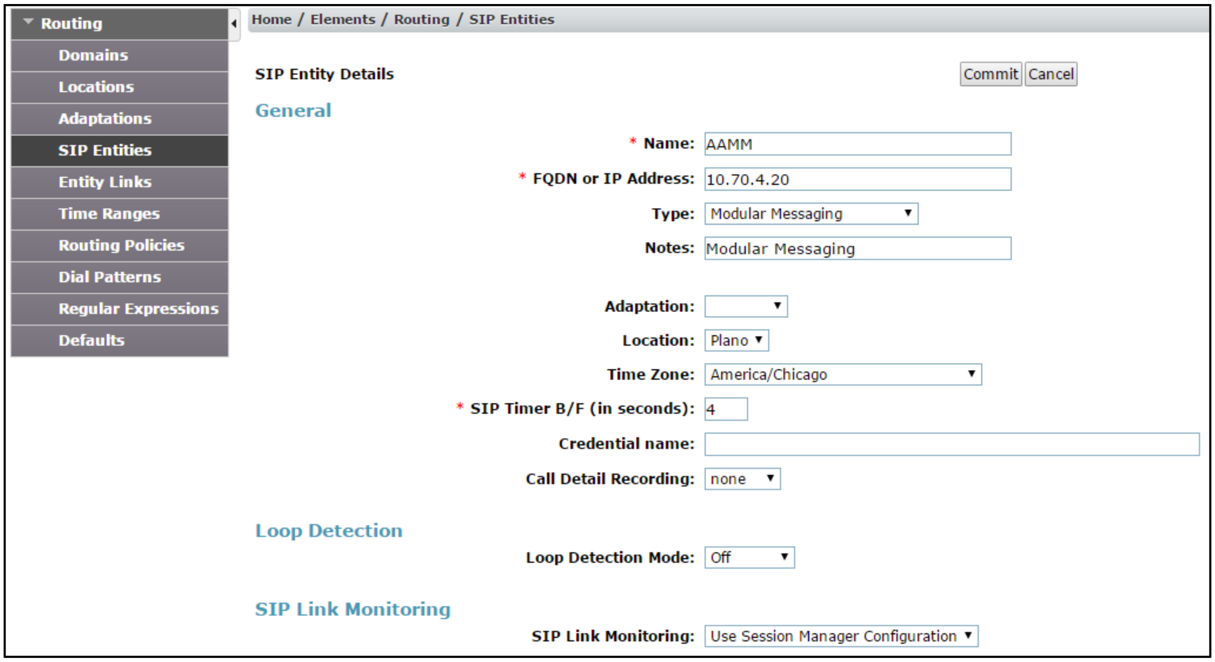
- Add the AvayaSBC Adaption created earlier, to the SIP Entity
- The link between the SM and the Avaya sBCe was configured as trusted using TCP protocol and port 5060

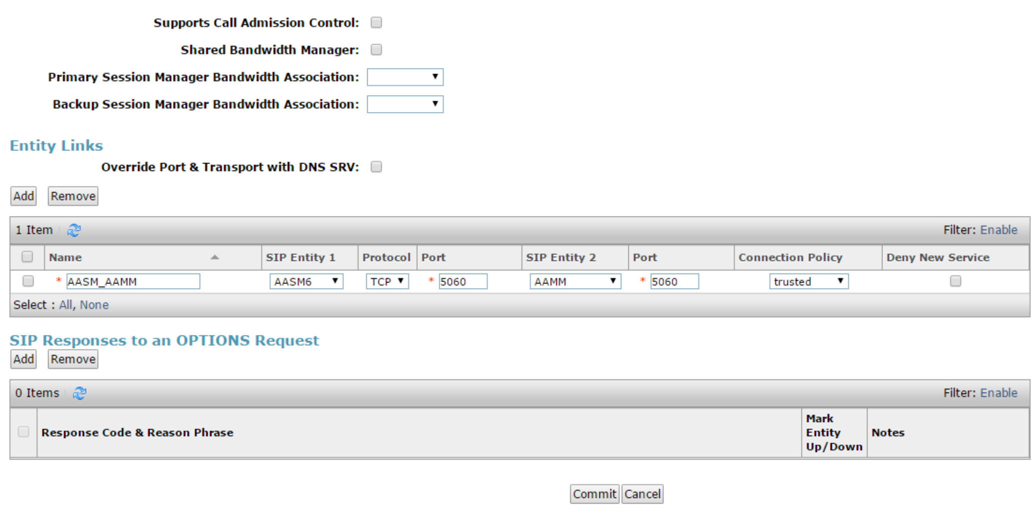
4.3.2.2 SIP Entity for Avaya CM



- The link between the Avaya SM and the CM was configured as trusted using TCP protocol and port 5060.

4.3.2.3 SIP Entity for Avaya MM





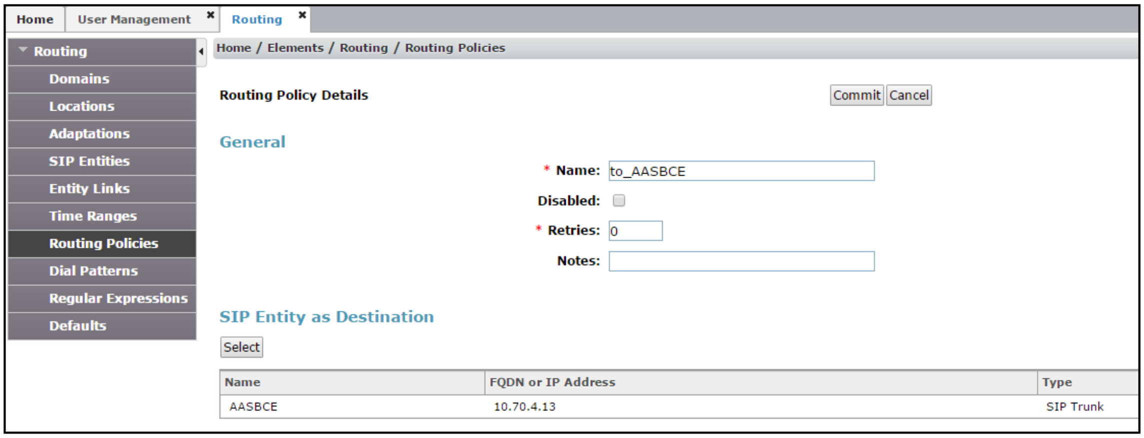
- The link between the Avaya SM and the Avaya MM was configured as trusted using TCP protocol and port 5060

4.3.3 Routing Policies

Navigate to **Routing > Routing Policies > New**

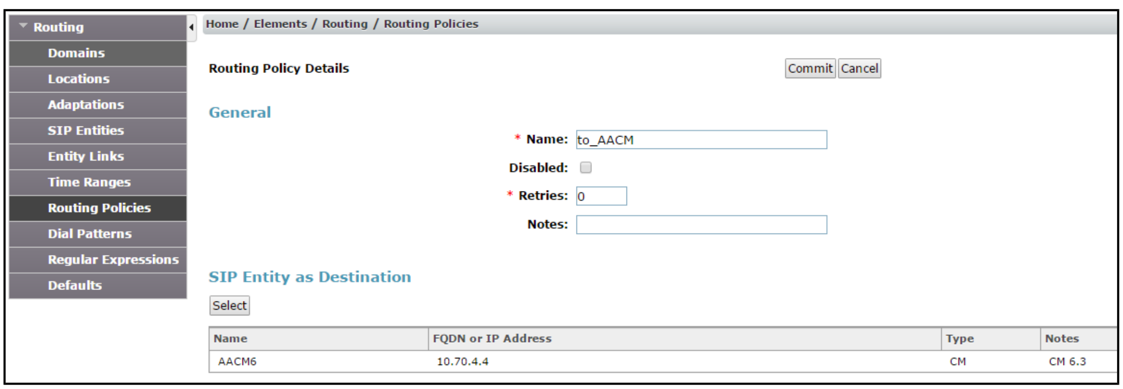
4.3.3.1 Routing Policy to Avaya SBCe

Create a routing policy to Avaya SBCe as shown below.



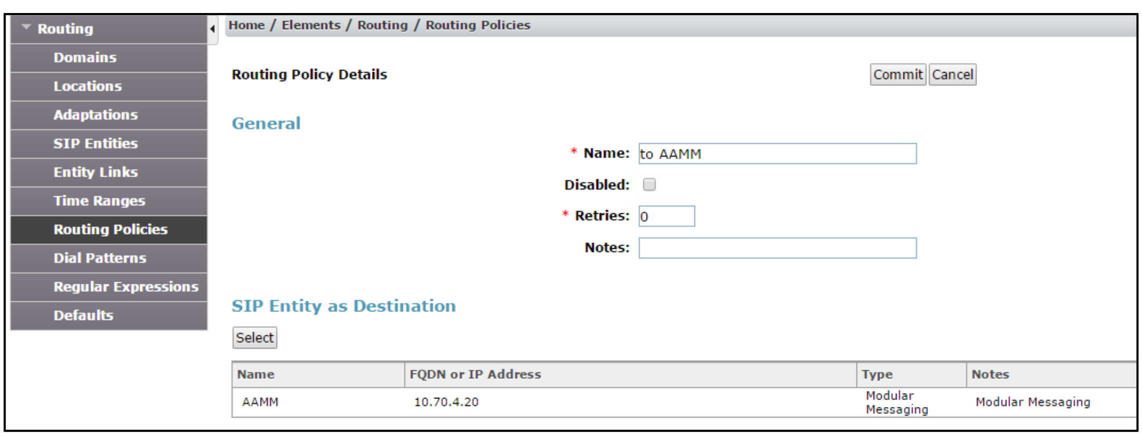
4.3.3.2 Routing Policy to Avaya CM

Create a routing policy to Avaya CM as shown below



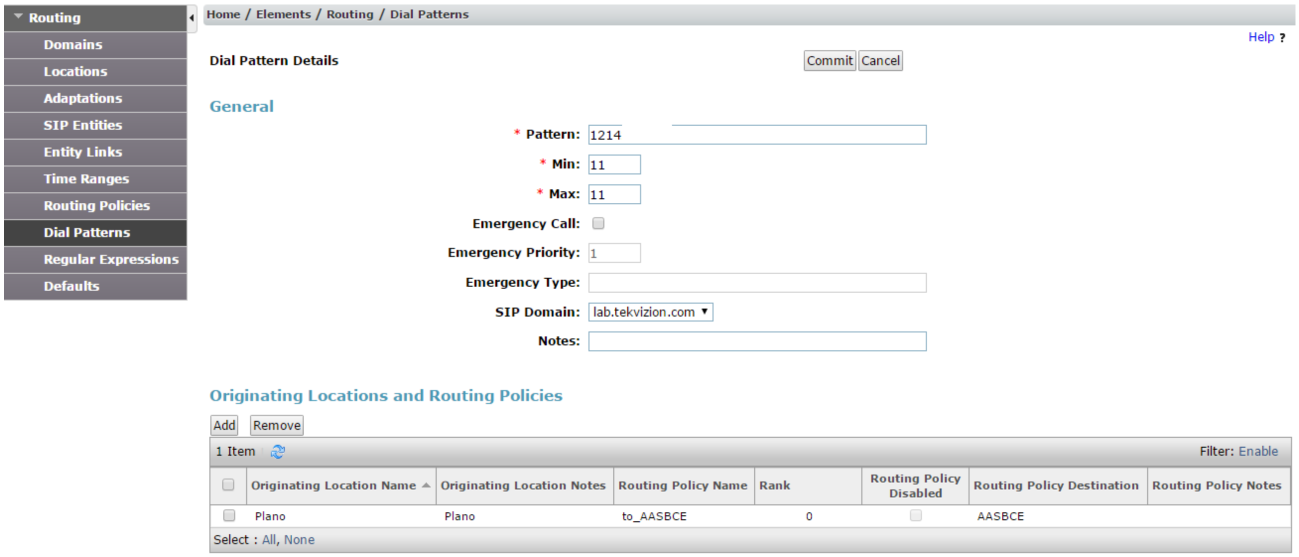
4.3.3.3 Routing Policy to Avaya MM

Create a routing policy to Avaya MM as shown below



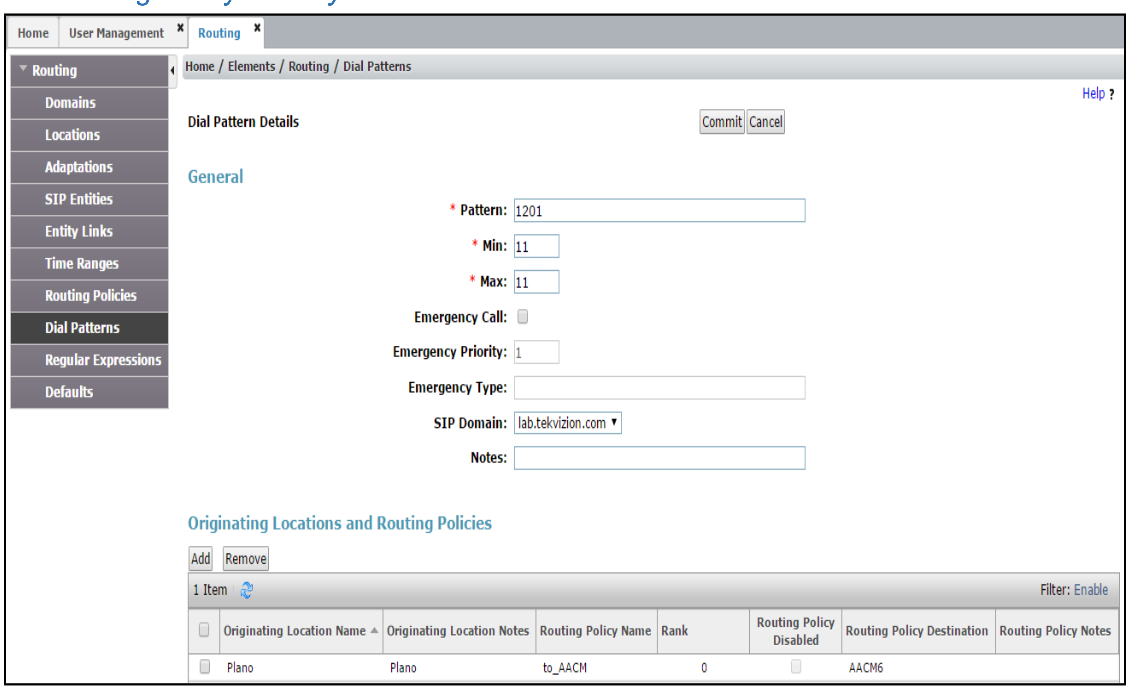
4.3.4 Dial Patterns

4.3.4.1 Routing Policy to Avaya SBCe



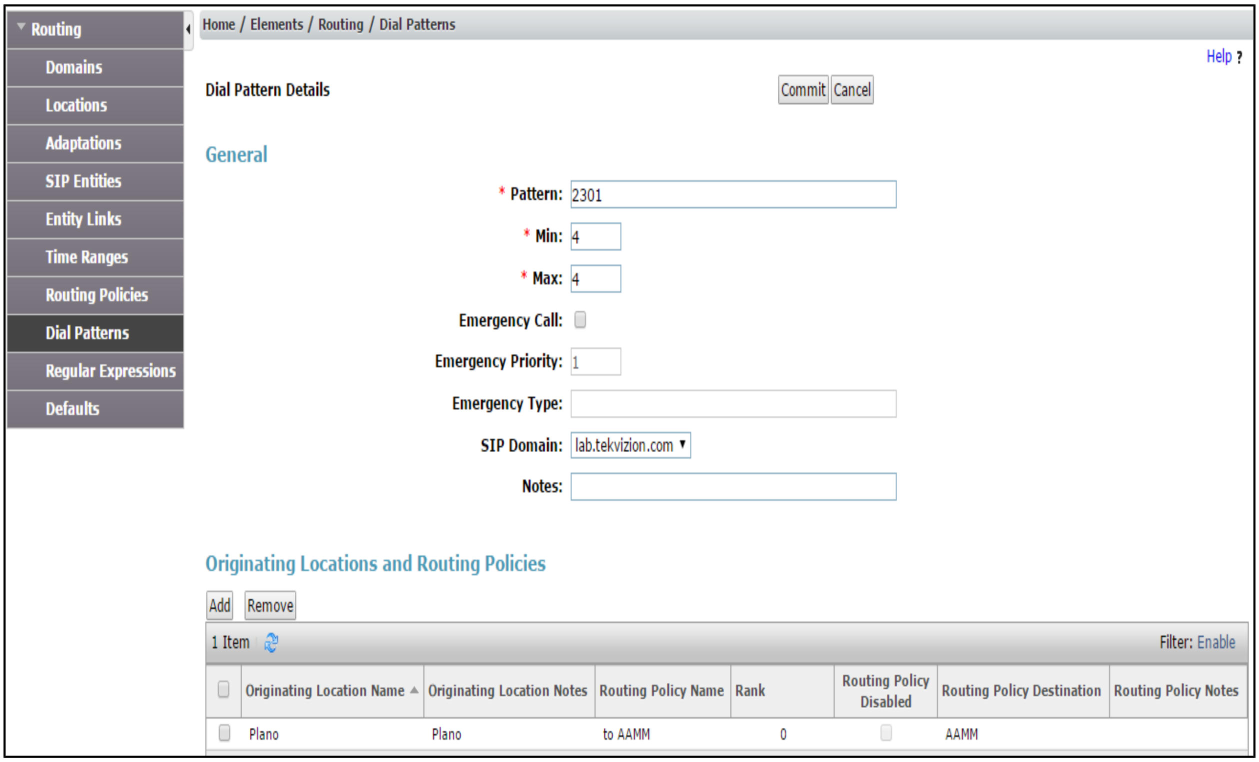
- Create a dial pattern to route the call to PSTN via Avaya SBCe and link the Routing Policy to Avaya SBCe as shown above.

4.3.4.2 Routing Policy to Avaya CM



- Create a dial pattern to route the call to Avaya Aura and link the Routing Policy to Avaya CM as shown above

4.3.4.3 Routing Policy to Avaya MM

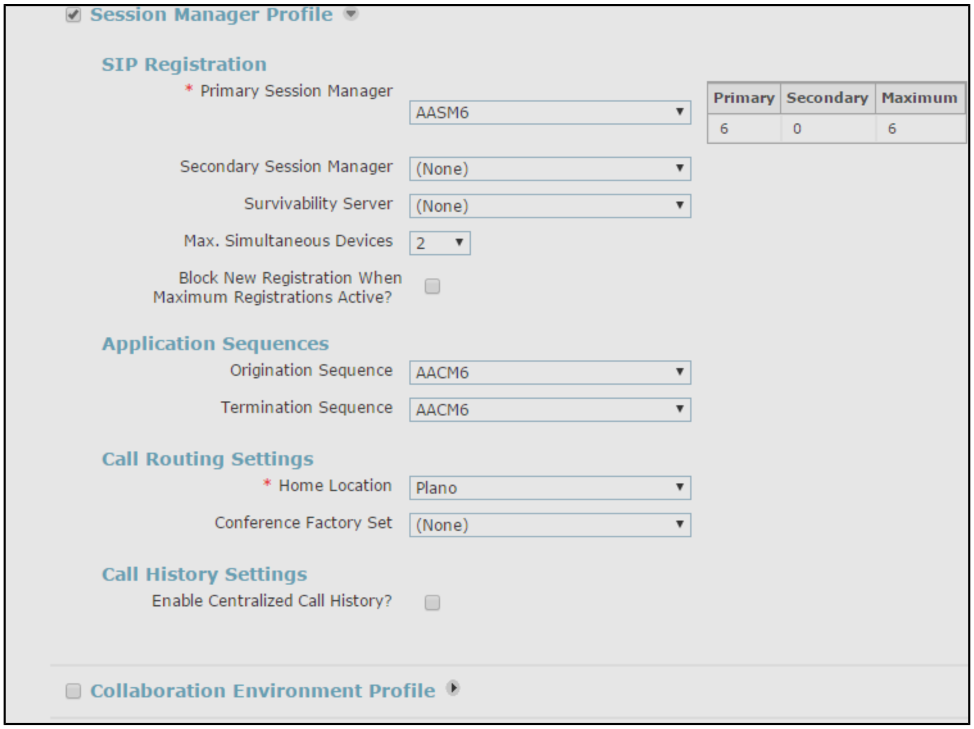


- Create a dial pattern to route the call to Avaya MM and link the Routing Policy to Avaya MM as shown above

4.3.5 SIP Extension

Create a SIP user profile as shown below.

Navigate to **User Management> Endpoints > Manage Users**



Session Manager Profile

SIP Registration

* Primary Session Manager: AASM6

Primary	Secondary	Maximum
6	0	6

Secondary Session Manager: (None)

Survivability Server: (None)

Max. Simultaneous Devices: 2

Block New Registration When Maximum Registrations Active?

Application Sequences

Origination Sequence: AACM6

Termination Sequence: AACM6

Call Routing Settings

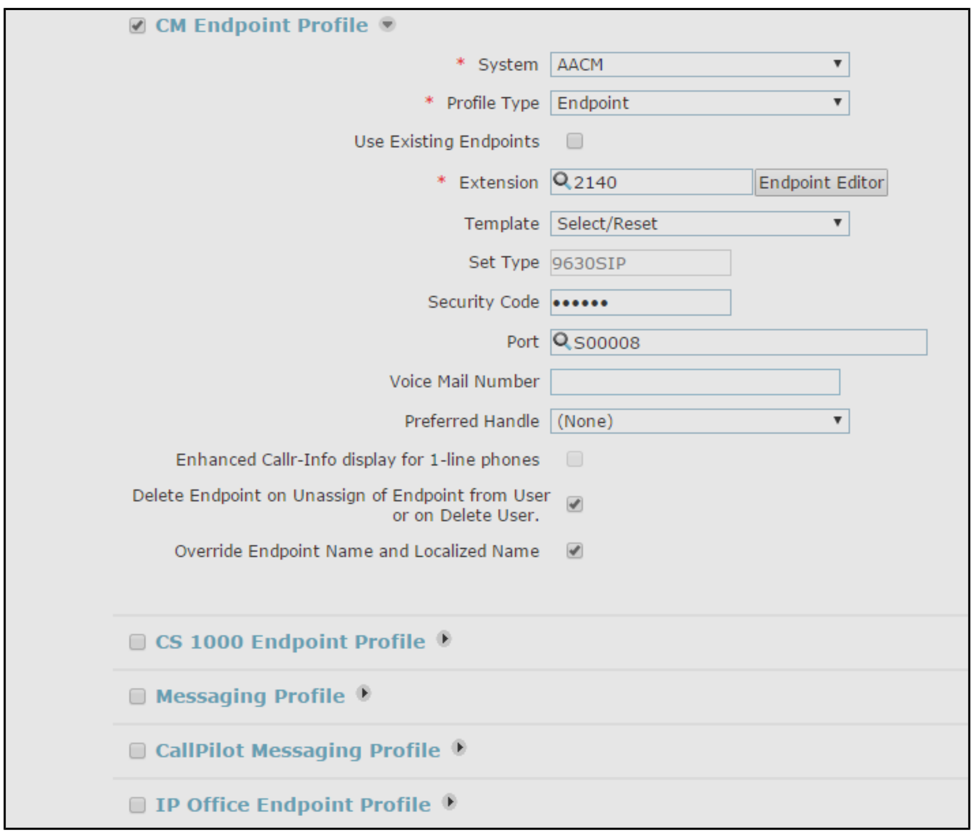
* Home Location: Plano

Conference Factory Set: (None)

Call History Settings

Enable Centralized Call History?

Collaboration Environment Profile



CM Endpoint Profile

* System: AACM

* Profile Type: Endpoint

Use Existing Endpoints

* Extension: 2140 Endpoint Editor

Template: Select/Reset

Set Type: 9630SIP

Security Code: ••••••

Port: S00008

Voice Mail Number

Preferred Handle: (None)

Enhanced Callr-Info display for 1-line phones

Delete Endpoint on Unassign of Endpoint from User or on Delete User.

Override Endpoint Name and Localized Name

CS 1000 Endpoint Profile

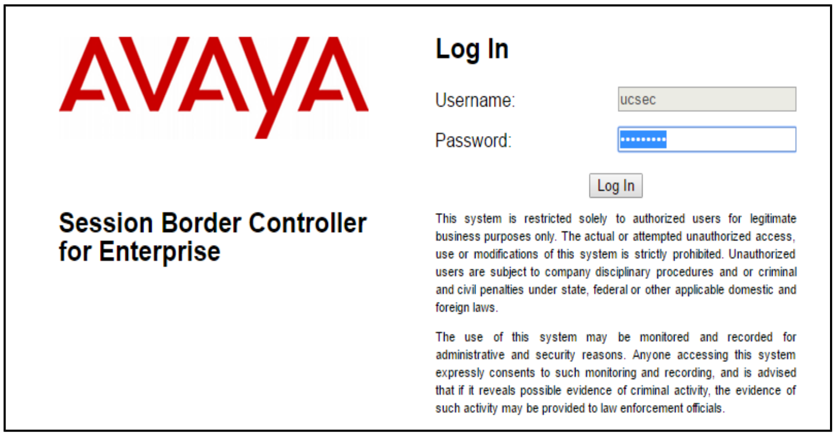
Messaging Profile

CallPilot Messaging Profile

IP Office Endpoint Profile

4.4 Configuring Avaya Session Border Controller for Enterprise

- Log into Avaya Session Border Controller for Enterprise (SBCE) web interface by typing "https://X.X.X.X/sbc".
- Enter the assigned Username and Password
- Click Log In

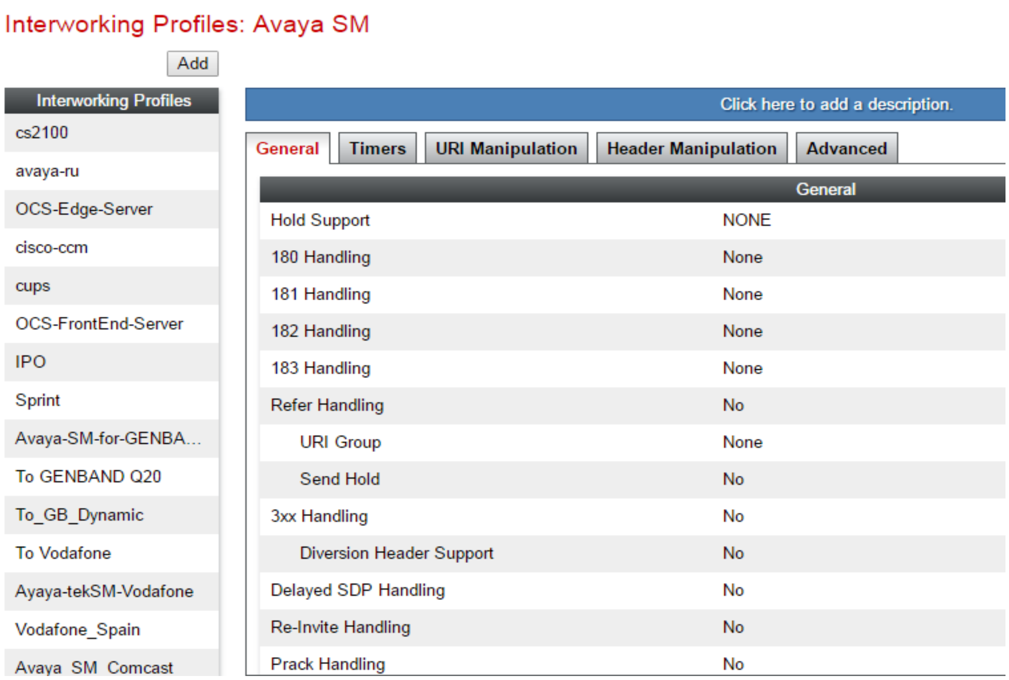


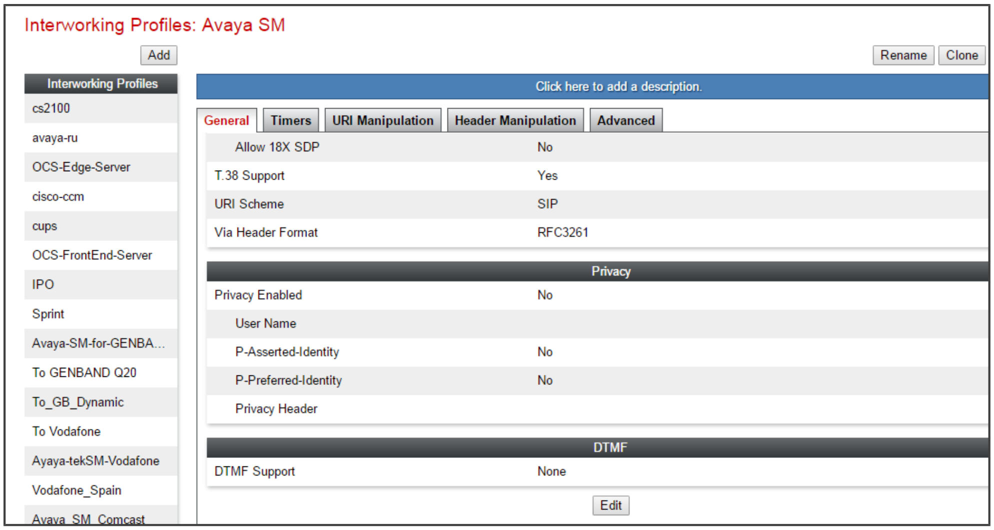
4.4.1 Global Profile

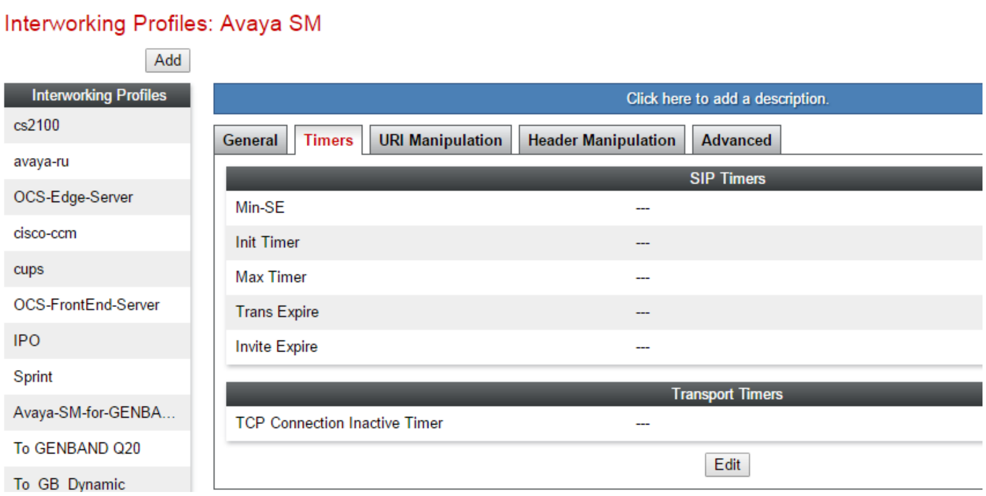
4.4.1.1 Server Interworking

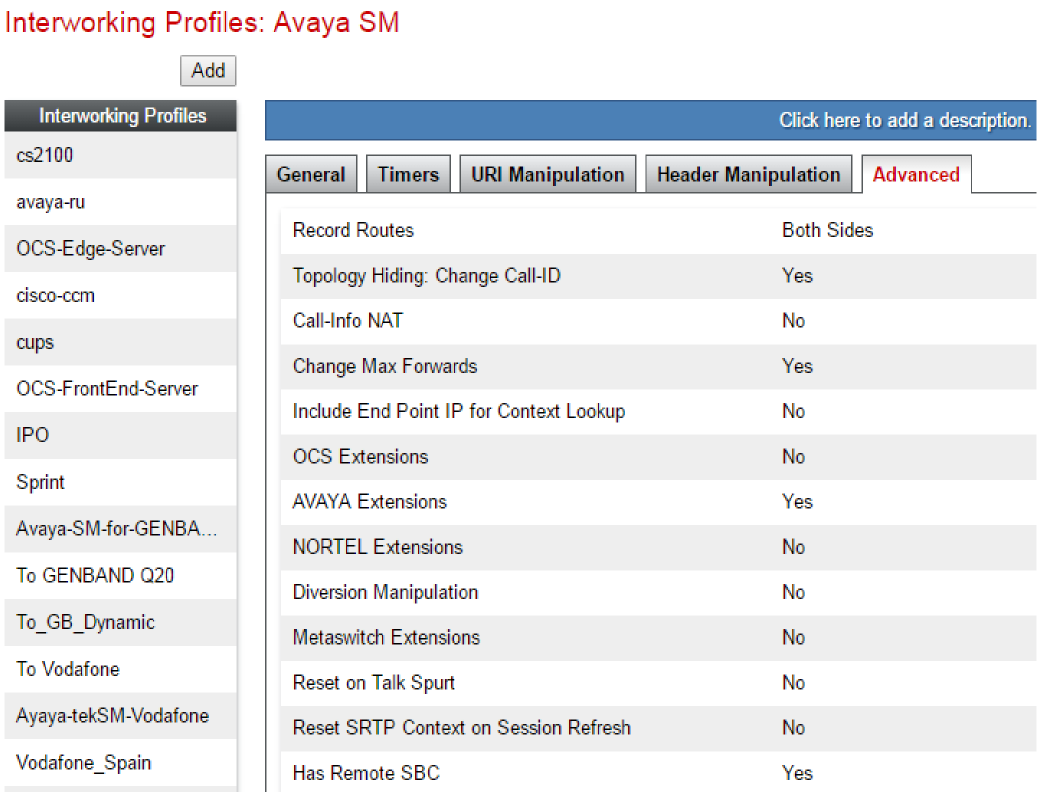
Navigate to System Management > Global Profiles > Server Interworking. Create a clone named AASM of predefined Interworking Profile avaya-ru as shown below.

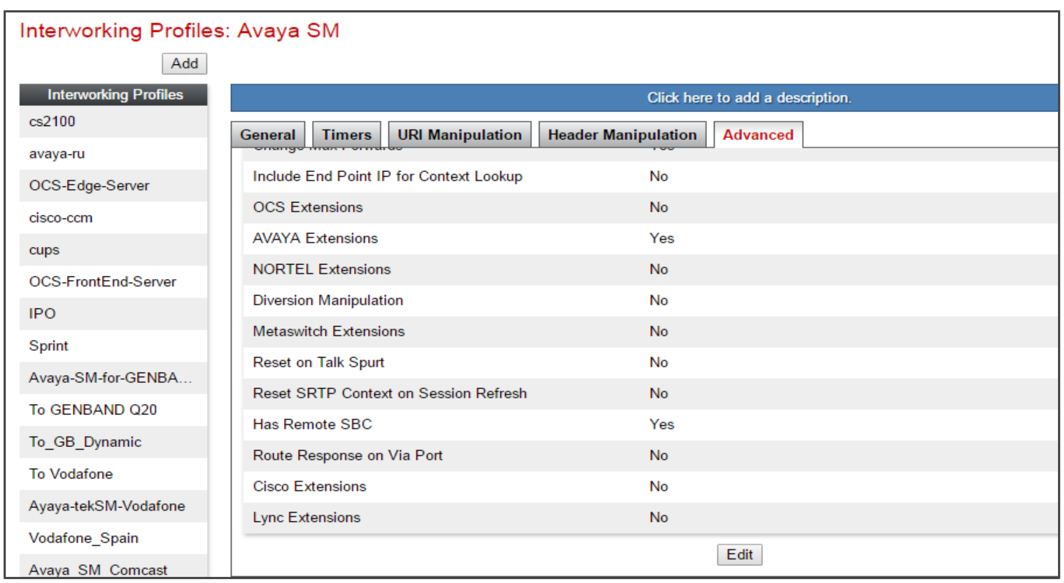
Create a Serving Interworking profile for Avaya SM as shown below.



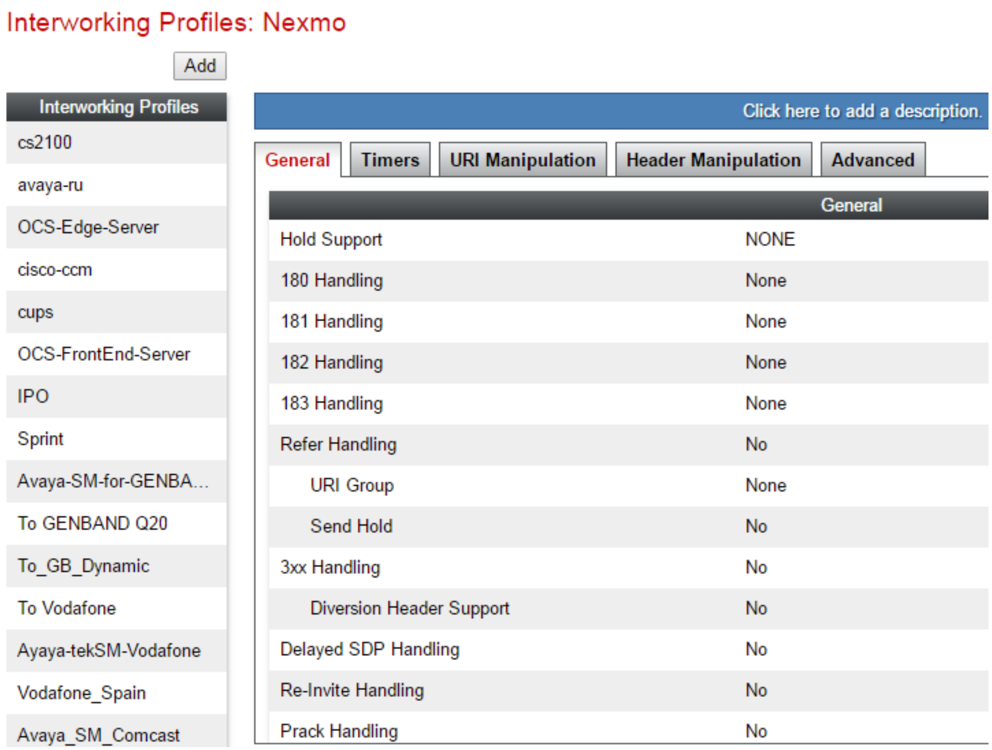

Interworking Profiles: Avaya SM
Add
Rename
Clone
Interworking Profiles
cs2100
avaya-ru
OCS-Edge-Server
cisco-ccm
cups
OCS-FrontEnd-Server
IPO
Sprint
Avaya-SM-for-GENBA...
To GENBAND Q20
To_GB_Dynamic
To Vodafone
Ayaya-tekSM-Vodafone
Vodafone_Spain
Click here to add a description.
General
Timers
URI Manipulation
Header Manipulation
Advanced
Allow 18X SDP No
T.38 Support Yes
URI Scheme SIP
Via Header Format RFC3261
Privacy
Privacy Enabled No
User Name
P-Asserted-Identity No
P-Preferred-Identity No
Privacy Header
DTMF
DTMF Support None
Edit

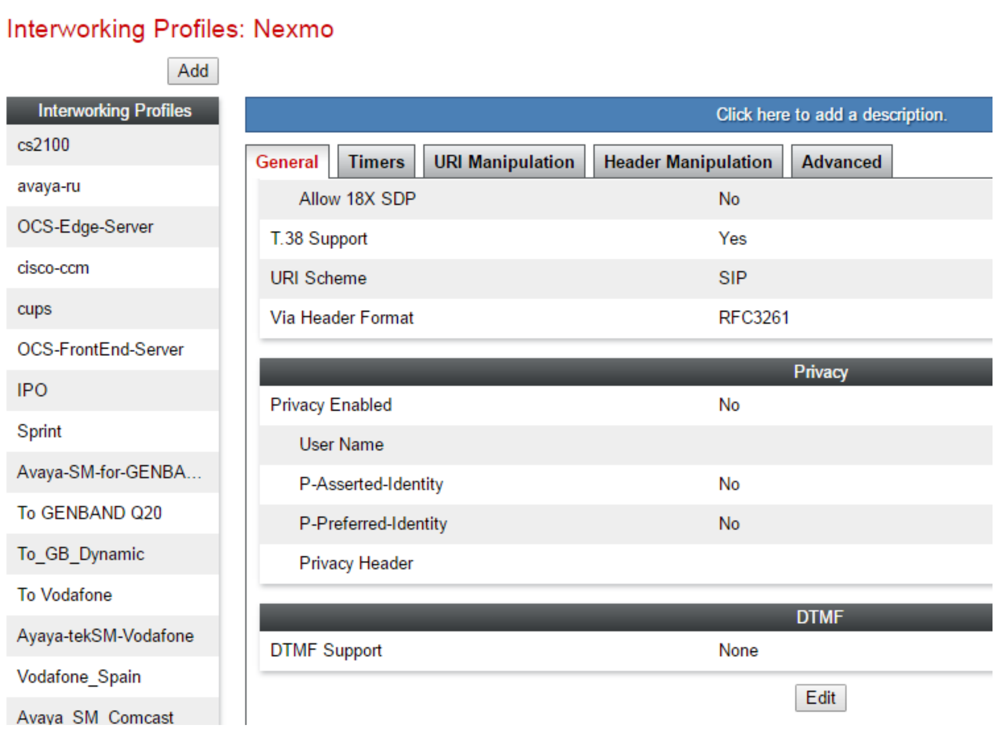

Interworking Profiles: Avaya SM
Add
Interworking Profiles
cs2100
avaya-ru
OCS-Edge-Server
cisco-ccm
cups
OCS-FrontEnd-Server
IPO
Sprint
Avaya-SM-for-GENBA...
To GENBAND Q20
Click here to add a description.
General
Timers
URI Manipulation
Header Manipulation
Advanced
SIP Timers
Min-SE ---
Init Timer ---
Max Timer ---
Trans Expire ---
Invite Expire ---
Transport Timers
TCP Connection Inactive Timer ---
Edit

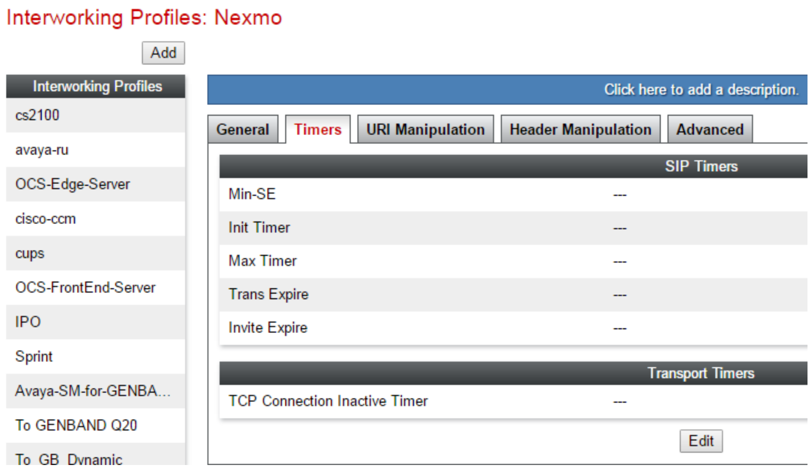

Interworking Profiles: Avaya SM
Add
Interworking Profiles
cs2100
avaya-ru
OCS-Edge-Server
cisco-ccm
cups
OCS-FrontEnd-Server
IPO
Sprint
Avaya-SM-for-GENBA...
To GENBAND Q20
To_GB_Dynamic
To Vodafone
Ayaya-tekSM-Vodafone
Vodafone_Spain
Click here to add a description.
General
Timers
URI Manipulation
Header Manipulation
Advanced
Record Routes Both Sides
Topology Hiding: Change Call-ID Yes
Call-Info NAT No
Change Max Forwards Yes
Include End Point IP for Context Lookup No
OCS Extensions No
AVAYA Extensions Yes
NORTEL Extensions No
Diversion Manipulation No
Metaswitch Extensions No
Reset on Talk Spurt No
Reset SRTP Context on Session Refresh No
Has Remote SBC Yes

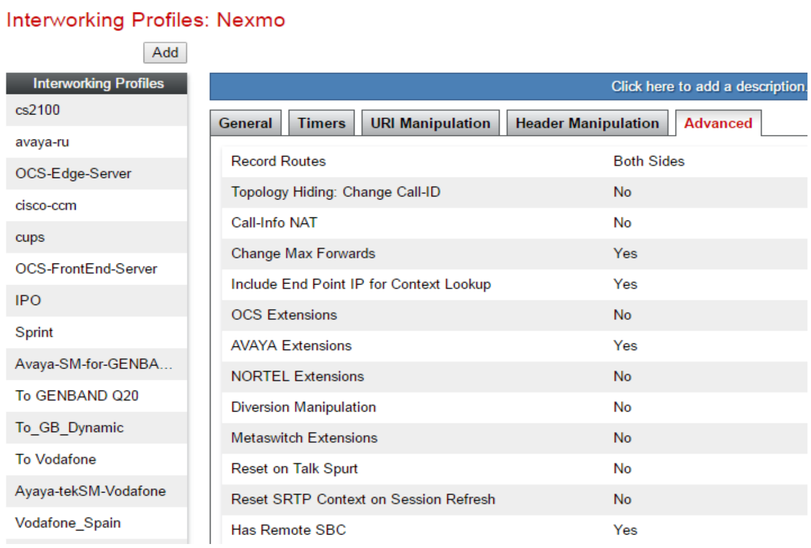

Interworking Profiles: Avaya SM
Add
Interworking Profiles
cs2100
avaya-ru
OCS-Edge-Server
cisco-ccm
cups
OCS-FrontEnd-Server
IPO
Sprint
Avaya-SM-for-GENBA...
To GENBAND Q20
To_GB_Dynamic
To Vodafone
Ayaya-tekSM-Vodafone
Vodafone_Spain
Avaya_SM_Comcast
Click here to add a description.
General
Timers
URI Manipulation
Header Manipulation
Advanced
Include End Point IP for Context Lookup No
OCS Extensions No
AVAYA Extensions Yes
NORTEL Extensions No
Diversion Manipulation No
Metaswitch Extensions No
Reset on Talk Spurt No
Reset SRTP Context on Session Refresh No
Has Remote SBC Yes
Route Response on Via Port No
Cisco Extensions No
Lync Extensions No
Edit

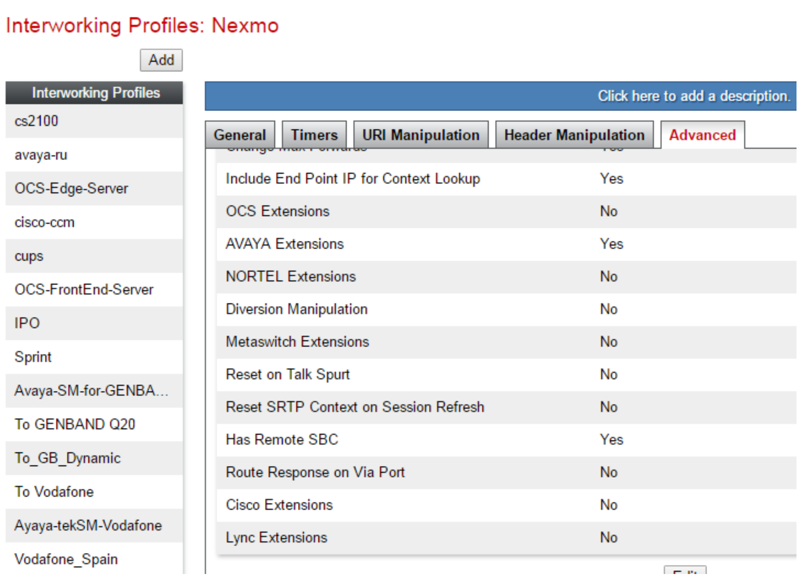
Create a Serving Interworking profile for Nexmo as shown below.






Interworking Profiles: Nexmo
Add
Interworking Profiles
cs2100
avaya-ru
OCS-Edge-Server
cisco-ccm
cups
OCS-FrontEnd-Server
IPO
Sprint
Avaya-SM-for-GENBA...
To GENBAND Q20
To_GB_Dynamic
Click here to add a description.
General
Timers
URI Manipulation
Header Manipulation
Advanced
SIP Timers
Min-SE ---
Init Timer ---
Max Timer ---
Trans Expire ---
Invite Expire ---
Transport Timers
TCP Connection Inactive Timer ---
Edit

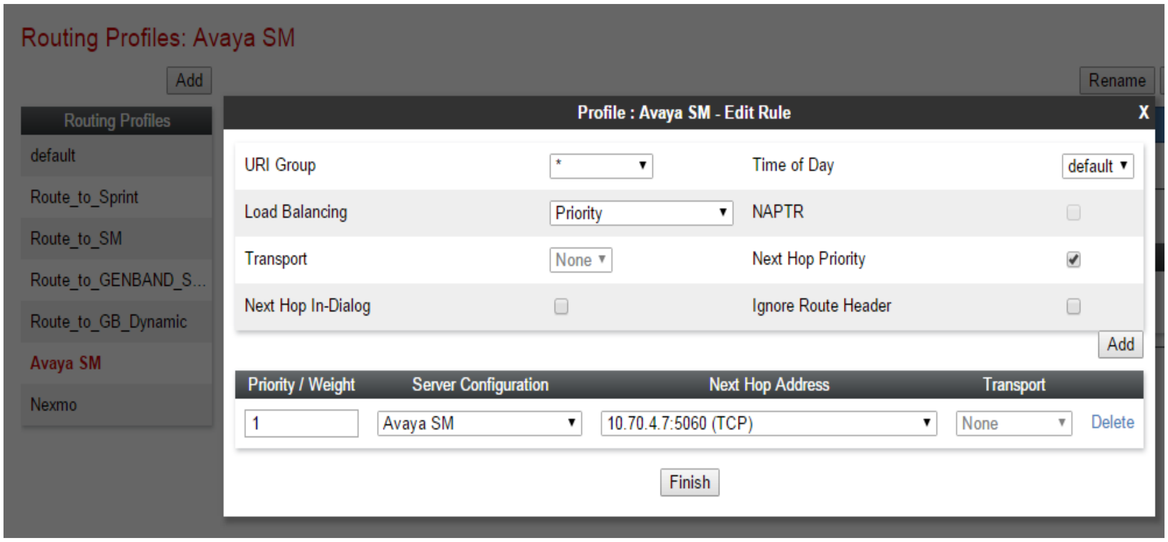

Interworking Profiles: Nexmo
Add
Interworking Profiles
cs2100
avaya-ru
OCS-Edge-Server
cisco-ccm
cups
OCS-FrontEnd-Server
IPO
Sprint
Avaya-SM-for-GENBA...
To GENBAND Q20
To_GB_Dynamic
To Vodafone
Ayaya-tekSM-Vodafone
Vodafone_Spain
Click here to add a description.
General
Timers
URI Manipulation
Header Manipulation
Advanced
Record Routes Both Sides
Topology Hiding: Change Call-ID No
Call-Info NAT No
Change Max Forwards Yes
Include End Point IP for Context Lookup Yes
OCS Extensions No
AVAYA Extensions Yes
NORTEL Extensions No
Diversion Manipulation No
Metaswitch Extensions No
Reset on Talk Spurt No
Reset SRTP Context on Session Refresh No
Has Remote SBC Yes


Interworking Profiles: Nexmo
Add
Interworking Profiles
cs2100
avaya-ru
OCS-Edge-Server
cisco-ccm
cups
OCS-FrontEnd-Server
IPO
Sprint
Avaya-SM-for-GENBA...
To GENBAND Q20
To_GB_Dynamic
To Vodafone
Ayaya-tekSM-Vodafone
Vodafone_Spain
Click here to add a description.
General
Timers
URI Manipulation
Header Manipulation
Advanced
Include End Point IP for Context Lookup Yes
OCS Extensions No
AVAYA Extensions Yes
NORTEL Extensions No
Diversion Manipulation No
Metaswitch Extensions No
Reset on Talk Spurt No
Reset SRTP Context on Session Refresh No
Has Remote SBC Yes
Route Response on Via Port No
Cisco Extensions No
Lync Extensions No

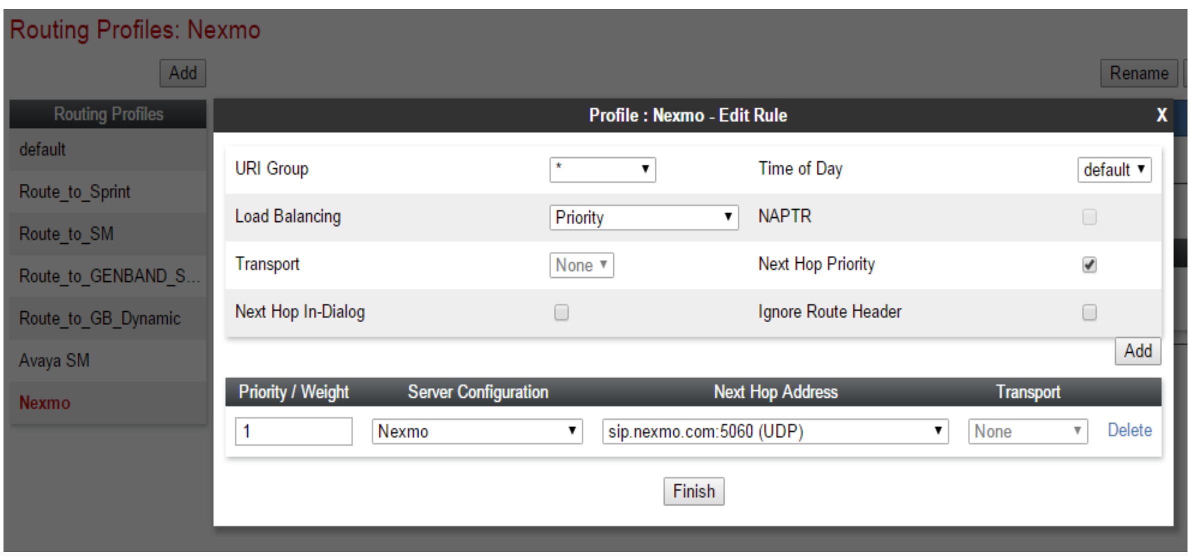
4.4.1.2 Routing

Navigate to **System Management > Global Profiles > Routing**

Creating a Routing profile for Avaya Session Manager as shown below.



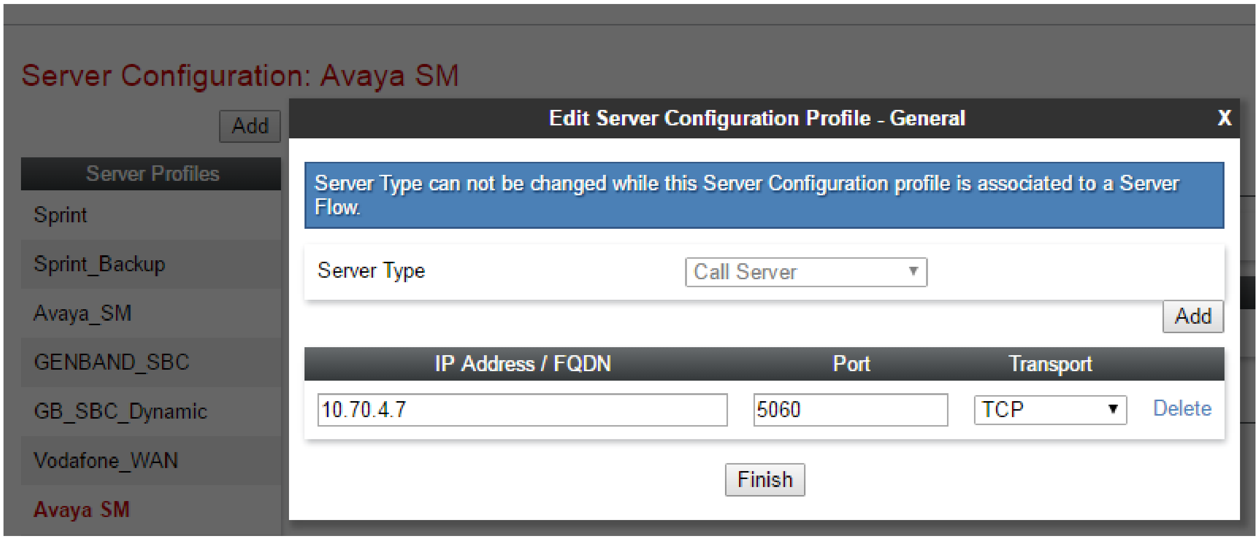
Creating a Routing profile for Nexmo as shown below.

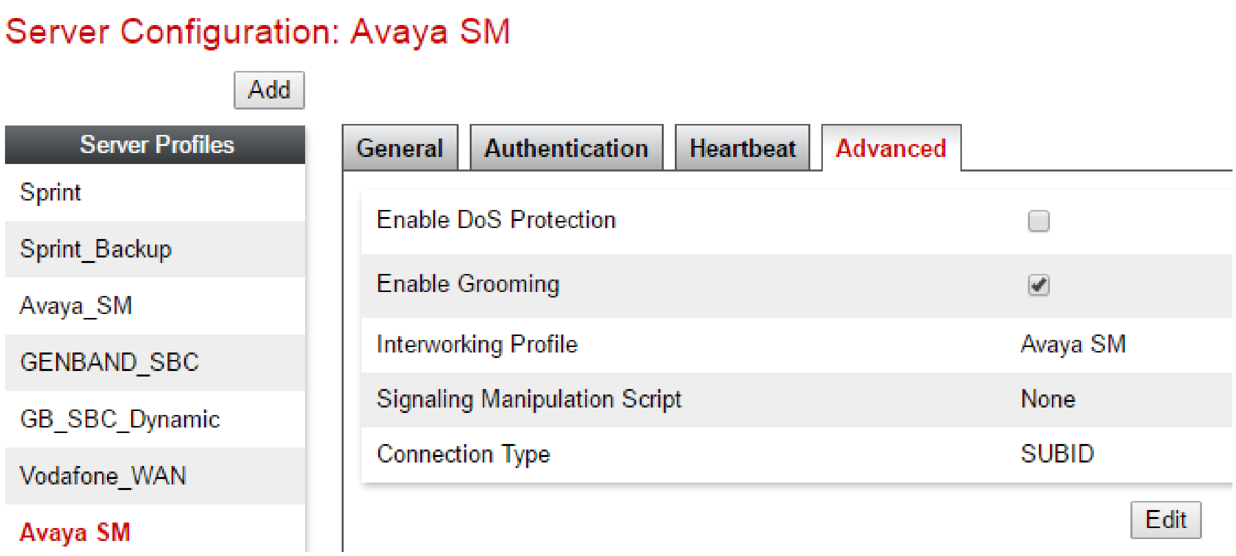


4.4.1.3 Server Configuration

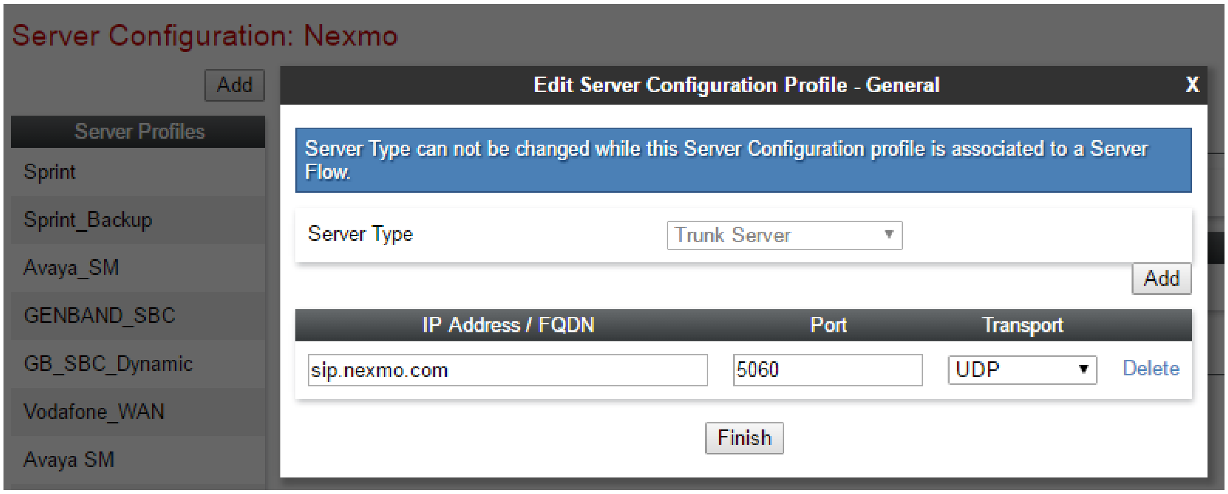
Navigate to **System Management > Global Profiles > Server Configuration**

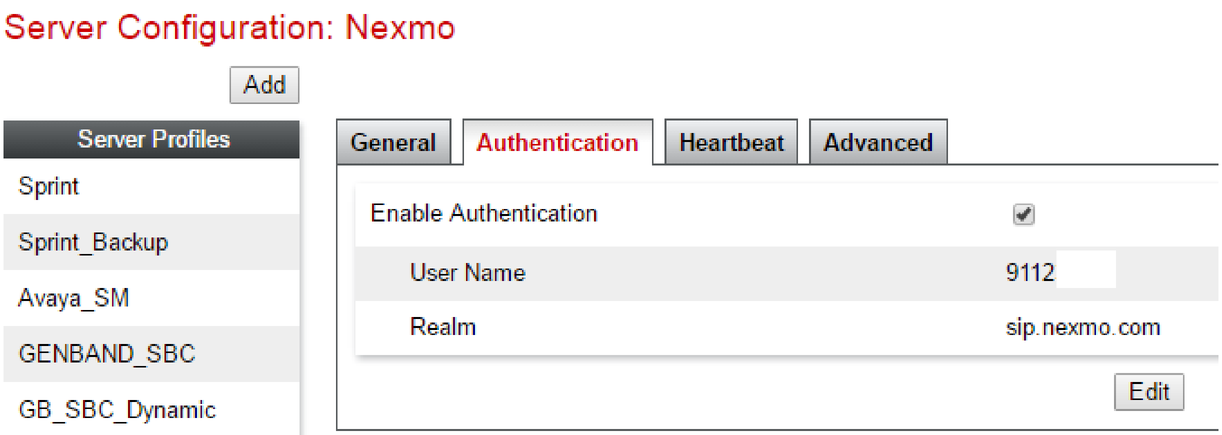
Create a Server configuration profile for Avaya Session Manager as shown below.

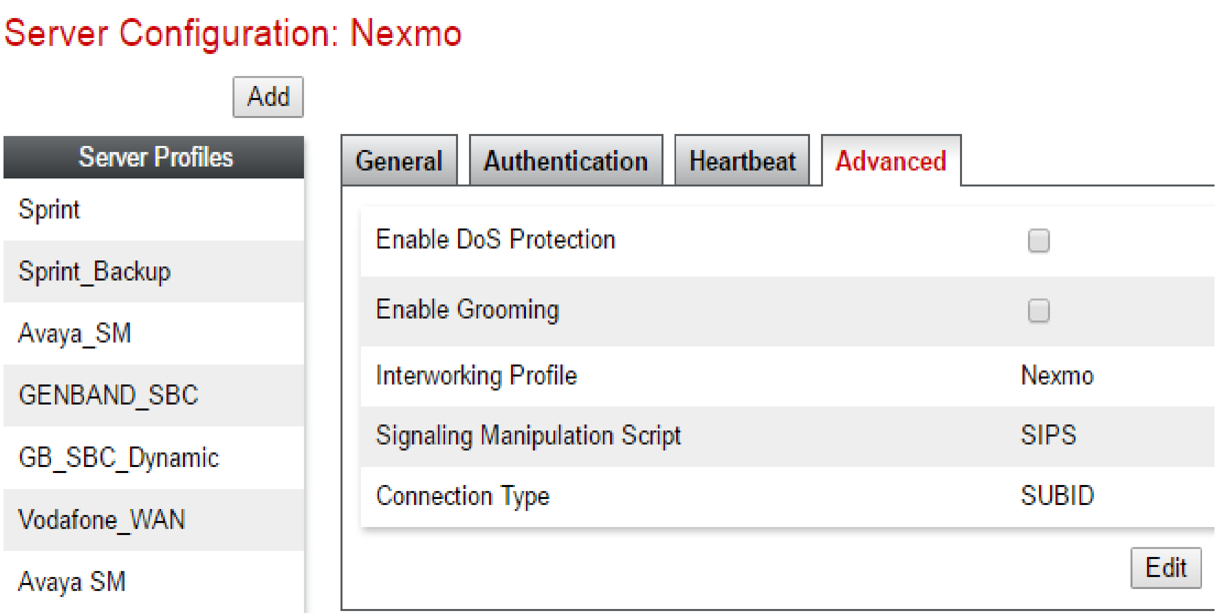




Create a Server configuration profile for Nexmo as shown below.



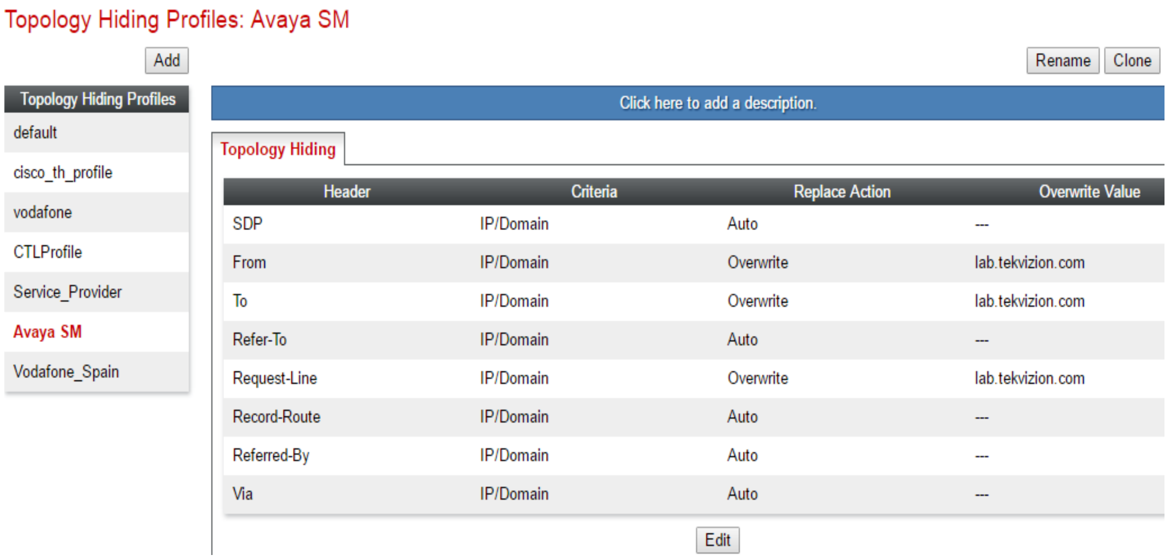




4.4.1.4 Topology Hiding

Navigate to **System Management > Global Profiles > Topology Hiding**

Creating a Topology hiding profile for Avaya Session Manager as shown below



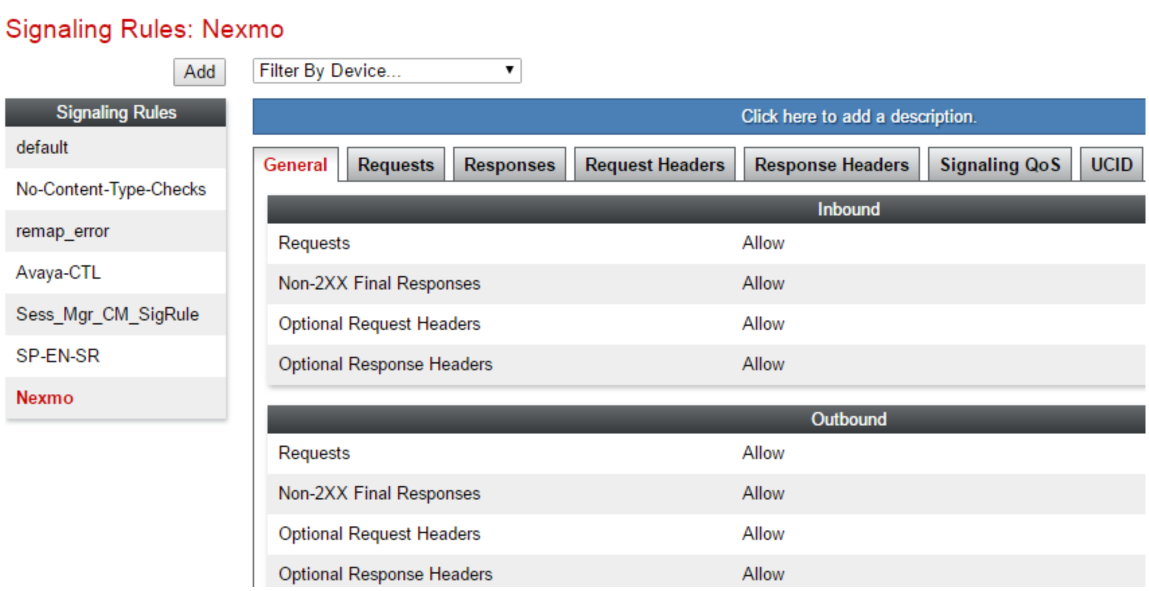
4.4.2 Domain Policies

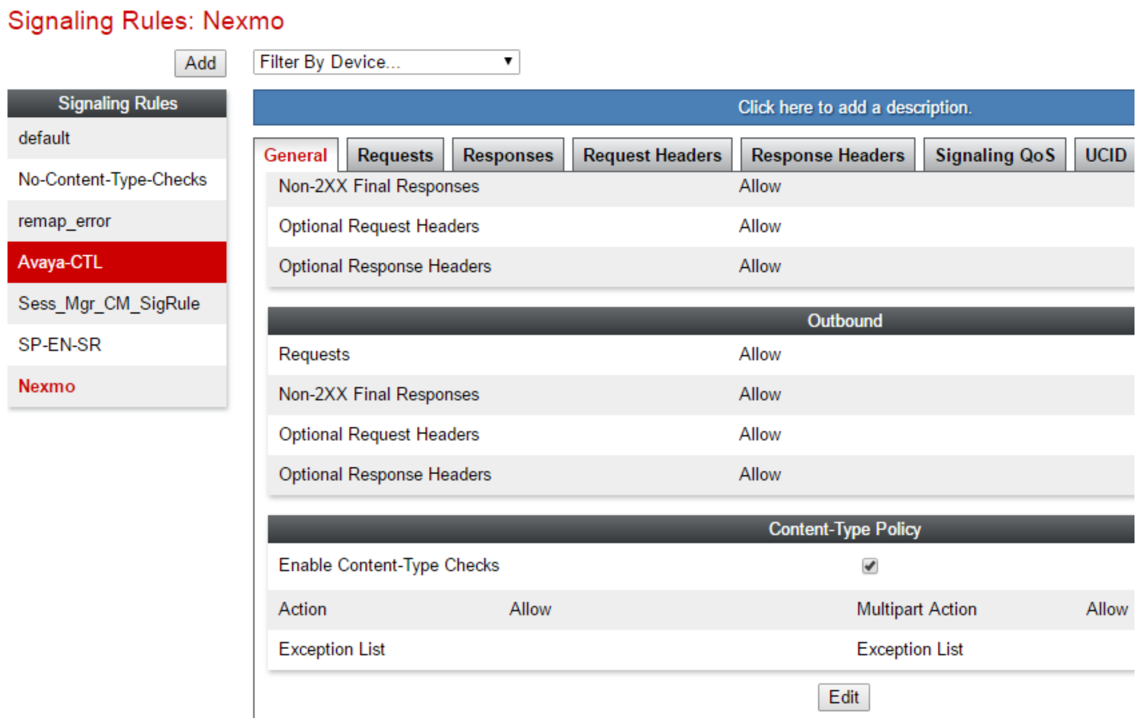
4.4.2.1 Signaling Rules

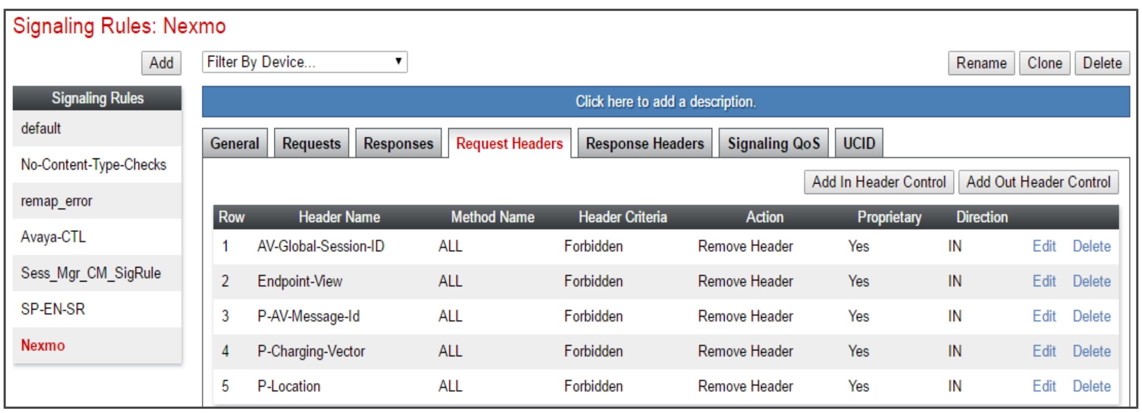
Signaling Rules define the actions to be taken (Allow, Block, Block with Response, etc.) for each type of SIP-specific signaling request and response message.

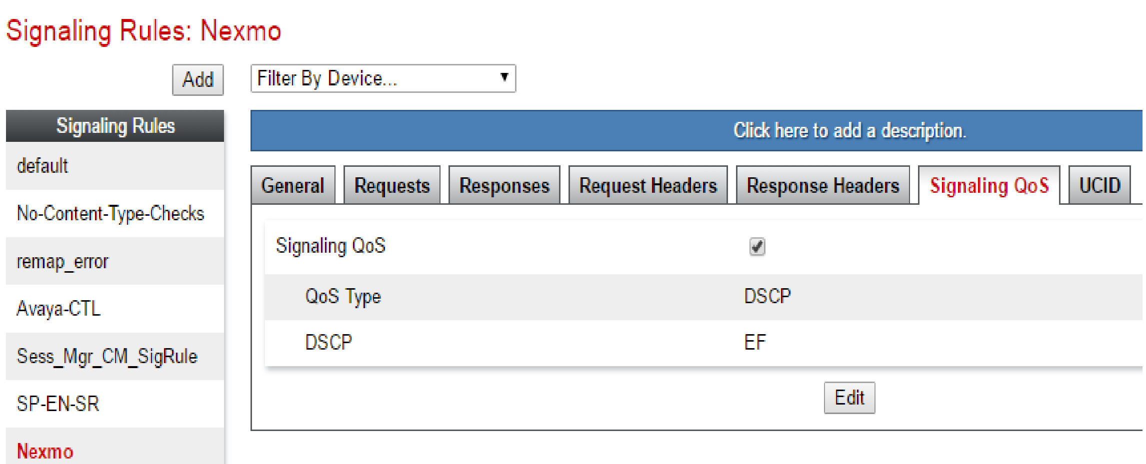
Headers such as P-Location, P-Charging-Vector and others are sent in SIP messages from Session Manager to the Avaya SBCe for egress to the Nexmo.

A Signaling Rule was created, to later be applied in the direction of the enterprise to block unwanted headers coming from Session Manager from being propagated to Nexmo.



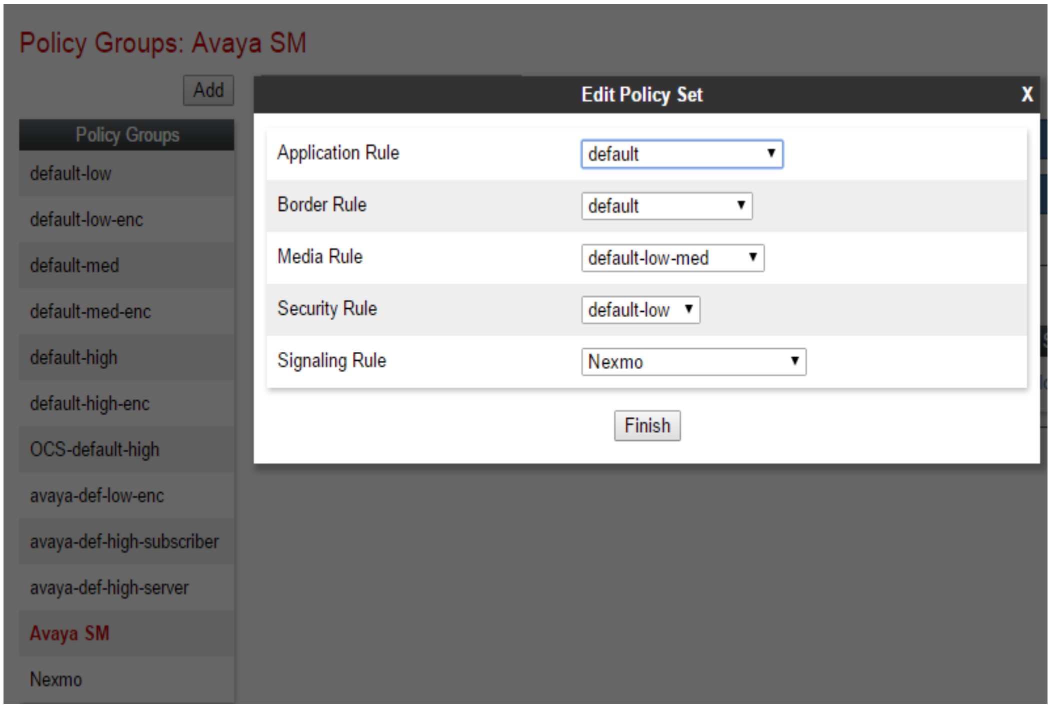

Signaling Rules: Nexmo
Add
Filter By Device...
Signaling Rules
default
No-Content-Type-Checks
remap_error
Avaya-CTL
Sess_Mgr_CM_SigRule
SP-EN-SR
Nexmo
Click here to add a description.
General
Requests
Responses
Request Headers
Response Headers
Signaling QoS
UCID
Non-2XX Final Responses Allow
Optional Request Headers Allow
Optional Response Headers Allow
Outbound
Requests Allow
Non-2XX Final Responses Allow
Optional Request Headers Allow
Optional Response Headers Allow
Content-Type Policy
Enable Content-Type Checks
Action Allow
Multipart Action Allow
Exception List
Exception List
Edit

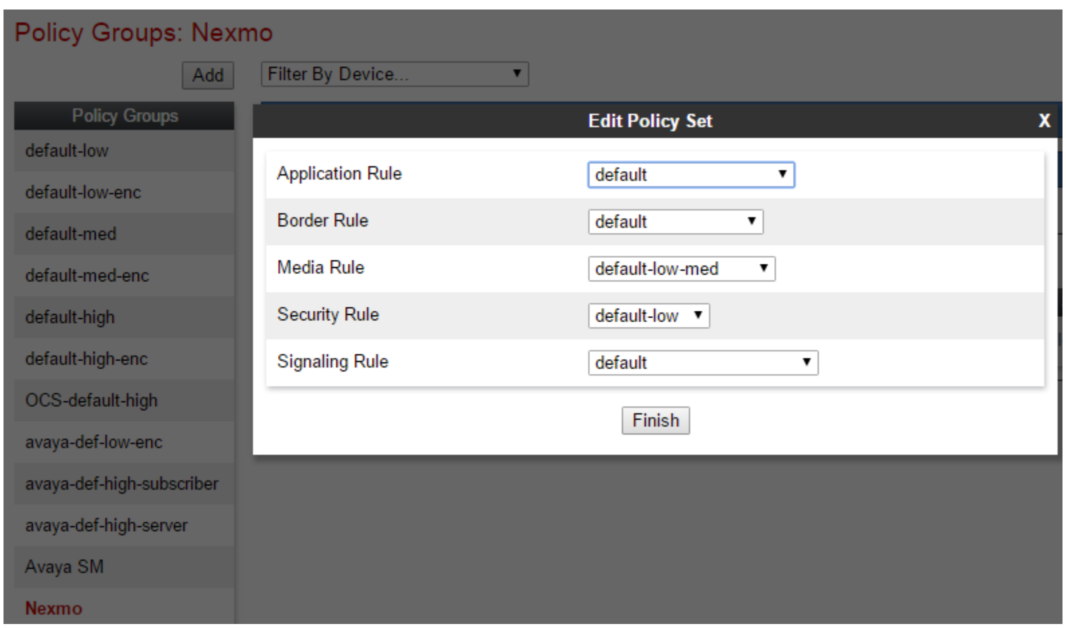

Signaling Rules: Nexmo
Add
Filter By Device...
Rename
Clone
Delete
Signaling Rules
default
No-Content-Type-Checks
remap_error
Avaya-CTL
Sess_Mgr_CM_SigRule
SP-EN-SR
Nexmo
Click here to add a description.
General
Requests
Responses
Request Headers
Response Headers
Signaling QoS
UCID
Add In Header Control
Add Out Header Control
Row Header Name Method Name Header Criteria Action Proprietary Direction
1 AV-Global-Session-ID ALL Forbidden Remove Header Yes IN Edit Delete
2 Endpoint-View ALL Forbidden Remove Header Yes IN Edit Delete
3 P-AV-Message-Id ALL Forbidden Remove Header Yes IN Edit Delete
4 P-Charging-Vector ALL Forbidden Remove Header Yes IN Edit Delete
5 P-Location ALL Forbidden Remove Header Yes IN Edit Delete


Signaling Rules: Nexmo
Add
Filter By Device...
Signaling Rules
default
No-Content-Type-Checks
remap_error
Avaya-CTL
Sess_Mgr_CM_SigRule
SP-EN-SR
Nexmo
Click here to add a description.
General
Requests
Responses
Request Headers
Response Headers
Signaling QoS
UCID
Signaling QoS
QoS Type DSCP
DSCP EF
Edit

4.4.2.2 End Point Policy Groups

End Point Policy group “Avaya SM” is created as shown below

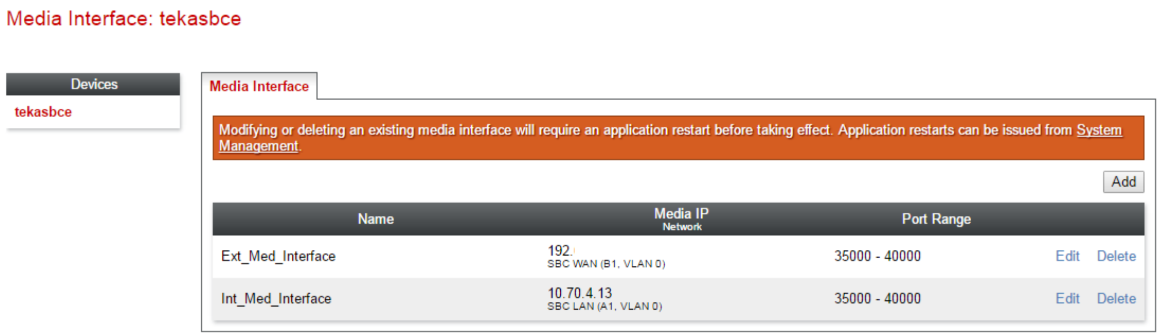




4.4.3 Device Specific Settings

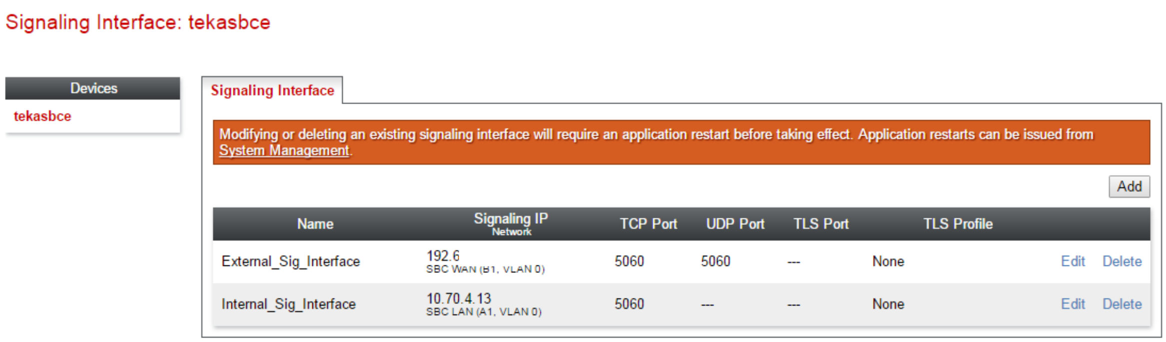
4.4.3.1 Media Interface

Navigate to **System Management > Device Specific Settings > Media Interface**. Create Internal and External Media Interface as shown below.



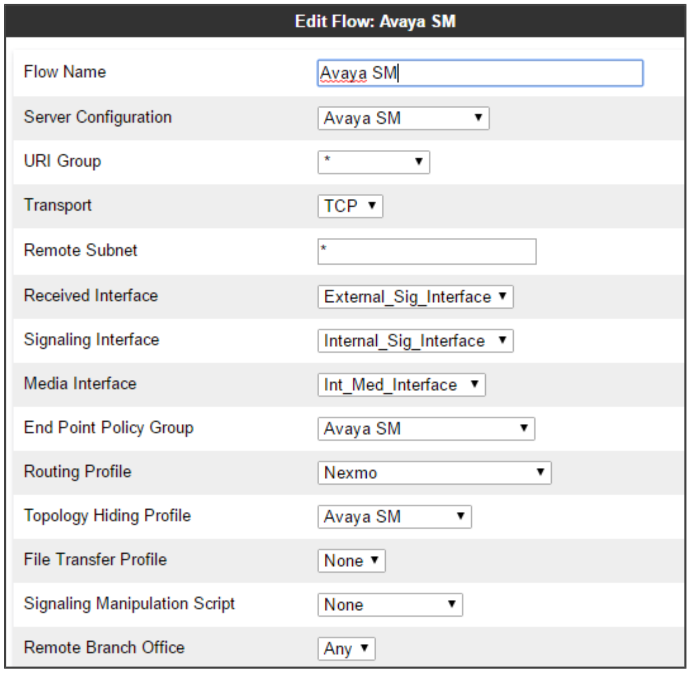
4.4.3.2 Signaling Interface

Navigate to **System Management > Device Specific Settings > Signaling Interface**. Create Internal and External Signaling Interface as shown below.

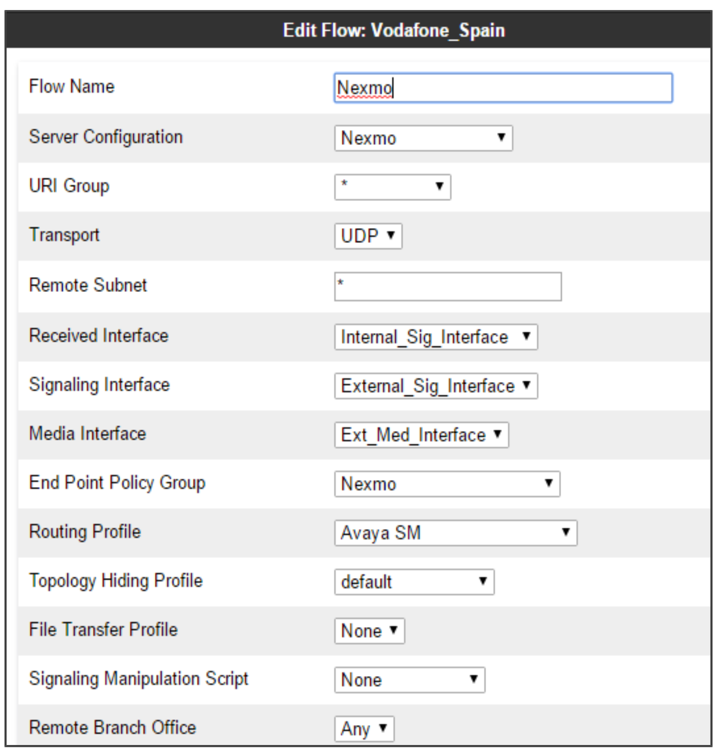


4.4.3.3 End Point Flows

Navigate to **System Management > Device Specific Settings > End Point Flows**. Select the Server Flows tab and click Add. Create a Server flow for Avaya Session Manager as shown below.



Select the Server Flows tab and click Add. Create a Server flow for Nexmo as shown below.

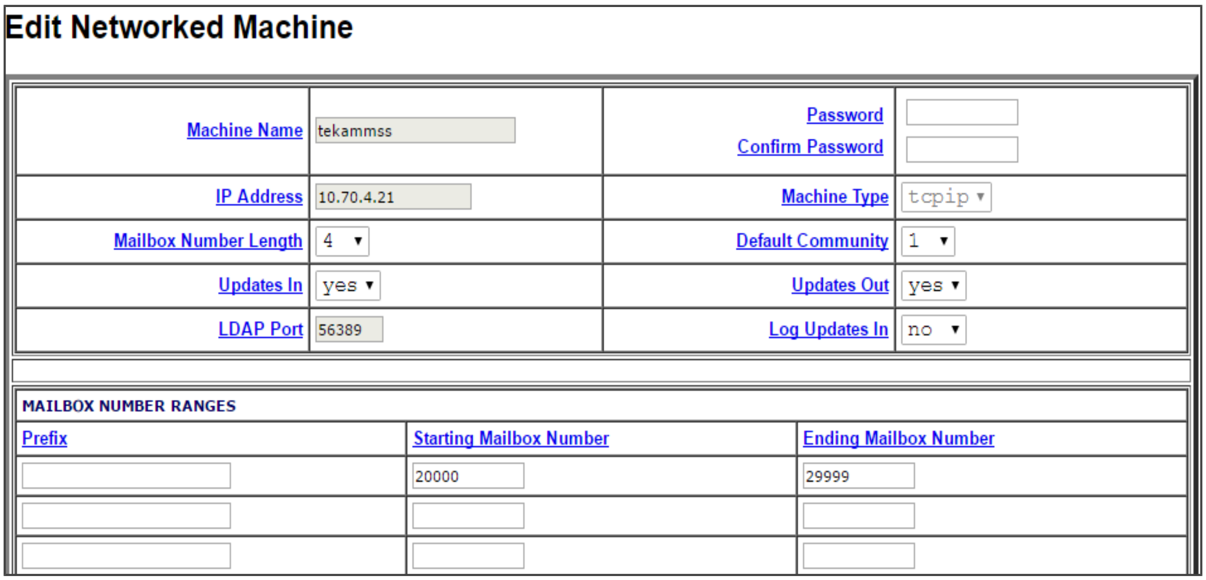


4.5 Avaya Modular Messaging

This section describes the steps for configuring the Avaya Modular Messaging to interoperate with Avaya Aura Session Manager via SIP trunking.

4.5.1.1 Messaging Server

Navigate to *Messaging Administration* > *Networked Machines* to configure Modular Messaging Server parameters as shown below.



Edit Networked Machine

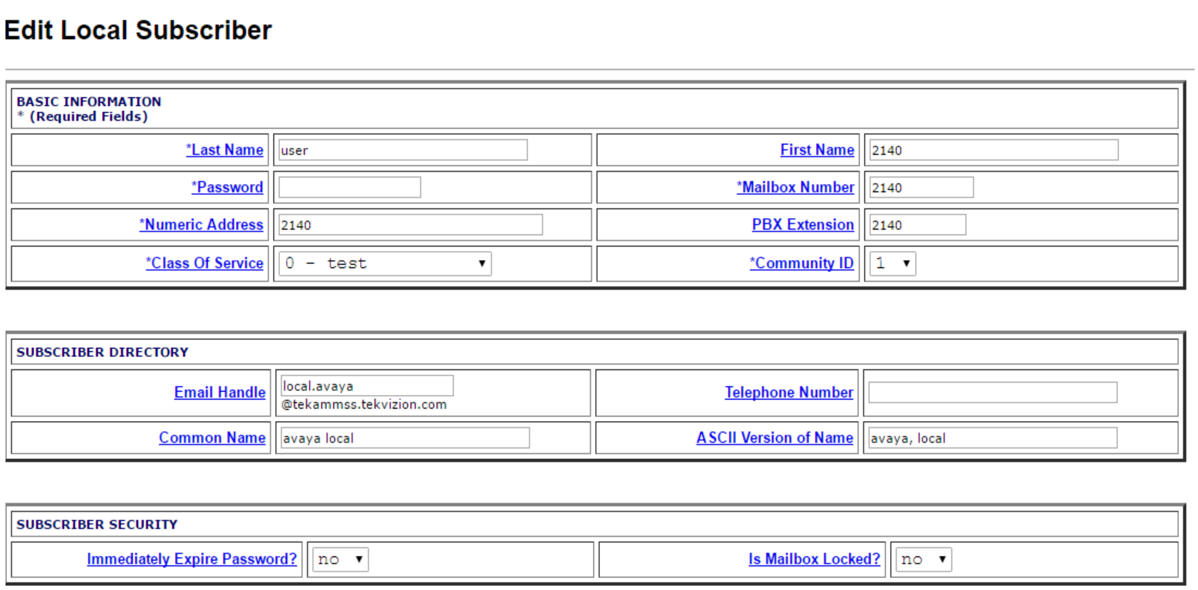
Machine Name	tekammss	Password Confirm Password	
IP Address	10.70.4.21	Machine Type	tcpip
Mailbox Number Length	4	Default Community	1
Updates In	yes	Updates Out	yes
LDAP Port	56389	Log Updates In	no

MAILBOX NUMBER RANGES

Prefix	Starting Mailbox Number	Ending Mailbox Number
	20000	29999

4.5.1.2 Subscriber

Navigate to **Messaging Administration > Subscriber Management**. Configure a subscriber for the Messaging server as shown below.



Edit Local Subscriber

BASIC INFORMATION
* (Required Fields)

*Last Name	user	First Name	2140
*Password		*Mailbox Number	2140
*Numeric Address	2140	PBX Extension	2140
*Class Of Service	0 - test	*Community ID	1

SUBSCRIBER DIRECTORY

Email Handle	local.avaya @tekammss.tekvizion.com	Telephone Number	
Common Name	avaya local	ASCII Version of Name	avaya, local

SUBSCRIBER SECURITY

Immediately Expire Password?	no	Is Mailbox Locked?	no

MAILBOX FEATURES			
Personal Operator Mailbox		Personal Operator Schedule	Always Active
VoiceMail Enabled	yes	Intercom Paging	paging is off

TUI MESSAGE ORDER			
TUI New Message Order	urgent first then newest	TUI Saved Message Order	urgent first then newest
TUI Deleted Message Order	urgent first then newest	TUI Admin Message Order	urgent first then newest

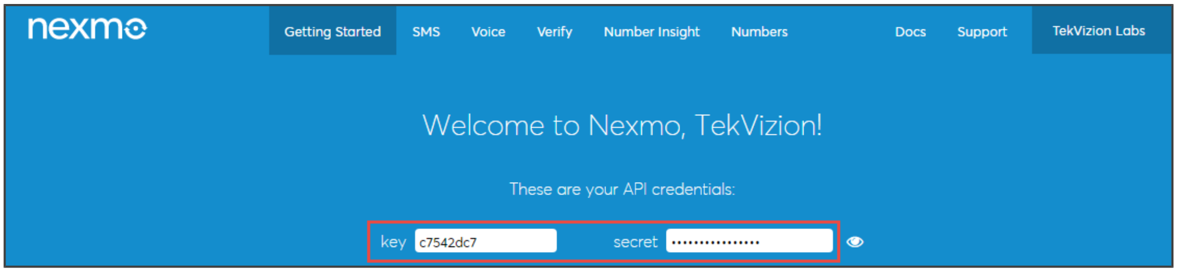
SECONDARY EXTENSIONS			
No Secondary Extensions	<--Add--	Secondary Extension	
	Delete	Caller Application	(none)

MISCELLANEOUS			
Miscellaneous1		Miscellaneous2	
Miscellaneous3		Miscellaneous4	

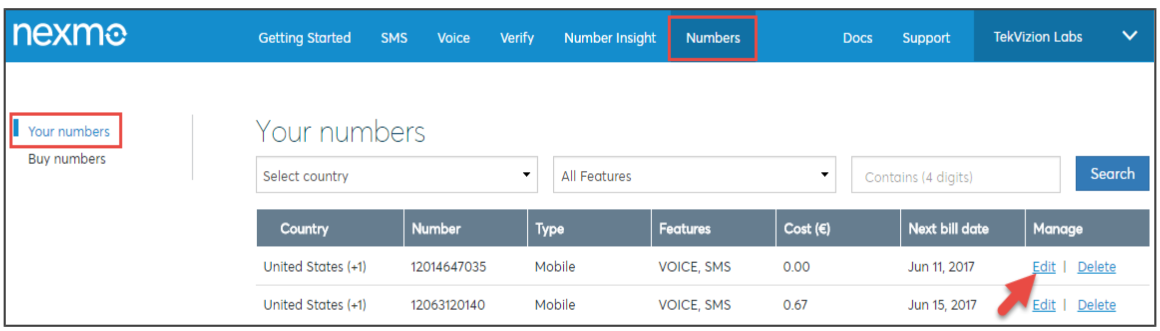
4.6 Nexmo Configuration

4.6.1 Configure Numbers in Nexmo Account

1. Login to the Nexmo account using the credentials provided at the time of registration. A **Key** and **Secret** will be displayed on the dashboard and this can be used as the username and password for Registration SIP Trunks.



2. In order to provide the URL to which the call has to be routed from Nexmo, navigate to the **Numbers** tab
3. Click **Edit** against each number as shown below



1. A pop-up will be displayed
2. Select the "**Forward to**" and provide the URL to which the calls route
3. Click **Update** to save the changes