Avaya Solution & Interoperability Test Lab

Application Notes for configuring Avaya Aura® Communication Manager R7.0.1, Avaya Aura® Session Manager R7.0.1 and Avaya Session Border Controller for Enterprise R7.1 with MiaRec - Issue 1.0

Abstract

These Application Notes describe the steps used to configure SIP-based Media Recording (SIPREC) between MiaRec and an Avaya SIP enabled Enterprise Solution. The Avaya platform consisted of Avaya Aura ® Communication Manager R7.0.1, Avaya Aura ® Session Manager R7.0.1 and Avaya Session Border Controller for Enterprise R7.1.

Readers should pay attention to **Section 2**, in particular the scope of testing as outlined in **Section 2.1** as well as any observations noted in **Section 2.2**, to ensure that their own use cases are adequately covered by this scope and results.

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 SIP-based Media Recording (SIPREC) between MiaRec and an Avaya SIP enabled Enterprise Solution. The Avaya solution consists of the following:

- Avaya Aura® Communication Manager R7.0.1 (Communication Manager)
- Avaya Aura® Session Manager R7.0.1 (Session Manager)
- Avaya Session Border Controller for Enterprise R7.1 (Avaya SBCE)

Note that the shortened names shown in brackets will be used throughout the remainder of the document.

MiaRec is a call recording and quality management solution. Using the SIPREC interface of Avaya SBCE, MiaRec provides centralized call recording solutions for the enterprises that use SIP Trunking services and Remote Workers.

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 Avaya SBCE. The enterprise site was configured to connect to the Service Provider's SIP Trunking service via SIP interface. MiaRec was recording calls to/from the enterprise site using the SIPREC interface on the Avaya SBCE.

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 call recording scenarios for the following:

- Recording of incoming calls to the enterprise site from Service Provider's SIP Trunk, calls made to SIP and H.323 telephones at the enterprise.
- Recording of outgoing calls from the enterprise site to remote destinations through the Service Provider's SIP Trunking service, calls made from SIP and H.323 telephones.
- Recording of incoming and outgoing calls to/from SIP Remote Worker.
- Recording of calls using the G.711A and G.729A codecs.
- Recording of call scenarios involving the user features such as hold and resume, transfer, conference, call forwarding, etc.
- Caller ID and DNIS presentation of recorded calls.
- Recording of call scenarios involving the call coverage and call forwarding for endpoints at the enterprise site.

- Transmission and response of SIP OPTIONS messages sent to MiaRec.
- Call recordings using combination of TCP/RTP and TLS/SRTP.

2.2. Test Results

Interoperability testing of the sample configuration was completed with successful results for the MiaRec solution with the following observations:

- Certain conference calls and transfer calls initiated from Remote Worker, result in duplicate recording on MiaRec. This is due to Avaya SBCE sending separate streams to MiaRec for each call leg. Avaya SBCE team is aware of this and is working towards resolution.

2.3. Support

For technical support on MiaRec products please contact MiaRec.
Email: support@miarec.com
Phone: 866-324-6717
Web: www.miarec.com

3. Reference Configuration

Figure 1 illustrates the test configuration. The test configuration shows an Enterprise site connected to the Simulated SIP Trunking service through the Avaya SBCE. Located at the Enterprise site is an Avaya SBCE, Session Manager, Communication Manager and MiaRec. Endpoints are Avaya 96x0 series and Avaya 96x1 Series IP Deskphones (with SIP and H.323 firmware and Avaya one-X® Communicator soft phone and Avaya Communicator for Windows running on laptop PCs. The Remote Workers are connecting to the Enterprise site through Avaya SBCE.

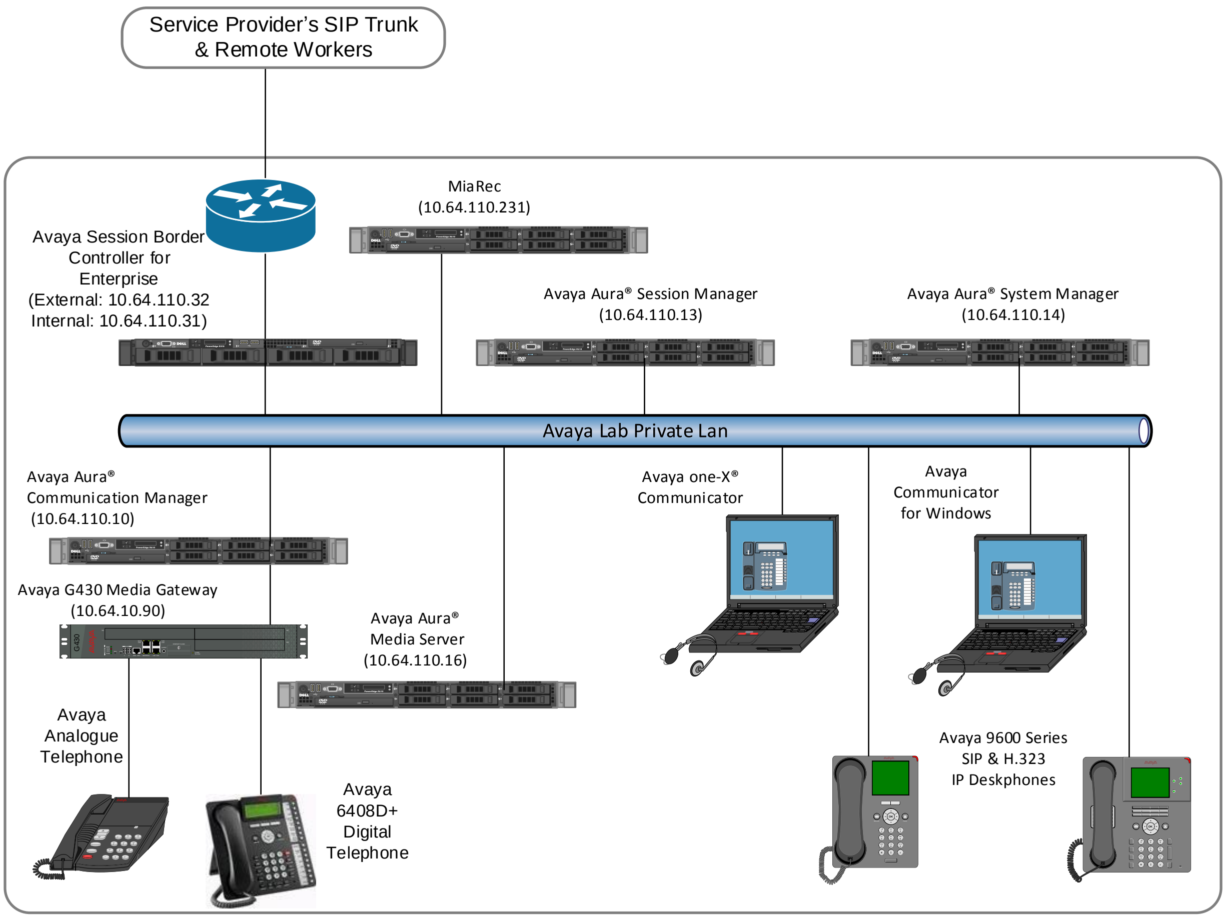


Figure 1: Test Setup MiaRec with Avaya Enterprise

4. Equipment and Software Validated

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

Equipment/Software	Release/Version
Avaya Aura® Session Manager running on a virtual platform	7.0.1.2
Avaya Aura® System Manager running on a virtual platform	7.0.1.2
Avaya Aura® Communication Manager running on a virtual platform	7.0.1.2.0.441.23523
Avaya Session Border Controller for Enterprise running on a virtual platform	7.1.0.1-07-12368
Avaya G450 Media Gateway	37.19.0
Avaya Aura® Media Server running on a virtual platform	7.7.0.236_2015.07.24
Avaya 96x0 Deskphone (H.323)	2_6_14_5
Avaya 96x1 Deskphone (SIP)	7.0.0 R39
Avaya 96x1 Deskphone (H.323)	3.230A
Avaya 6408D+ Digital Telephone	-
Avaya 6211 Analogue Telephone	-
Avaya one-X® Communicator running on Windows 10 PC	6.2.7.03-SP7
Avaya Communicator for Windows running on Windows 10 PC	2.1.2.75
MiaRec running on a virtual platform	Recorder: 5.2.1.155 Web UI: 5.2.0.2037

5. Configure Avaya Aura® Communication Manager

This section describes the steps for configuring Communication Manager. All configuration in this section is performed via a SAT terminal. Though Communication Manager and Session Manager do not directly integrate with MiaRec in the current setup, configuration is provided for reference.

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 authorised 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 Service Provider's SIP Trunking service and any other SIP trunks used.

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

IP PORT CAPACITIES                                              USED
                     Maximum Administered H.323 Trunks: 12000 0
           Maximum Concurrently Registered IP Stations: 18000 0
             Maximum Administered Remote Office Trunks: 12000 0
Maximum Concurrently Registered Remote Office Stations: 18000 0
              Maximum Concurrently Registered IP eCons: 128   0
  Max Concur Registered Unauthenticated H.323 Stations: 100   0
                        Maximum Video Capable Stations: 36000 0
                   Maximum Video Capable IP Softphones: 18000 0
                       Maximum Administered SIP Trunks: 12000 10
  Maximum Administered Ad-hoc Video Conferencing Ports: 12000 0
   Maximum Number of DS1 Boards with Echo Cancellation: 522   0
```

On **Page 5**, verify that **IP Trunks** field is set to **y**.

```
display system-parameters customer-options                      Page   5 of  12
                                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 Session Manager. In this case, **asm** and **10.64.110.13** are the **Name** and **IP Address** for the Session Manager SIP interface. Also note the **procr** IP address 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
ams                 10.64.110.16
asm                 10.64.110.13
default             0.0.0.0
procr               10.64.110.10
procr6              ::
```

5.3. Administer IP Network Region

Use the **change ip-network-region n** command where **n** is the chosen value of the configuration for the SIP Trunk. 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 SIP Trunk call is shuffled, the media stream is established directly between the enterprise end-point and the internal media interface of the Avaya SBCE.
- 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.
- The rest of the fields can be left at default values.

```
change ip-network-region 1                                        Page   1 of  20
                               IP NETWORK REGION
  Region: 1
Location:          Authoritative Domain: avaya.com
    Name: Trunk                 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
```

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** by typing **change ip-codec set n w**here **n** is the chosen value of the configuration for the SIP Trunk. Enter the list of audio codecs eligible to be used in order of preference. For the interoperability test the codecs supported by MiaRec were configured, namely **G.711A** and **G.729A** (other supported codecs by MiaRec are G.722, G.726-32k and GSM). Also, configure the **Media Encryption** as shown below.

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

     Media Encryption                     Encrypted SRTCP: enforce-unenc-srtcp
 1: 1-srtp-aescm128-hmac80
 2: 2-srtp-aescm128-hmac32
 3: none
 4:
 5:
```

5.5. Administer SIP Signaling Groups

This signalling group (and trunk group) will be used for inbound and outbound calls to the Service Provider's SIP Trunking service. Configure the **Signaling Group** using the **add signaling-group n** command as follows:

- Set **Group Type** to **sip**.
- Set **Transport Method** to **tls**.
- Set **Peer Detection Enabled** to **y** allowing 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 **Session_Manager** as defined in the **IP Node Names** form shown in **Section 5.2**).
- Set **Near-end Listen Port** and **Far-end Listen Port** as required. The standard value for TCP is **5060**, though **5061** was used in test to separate the SIP Trunk from the SIP endpoints on the Session Manager (See **Section 6.5**).
- 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 **2**).
- Set **Far-end Domain** to **avaya.com.**
- Set **Direct IP-IP Audio Connections** to **y**.
- Leave **DTMF over IP** at default value of **rtp-payload** (Enables **RFC2833** for DTMF transmission from Communication Manager).

Note: 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: tls
        Q-SIP? n
     IP 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? y
   Near-end Node Name: procr                 Far-end Node Name: asm
 Near-end Listen Port: 5061                Far-end Listen Port: 5061
                                        Far-end Network Region: 1

Far-end Domain: avaya.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? y
         Enable Layer 3 Test? y                     Initial IP-IP Direct Media? y
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 n** command, where **n** is an available trunk group for the SIP Trunk. On **Page 1** of this form:

- Set the **Group Type** field to **sip**.
- Choose a descriptive **Group Name**.
- Specify a trunk access code (**TAC**) consistent with the dial plan.
- The **Direction** is set to **two-way** to allow incoming and outgoing calls.
- Set the **Service Type** field to **tie**.
- Specify the 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  22
                                TRUNK GROUP

Group Number: 1                   Group Type: sip           CDR Reports: y
  Group Name: asm                        COR: 1        TN: 1        TAC: 101
   Direction: two-way        Outgoing Display? y
 Dial Access? n                                    Night Service:
Queue Length: 0
Service Type: tie                    Auth Code? n
                                              Member Assignment Method: auto
                                                       Signaling Group: 1
                                                     Number of Members: 10
```

On **Page 2** of the trunk-group form, the Preferred **Minimum Session Refresh Interval (sec)** field should be set to a value mutually agreed with MiaRec to prevent unnecessary SIP messages during call setup. During testing, a value of **900** was used that sets Min-SE to 1800 in the SIP signalling.

```
add trunk-group 1                                                     Page   2 of  22
      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
```

On **Page 3**, set the **Numbering Format** field to **private**. This allows delivery of CLI in formats other than E.164 with leading "+".

```
add trunk-group 1                                                    Page   3 of  22
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
```

5.7. Administer Calling Party Number Information

Use the **change private-unknown-numbering** command to configure Communication Manager to send the calling party number in the format required. In test, calling party numbers were sent as Communication Manager extension numbers to be modified in Session Manager. These calling party numbers are sent in the SIP From, Contact and PAI headers. The numbers are displayed on display-equipped PSTN telephones with any reformatting performed in the network.

```
change private-numbering 0                                           Page   1 of   2
                           NUMBERING - PRIVATE FORMAT

Ext Ext               Trk        Private          Total
Len Code              Grp(s)     Prefix           Len
 5  1                 1                           5     Total Administered: 2
                                                           Maximum Entries: 540
```

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 Avaya SBCE to the Service Provider's 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:
                       Answer Back Access Code: #25
                         Attendant Access Code:
      Auto Alternate Routing (AAR) Access Code: 8
    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 1. A sample of dial pattern is 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 1. Note that exact maximum number lengths should be used where possible to reduce post-dial delay. Calls are sent to **Route Pattern 1**.

```
                              ARS DIGIT ANALYSIS TABLE
                                    Location: all            Percent Full: 0

          Dialed           Total      Route     Call   Node  ANI
          String          Min  Max  Pattern    Type   Num   Reqd
    1                     11   11     1        natl         n
```

Use the **change route-pattern n** command, where **n** 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**. **Numbering Format** is applied to CLI and is used to set TDM signalling parameters such as type of number and numbering plan indicator. This doesn't have the same significance in SIP calls and during testing it was set to **lev0-pvt** to ensure that calling party number was not prefixed with a leading "+".

```
change route-pattern 1                                              Page   1 of   3
                    Pattern Number: 1      Pattern Name: SIP_Endpoints
    SCCAN? n    Secure SIP? n     Used for SIP stations? 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 Sub  Numbering LAR
    0 1 2 M 4 W     Request                                 Dgts Format
 1: y y y y y n  n            rest                               lev0-pvt  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
```

Save the Communication Manager configuration by entering **save translation**.

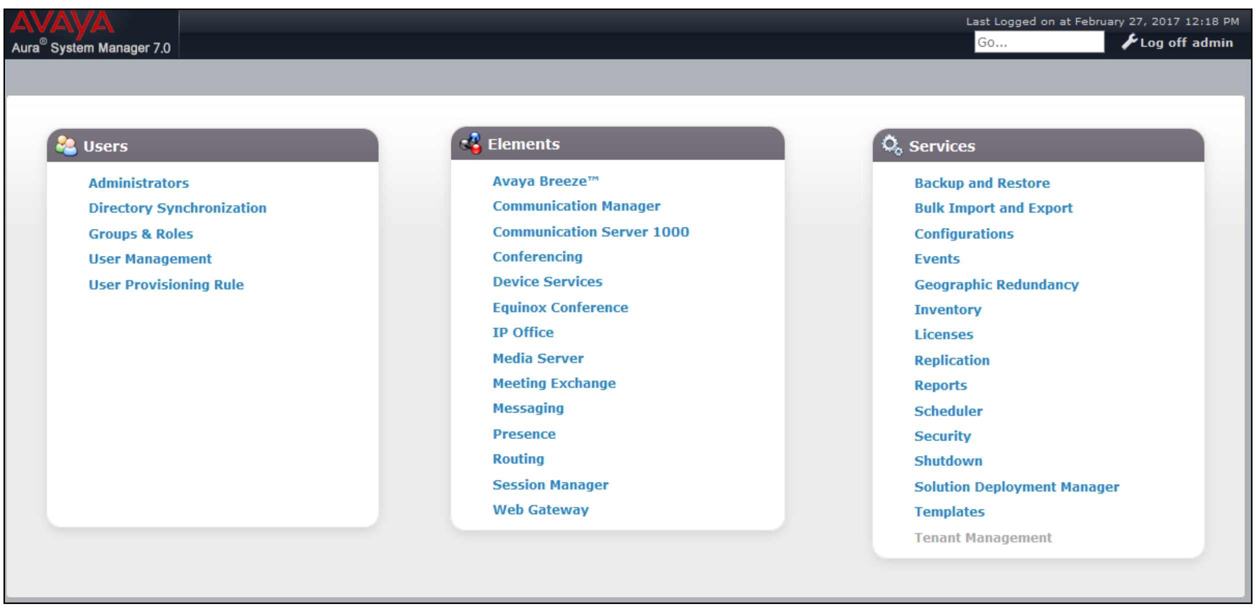
6. Configuring Avaya Aura® Session Manager

This section provides the procedures for configuring Session Manager. Session Manager is configured by opening a web browser to System Manager. The procedures include the following areas:

- Log in to Avaya Aura® System Manager
- Administer SIP Domain
- Administer Locations
- Administer Adaptations
- 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

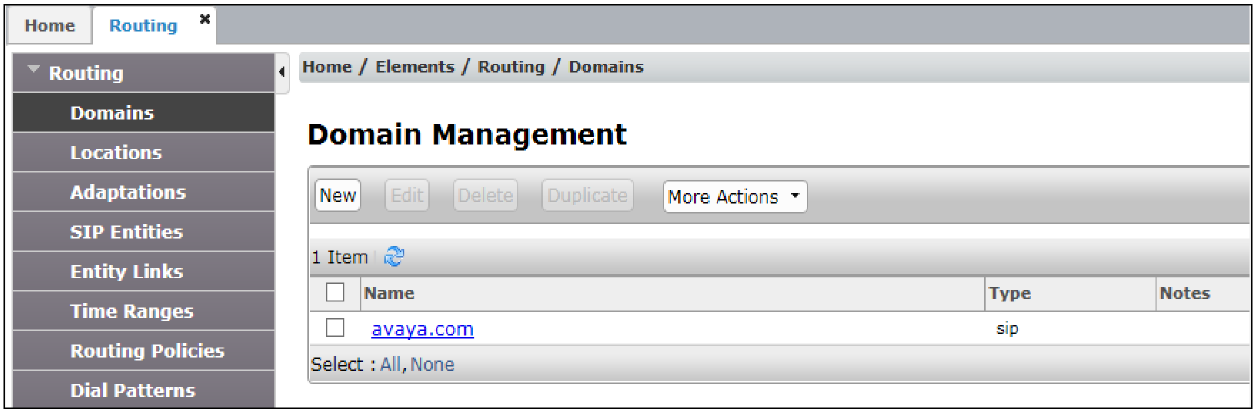
6.1. Log in to Avaya Aura® System Manager

Access the System Manager using a web browser and 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

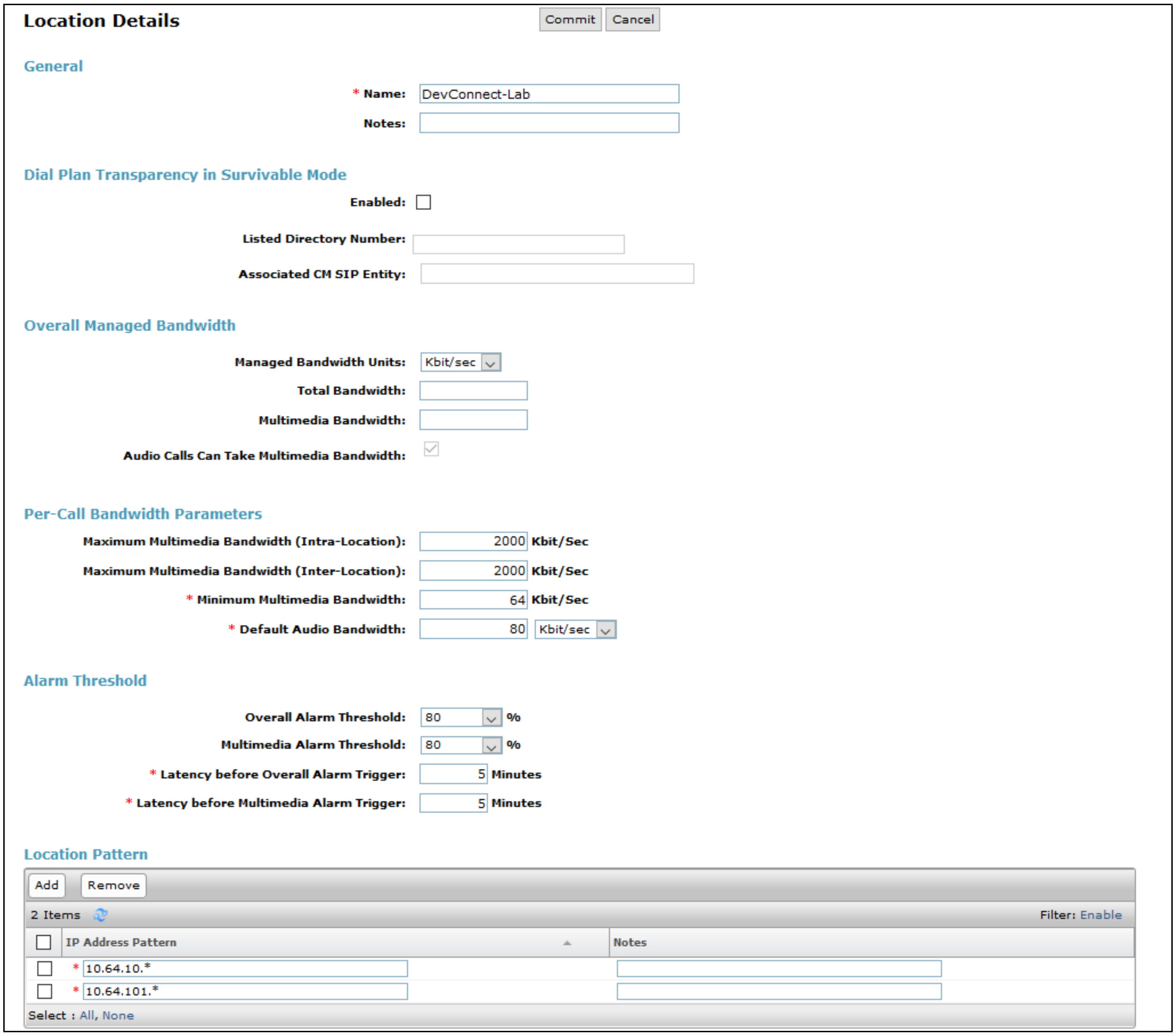
To add the SIP domain that will be used with Session Manager, select **Routing** from the **Home** tab menu and in the resulting tab select **Domains** from left hand menu. Click the **New** button to create a new SIP domain entry. In the **Name** field enter the domain; this will be the same as specified in the Authoritative Domain specified in the IP Network Region on 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 (not shown). Click **Commit** to save changes.



Note: If the existing domain name used in the enterprise equipment does not match that used in the network, Topology Hiding in the Avaya SBCE can be used to change it (see **Section 7.8**).

6.3. Administer Locations

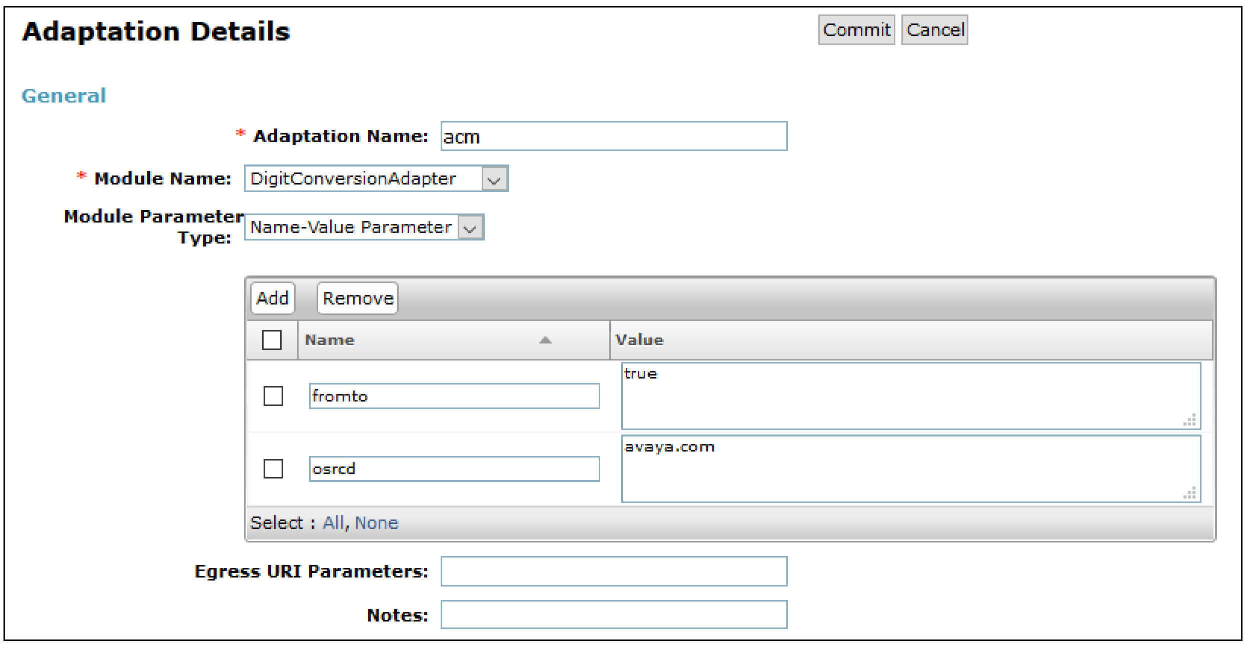
Locations can be used to identify logical and/or physical locations where SIP Entities reside for the purposes of bandwidth management. One location is added to the sample configuration for all of the enterprise SIP entities. On the **Routing** tab select **Locations** from the left hand menu (not shown). 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 Adaptation

An Adaptation needs to be added to ensure that the From header contains proper hostname. To add an Adaptation, select **Adaptations** on the left pane and select **New** (not shown). Configure the Adaptation as follows:

- In the **Adaptation Name** field enter an informative name.
- In the **Module Name** select **DigitConversionAdapter.**
- Select the **Add** button to add adaptation parameters. Following two values were configured during Compliance Testing.
  - fromto=true
  - osrcd=Avaya.com



6.5. Administer SIP Entities

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

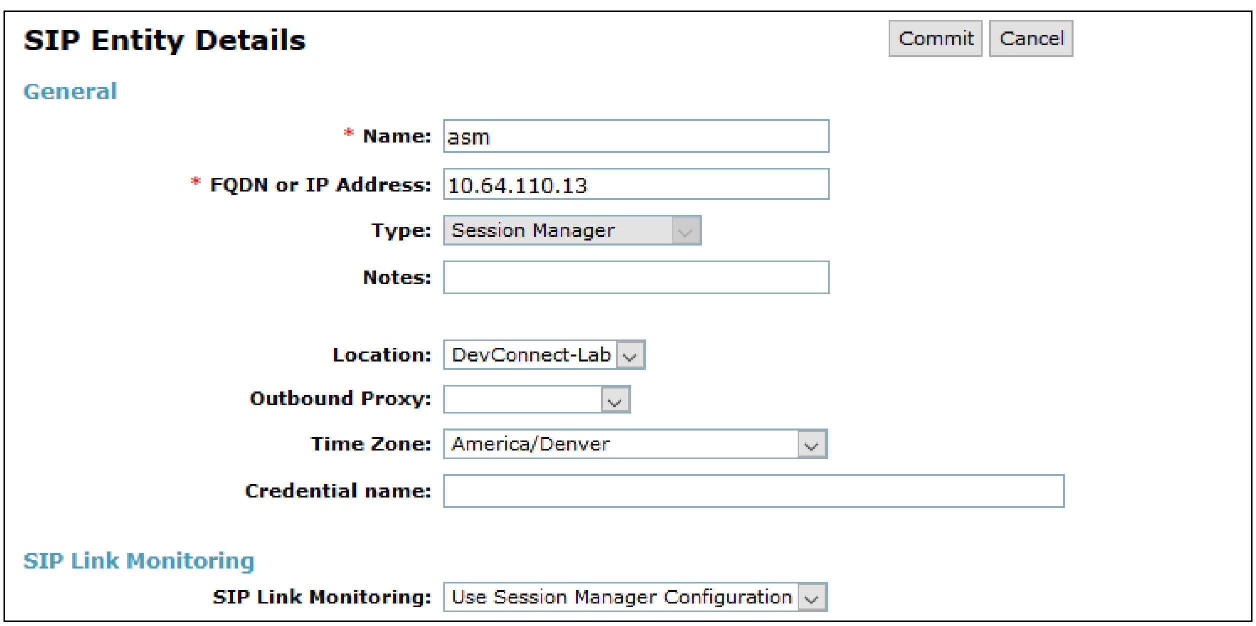
- In the **Name** field enter an informative name.
- In the **FQDN or IP Address** field enter the IP address of Session Manager or the signalling interface on the connecting system.
- In the **Type** field use **Session Manager** for a Session Manager SIP entity, **CM** for a Communication Manager SIP entity and **SIP Trunk** for the Avaya SBCE SIP entity.
- In the **Adaptation** field (not available for the Session Manager SIP Entity), select the appropriate Adaptation from the drop down menu.
- 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 Session Border Controller for Enterprise (Avaya SBCE) SIP Entity.

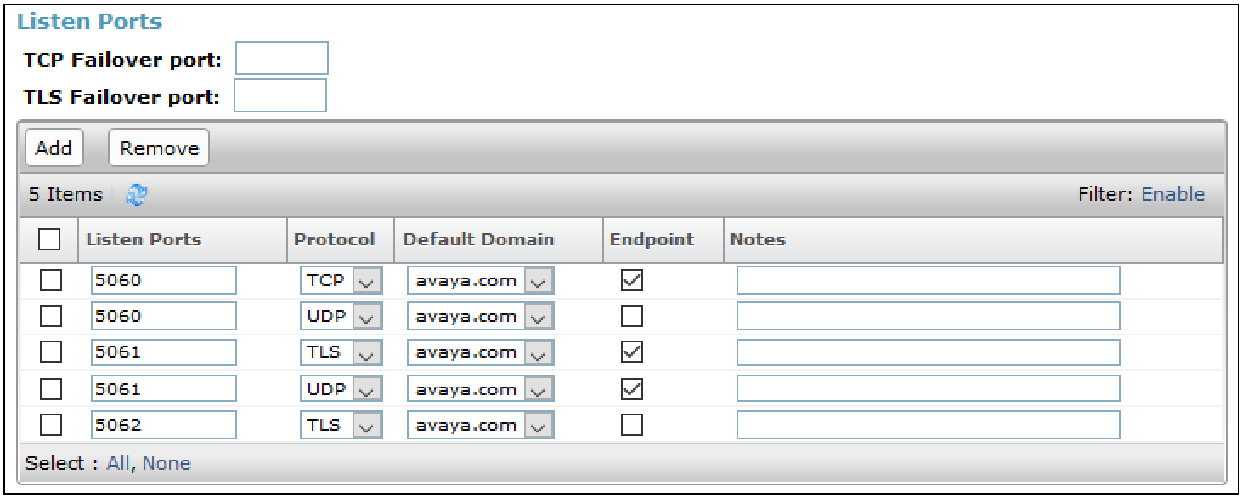
6.5.1. Avaya Aura® Session Manager SIP Entity

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.



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.
- Note that the **Endpoints** boxes were checked to allow SIP Endpoints to register on the specified ports.

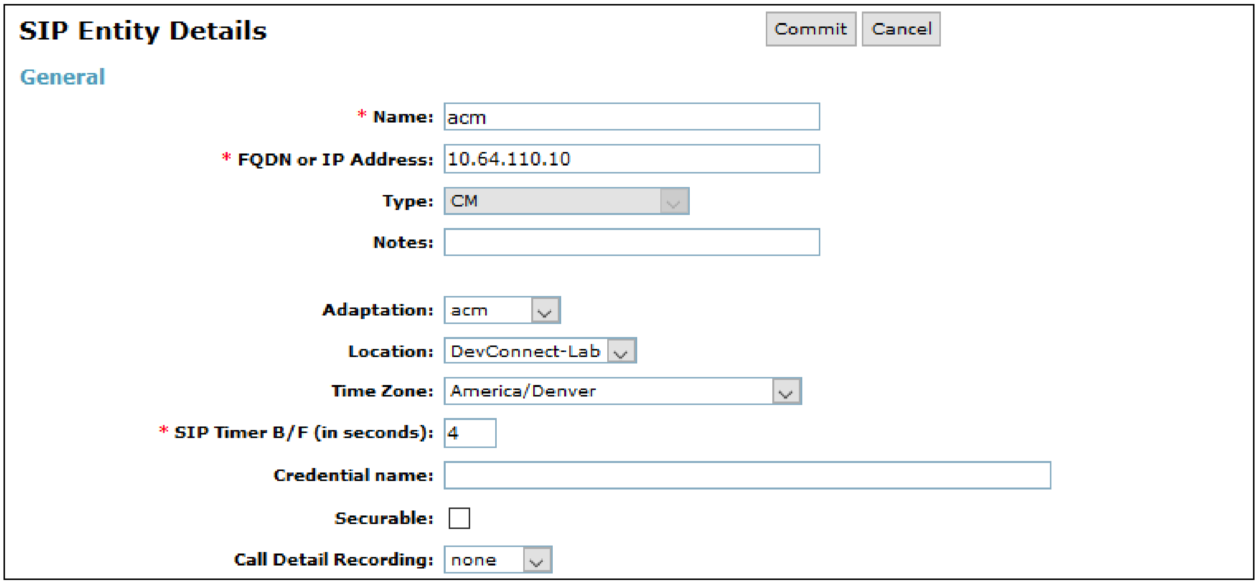

Listen Ports
TCP Failover port:
TLS Failover port:
Add Remove
5 Items Filter: Enable

Listen Ports	Protocol	Default Domain	Endpoint	Notes
5060	TCP	avaya.com	☑	
5060	UDP	avaya.com	☐	
5061	TLS	avaya.com	☑	
5061	UDP	avaya.com	☑	
5062	TLS	avaya.com	☐	

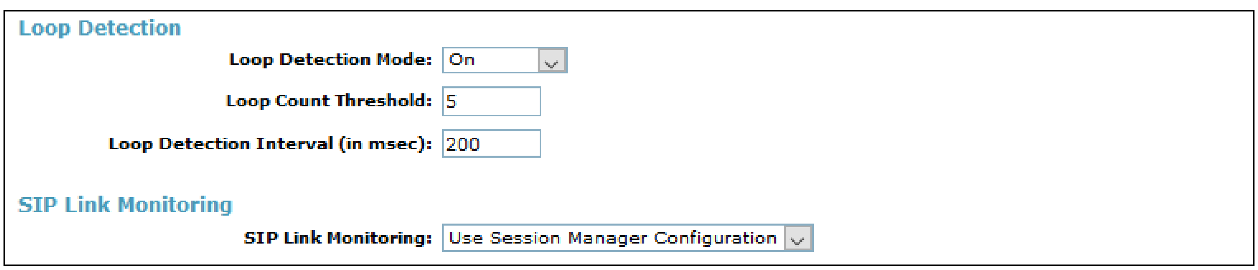
Select : All, None

6.5.2. Avaya Aura® Communication Manager SIP Entities

The following screen shows one of the SIP entities for Communication Manager which is configured as an Evolution Server. This SIP Entity is used for the SIP Trunk. 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**.

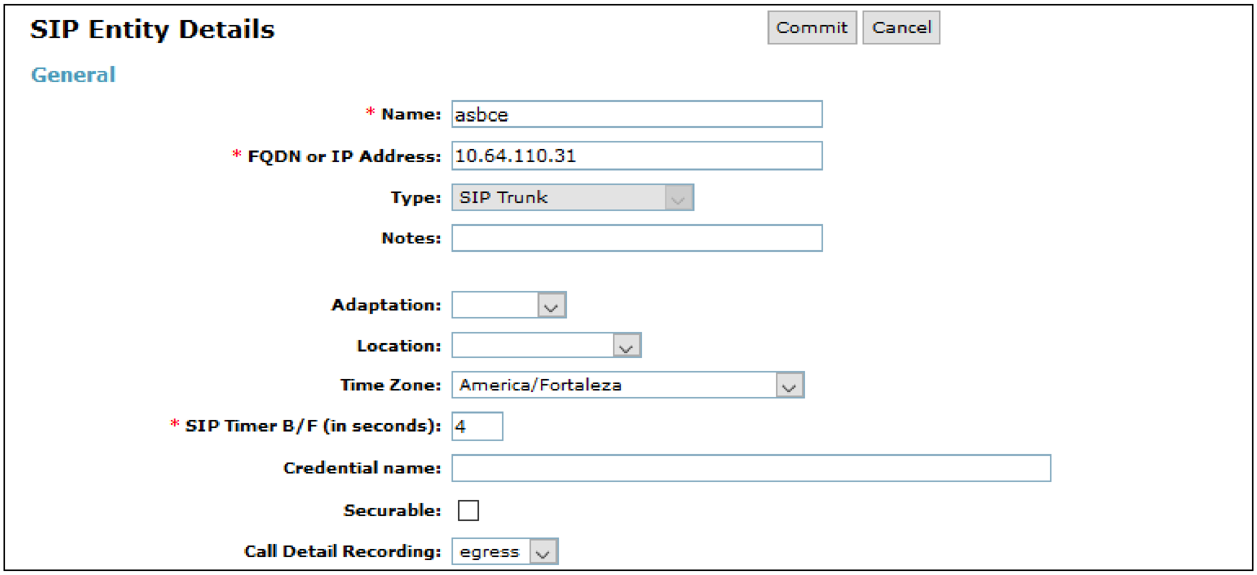


Other parameters can be set for the SIP Entity as shown in the following screenshot, but for test, these were left at default values.



6.5.3. Avaya Session Border Controller for Enterprise SIP Entity

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



6.6. 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 6.5**.
- 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.

Entity Links

New | Edit | Delete | Duplicate | More Actions

13 Items — Filter: Enable

	Name	SIP Entity 1	Protocol	Port	SIP Entity 2	DNS Override	Port	Connection Policy	Deny New Service	Notes
☐	asm_aaep_5060_TCP	asm	TCP	5060	aaep	☐	5060	trusted	☐	
☐	asm_abrz_5060_TCP	asm	TCP	5060	abrz	☐	5060	trusted	☐	
☐	asm_abrz_5061_TLS	asm	TLS	5061	abrz	☐	5061	trusted	☐	
☐	asm_abrz-ps_5061_TLS	asm	TLS	5062	abrz-ps	☐	5061	trusted	☐	
☐	asm_abrz-ps-cluster_5061_TLS	asm	TLS	5061	abrz-ps-cluster	☐	5061	trusted	☐	
☐	asm_acm_5061_TLS	asm	TLS	5061	acm	☐	5061	trusted	☐	
☐	asm_acmm_5060_TCP	asm	TCP	5060	acmm	☐	5060	trusted	☐	
☐	asm_ams_5060_TCP	asm	TCP	5060	ams	☐	5060	trusted	☐	
☐	asm_asbce_5061_TLS	asm	TLS	5061	asbce	☐	5061	trusted	☐	
☐	asm_ipo_5060_UDP	asm	UDP	5060	ipo	☐	5060	trusted	☐	
☐	asm_msm_5060_TCP	asm	TCP	5060	msm	☐	5060	trusted	☐	
☐	asm_mxbridge_5060_TCP	asm	TCP	5060	mxbridge	☐	5060	trusted	☐	
☐	asm_sipp_5060_TCP	asm	TCP	5060	sipp	☐	5060	trusted	☐	

Select : All, None

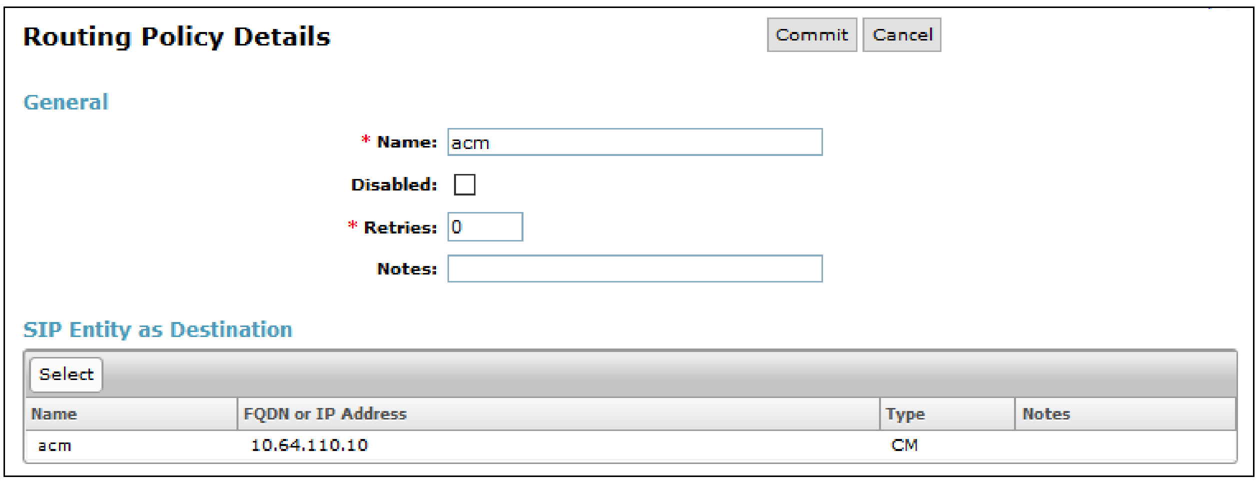
6.7. Administer Routing Policies

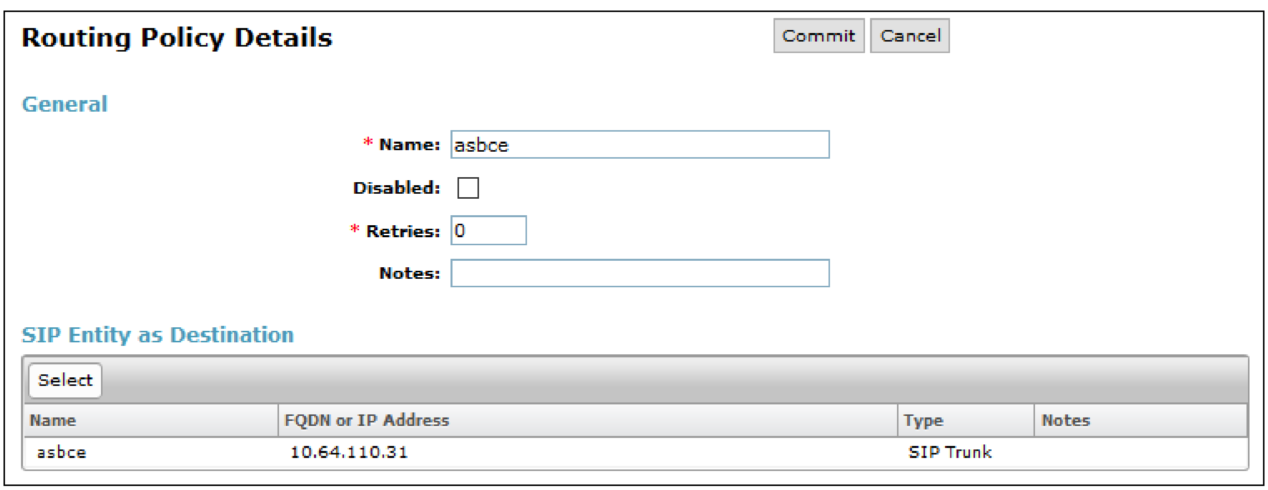
Routing policies must be created to direct how calls will be routed to a system. To add a routing policy, select **Routing Policies** on the left panel menu and then click on the **New** button (not shown).

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 calls inbound from the SIP Trunk to Communication Manager and ASBCE.





6.8. 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).

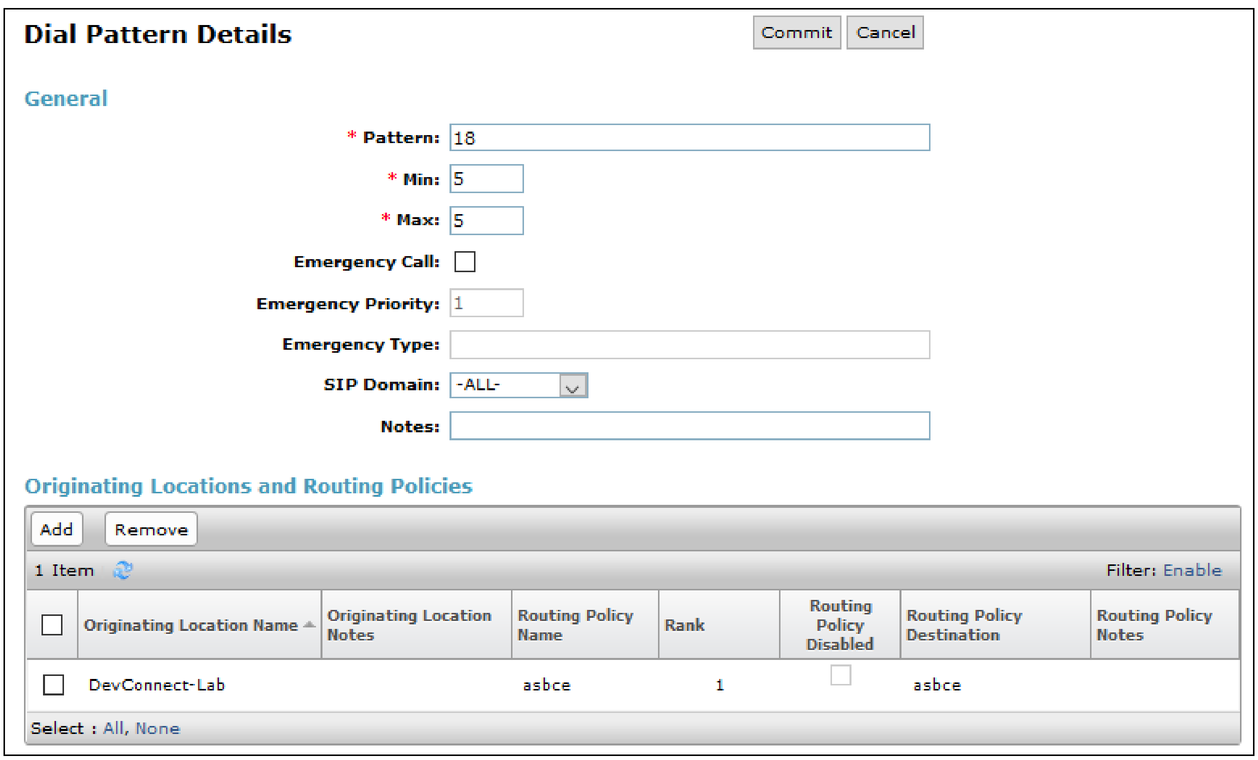
Under **General**:

- 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**.

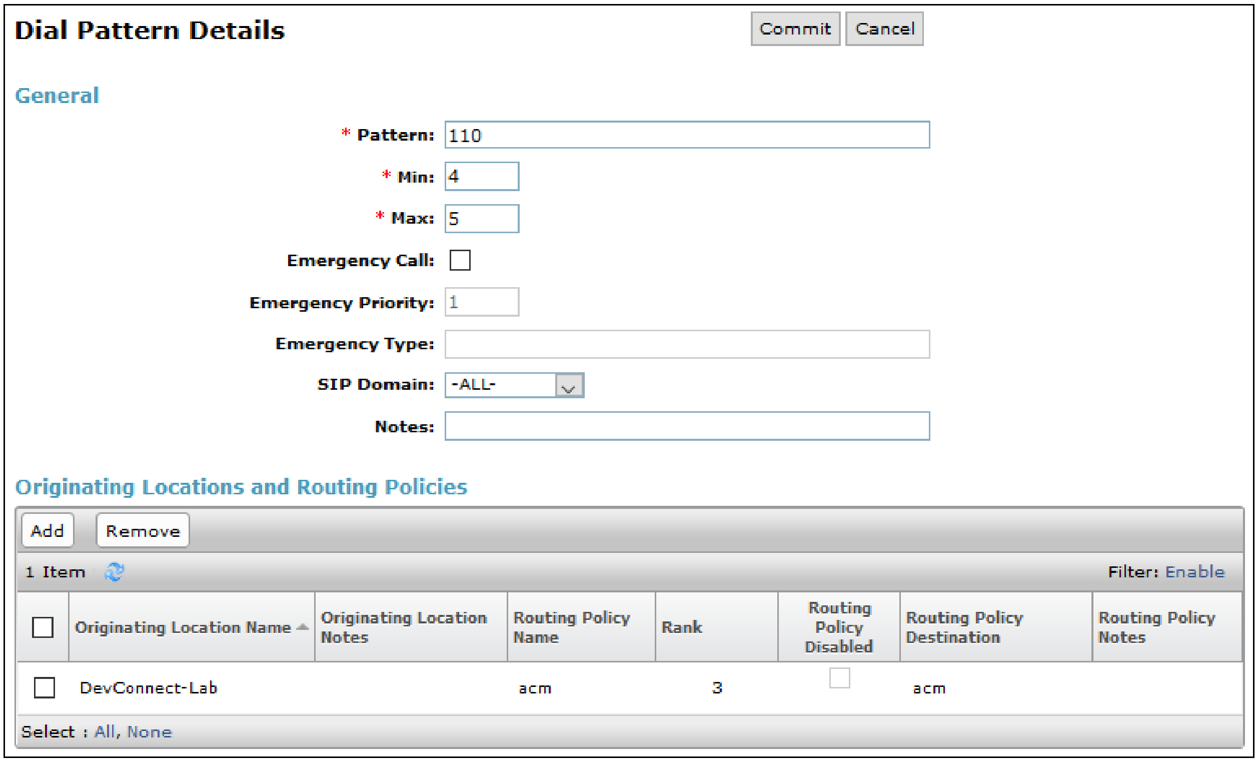
Under **Originating Locations and Routing Policies**:

- Click **Add**, in the resulting screen (not shown).
- Under **Originating Location**, select the location defined in **Section 6.3** or **ALL**.
- Under **Routing Policies** select one of the routing policies defined in **Section 6.7**.
- Click **Select** button to save.

The following screen shows an example dial pattern configured for the Avaya SBCE which will route the calls out to Service Provider's SIP Trunking service.



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



Dial Pattern Details

Commit Cancel

General

* Pattern: 110

* Min: 4

* Max: 5

Emergency Call:

Emergency Priority: 1

Emergency Type:

SIP Domain: -ALL-

Notes:

Originating Locations and Routing Policies

Add Remove

1 Item Filter: Enable

	Originating Location Name	Originating Location Notes	Routing Policy Name	Rank	Routing Policy Disabled	Routing Policy Destination	Routing Policy Notes
	DevConnect-Lab		acm	3		acm	

Select : All, None

6.9. Administer Application for Avaya Aura® Communication Manager

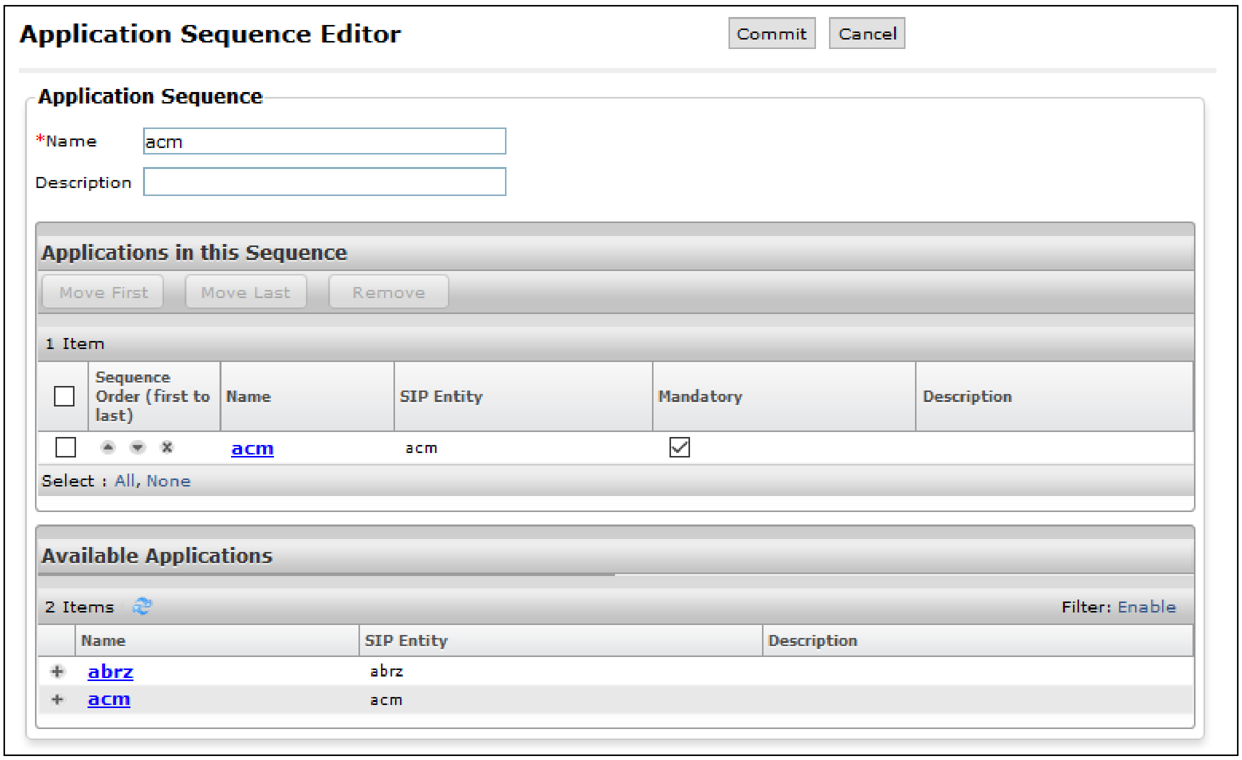
The Application for Communication Manager would normally be defined at system installation, but is shown here for reference. From the **Home** tab select **Session Manager** from the menu. In the resulting tab from the left panel menu select **Application Configuration → Applications** and click **New** (not shown).

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

6.10. Administer Application Sequence for Avaya Aura® Communication Manager

The Application Sequence for Communication Manager would normally be defined at system installation, but is shown here for reference. From the left panel navigate to **Session Manager → Application Configuration → Application Sequences** and click on **New** (not shown).

- 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**.

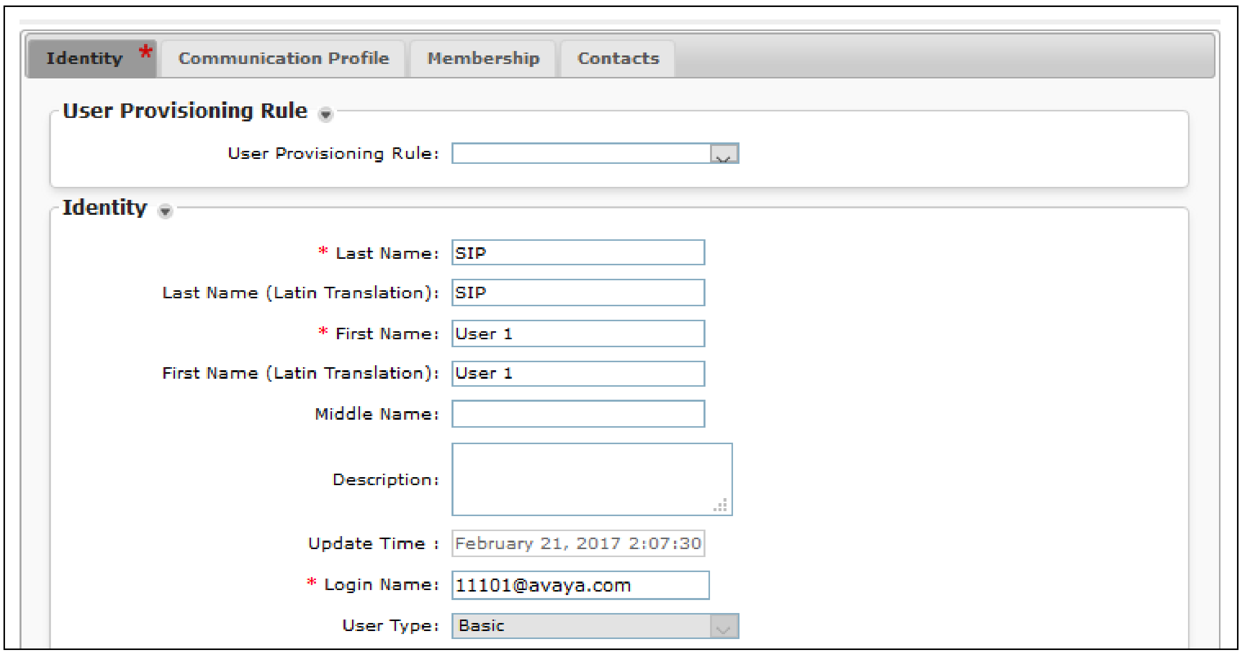


6.11. Administer SIP Extensions

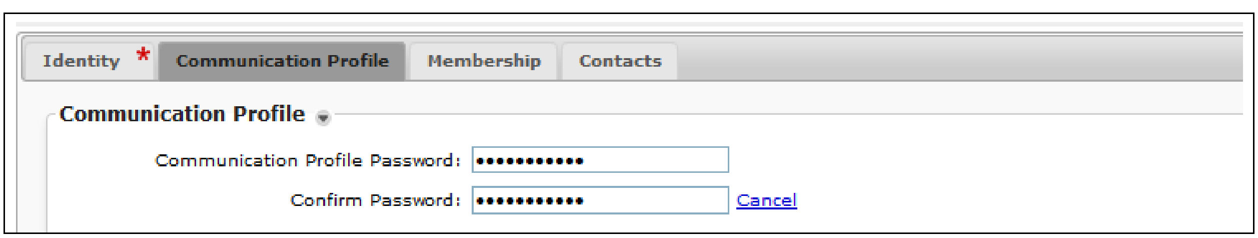
SIP extensions are registered with the Session Manager and use Communication Manager for their feature and configuration settings. From the **Home** tab select **User Management** from the menu. Then select **Manage Users** and click **New** (not shown).

On the **Identity** tab:

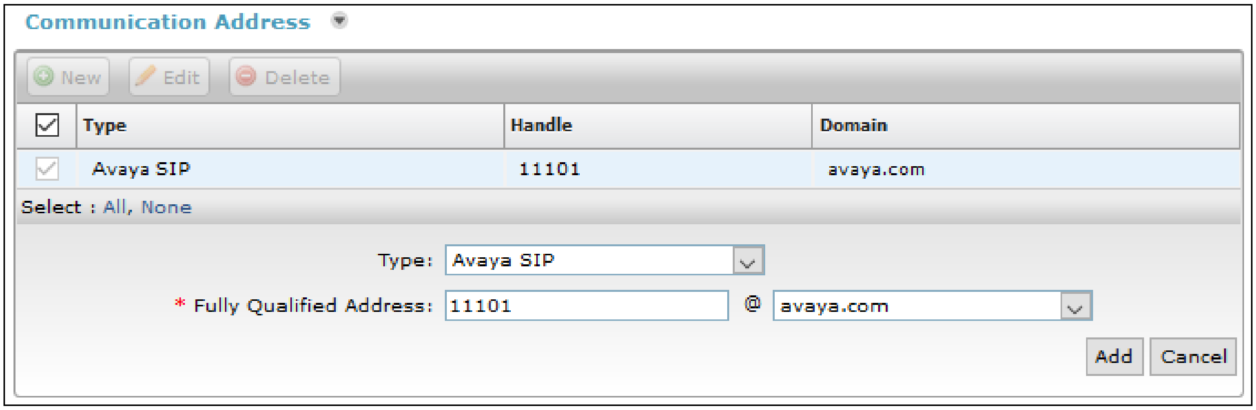
- Enter the user's name in the **Last Name** and **First Name** fields.
- In the **Login Name** field enter a unique system login name in the form of user@domain e.g. **11101@avaya.com** which is used to create the user's primary handle.
- The **Authentication Type** should be **Basic**.
- Set the **Language Preference** and **Time Zone** as required.



On the **Communication Profile** tab, enter a numeric **Communication Profile Password** and confirm it.

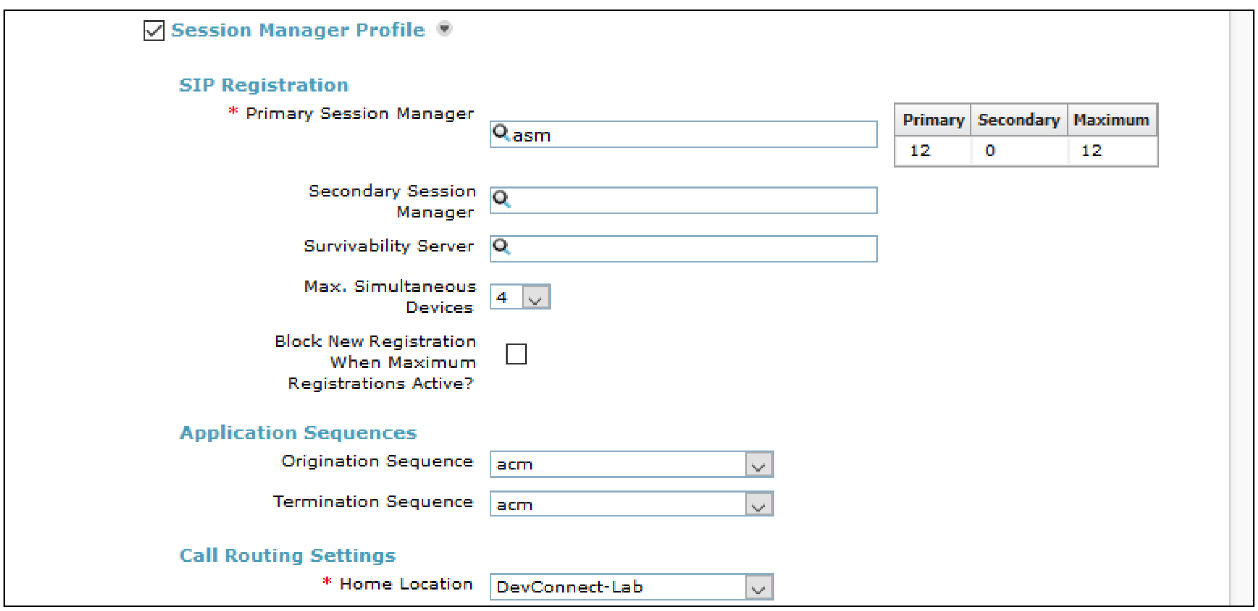


Expand the **Communication Address** section and click **New**. For the **Type** field select **Avaya SIP** from the drop-down menu. In the **Fully Qualified Address** field, enter an extension number and select the relevant domain from the drop-down menu. Click the **Add** button.



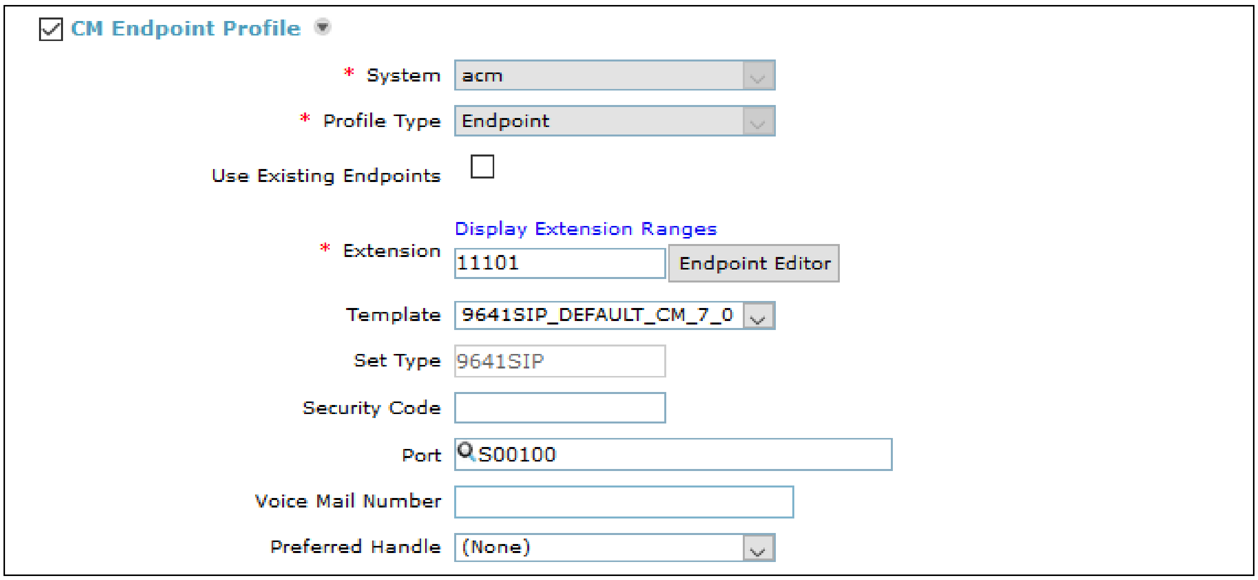
Expand the **Session Manager Profile** section.

- 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 Sequence** field configured in **Section 6.10**.
- Select the appropriate application sequence from the drop-down menu in the **Termination Sequence** field configured in **Section 6.10**.
- Select the appropriate location from the drop-down menu in the **Home Location** field.



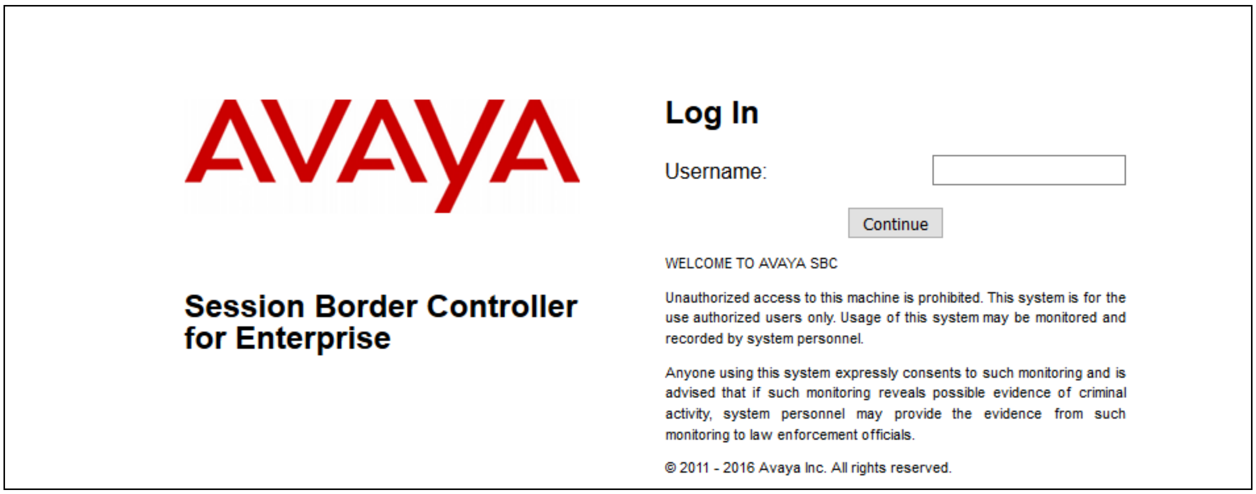
Expand the **Endpoint Profile** section.

- Select Communication Manager Element 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.
- In the **Port** field **IP** is automatically inserted.
- Select the **Delete Endpoint on Unassign of Endpoint from User or on Delete User** check box.
- Select **Commit** (not shown) to save changes and System Manager will add the Communication Manager user configuration automatically.



7. Configure Avaya Session Border Controller for Enterprise

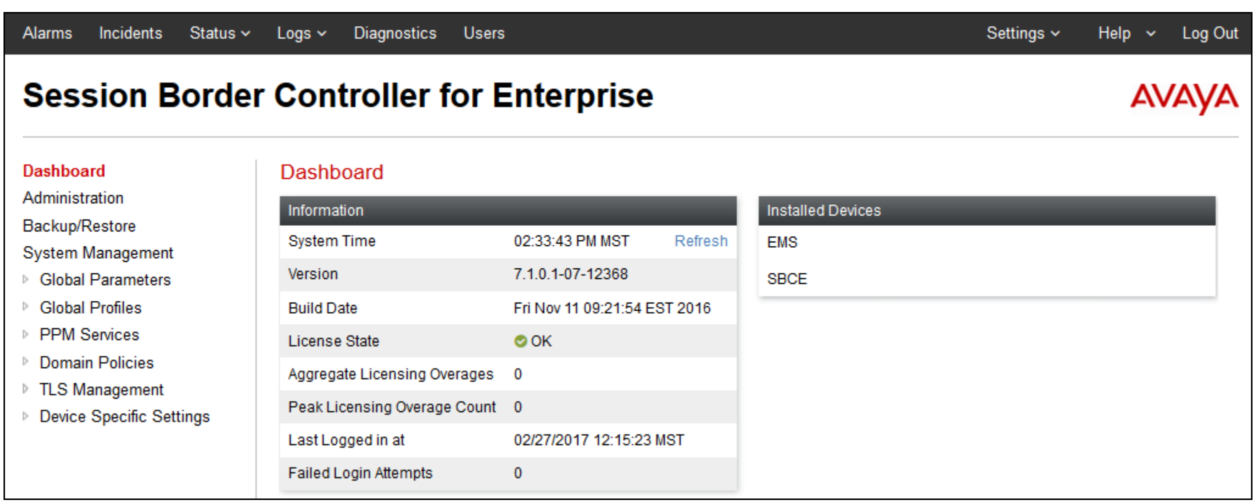
This section describes the configuration of the Avaya Session Border Controller for Enterprise (Avaya SBCE). The Avaya SBCE provides security and manipulation of signalling to provide an interface to the Service Provider's SIP Trunk that is standard where possible and adapted to the Service Provider's SIP implementation where necessary. Avaya SBCE also provides the SIPREC interface that is used by MiaRec to record calls. Configuration related to Session Manager and Service Provider's SIP Trunk is not shown in this document.



7.1. Access Avaya Session Border Controller for Enterprise

Access the Session Border Controller using a web browser by entering the URL **https://<ip-address>**, where **<ip-address>** is the private IP address configured at installation. A log in screen is presented. Log in using the appropriate username and password.

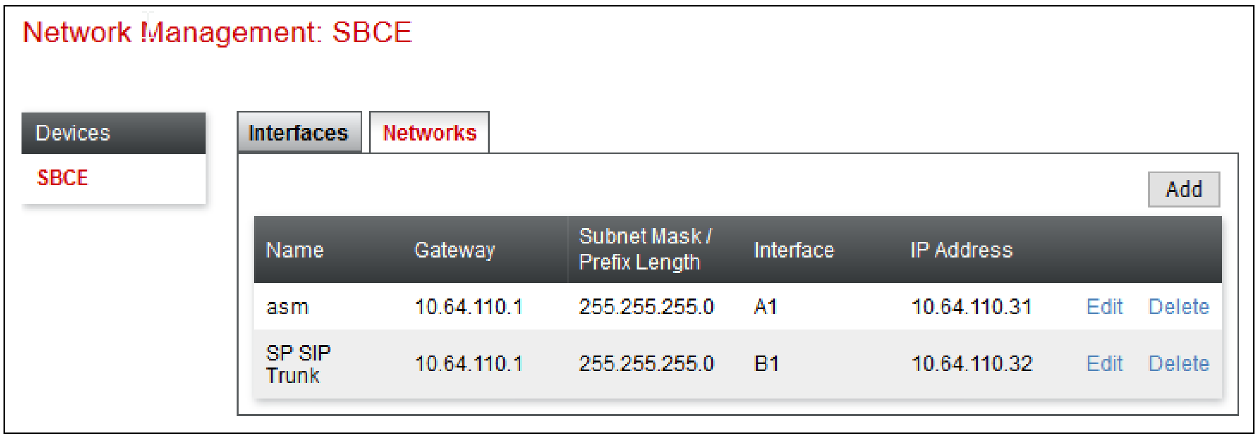
Once logged in, a dashboard is presented with a menu on the left-hand side. The menu is used as a starting point for all configuration of the Avaya SBCE.



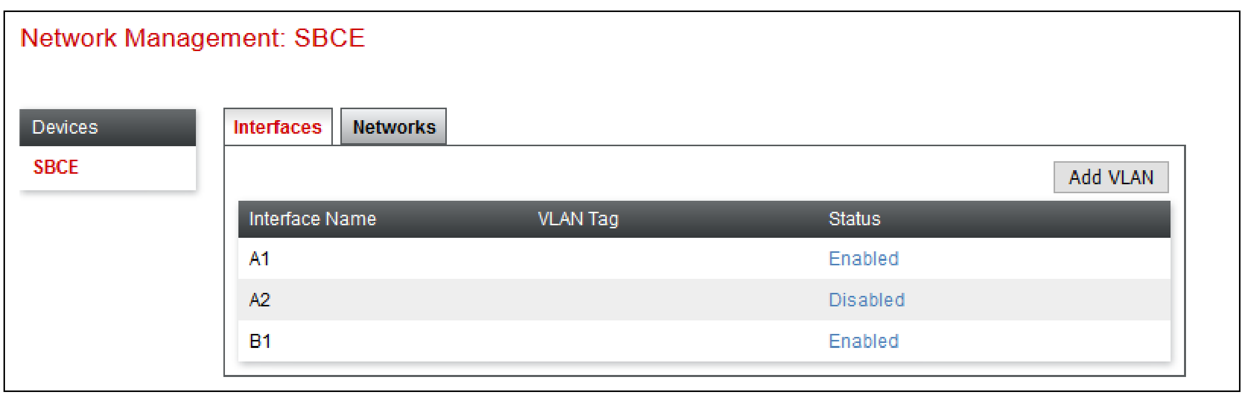
7.2. Define Network Management

Network information is required on the Avaya SBCE to allocate IP addresses and subnet masks to the interfaces. Note that only the **A1** and **B1** interfaces are used, typically the **A1** interface is used for the internal side and **B1** is used for external. Each side of the Avaya SBCE can have only one physical interface assigned.

To define the network information, navigate to **Device Specific Settings → Network Management** in the main menu on the left hand side and click on **Add**. The following interfaces were added for Session Manager and Service Provider's SIP Trunk.



Select the **Interface Configuration** tab and click on the **Status** of the physical interface to toggle it. A status of **Disabled** will be changed to **Enabled**.



Note: to ensure that the Avaya SBCE uses the interfaces defined, the Application must be restarted.

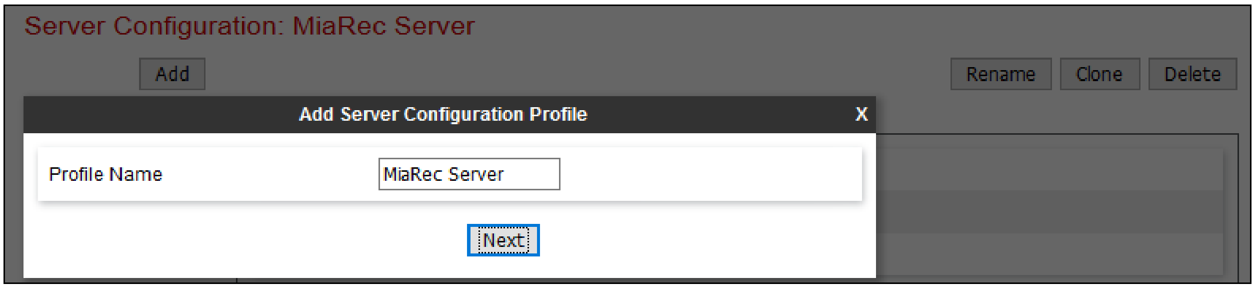
- Click on **System Management** in the main menu (not shown).
- Select **Restart Application** indicated by an icon in the status bar (not shown).

A status box will appear (not shown) that will indicate when the application has restarted.

7.3. Define Servers

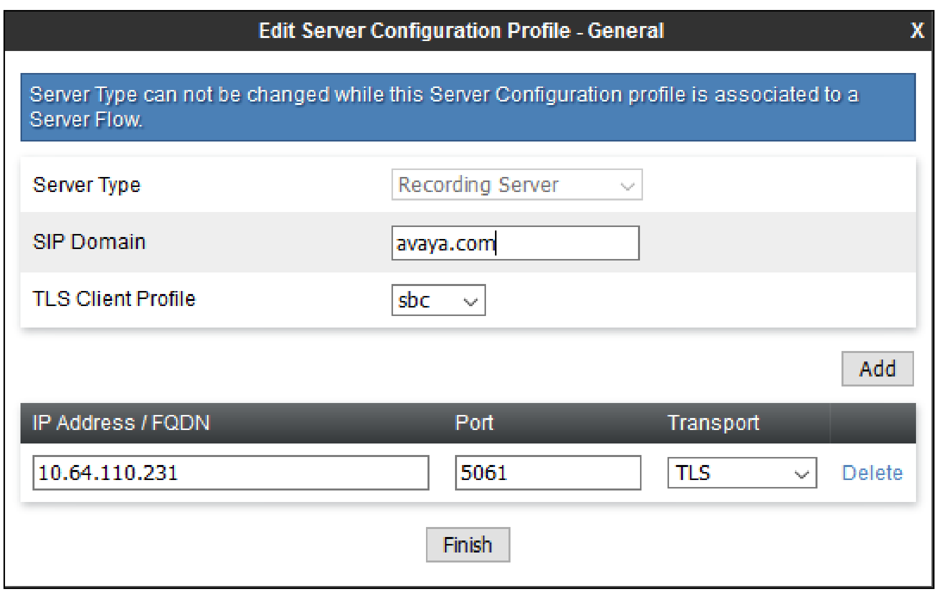
A server definition is required for each server connected to the Avaya SBCE. In this case, the MiaRec is configured as a Recording Server.

To define the MiaRec Recording Server, navigate to **Global Profiles → Server Configuration** in the main menu on the left hand side. Click on **Add** and enter an appropriate name in the pop-up menu.

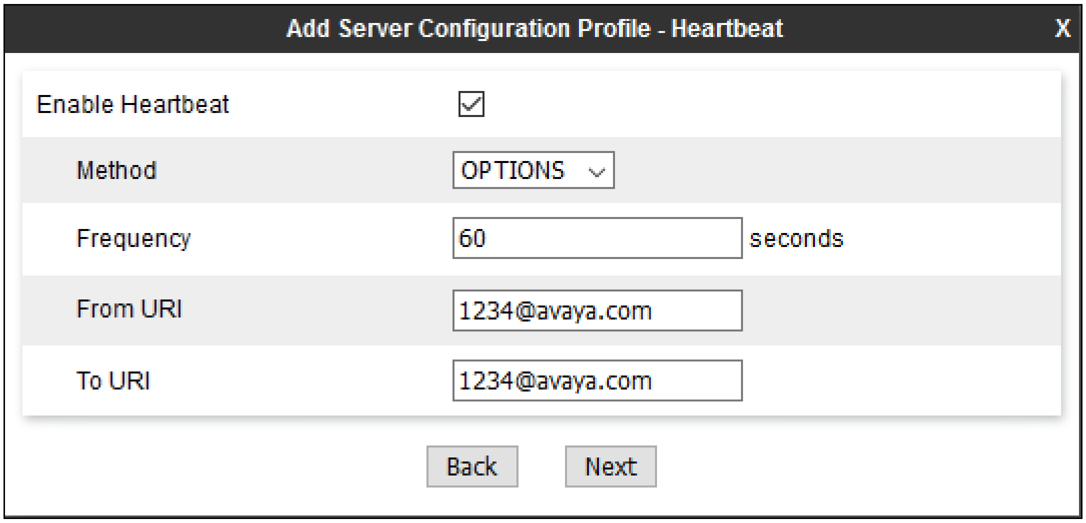


Click on **Next** and enter details in the dialogue box.

- In the **Server Type** drop down menu, select **Recording Server**.
- In the **SIP Domain** type in the domain configured in **Section 6.2**.
- Select a configured **TLS** profile for **TLS Client Profile**.
- Click on **Add** to enter an IP address.
- In the **IP Addresses / FQDN** box, type the MiaRec recording server interface address.
- In the **Port** box, enter the port to be used for the TLS listening port configured on the MiaRec (shown in the step 8).
- In the **Transport** drop down menu, select **TLS**.
- Click on **Next**.



Click on **Next** and configure as follows.

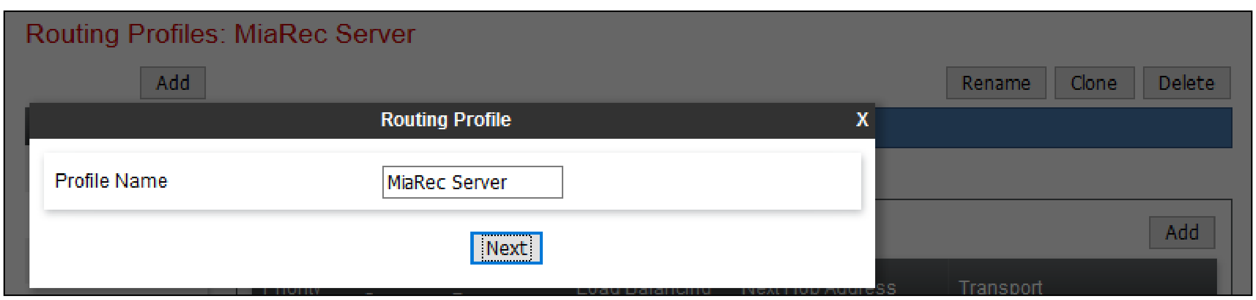


Select **Next** and then **Finish** (not shown).

7.4. Define Routing

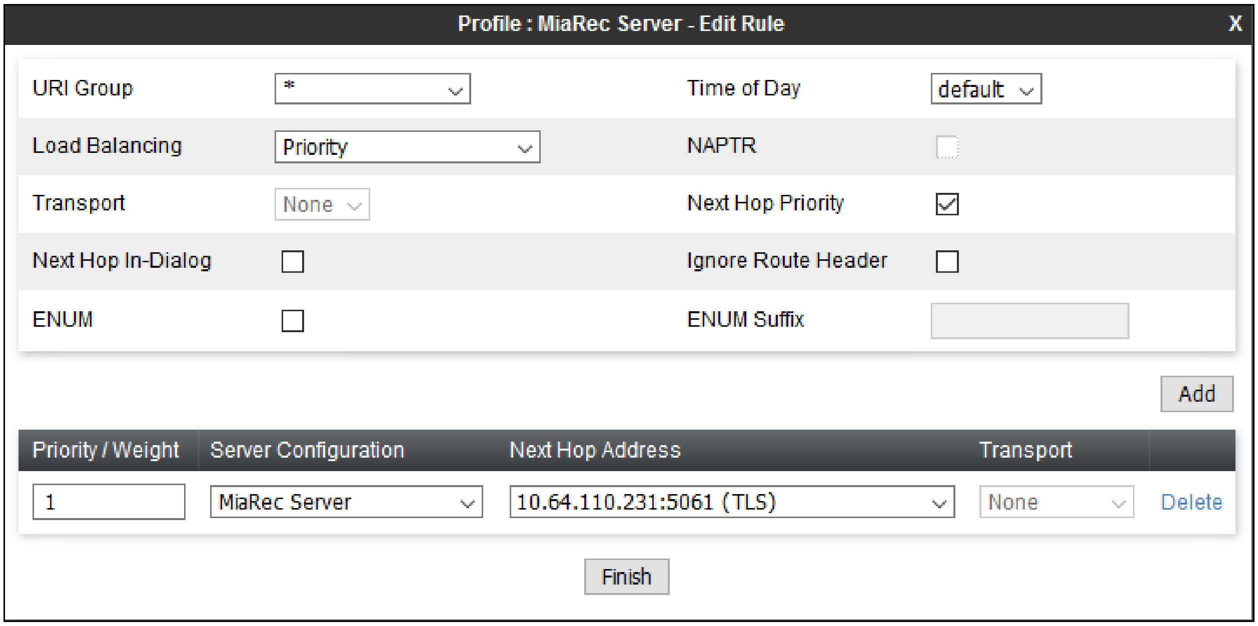
Routing information is required for routing recordings to MiaRec. The IP addresses and ports defined here will be used as the destination addresses for signalling.

To define routing to the MiaRec SIP Trunk, navigate to **Global Profiles → Routing** in the main menu on the left hand side. Click on **Add** and enter an appropriate name in the dialogue box.



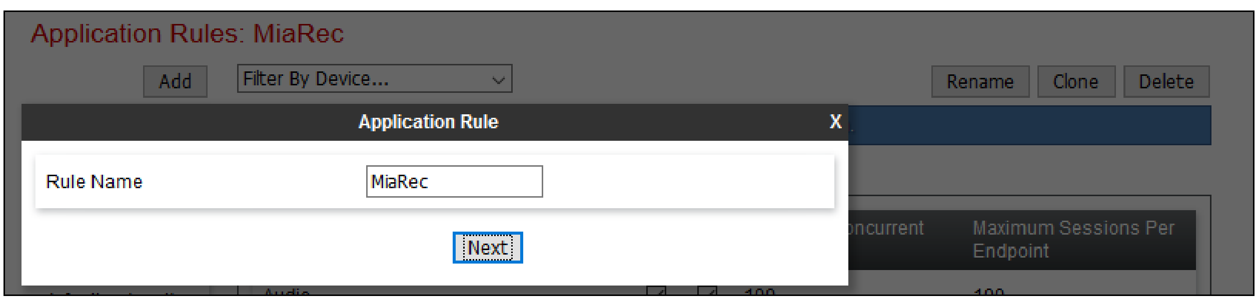
Click on **Next** and enter details for the Routing Profile:

- Click on **Add** to specify the IP address for the MiaRec SIP trunk.
- Assign a priority in the **Priority / Weight** field, during testing a value of **1** was used.
- Select the Server Configuration defined in **Section 7.3** in the **Server Configuration** drop down menu. This automatically populates the **Next Hop Address** field.
- Click **Finish**.

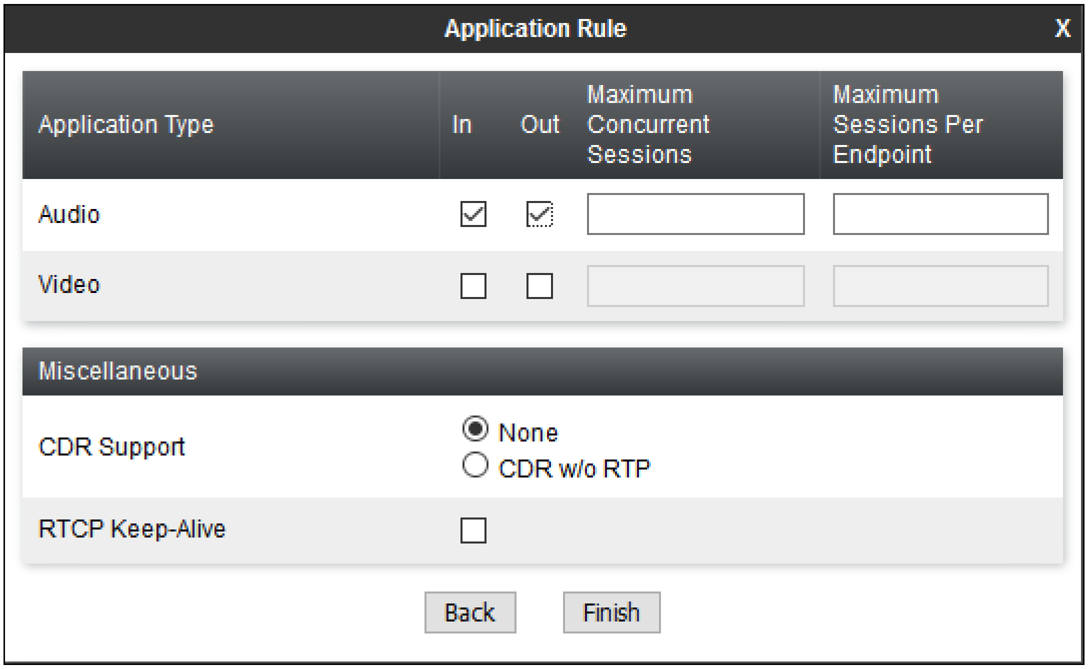


7.5. Define Application Rules

An application rules needs to be defined for MiaRec. To create a new Application Rules, navigate to **Domain Policies → Application Rules.** Click on **Add** and enter an appropriate name in the pop-up menu and select **Next**.

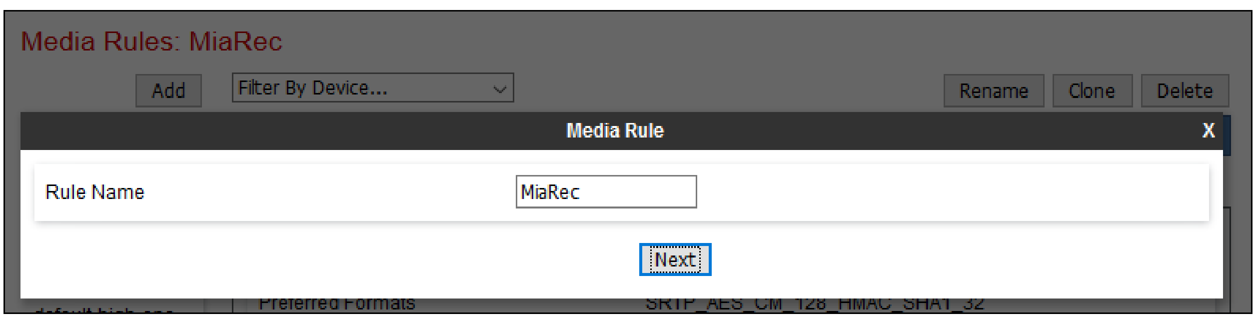


On the **Application Rule** pop-up windows check **In** and **Out** boxes for **Audio**, and select **Finish**.

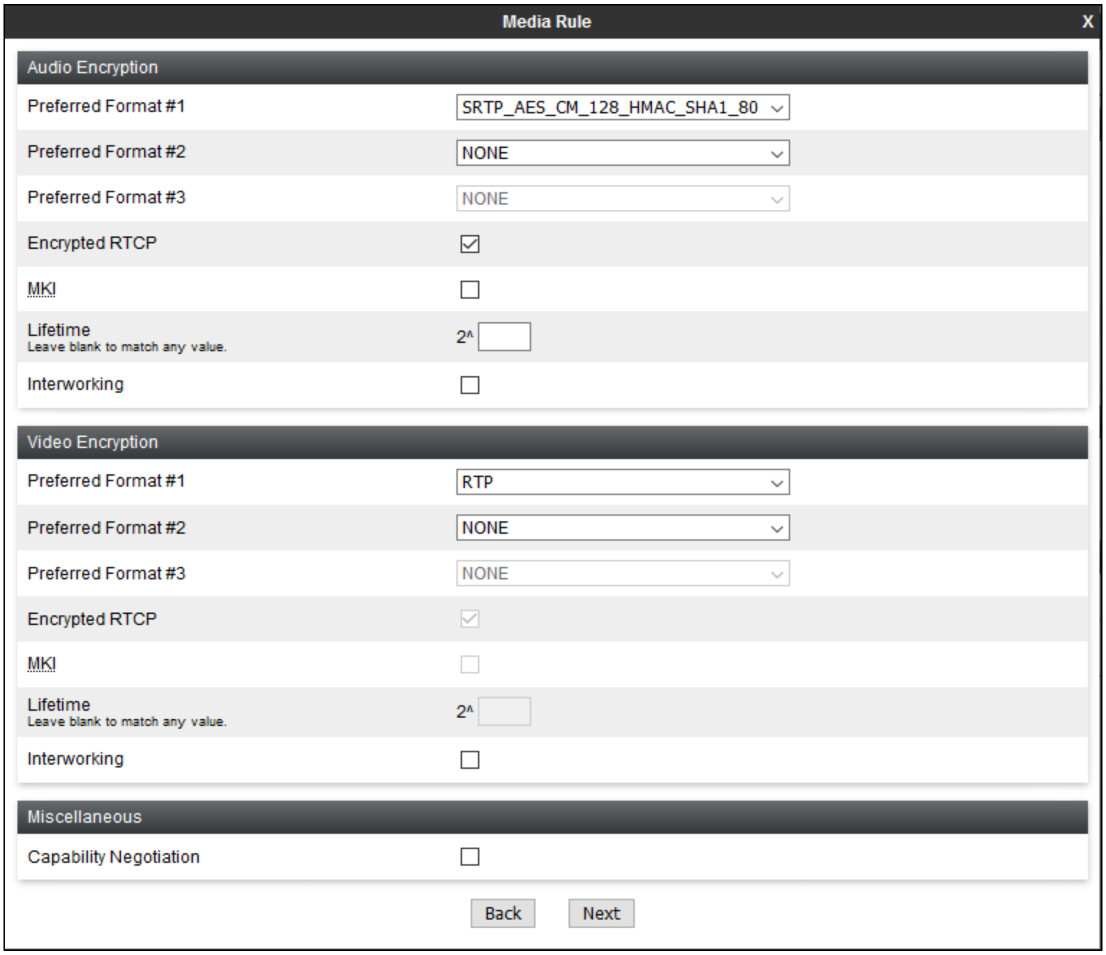


7.6. Define Media Rules

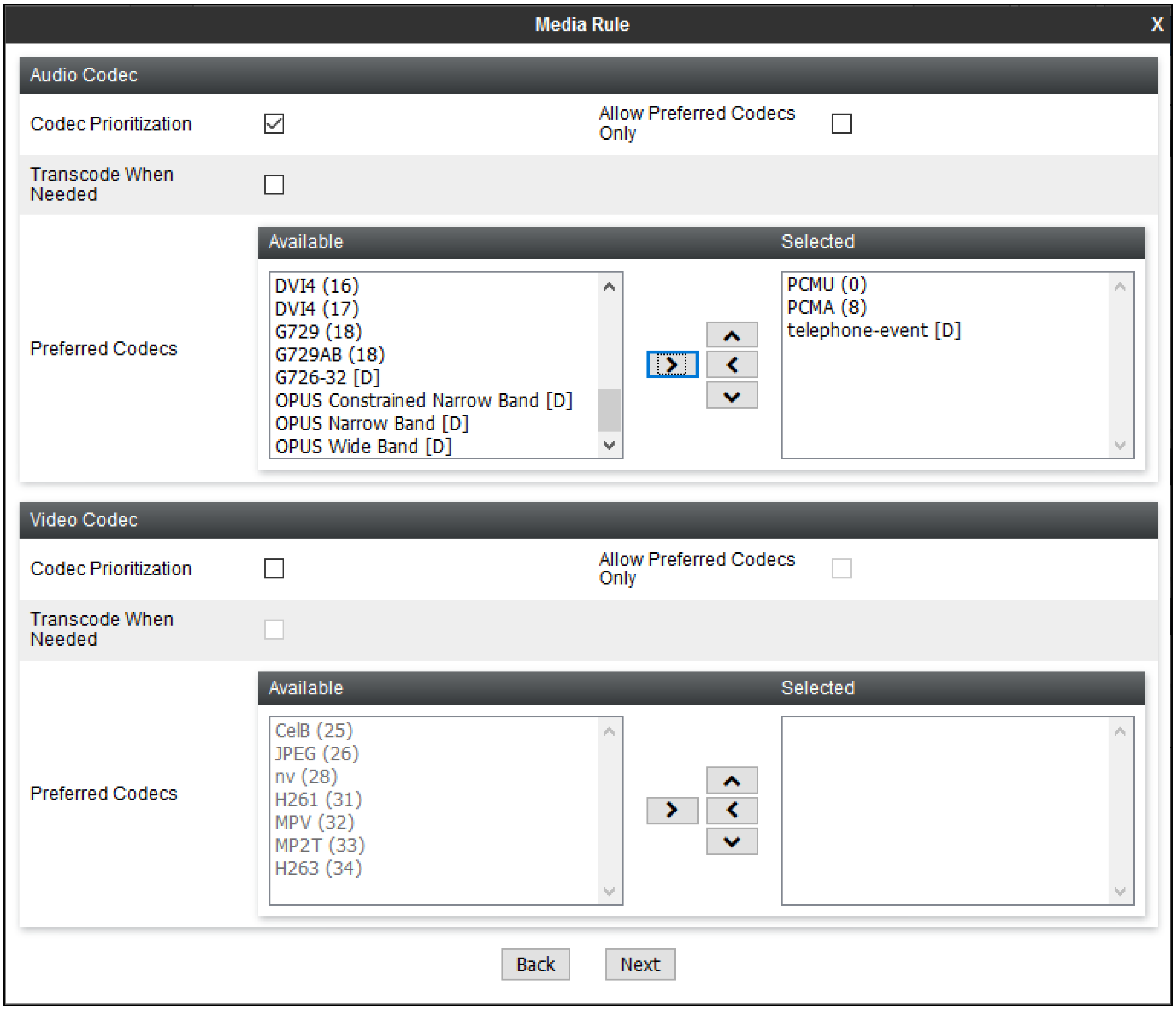
Audio formats need to be specified for MiaRec. To create a Media Rule for MiaRec, navigate to **Domain Policies → Media Rules.** Click on **Add** and enter an appropriate name in the pop-up menu and select **Next**.



On the **Media Rule** pop-up**,** under **Audio Encryption,** select a **Preferred Format #1** and select continue. If using, SRTP select **SRTP_AES_CM_128_128_HMAC_SHA1_80** or for RTP select **RTP,** select **Next**.

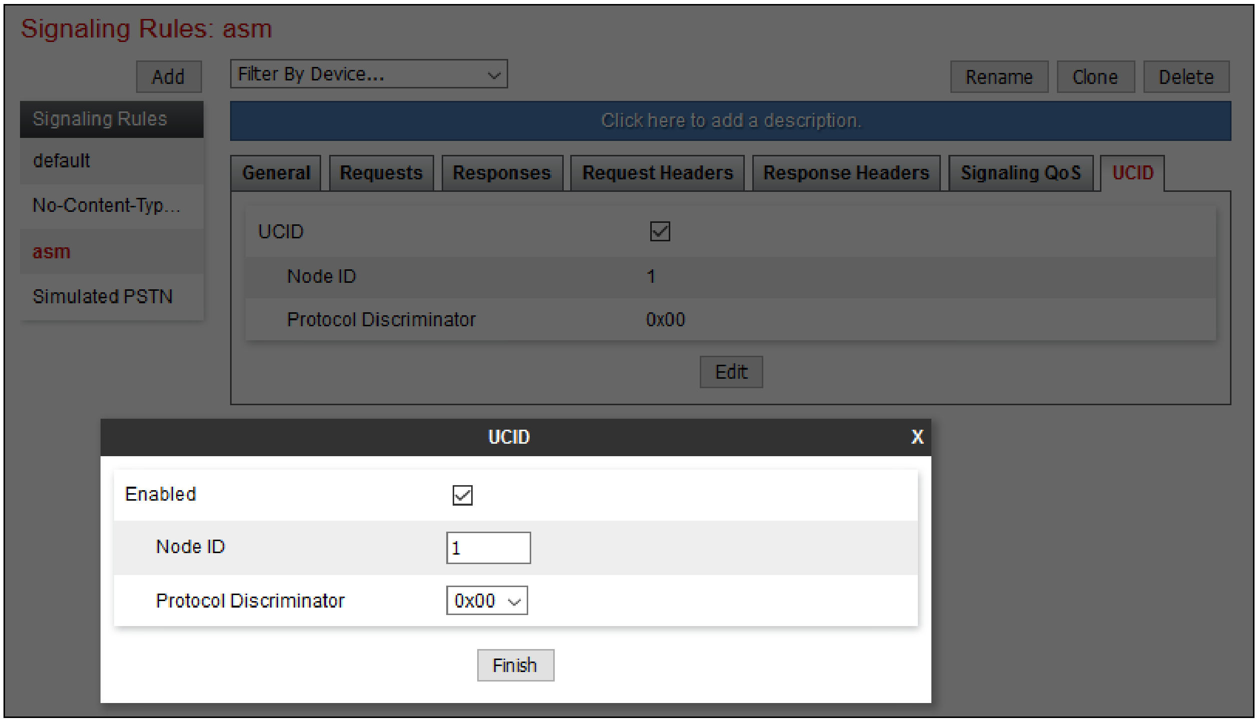


On the **Media Rule** pop-up, under the **Audio Codec** section, select box for **Codec Prioritization**. For **Preferred Codecs** select **PCMU**, **PCMA** and **telephone-event**, and click **>.** Select **Next** and **Finish** to save the configuration (not shown).



7.7. Configure UCID

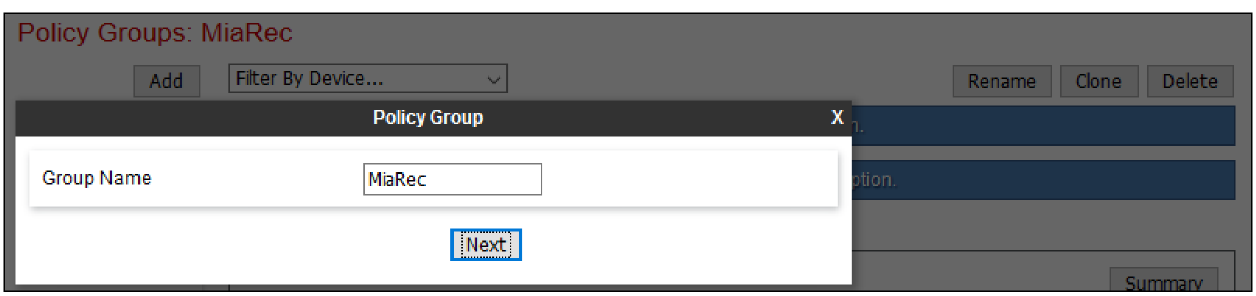
UCID needs to be enabled for Signaling Rules that are defined for Session Manager and MiaRec. Navigate to **Domain Policies → Signaling Rules.** Select the policy for Session Manager and select the **UCID** tab. Click **Edit**, check box for **Enabled** and type in a unique value in **Node ID** field. Select **Finish** to save configuration.



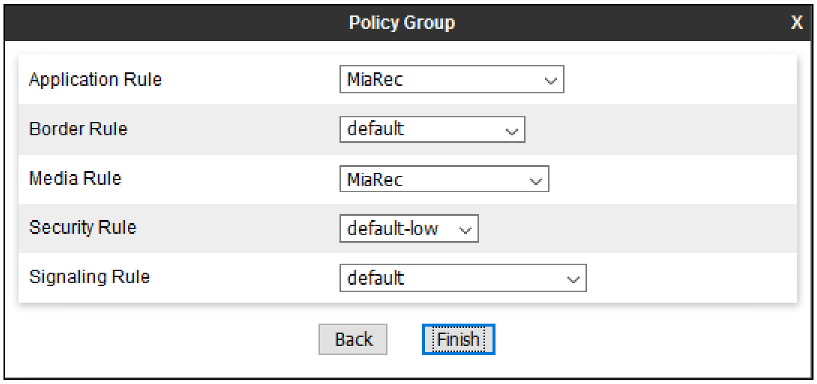
Perform similar steps for MiaRec signaling rule.

7.8. Define End Point Policy Group

To define an End Point Policy Group for MiaRec, navigate to **Domain Policies → End Point Policy Group** and select **Add**. Click on **Add** and enter an appropriate name in the pop-up menu and select **Next**.

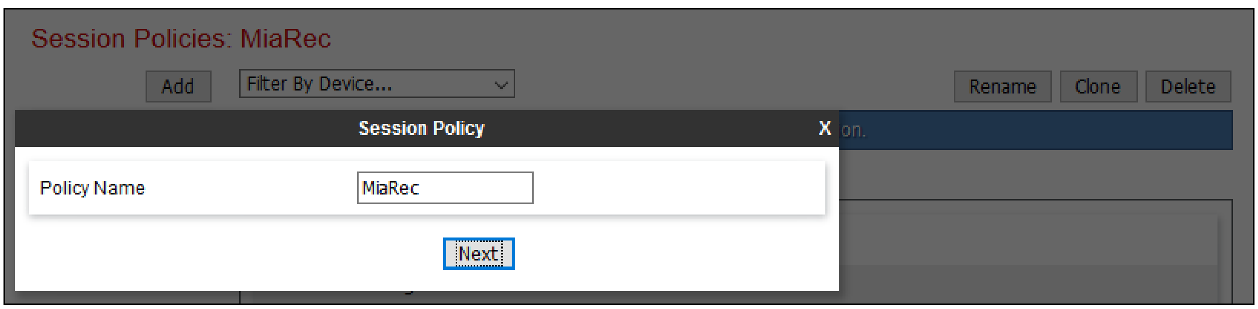


On the **Policy Group** pop-up, select the **Application Rule** defined in **Section 7.5** and select the **Media Rule** defined in **Section 7.6**. Select **Finish** to save configuration.

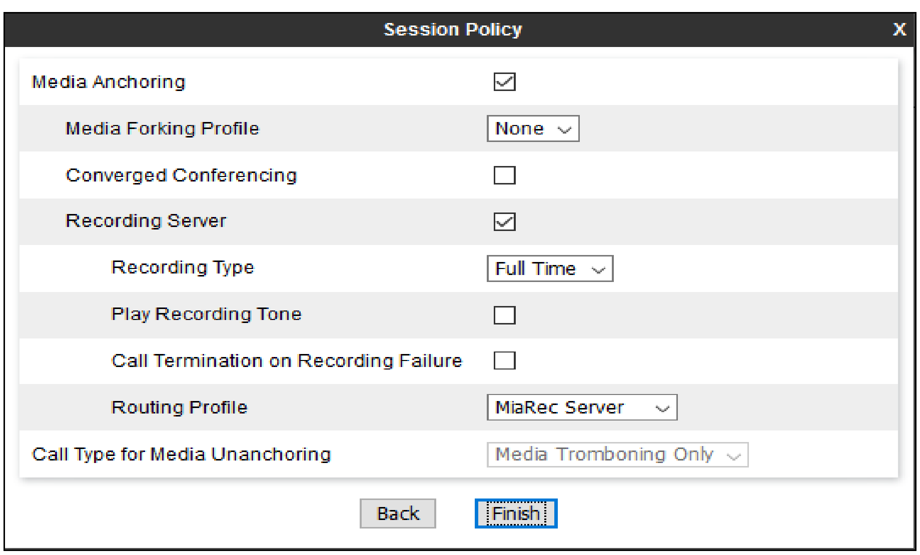


7.9. Define Session Policies

To define Session Policy for MiaRec, navigate to **Domain Policies → Session Policies** and select **Add**. Click on **Add** and enter an appropriate name in the pop-up menu and select **Next**.

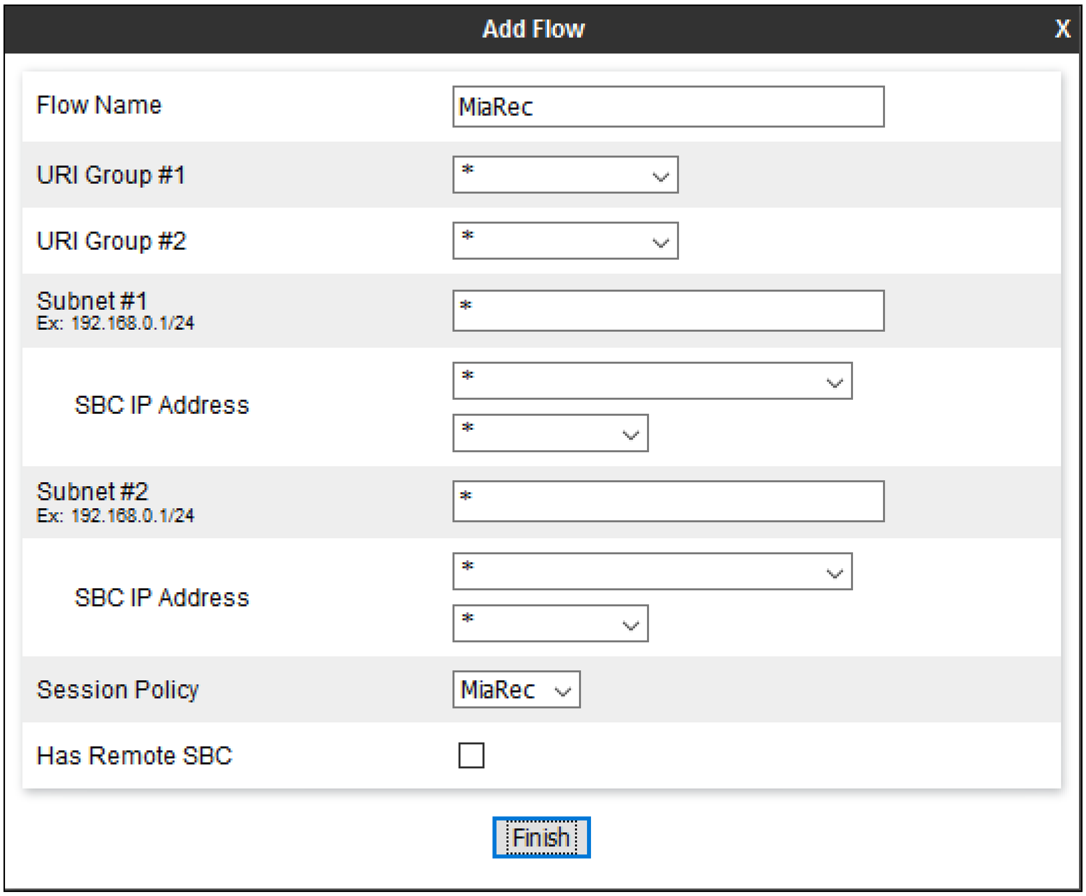


On the **Session Policy** pop-up, select box for **Media Anchoring** and **Recording Server.** For **Routing Profile** select the Routing profile configured in **Section 7.4**.



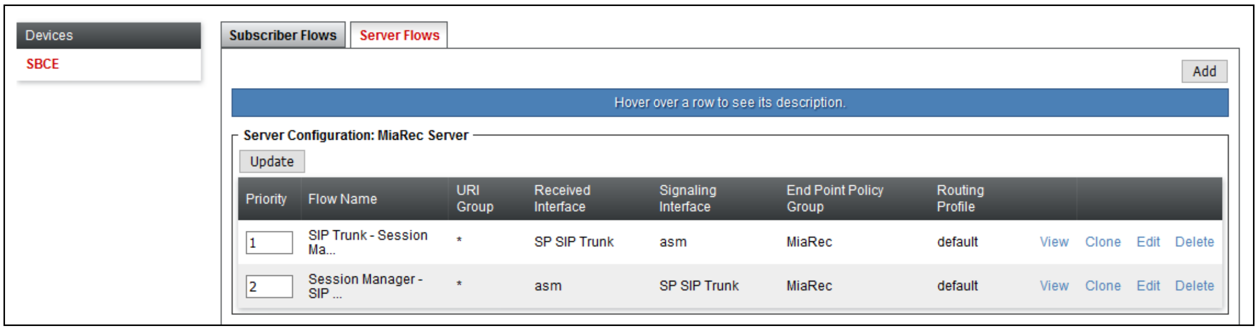
7.10. Define Session Flows

To define Session Policy for MiaRec, navigate to **Device Specific Settings → Session Flows** and select **Add**. Click on **Add** and enter an appropriate **Flow Name** in the pop-up menu and select the **Session Policy** defined in **Section 7.9**. Select **Finish** to save the configuration.

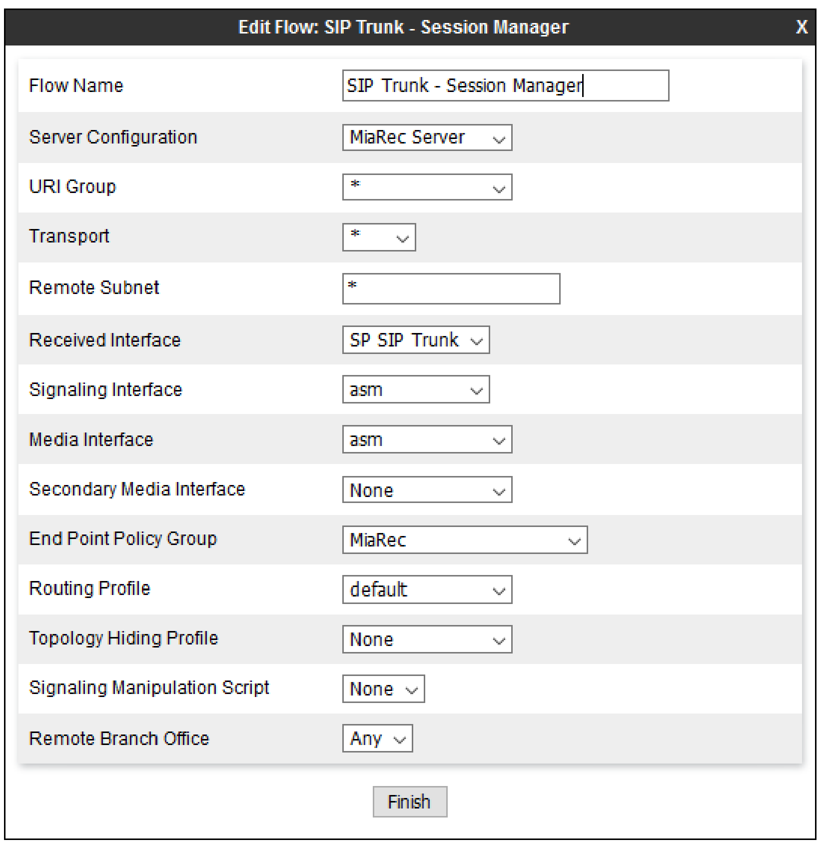


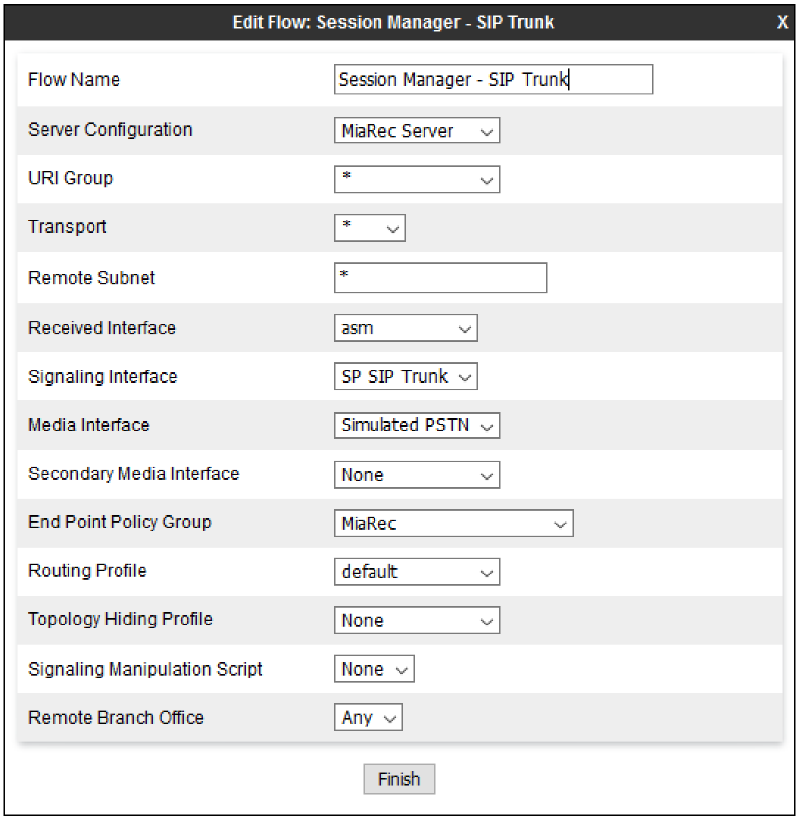
7.11. Server Flows

Server Flows combine the previously defined profiles for Session Manager and Service Provider's SIP Trunk. These End Point Server Flows allow calls to be recorded by MiaRec when they are passing through Avaya SBCE to the Service Provider's SIP Trunk. Navigate to **Device Specific Setting → End Point Flows → Server Flows**. There were two Server Flows added for MiaRec, one to record calls coming in from Service Provider's SIP Trunking service and another for calls coming in from Session Manager. The screen capture below displays the configured Session Flows. Configure the fields as shown in the screen capture.

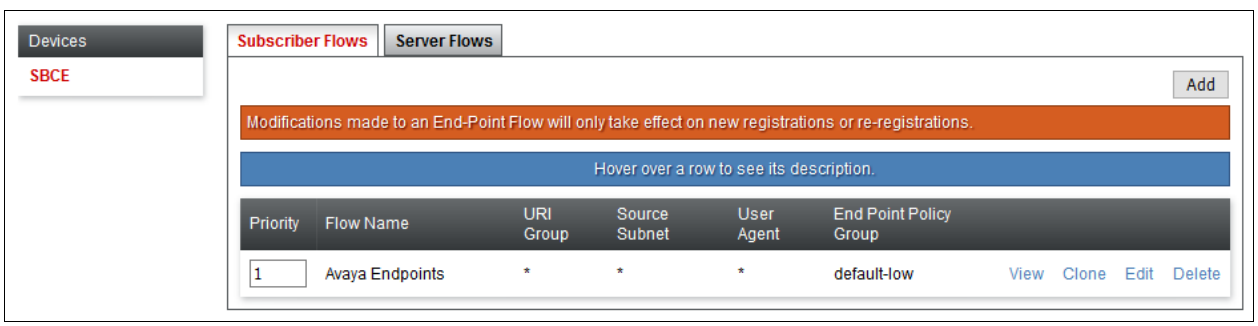


Screen captures for configuration of each Server Flow are as shown below:



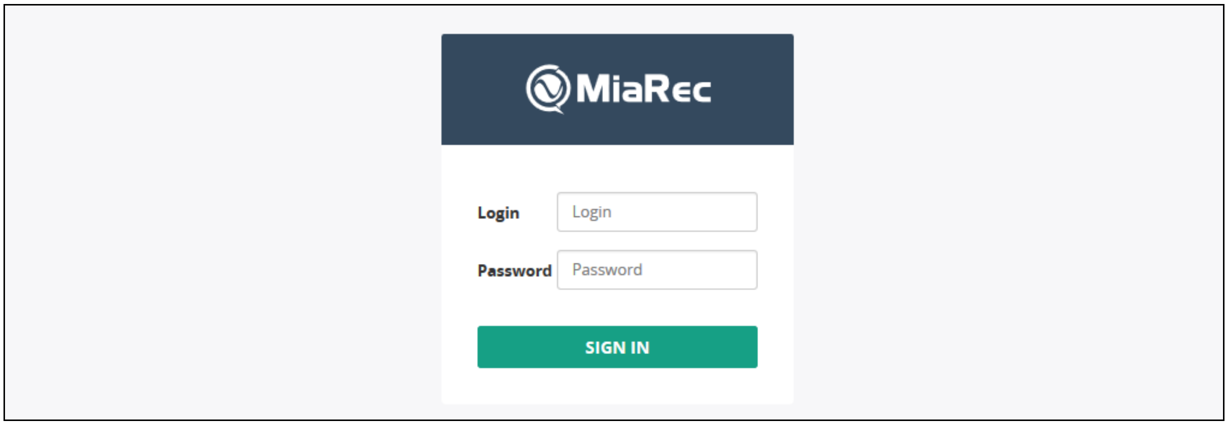


Additionally, for a **Subscriber Flow** was added for Remote Workers, as shown below. The Subscriber Flow allows Remote Workers to register to Session Manager via Avaya SBCE and also SIPREC recordings for MiaRec.

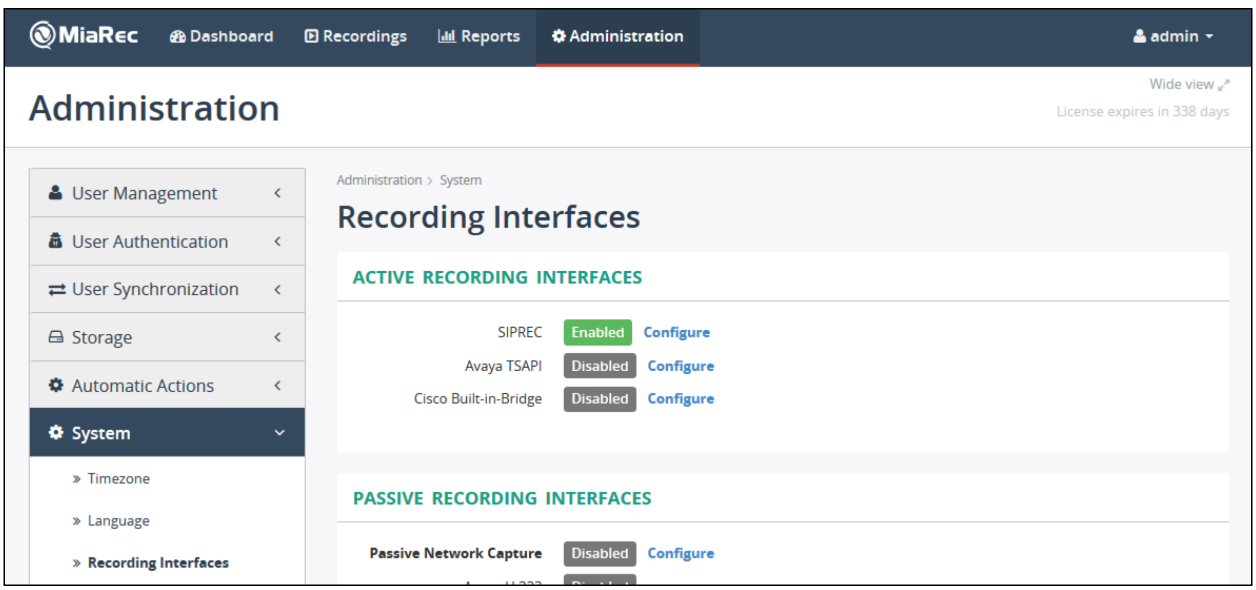


8. Configure the MiaRec

MiaRec was deployed as a virtual machine on a virtualization platform. Configuration for MiaRec is performed via MiaRec web user interface which can be accessed through a browser. Point the browser to http://<ip-address>, where ip-address is the IP Address of MiaRec. Log on using appropriate credentials.



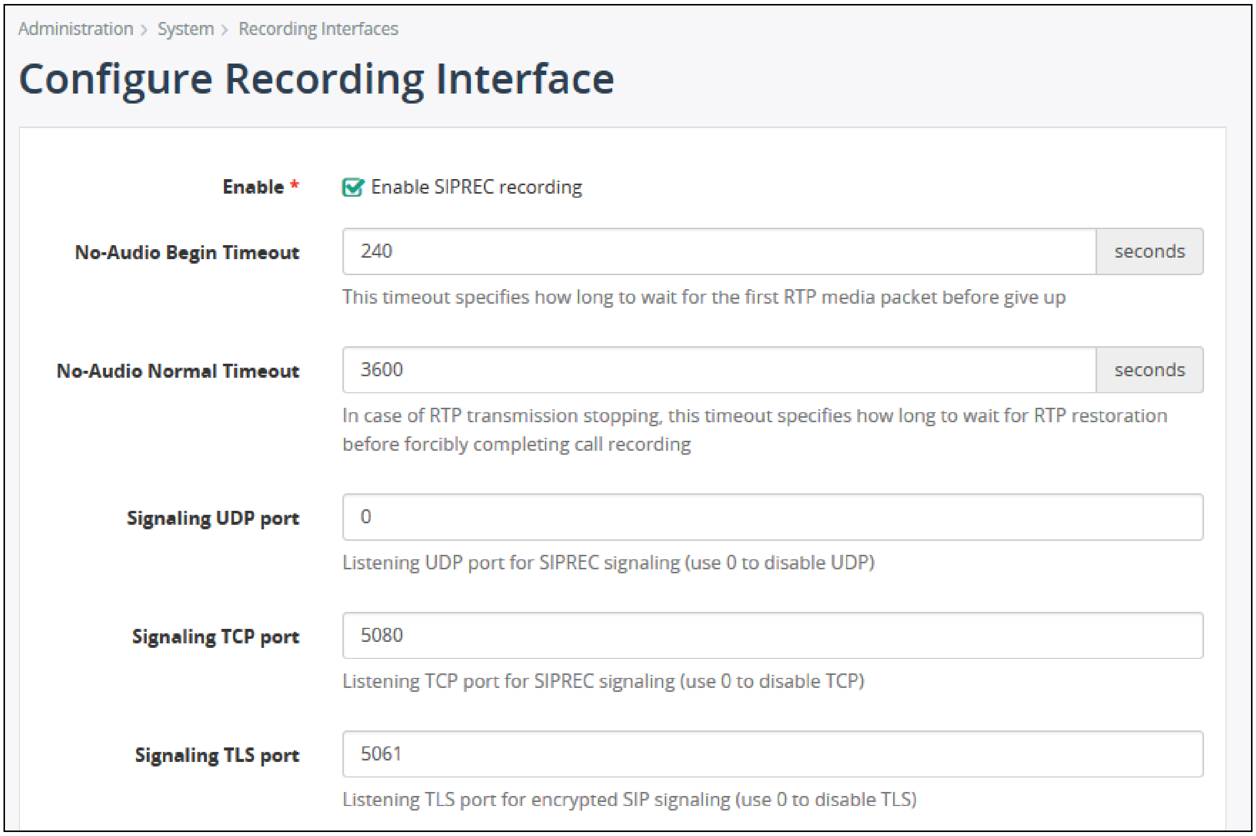
Navigate to **Administration → System → Recording Interfaces** and select **Configure** for **SIPREC**.



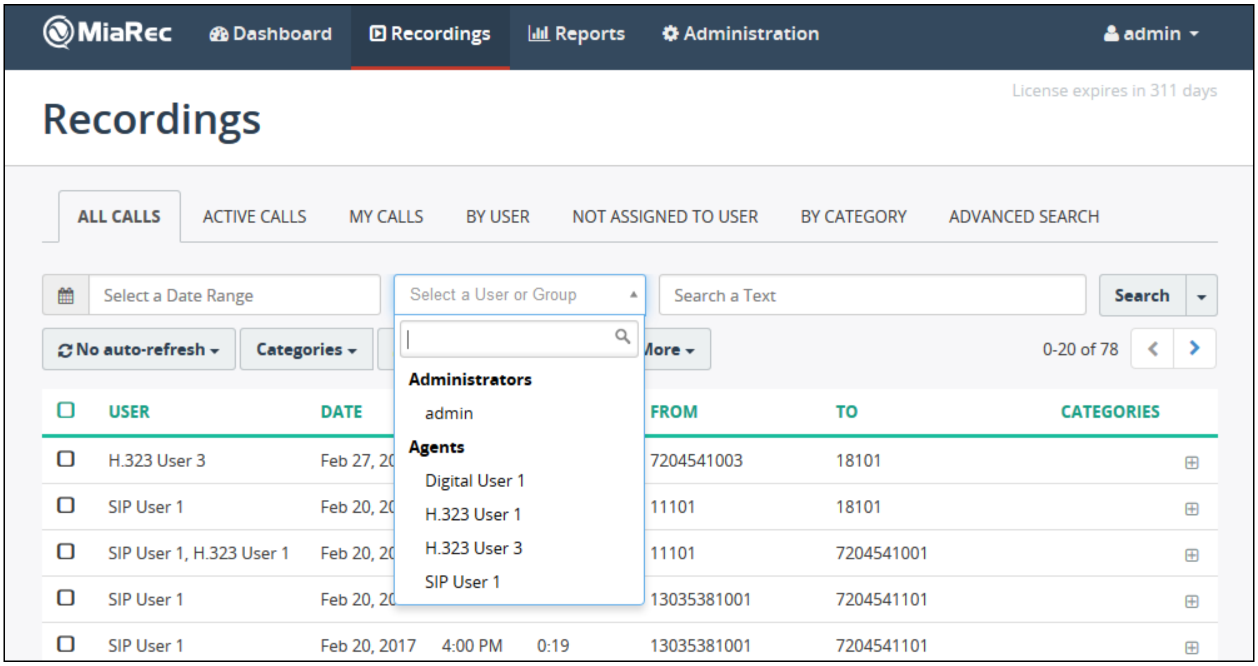
On the **Configure Recording Interface** page:

- Check box for **Enable SIPREC recording**.
- Type in port values for the signaling port depending on whether TCP or TLS is being used.

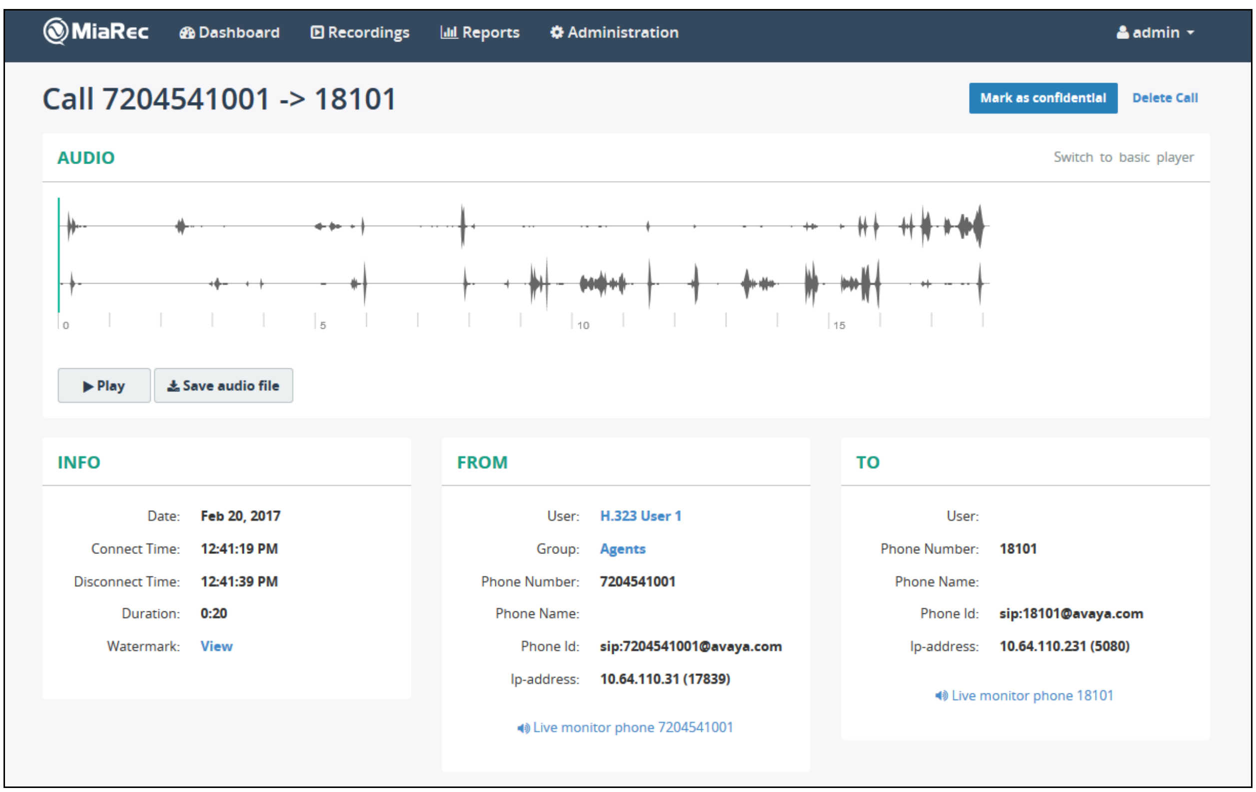
Select **Save** once done (not shown).



To access the call recordings, select **Recordings** on the MiaRec web interface:



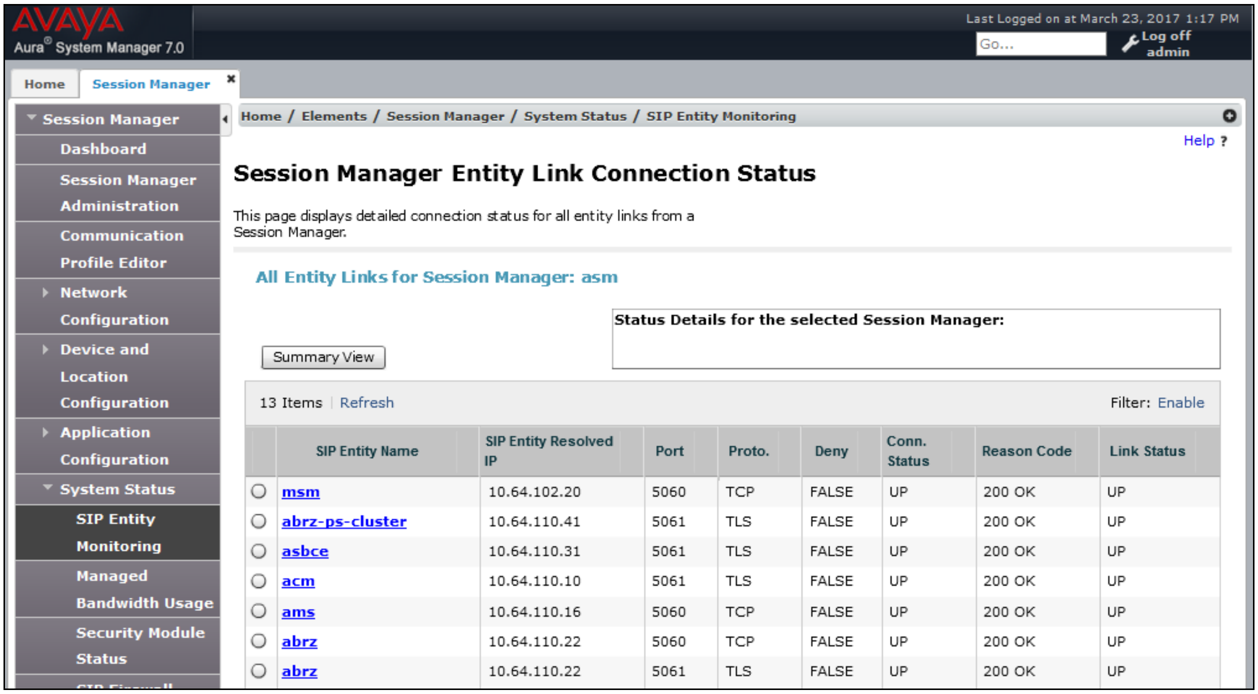
Select a recording to view the details and playback.



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 Entities from the list and observe if the **Conn Status** and **Link Status** are showing as **UP**.



2. From 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 T00011   in-service/idle      no
0001/002 T00012   in-service/idle      no
0001/003 T00013   in-service/idle      no
0001/004 T00014   in-service/idle      no
0001/005 T00015   in-service/idle      no
0001/006 T00016   in-service/idle      no
0001/007 T00017   in-service/idle      no
0001/008 T00018   in-service/idle      no
0001/009 T00019   in-service/idle      no
0001/010 T00020   in-service/idle      no
```

3. Verify that endpoints at the enterprise site can place calls to the Service Provider's SIP Trunk and that the calls are being recorded by MiaRec.
4. Verify that endpoints at the enterprise site can receive calls from the Service Provider's SIP Trunk and that the calls are being recorded by MiaRec.
5. Verify that the Remote Worker endpoints can place calls to the endpoints at the enterprise site and that the calls are being recorded by MiaRec.
6. Verify that the endpoints at the enterprise can place calls to the Remote Worker endpoints and that the calls are being recorded by MiaRec.
7. Verify that the Remote Worker endpoints can place calls to other Remote Workers and that the calls are being recorded by MiaRec.
8. Verify that the Remote Worker endpoints can place calls to the Service Provider's SIP Trunk and that the calls are being recorded by MiaRec.
9. Verify that the Remote Worker endpoints can receive calls from the Service Provider's SIP Trunk and that the calls are being recorded by MiaRec.
10. Should issues arise with the call recording, use the Avaya SBCE trace facility to check that the OPTIONS requests sent from the Avaya SBCE to the MiaRec server are receiving a response.

10. Conclusion

These Application Notes describe the configuration necessary to record calls using MiaRec in the Avaya Aura Platform consisting of Avaya Aura ® Communication Manager R7.0.1, Avaya Aura ® Session Manager 7.0.1 and Avaya Session Border Controller for Enterprise R7.1. The MiaRec call recording and quality management solutions help businesses to easily record, analyze and access important interactions to meet regulatory compliance requirements, enhance customer service and increase agent productivity. The software 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] *Migrating and Installing Avaya Appliance Virtualization Platform*, Release 7.0.1, Aug 2016

[2] *Upgrading and Migrating Avaya Aura® applications to 7.0.1 from System Manager*, Release 7.0.1, Aug2016

[3] *Deploying Avaya Aura® applications*, Release 7.0, Dec 2015

[4] *Deploying Avaya Aura® Communication Manager*, Oct 2016

[5] *Administering Avaya Aura® Communication Manager*, Release 7.0.1, May 2016

[6] *Deploying Avaya Aura® System Manager*, Release 7.0.1 Aug 2016

[7] *Upgrading Avaya Aura® Communication Manager*, Release 7.0.1, Oct 2016

[8] *Upgrading Avaya Aura® System Manager to Release 7.0.1*, Aug 2016.

[9] *Administering Avaya Aura® System Manager for Release 7.0.1*, Nov 2016

[10] *Deploying Avaya Aura® Session Manager*, Release 7.0.1 Nov 2016

[11] *Upgrading Avaya Aura® Session Manager* Release 7.0.1, Nov 2016

[12] *Administering Avaya Aura® Session Manager* Release 7.0.1, May 2016

[13] *Deploying Avaya Session Border Controller for Enterprise*, Release 7.0, August 2015

[14] *Upgrading Avaya Session Border Controller for Enterprise*, Release 7.0, August 2015

[15] *Administering Avaya Session Border Controller for Enterprise*, Release 7.0, Jan 2016

[16] *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.