Avaya Solution & Interoperability Test Lab

Application Notes for AudioCodes Mediant 500Li Analog Gateway with Avaya Aura® Communication Manager, Avaya Aura® Session Manager, and Avaya Session Border Controller for Enterprise – Issue 1.0

Abstract

These Application Notes contain instructions for configuring AudioCodes Mediant 500Li Analog Gateway R7.20AN in the Avaya enterprise and outside the enterprise in a remote worker configuration. The enterprise environment incorporates Avaya Session Border Controller for Enterprise 8.1, Avaya Aura® Communication Manager 8.1, and Avaya Aura® Session Manager 8.1. Compliance testing was conducted to verify interoperability. The AudioCodes Mediant 500Li GE/GE/8FXS 8 Port FXS Analog Gateway was used for testing.

Readers should pay attention to **Section 2**, in particular the scope of testing as outlined in **Section 2.1** as well as the 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.

Table of Contents

1. Introduction

The AudioCodes Mediant 500Li Analog Gateway (Mediant 500Li) implements voice technology that connects analog telephones and fax machines to IP-based enterprise PBX systems. In the compliance test, AudioCodes Mediant 500Li provided SIP access to analog devices to verify interoperability within an enterprise Avaya Aura® IP Telephony Environment. AudioCodes Mediant 500Li registers to Avaya Aura® Session Manager when located within the enterprise. AudioCodes Mediant 500Li registers to Avaya Aura® Session Manager through Avaya Session Border Controller for Enterprise (SBCE) when located outside the enterprise as a Remote Worker. The AudioCodes Mediant 500Li GE/GE/8FXS 8 Port FXS Analog Gateway was used for testing.

2. General Test Approach and Test Results

Interoperability compliance testing focused on verifying various inbound and outbound call flows between AudioCodes Mediant 500Li, Communication Manager, Session Manager, and Session Border Controller for Enterprise

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.

Avaya recommends our customers implement Avaya solutions using appropriate security and encryption capabilities enabled by our products. The testing referenced in these DevConnect Application Notes included the enablement of supported encryption capabilities in the Avaya products. Readers should consult the appropriate Avaya product documentation for further information regarding security and encryption capabilities supported by those Avaya products.

Support for these security and encryption capabilities in any non-Avaya solution component is the responsibility of each individual vendor. Readers should consult the appropriate vendor-supplied product documentation for more information regarding those products.

For the testing associated with these Application Notes, the interface between Avaya systems and AudioCodes Mediant 500Li employed TLS connectivity with SRTP.

2.1. Interoperability Compliance Testing

AudioCodes Mediant 500Li registered analog lines as SIP users on Session Manager. AudioCodes Mediant 500Li registered analog lines as SIP users to Session Manager through Avaya Session Border Controller for Enterprise as a Remote Worker. SIP, TLS, and SRTP were utilized during this test effort. Note that compliance testing only verified the analog lines provided by the gateway and no other features on Mediant 500Li. The following analog line features and functionalities were covered during compliance testing:

- Incoming calls from Avaya SIP/H.323 endpoints and PSTN to AudioCodes Mediant 500Li (with the Mediant 500Li located in the enterprise as well as in a remote worker location)
- Outgoing calls from AudioCodes Mediant 500Li to Avaya SIP/H.323 endpoints and PSTN (with the Mediant 500Li located in the enterprise as well as in a remote worker location)
- SIP signaling using TLS
- Voice codecs G.711U, G.711A and G.729AB using SRTP
- Incoming and outgoing faxes using encrypted G.711 (in pass-through mode)
- DTMF tone transmission with RFC2833
- Calls using various Avaya endpoints, including analog, digital, H.323, and SIP
- Basic features including Hold/Resume, DTMF transmission, Voicemail with Message Waiting Indicator (MWI)

2.2. Test Results

All test cases passed. The following observations were noted during the compliance testing:

- Mediant 500Li does not support encrypted T.38 FAX. Testing verified encrypted G.711 Fax in pass-through mode.
- TLS/SRTP testing employed a mandatory media encryption configuration. Mediant 500Li does not currently support attribute capability negotiation as defined in RFC5939. To use Mediant 500Li preferable media encryption, message manipulations would have to be configured to remove the acap: attribute from the SRTP line.
- Voicemail MWI was verified using stutter tone message waiting notification.

2.3. Support

Technical support for AudioCodes Mediant 500Li Analog Gateway can be obtained through the following:

- Phone:
  - Americas: +1-732-652-1085 or 1-800-735-4588
  - Rest of the World: 800-44422444 or 972-3-9764343
- Web: https://services.audiocodes.com
- E-Mail: support@audiocodes.com

3. Reference Configuration

AudioCodes Mediant 500Li is shown below in the Enterprise or configured as Remote Worker.

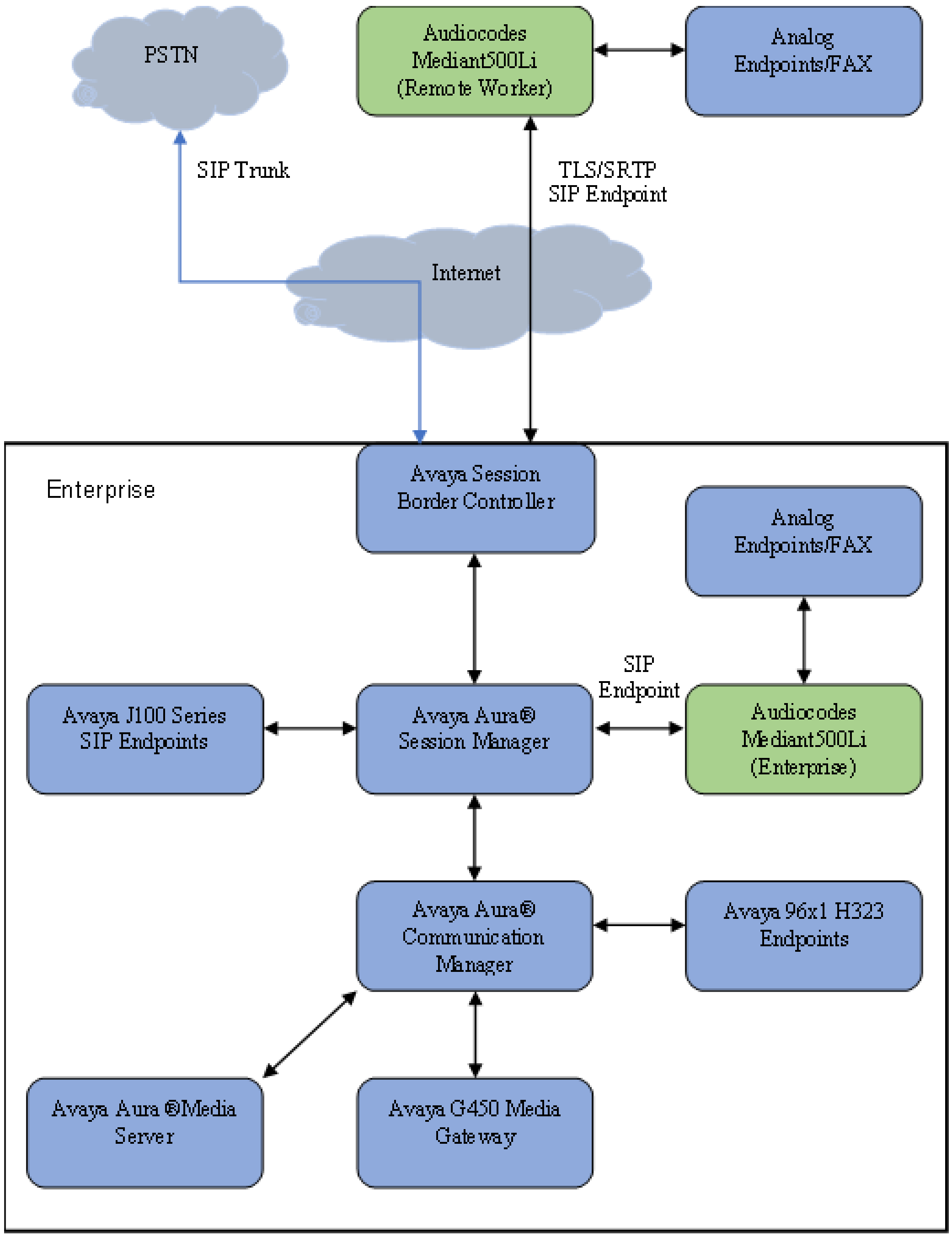


Figure 1: AudioCodes Mediant 500Li Analog Gateway Configuration

4. Equipment and Software Validated

The following equipment and software were used for interoperability testing:

Equipment/Software	Release/Version
Avaya Aura® Communication Manager running on Virtual Machine	8.1.3.0.1.890.26685
Avaya Aura® Session Manager running on Virtual Machine	8.1.3.0.813014
Avaya Aura® System Manager running on Virtual Machine	8.1.3.0.813014
Avaya Session Border Controller for Enterprise running on Virtual Machine	8.1.2.0-19809
Avaya G450 Media Gateway	41.34.1
Avaya Aura® Media Server	8.0.2.163
Avaya 1408 Digital Phone	NA
Avaya 6220 Analog Phone	NA
Avaya 9641G H.323 Deskphone	6.8.3.04
Avaya J179 SIP Deskphone	4.0.9.0.4
AudioCodes Mediant 500Li Analog Gateway	7.20AN.456.539

5. Configure Avaya Aura® Communication Manager

This section provides steps for configuring Communication Manager. All configuration for Communication Manager is done through System Access Terminal (SAT).

5.1. Verify Avaya Aura® Communication Manager License

Use the **display system-parameters customer-options** command to verify options.

On **Page 1**, verify that the number of OPS stations allowed in the system is sufficient for the number of Mediant 500Li analog lines (as SIP endpoints) that will be deployed.

```
display system-parameters customer-options                      Page   1 of  12
                                OPTIONAL FEATURES

     G3 Version: V18                             Software Package: Enterprise
       Location: 2                                System ID (SID): 1
       Platform: 28                               Module ID (MID): 1

                                                             USED
                                Platform Maximum Ports: 48000    111
                                      Maximum Stations: 36000     86
                              Maximum XMOBILE Stations: 36000      0
                    Maximum Off-PBX Telephones - EC500: 41000      0
                    Maximum Off-PBX Telephones -   OPS: 41000     51
                    Maximum Off-PBX Telephones - PBFMC: 41000      0
                    Maximum Off-PBX Telephones - PVFMC: 41000      0
                    Maximum Off-PBX Telephones - SCCAN:     0      0
                         Maximum Survivable Processors: 313   0

        (NOTE: You must logoff & login to effect the permission changes.)
```

On **Page 2,** verify that there is sufficient capacity for SIP trunks by comparing **Maximum Administered SIP Trunks** field with corresponding **USED** column field.

```
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:  2400  6
             Maximum Administered Remote Office Trunks: 12000  0
Maximum Concurrently Registered Remote Office Stations:  2400  0
              Maximum Concurrently Registered IP eCons:  128   0
         Max Concur Reg Unauthenticated H.323 Stations:   100  0
                        Maximum Video Capable Stations:  36000 2
                   Maximum Video Capable IP Softphones:  2400 23
                       Maximum Administered SIP Trunks: 12000 10
  Maximum Administered Ad-hoc Video Conferencing Ports: 12000  0
   Maximum Number of DS1 Boards with Echo Cancellation: 688  0





        (NOTE: You must logoff & login to effect the permission changes.)
```

On **Page 5,** verify **Media Encryption Over IP** 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? y
                  Enhanced EC500? y       ISDN/SIP Network Call Redirection? y
    Enterprise Survivable Server? n                          ISDN-BRI Trunks? y
       Enterprise Wide Licensing? n                                 ISDN-PRI? y
              ESS Administration? y               Local Survivable Processor? n
          Extended Cvg/Fwd Admin? y                     Malicious Call Trace? y
     External Device Alarm Admin? y                 Media Encryption Over IP? y
  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
        (NOTE: You must logoff & login to effect the permission changes.)
```

5.2. Administer IP Network Region

Use the **change ip-network-region *n*** command to configure a network region, where ***n*** is an existing network region.

5.2.1. IP Network Region for Voice and Fax Calls

Configure this network region as follows:

- Set **Name** to an appropriate value
- Set **Location** to **1**
- Set **Codec Set** to that administered in **Section 5.3**, e.g., **1**
- Set **Intra-region IP-IP Direct Audio** to **yes**
- Set **Inter-region IP-IP Direct Audio** to **yes**
- Enter an **Authoritative Domain**, e.g., **avaya.com**

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

5.3. Administer IP Codec Set

Use the **change ip-codec-set *n*** command to configure IP codec set, where ***n*** is the codec set used in **Section 5.2.1** IP Network Region administration.

5.3.1. IP Codec set for Voice and Fax Calls

Configure this codec set as follows, on **Page 1**:

- Set **Audio Codec 1, 2 and 3** to **G.711MU, G.711A, G.729AB** respectively
- Set **Media Encryption 1:** to **1-srtp-aescm128-hmac80 and Media Encryption 2: to 2-srtp-aescm128-hmac32**
- Set **Encrypted SRTCP** to **enforce-unenc-srtcp**

Note: G.711MU, G.711A and G.729AB codecs were used during compliance testing

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

                          IP MEDIA PARAMETERS
    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.729AB          n          2        20
 4:
 5:
 6:
 7:

     Media Encryption                  Encrypted SRTCP: enforce-unenc-srtcp
 1: 1-srtp-aescm128-hmac80
 2: 2-srtp-aescm128-hmac32
 3:
 4:
 5:
```

On **Page 2**:

FAX settings allow the use of encrypted G.711 Fax Mode. Encrypted T.38 is not supported as noted in **Section 2.2**.

- Set **FAX Mode** to **pass-through**

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

                          IP MEDIA PARAMETERS

                              Allow Direct-IP Multimedia? y
               Maximum Call Rate for Direct-IP Multimedia: 10240:Kbits
      Maximum Call Rate for Priority Direct-IP Multimedia: 10240:Kbits

                                                                  Packet
                          Mode                    Redundancy      Size(ms)
    FAX                   pass-through             0          ECM: y
    Modem                 off                      0
    TDD/TTY               US                       3
    H.323 Clear-channel   n                        0
    SIP 64K Data          n                        0              20
```

5.4. Administer IP Node Names

Use the **change node-names ip** command to add an entry for Session Manager. For compliance testing, **sm81** and **10.64.110.212** entry was added.

```
change node-names ip
                                  IP NODE NAMES
    Name              IP Address
aes81                 10.64.110.215
aes811                10.64.110.209
ams81                 10.64.110.214
aura_cms18            10.64.110.20
cms19                 10.64.110.225
default               0.0.0.0
procr                 10.64.110.213
procr6                ::
remotecms191          10.64.110.226
sm81                  10.64.110.212


( 10 of 10   administered node-names were displayed )
Use 'list node-names' command to see all the administered node-names
Use 'change node-names ip xxx' to change a node-name 'xxx' or add a node-name
```

5.5. Administer SIP Signaling Group

Use the **add signaling-group *n*** command to add a new signaling group, where ***n*** is an available signaling group number.

5.5.1. Signaling Group for Voice and Fax Calls

Configure this signaling group as follows:

- Set **Group Type** to **sip**
- Set **Transport Method** to **tls**
- Set **Near-end Node Name** to **procr**
- Set **Near-end Listen Port** to **5061**
- Set **Far-end Node Name** to the configured Session Manager name in **Section 5.4** e.g., **sm81**
- Set **Far-end Listen Port** to **5061**
- Set **Far-end Network region** to the configured IP network region in **Section 5.2.1** e.g., **1**
- Enter a **Far-end Domain,** e.g., **avaya.com**
- Set **Direct IP-IP Audio Connections** to **y**
- Set **Initial IP-IP Direct Media** to **y**
- Set **DTMF over IP** to **rtp-payload**

Communication Manager supports DTMF transmission using RFC 2833.

```
add signaling-group 1                                       Page   1 of   3
                                SIGNALING GROUP

 Group Number: 1              Group Type: sip
  IMS Enabled? n        Transport Method: tls
        Q-SIP? n
     IP Video? y          Priority Video? n        Enforce SIPS URI for SRTP? n
  Peer Detection Enabled? y  Peer Server: SM                        Clustered? n
 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: sm81
 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): 65                     IP Audio Hairpinning? n
         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

Use the **add trunk-group *n*** command to add a trunk group, where ***n*** is an available trunk group number.

5.6.1. Trunk Group for Voice and Fax Calls

Configure this trunk group as follows, on **Page 1**:

- Set **Group Type** to **sip**
- Enter an appropriate **Group Name** e.g., **SM Trunk 1**
- Enter a valid **TAC** e.g., **101**
- Set **Service Type** to **tie**
- Enter **Signaling Group** value to the signaling group configured in **Section 5.5.1** e.g., **1**
- Enter a desired number in **Number of Members** field

```
change trunk-group 1                                            Page   1 of  5
                                TRUNK GROUP

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

On **Page 3**:

- Set **Numbering Format** to **private**

```
TRUNK FEATURES
          ACA Assignment? n            Measured: both
                                                   Maintenance Tests? y



   Suppress # Outpulsing? n  Numbering Format: private
                                              UUI Treatment: shared
                                            Maximum Size of UUI Contents: 128
                                               Replace Restricted Numbers? n
                                              Replace Unavailable Numbers? n


                                Modify Tandem Calling Number: no
               Send UCID? y



 Show ANSWERED BY on Display? y

 DSN Term? n                                                Hold/Unhold
```

5.7. Administer Route Pattern

Use the **change route-pattern *n*** command to configure a route pattern, where ***n*** is an available route pattern.
Configure this route pattern as follows:

- Type an appropriate name in **Pattern Name** field
- For line 1, set **Grp No** to the trunk group configured in **Section 5.6.1** e.g., **1**
- For line 1, set **FRL** to **0**
- For line 1, set **Numbering Format** to **lev0-pvt**

```
change route-pattern 1                                          Page   1 of   3
                    Pattern Number: 1      Pattern Name: main
    SCCAN? n    Secure SIP? y     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
```

5.8. Administer Private Numbering

Use the **change private-numbering 1** command to define the calling party number to send to Session Manager and configure private numbering as follows. For compliance testing, 5-digit extensions beginning with 7 are routed over trunk group 1 which resulted in a 5-digit calling party number.

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

Ext Ext               Trk        Private          Total
Len Code              Grp(s)     Prefix           Len
 5  5                                             5  Total Administered: 2
 5  7                                             5     Maximum Entries: 540
```

5.9. Administer AAR Analysis

Use the **change aar analysis *n*** command to configure routing for extensions starting with ***n***. For compliance testing, extensions starting with **701** were used for both voice and fax calls.

- Set **Dialed String** to starting digits of extensions that will be used e.g., **701**
- Set **Min** and **Max** to **5** for 5-digit extensions
- Set **Route Pattern** to pattern configured in **Section 5.7**, e.g., **1**
- Set **Call Type** to **lev0**

Note: The extension range used in this step needs an entry to the dial plan.

```
change aar analysis 7
                            AAR DIGIT ANALYSIS TABLE
                                   Location: all           Percent Full: 0

          Dialed           Total     Route     Call   Node  ANI
          String          Min  Max  Pattern    Type   Num   Reqd
    701                   5    5     1         lev0         n
```

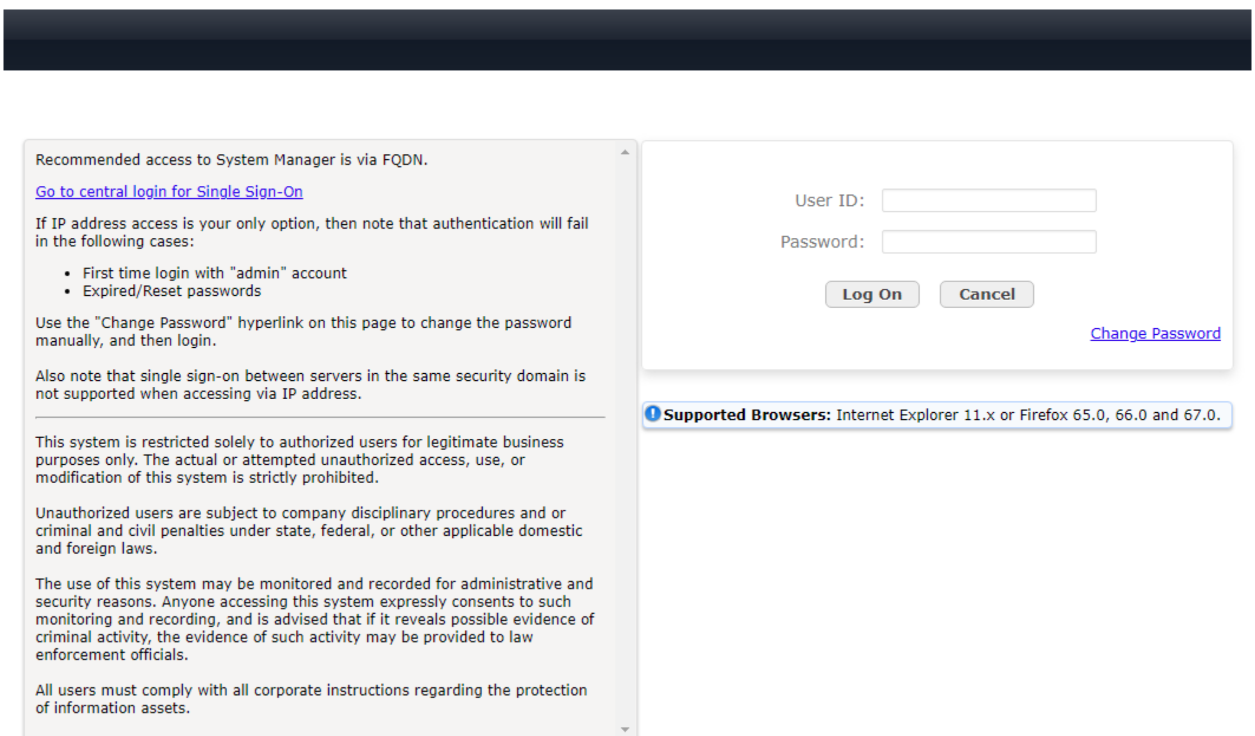
6. Configure Avaya Aura® Session Manager

This section provides the procedures for configuring Session Manager. It is assumed that the basic configuration is already in place. This section discusses the following areas:

- Add SIP Domain
- Add Location
- Add Adaptations
- Add SIP Entities
- Add Entity Links
- Add Routing Policy
- Add Dial Patterns
- Add Users

Note: The sections that reference configuration related to the Avaya Session Border Controller for Enterprise are only needed if the Median500Li is located outside the enterprise environment and registering through the internet as remote workers.

Access Session Manager Administration web interface by entering **http://<ip-address>/SMGR** in a web browser, where <ip-address> is the IP address of System Manager. Log in with the appropriate credentials.

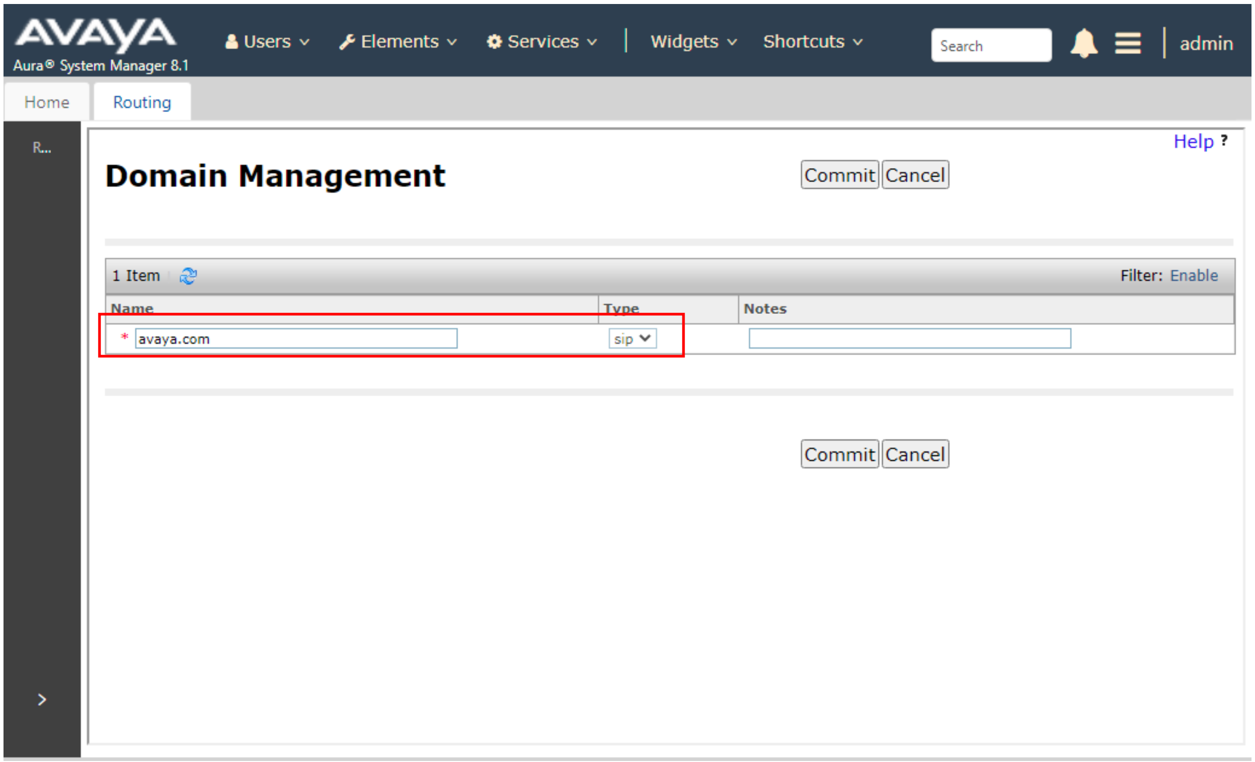


6.1. Add SIP Domain

Navigate to **Elements → Routing → Domains**, click on **New** button (not shown) and configure as follows:

- In **Name** field type in a domain (authoritative domain used in **Section 5.2.1**) e.g., **avaya.com**
- Set **Type** to **sip**

Click **Commit** to save changes.



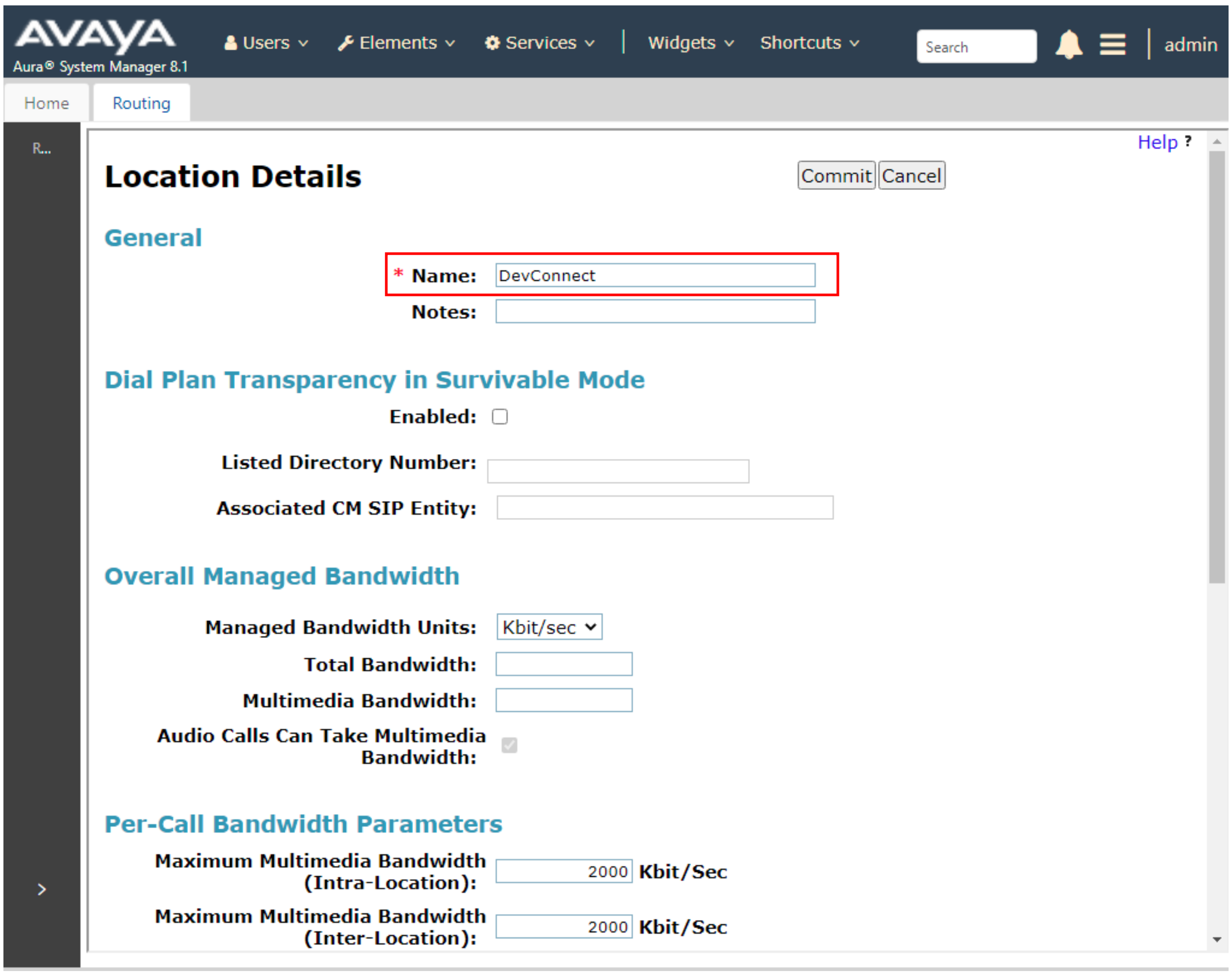
6.2. Add Location

Navigate to **Elements → Routing → Location**, click on the **New** button (not shown) and configure as follows.

Under **General**:

- Type in a descriptive **Name** e.g., **DevConnect**
- Under **Location Pattern** click on **Add** (not shown)
- Type in **IP Address Pattern** for applicable subnets, e.g., **10.64.***

Click **Commit** to save changes.



6.3. Add Adaptations

Note: The configuration in this section is only needed if Mediant 500Li is located outside the enterprise environment and registering through the internet as remote workers.

Add an adaptation to convert incoming domains from an IP address to the pertinent domain. Select **Adaptations → Adaptations** from the left pane and click **New** (not shown) to add a new adaptation for IPC.

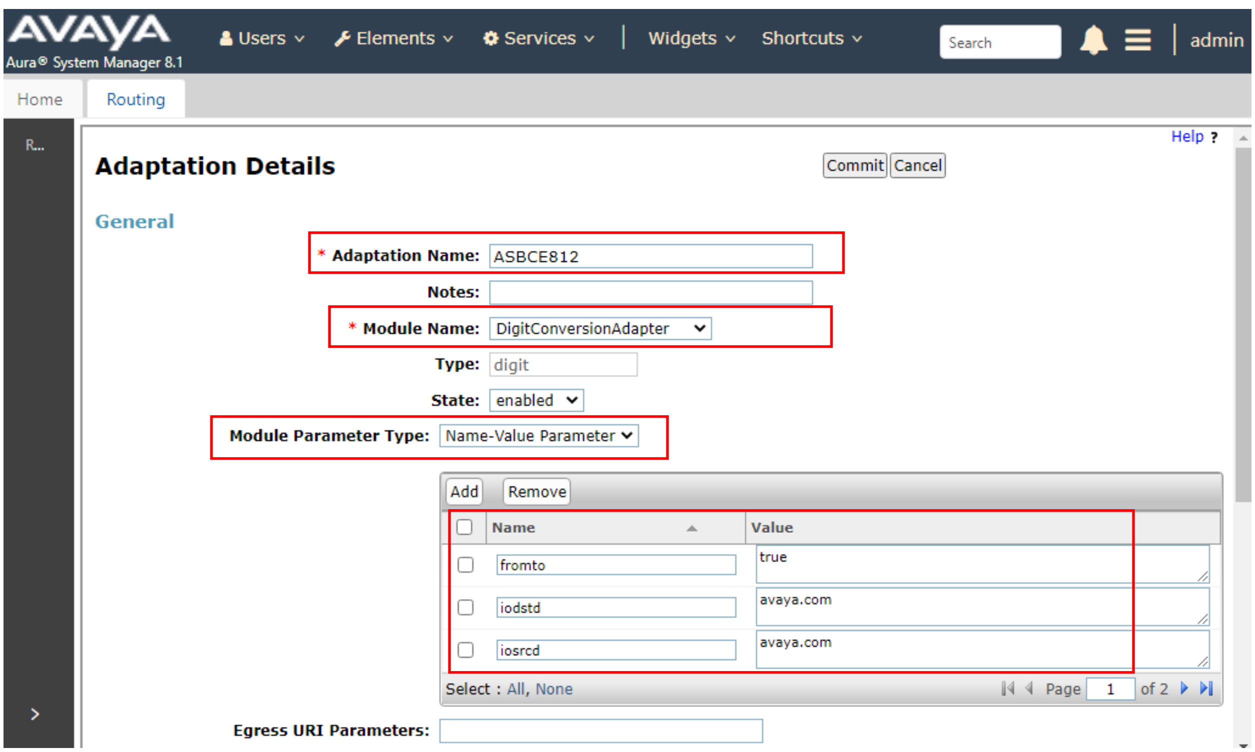
The **Adaptation Details** screen is displayed. Enter the following values for the specified fields:

- **Adaptation Name:** A descriptive name, e.g., **ASBCE812**.
- **Module Name:** **DigitConversionAdapter**
- **Module Parameter Type:** **Name-Value Parameter**

Click **Add** to add the adaptation name value pairs as specified:

- **fromto:** **true**
- **iodstd:** The pertinent domain name, e.g., **avaya.com**
- **iosrcd:** The pertinent domain name, e.g., **avaya.com**
- **odstd:** The pertinent domain name, e.g., **avaya.com** (not shown)
- **osrcd:** The pertinent domain name, e.g., **avaya.com** (not shown)

Click **Commit** to save changes.



NOTE: SIP message manipulation to modify the incoming domain can be done through AudioCodes administration. Interoperability testing employed this adaptation too.

6.4. Add SIP Entities

Add a SIP entity for Communication Manager and for Session Border Controller for Enterprise.

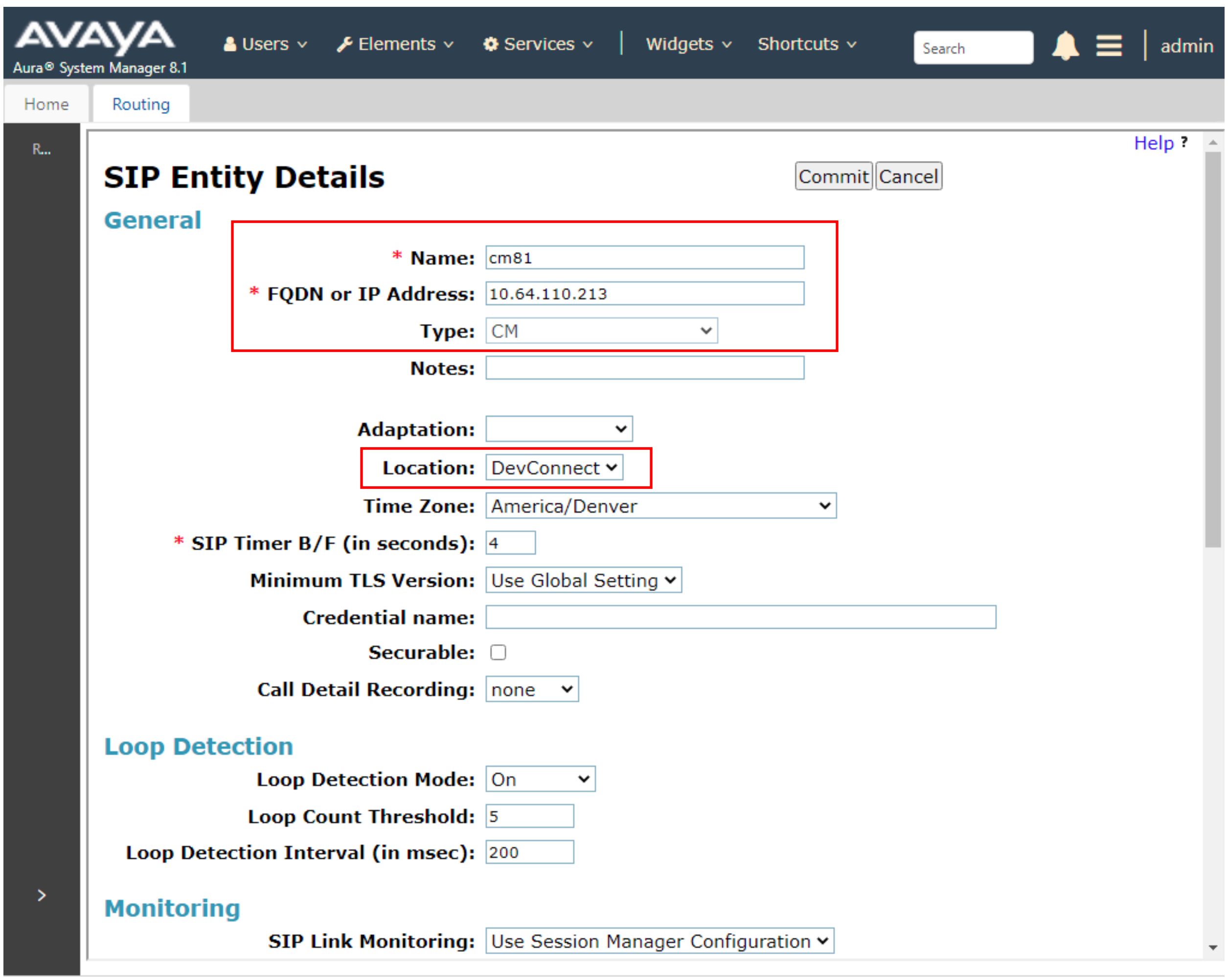
6.4.1. Communication Manager

Add Communication Manager as a SIP Entity. Navigate to **Elements → Routing → SIP Entities,** click on **New** (not shown) and configure as follows:

- Type in a descriptive name in **Name** field, e.g., **cm81**
- Type in the IP address or FQDN of Communication Manager in **FQDN or IP Address** field, e.g., **10.64.110.213**
- Set **Type** to **CM**
- Set **Location** to the location configured in **Section 6.2**, e.g., **DevConnect**

Click **Commit** to save changes.

Note: It is assumed that SIP Entity for Session Manager has been already configured.



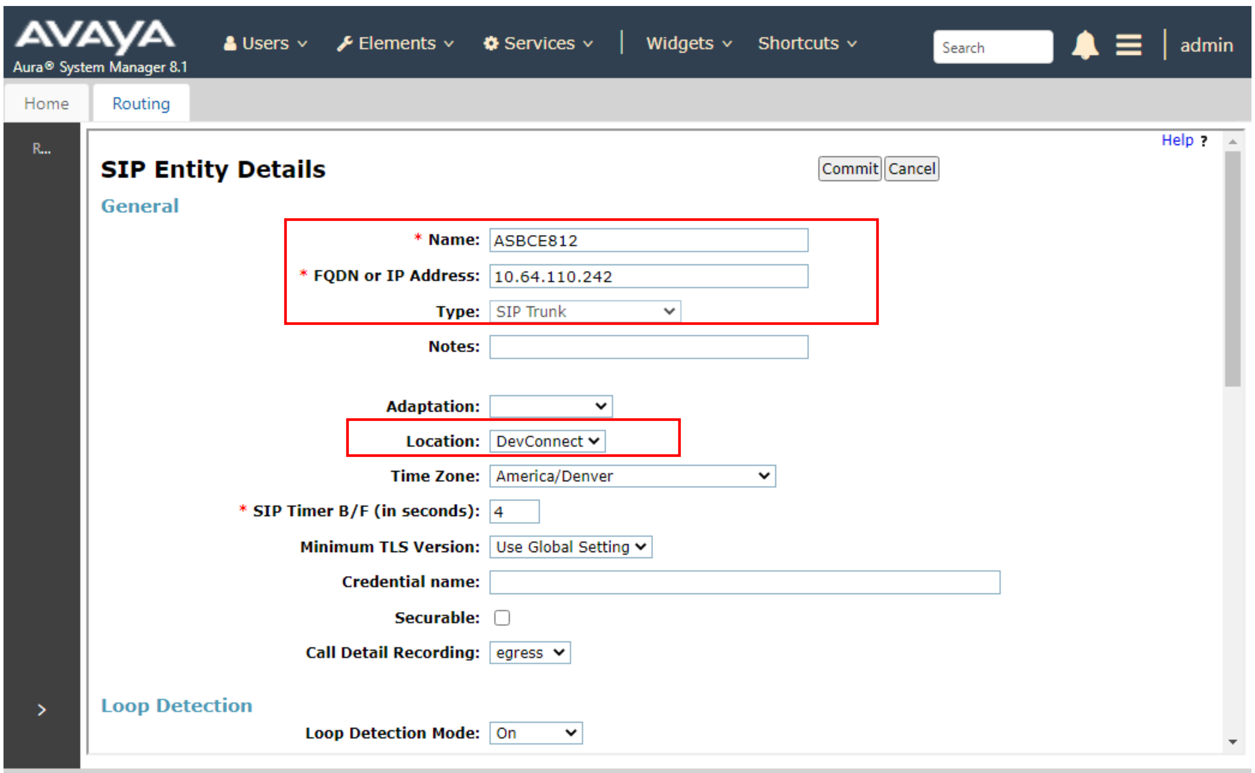
6.4.2. Avaya Session Border Controller for Enterprise

Note: The configuration in this section is only needed if Mediant 500Li is located outside the enterprise environment and registering through the internet.

Add Session Border Controller as a SIP Entity. Navigate to **Elements → Routing → SIP Entities,** click on **New** (not shown) and configure as follows:

- Type in a descriptive name in **Name** field, e.g., **ASBCE812**
- Type in the IP address of the internal SBCE Interface from **Section 7.2** in **FQDN or IP Address** field, e.g., **10.64.110.242**
- Set **Type** to **SIP Trunk**
- Set **Location** to the location configured in **Section 6.2**

Click **Commit** to save changes.



6.5. Add Entity Links

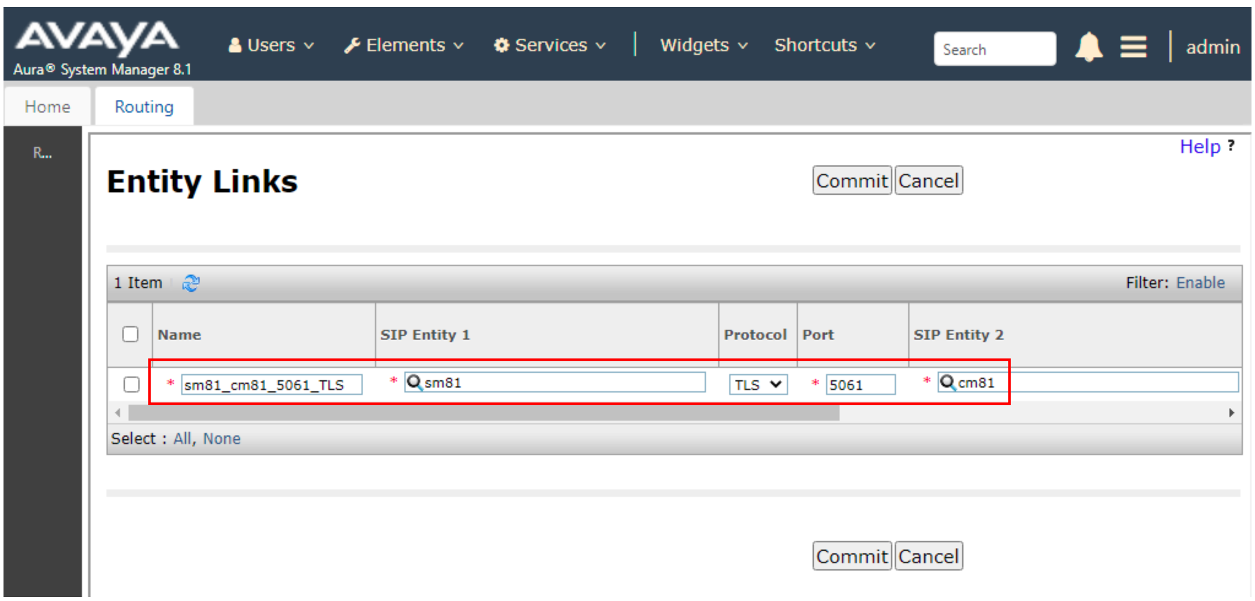
Add entity links between Communication Manager and Session Manager and between SBCE and Session Manager.

6.5.1. Communication Manager

Add an entity link between Communication Manager and Session Manager. Navigate to **Elements → Routing → Entity Links**, click on **New** (not shown) and configure as follows:

- Type in a descriptive name in **Name** field
- Set **SIP Entity 1** to the name of Session Manager SIP Entity e.g., **sm81**
- Set **SIP Entity 2** to Communication Manager SIP Entity configured in **Section 6.4.1** e.g., **cm81**
- Set **Protocol** to **TLS**
- Set **Port** to **5061**

Click **Commit** to save changes.



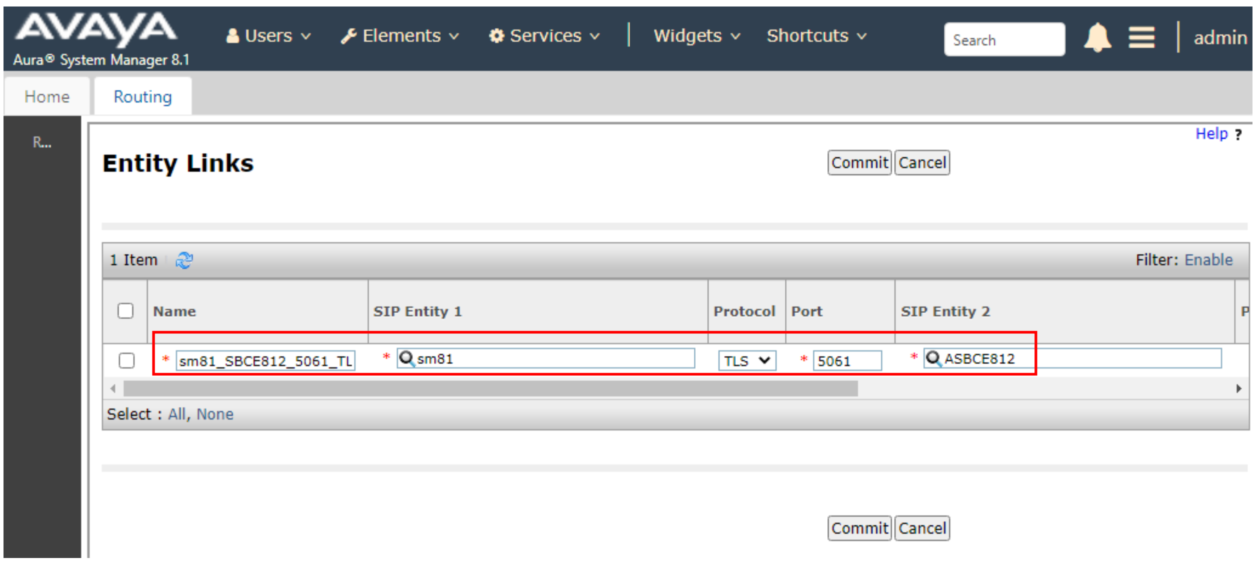
6.5.2. Session Border Controller for Enterprise

Note: The configuration in this section is only needed if Mediant 500Li is located outside the enterprise environment and registering through the internet as remote workers.

Add an Entity link between SBCE and Session Manager. Navigate to **Elements → Routing → Entity Links**, click on **New** (not shown) and configure as follows:

- Type in a descriptive name in **Name** field
- Set **SIP Entity 1** to the Session Manager Entity name e.g., **sm81**
- Set **SIP Entity 2** to the SBCE SIP Entity name from **Section 6.4.2** e.g., **ASBCE812**
- Set **Protocol** to **TLS**
- Set **Port** to **5061**

Click **Commit** to save changes.

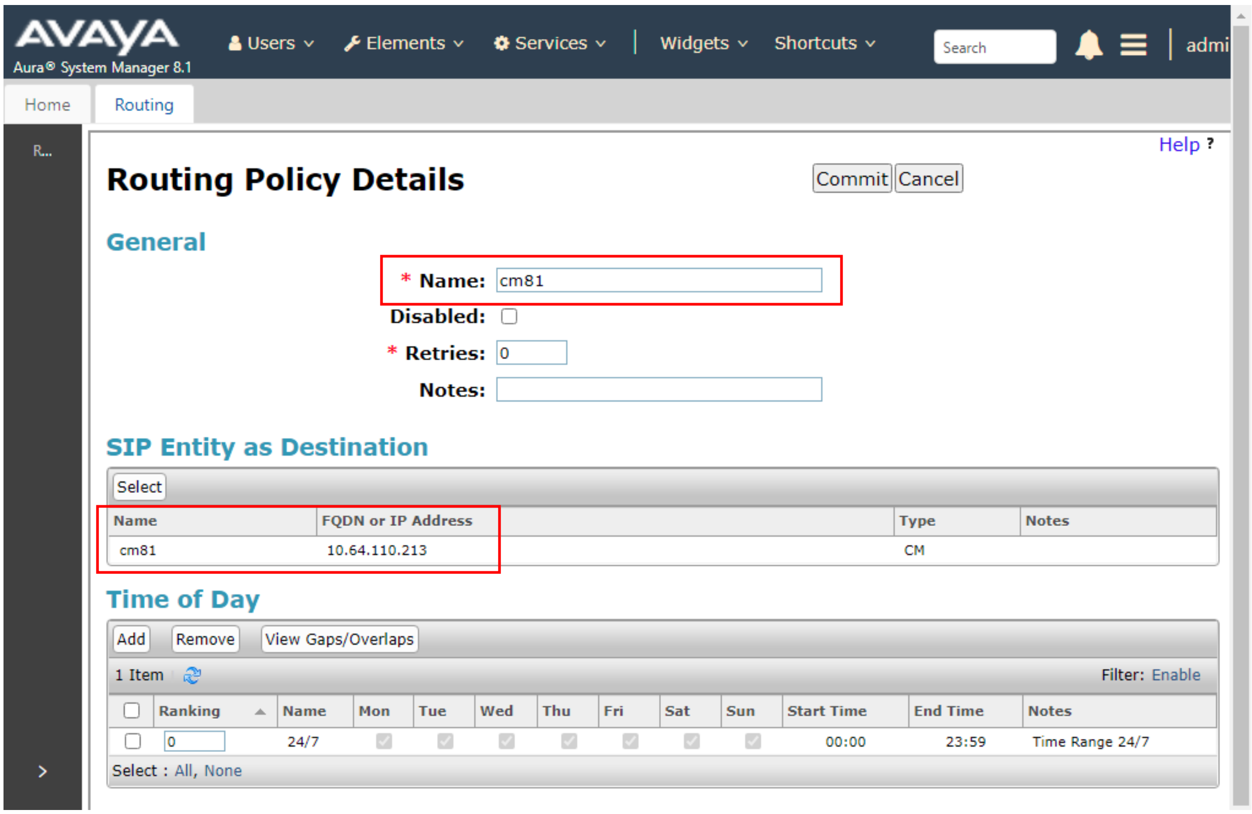


6.6. Add Routing Policy

Routing policies describe the conditions under which calls will be routed to the SIP Entities specified in **Section 6.4**. Navigate to **Elements → Routing → Routing Policies**, click on **New** (not shown) and configure as follows:

- Type in a descriptive name in **Name** field
- Under **SIP Entity as Destination**, click on **Select**. Select Communication Manager SIP entity added in **Section 6.4.1** e.g., **cm81**

Click **Commit** to save changes.



6.7. Add Dial Patterns

Dial patterns are defined to direct calls to the appropriate SIP Entity. Navigate to **Elements → Routing → Dial Patterns**, click on **New** (not shown) and configure as follows:

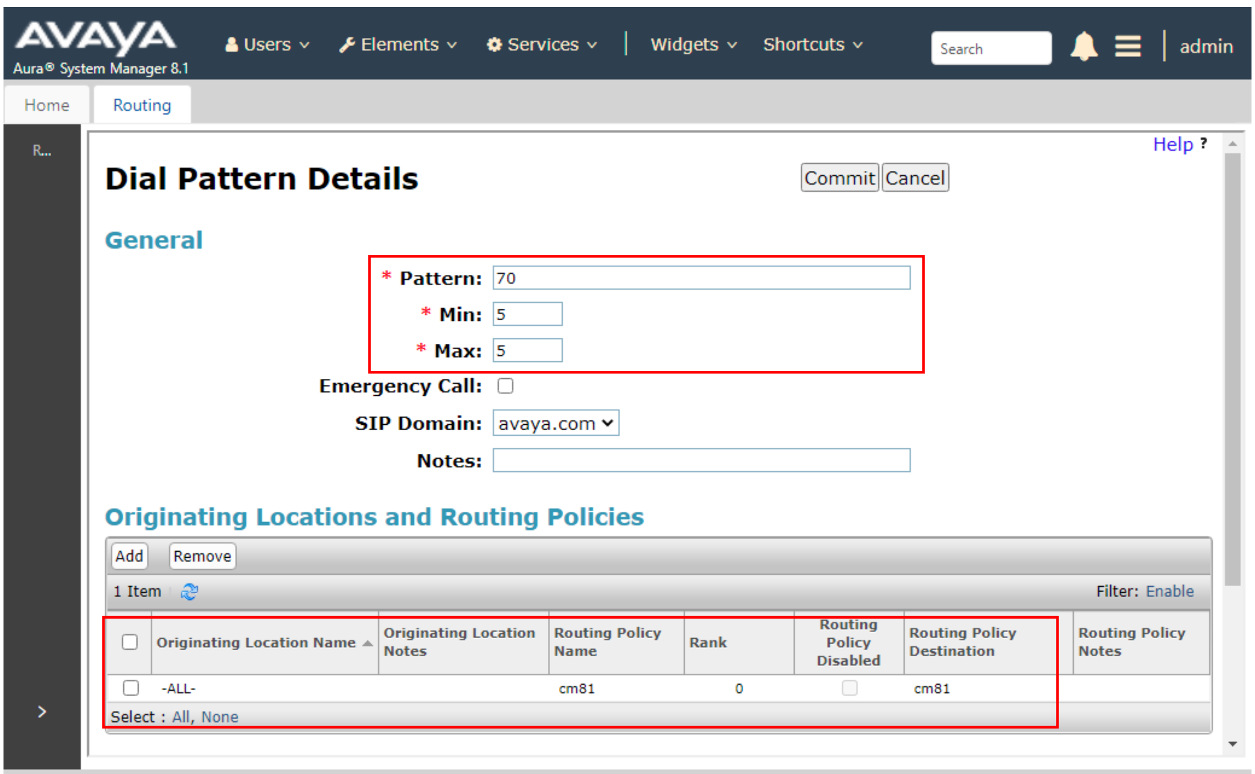
Under **General:**

- Set **Pattern** to prefix of dialed number e.g., **70**
- Set **Min** to minimum length of dialed number e.g., **5**
- Set **Max** to maximum length of dialed number e.g., **5**

Under **Originating Locations and Routing Policies:**

- Click **Add** and select the location configured in **Section 6.2** for **Originating Location**. Interoperability testing used **-ALL-** in this case
- Select the Communication Manager routing policy administered in **Section 6.6** for **Routing Policies** e.g., **cm81**

Click **Commit** to save changes.



6.8. Add Users

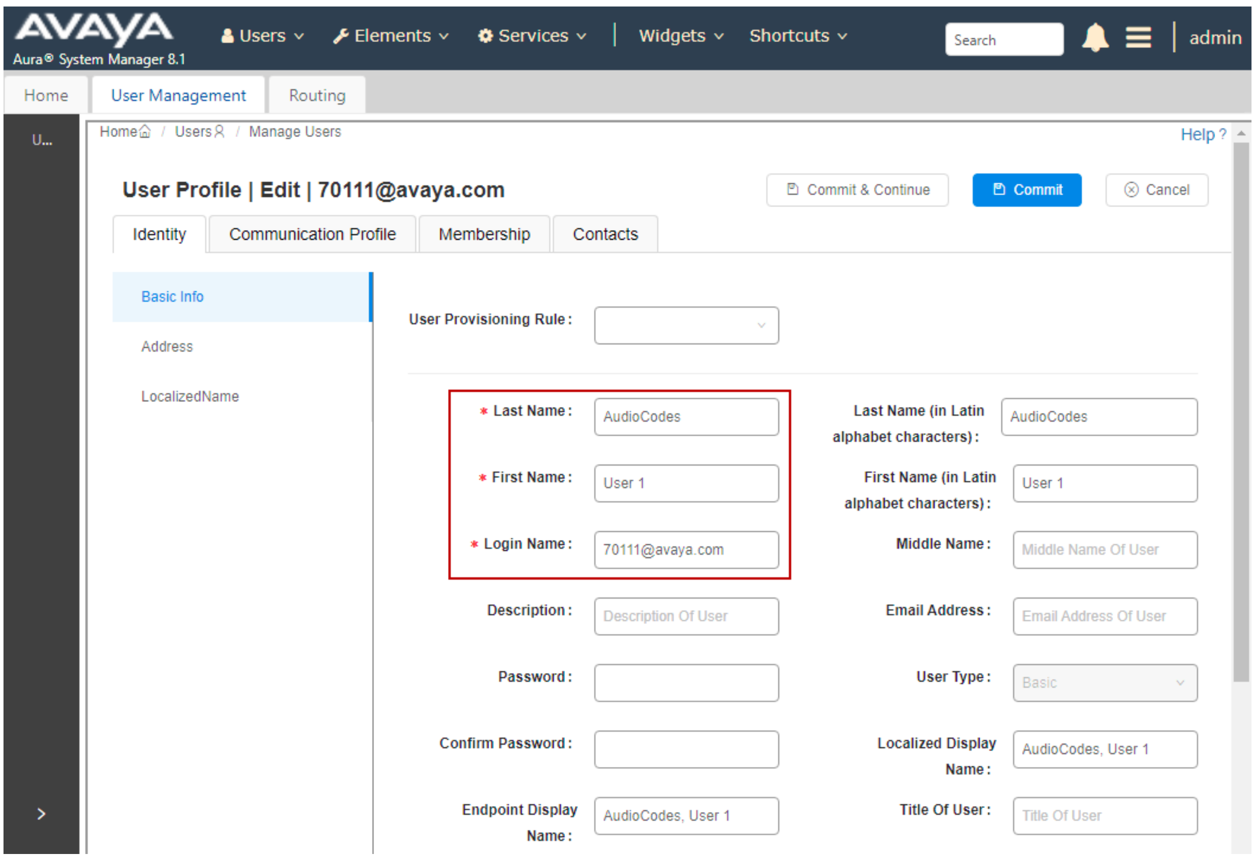
For each analog line on AudioCodes Mediant 500Li, a user needs to be added on Session Manager. Information in this section will be used by AudioCodes Mediant 500Li for registering to Session Manager.

Navigate to **Users →User Management →Manage Users** to display the **User Management** screen (not shown). Click **+ New** to add a user.

6.8.1. Identity

Enter values for the following required attributes for a new SIP user in the **New User Profile** screen:

- Enter appropriate name for **Last Name**, e.g., **AudioCodes**
- Enter appropriate name for **First Name**, e.g., **User 1**
- Enter <extension>@<sip domain> for the **Login Name, e.g., 70111@avaya.com**)



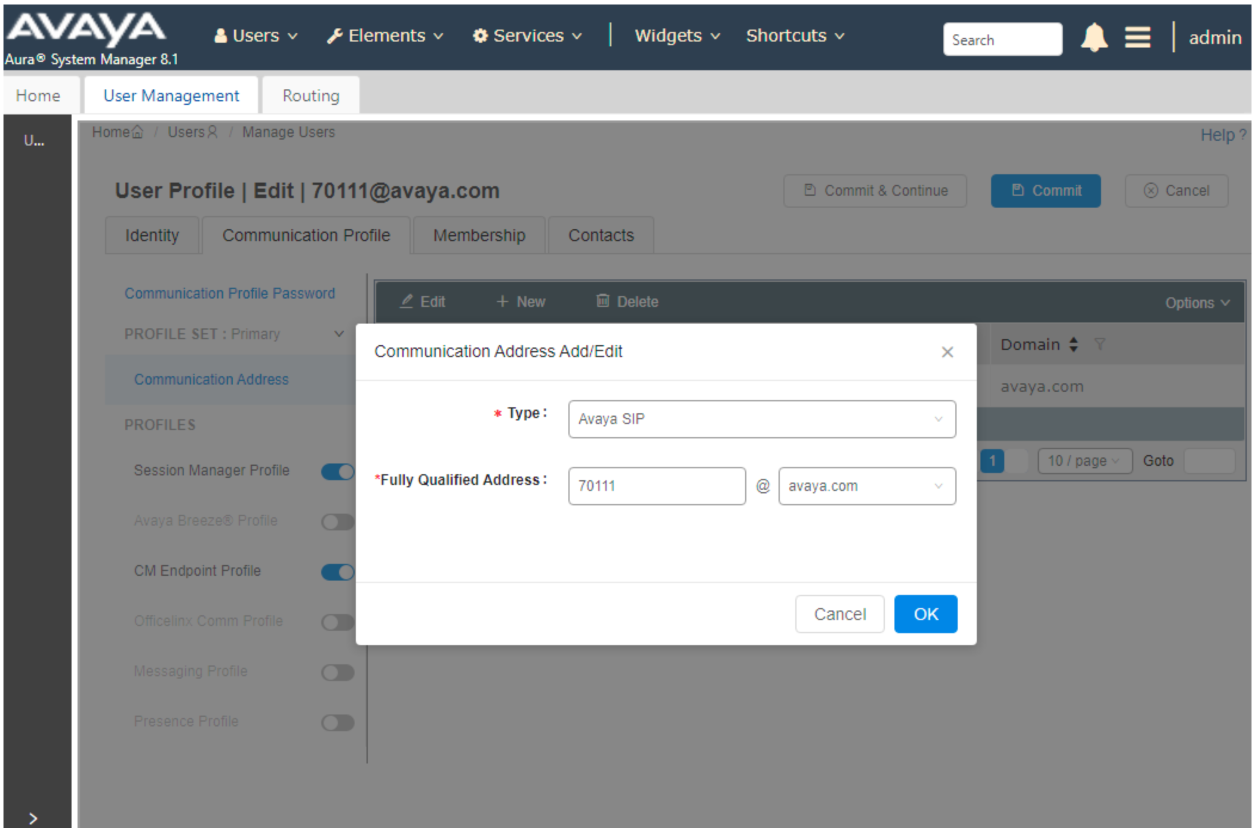
Press **Commit & Continue** after making entries or selections.

6.8.2. Communication Address

Select **Communication Address** in the left list and click **+ New** (not shown).
Enter the following attributes for the **Communication Address**:

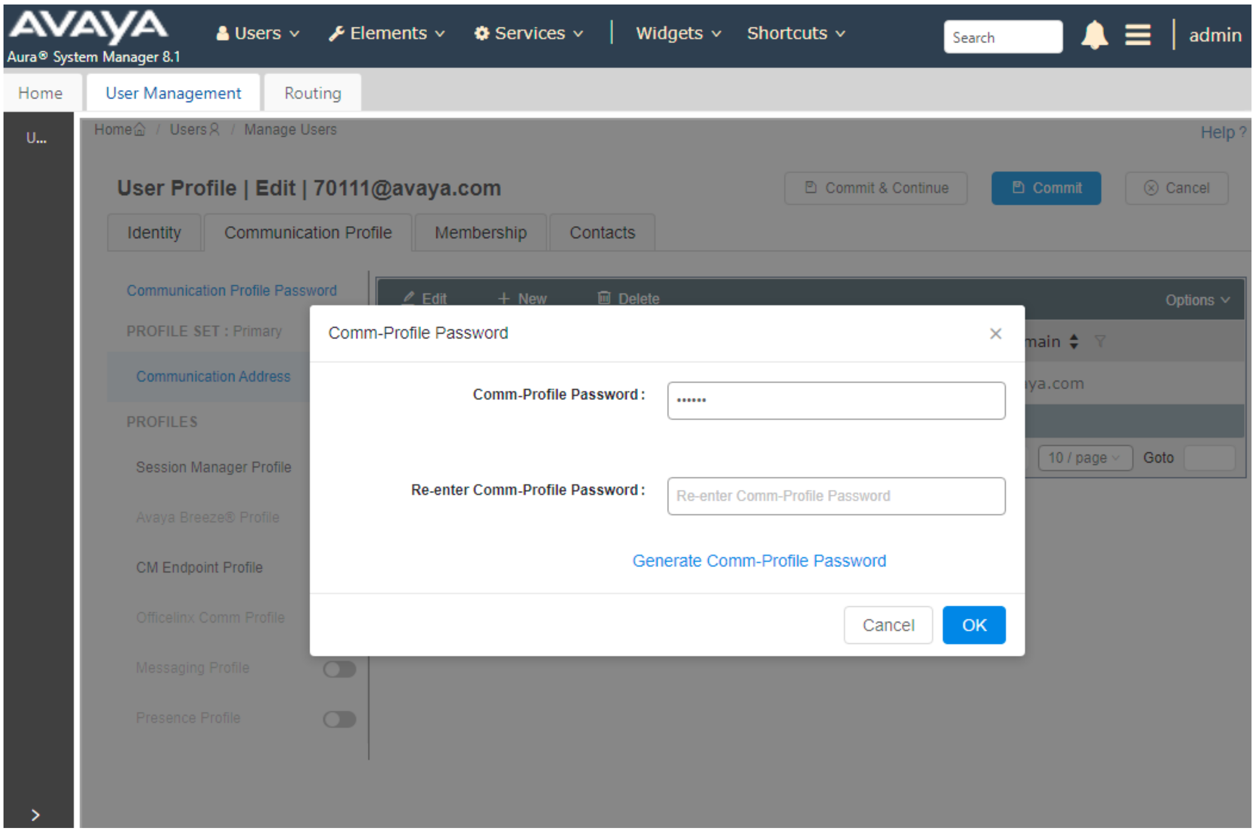
- Select **Avaya SIP** from the drop-down list for **Type**
- Enter the extension number for **Fully Qualified Address,** e.g.,**70111**
- Enter the **domain** (e.g., **avaya.com**)

Click **OK**.



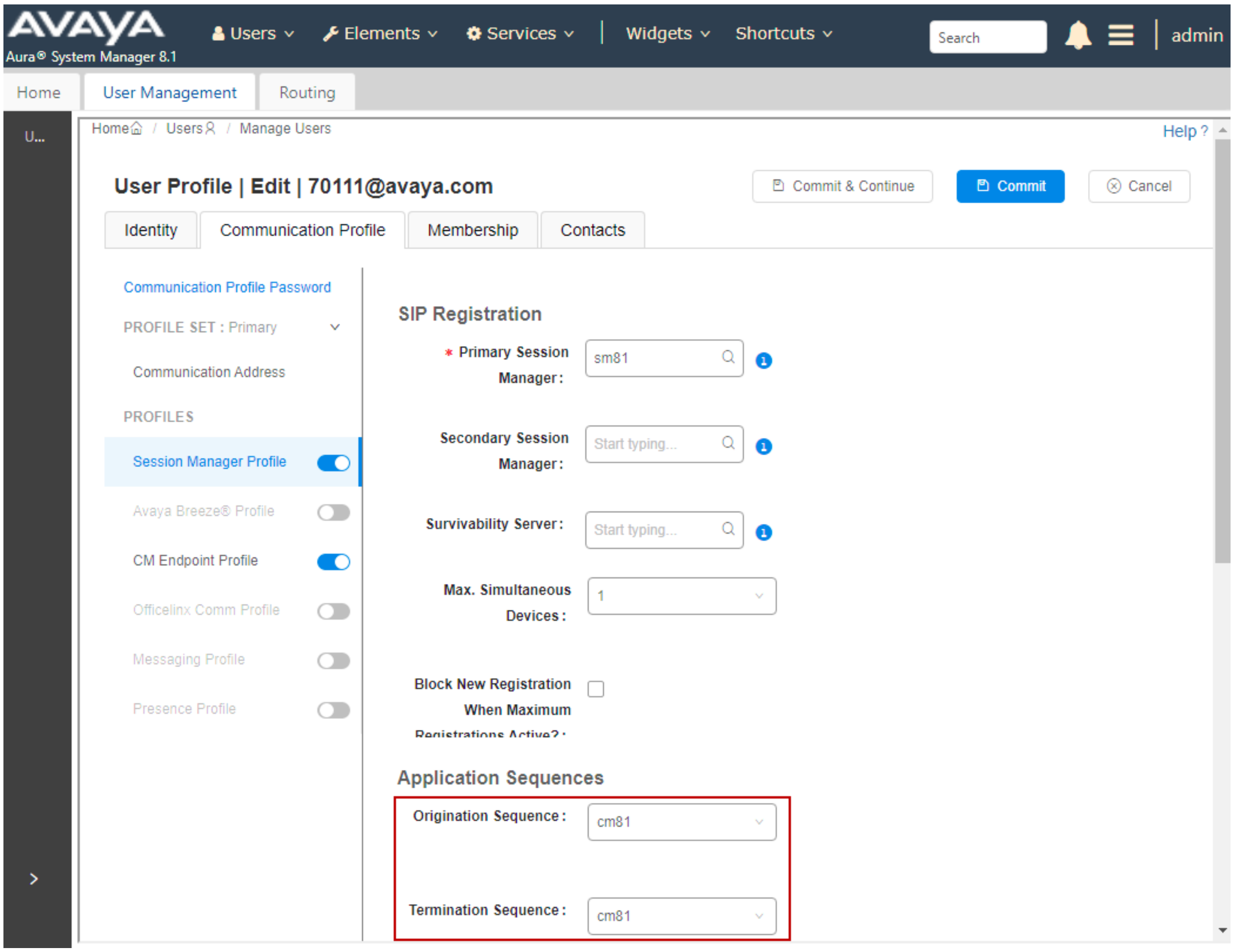
6.8.3. Communication Profile

Click the **Communication Profile** tab and in the **Comm-Profile Password** and **Re-enter Comm-Profile Password** fields, enter a numeric password. This will be used to register the device during login. Click **OK**.



6.8.4. Session Manager Profile

Click on the **Session Manager Profile** slide button. For **Primary Session Manager**, **Origination Sequence**, **Termination Sequence,** and **Home Location** (not shown), select the values corresponding to the applicable Session Manager and Communication Manager application sequences. Retain the default values in the remaining fields.

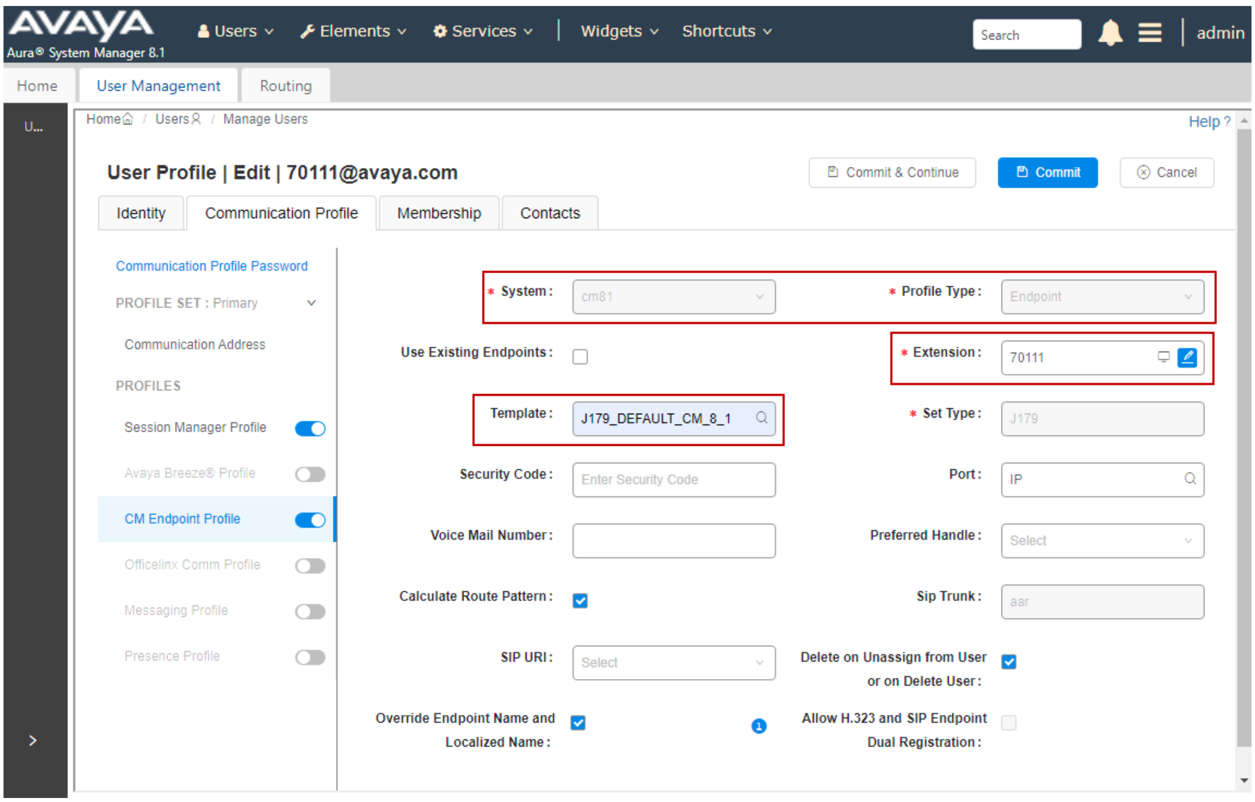


6.8.5. CM Endpoint Profile

Click on the **CM Endpoint Profile** slide button. Fill in the following fields:

- Select the relevant Communication Manager SIP Entity for **System** e.g., **cm81**
- Select **Endpoint** for **Profile Type**
- Select **J179_DEFAULT_CM_8_1** for **Template**
- Enter the **Extension** number (e.g., **70111**)

Click on **Endpoint Editor** in the Extension field to edit Communication Manager settings. Input the appropriate **coverage path 1** number(not shown) to route unanswered calls to voicemail. Click **Done** to close the Endpoint Editor. Click **Commit**.



7. Administer Session Border Controller for Enterprise

SBCE provides an edge capability to allow remote worker registration external to the private enterprise network. Remote workers interact with the external interface of SBCE while the SBCE internal interface is shielded from the public external interface.

Note: The configuration in this section is only needed if the Median500Li is located outside the enterprise environment and registering through the internet.

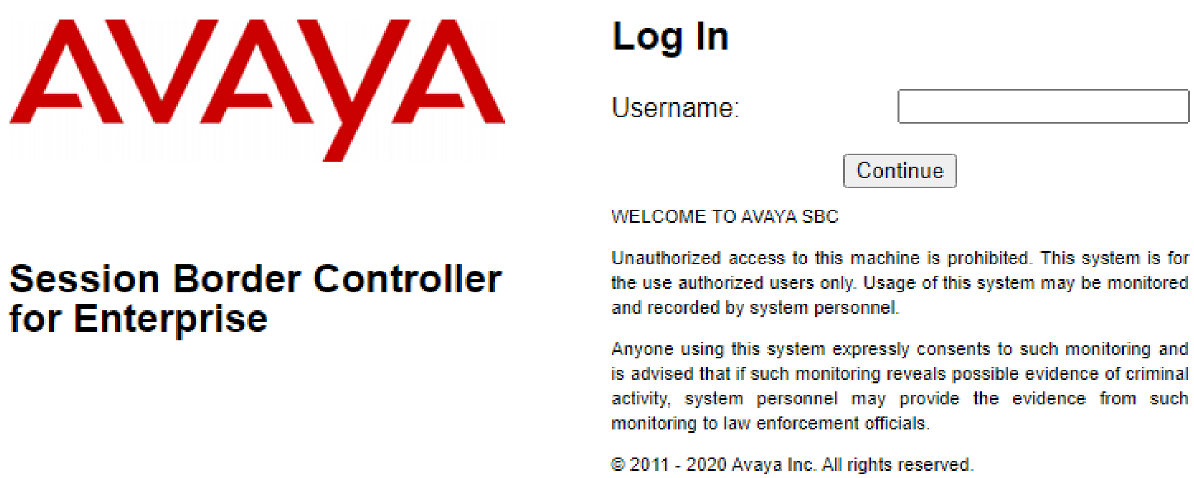
The configuration steps on SBCE include the following:

- Launch SBCE web interface
- Administer Network Management
- Administer Server TLS Profiles
- Administer SIP Servers
- Administer Routing Profiles
- Administer Media Rules
- Administer End Point Policy Groups
- Administer Media Interfaces
- Administer Signaling Interfaces
- Administer End Point Flows

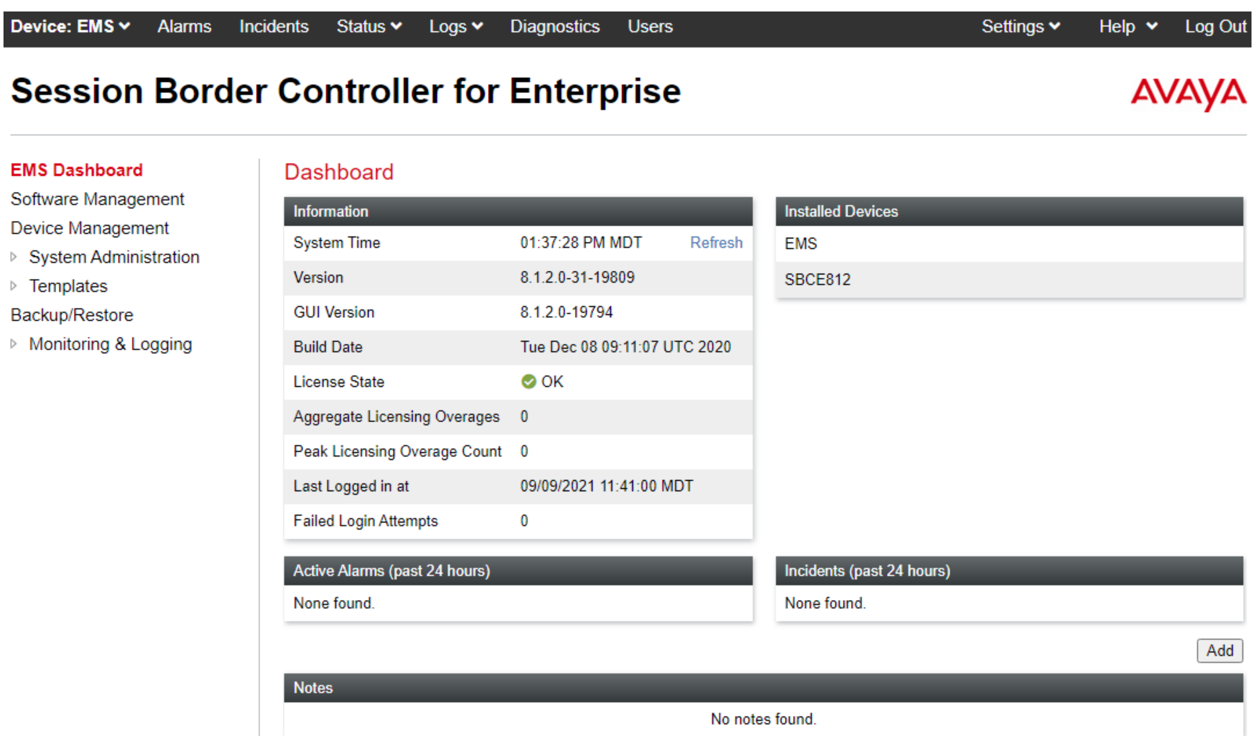
The SBCE administration tasks will either be stepped through or displayed as administered.

7.1. Launch SBCE Web Interface

Access the SBCE web interface by using the URL **https://<*ip-address*>/sbc** in an Internet browser window, where **<*ip-address*>** is the IP address of the SBCE management interface. The screen below is displayed. Log in using the appropriate credentials.

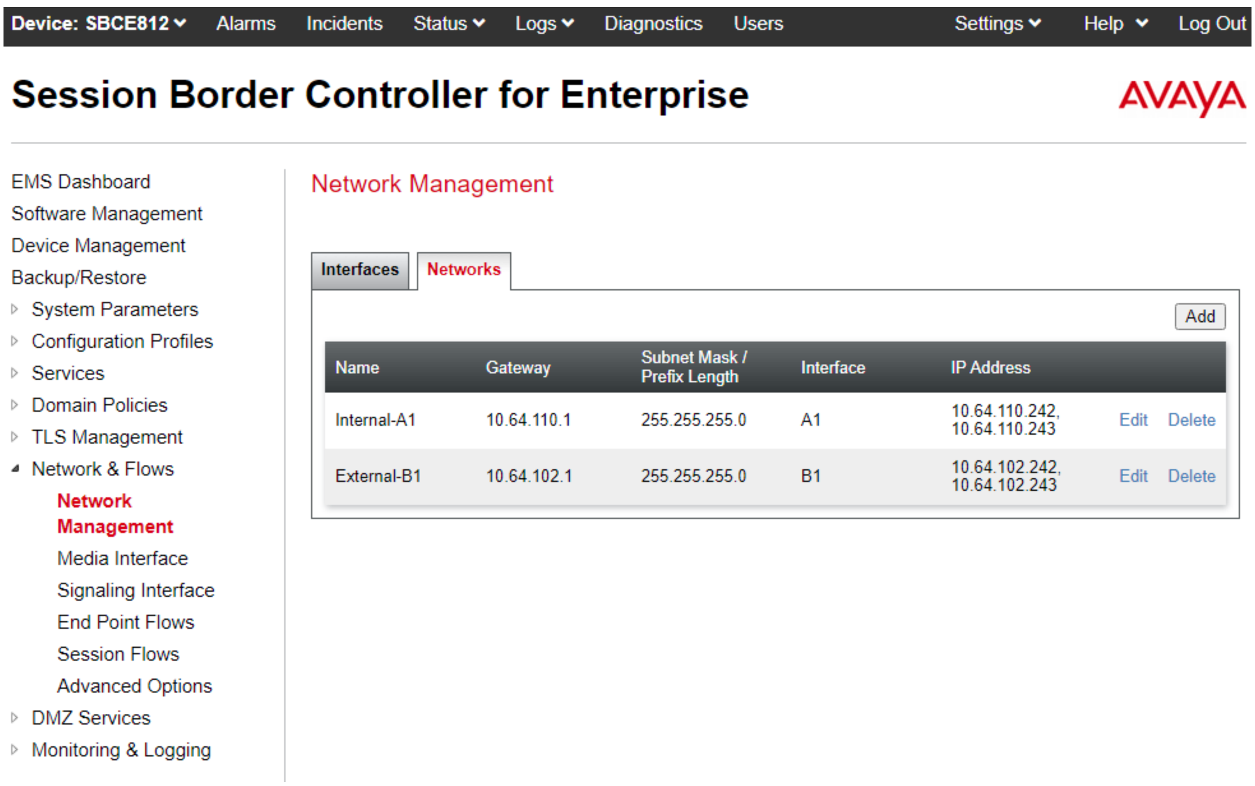


After logging in, the Dashboard will appear as shown below. All configuration screens of the SBCE are accessed by navigating the menu tree in the left pane. Select **Device → SBCE** from the top menu.



7.2. Administer Network Management

The Network Management screen is where the network interface settings are configured and enabled. During the SBCE installation process, certain network-specific information is defined such as device IP address(es), public IP address(es), netmask, gateway, etc., to interface the device to the network. It is this information that populates the various Network Management tab displays, which can be edited as needed to optimize device performance and network efficiency. Navigate to **Networks & Flows** → **Network Management**. On the Networks tab, select **Add** to add a new interface entry, or **Edit** to add or change IP addresses on an existing interface.
The following screen shows the enterprise interface assigned to **A1** and the interface towards the Remote Workers assigned to **B1**.



The following IP addresses and associated interfaces are used for remote workers in the reference configuration:

- 10.64.110.243: IP Address of Internal Interface A1 (Remote Workers media traffic)
- 10.64.102.243: IP Address of External Interface B1 (Remote Workers media traffic)

The IP address of 10.64.102.243 assigned to the external interface B1 is used for remote worker proxy registration. Click on the **Interfaces** tab (not shown) and verify the A1 and B1 interfaces are enabled. To enable an interface, click the corresponding **Disabled** link under the Status column to change it to **Enabled**.

7.3. Administer TLS Profiles

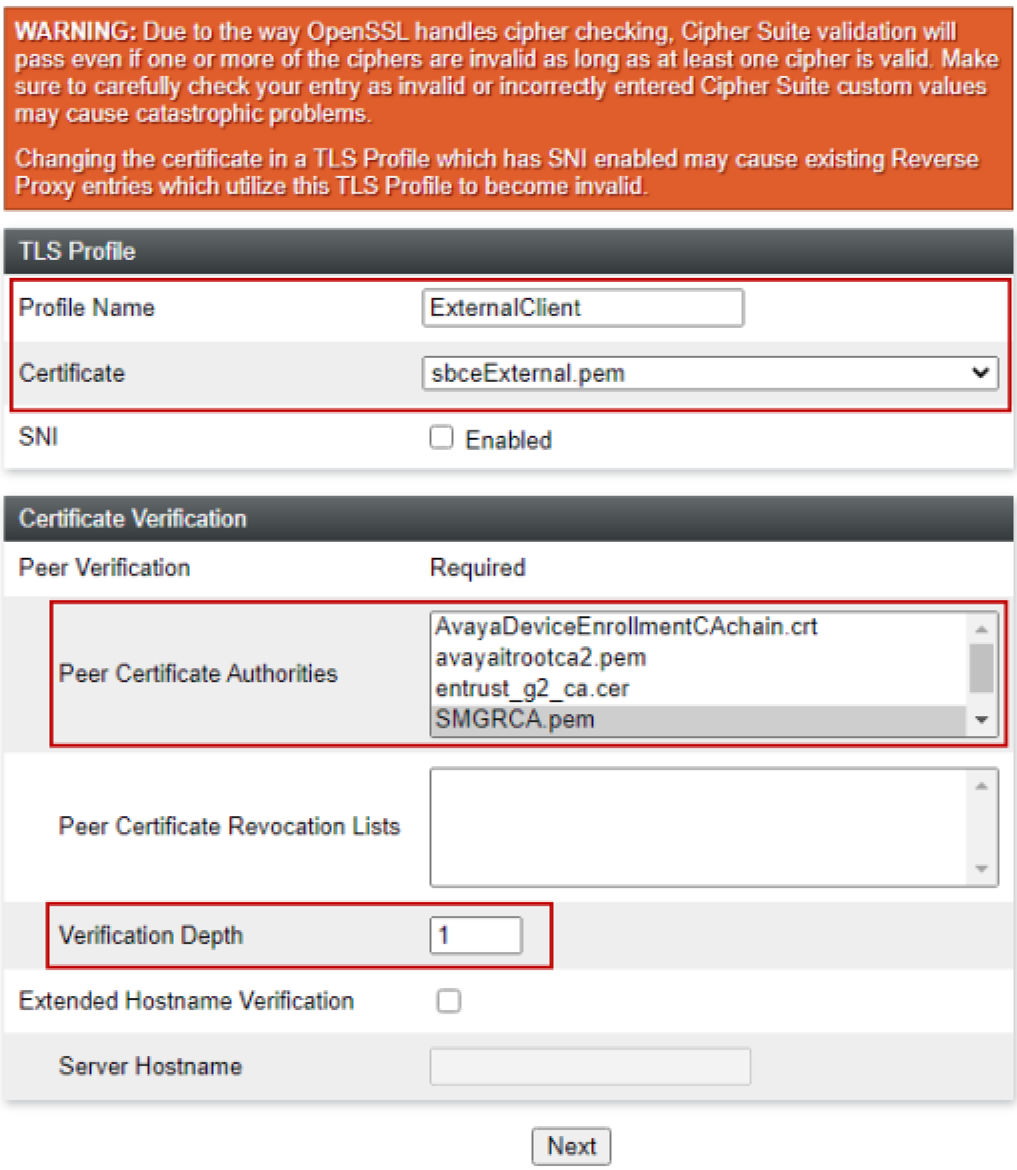
TLS profiles are created to assign identity certificates and CA certificates to SIP Servers. Certificate creation is not covered in these application notes. Details are provided in References **[2]** and **[4]** in **Section 11.** Both the internal and external interface SBCE identity certificates and their CA certificates should be installed in SBCE and available for use in TLS profiles.

7.3.1. Configure Avaya SBCE TLS Client Profiles

Select **TLS Management → Client Profiles** from the left-hand menu to create a SBCE TLS Client Profile. Click **Add.** Configure as follows:

- **Profile Name:** Input an appropriate name e.g., **ExternalClient**
- **Certificate:** Select the certificate for the external SBCE interface e.g., **sbceExternal.pem**
- **Peer Verification:** Set to **Required** by default
- **Peer Certificate Authorities:** Select the CA certificate e.g., **SMGRCA.pem**
- **Verification Depth:** Input **1**

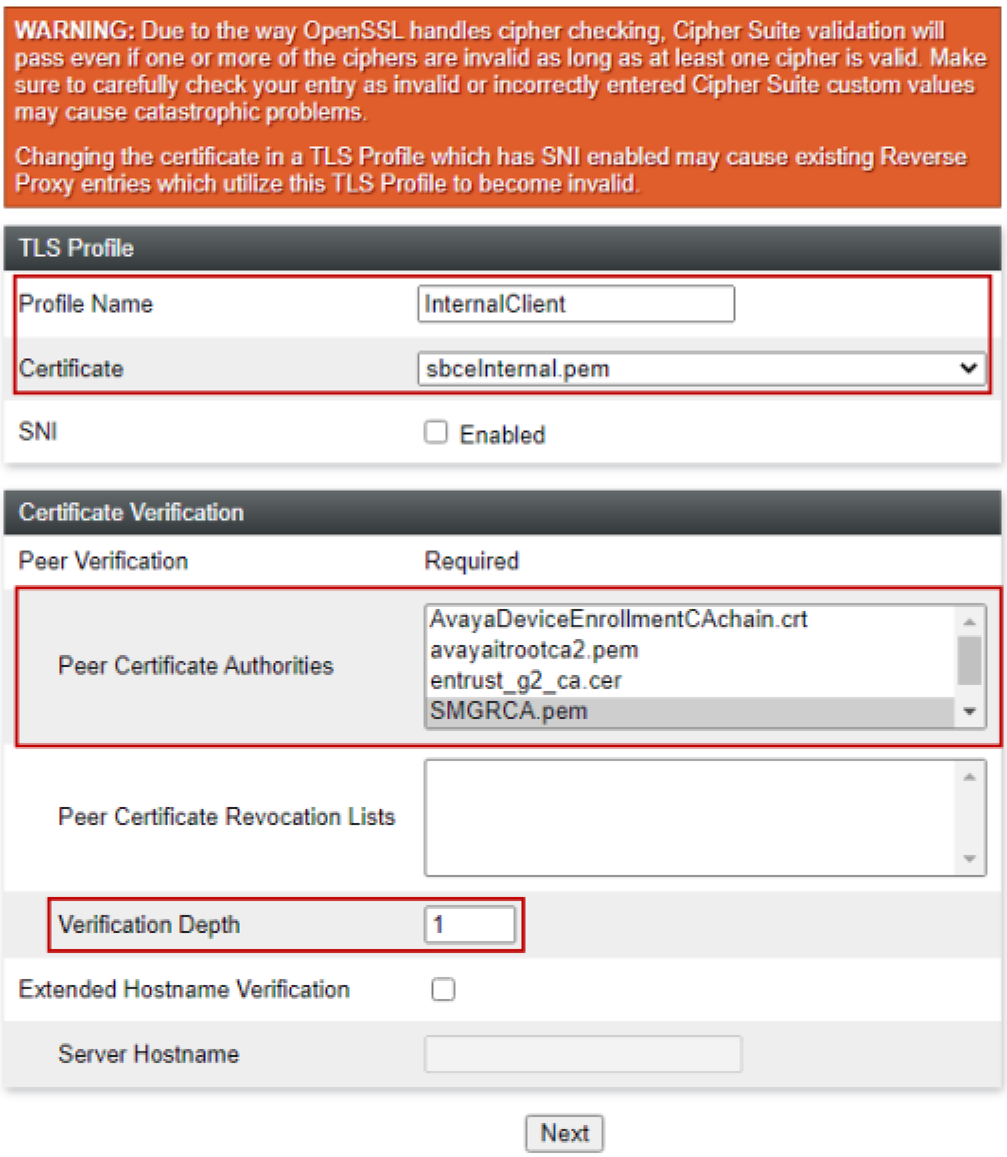
Click **Next**. Accept default values for the next screen and click **Finish**. The default TLS version is **TLS 1.2.**



Create a SBCE TLS Client Profile for the internal SBCE interface. Configure as follows:

- **Profile Name:** Input an appropriate name e.g., **InternalClient**
- **Certificate:** Select the certificate for the internal SBCE interface e.g., **sbceInternal.pem**
- **Peer Verification:** Set to **Required** by default
- **Peer Certificate Authorities:** Select the CA certificate e.g., **SMGRCA.pem**
- **Verification Depth:** Input **1**

Click **Next**. Accept default values for the next screen and click **Finish**. The default TLS version is **TLS 1.2.**

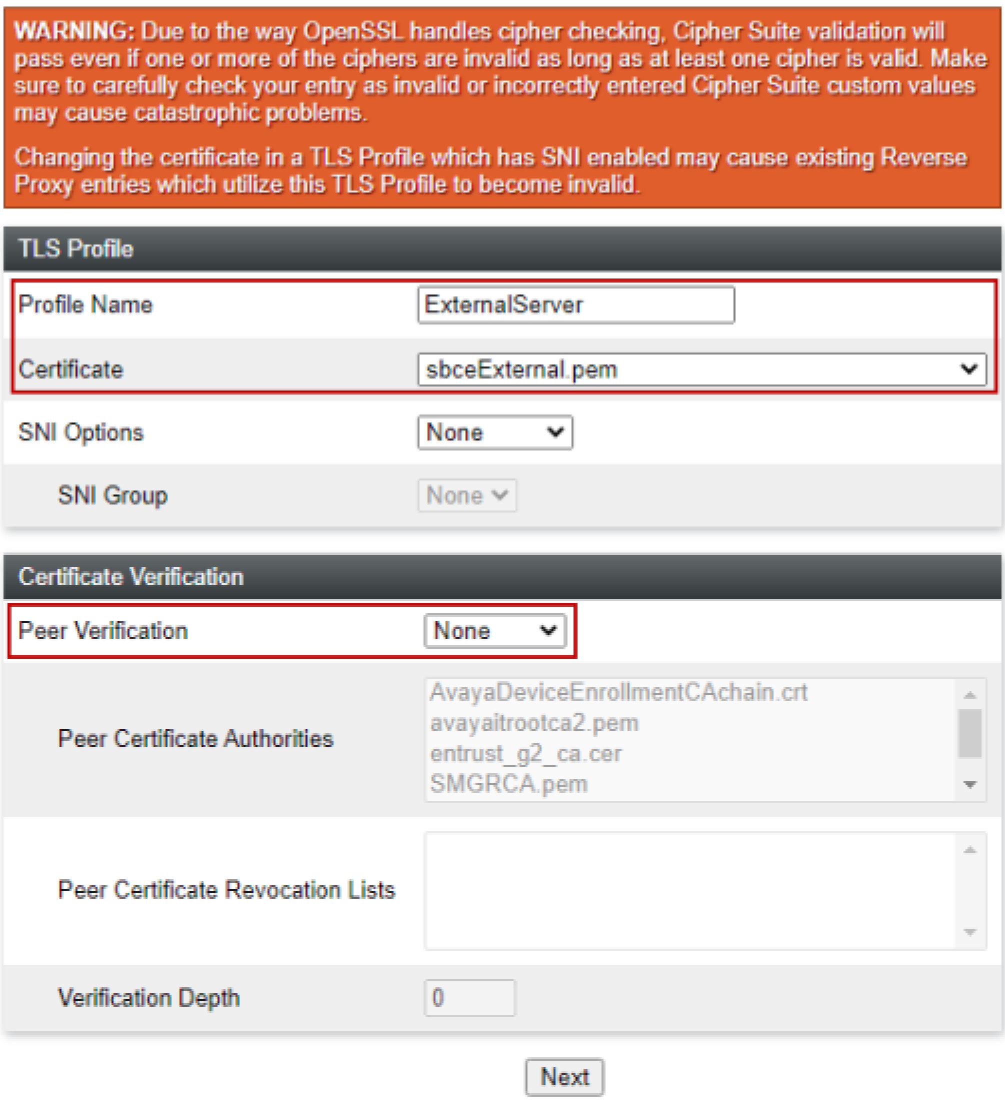


7.3.2. Configure Avaya SBCE TLS Server Profiles

Select **TLS Management → Server Profiles** from the left-hand menu to create an external SBCE TLS Server Profile. Click **Add.** Configure as follows:

- **Profile Name:** Input an appropriate name e.g., **External Server**
- **Certificate:** Select the certificate for the external SBCE interface e.g., **sbceExternal.pem**
- **Peer Verification:** Set to **None**

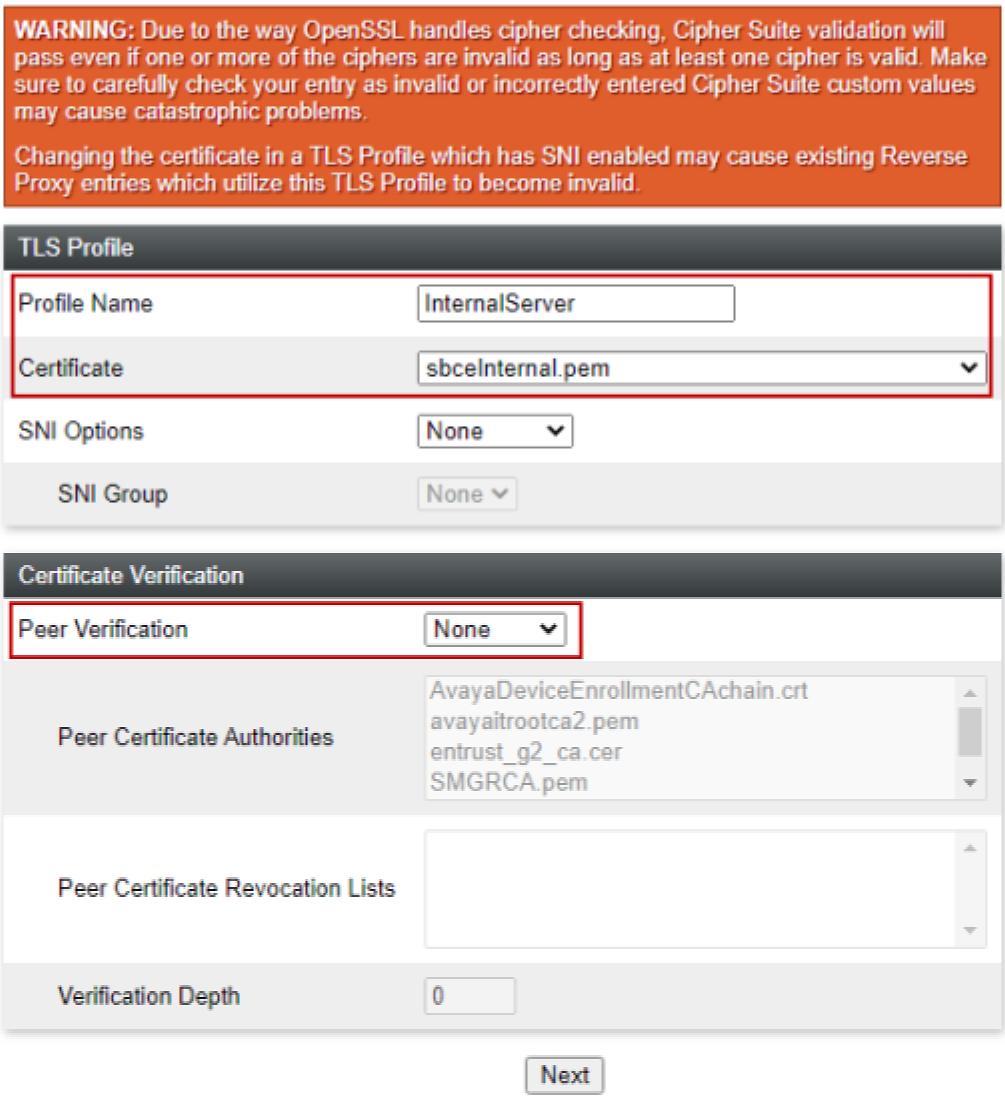
Click **Next**. Accept default values for the next screen and click **Finish**. The default TLS version is **TLS 1.2.**



Select **TLS Management** → **Server Profiles** from the left-hand menu to create an internal SBCE TLS Server Profile. Click **Add.** Configure as follows:

- **Profile Name:** Input an appropriate name e.g., **InternalServer**
- **Certificate:** Select the certificate for the external SBCE interface e.g., **SBCEInternal.pem**
- **Peer Verification:** Set to **None**

Click **Next**. Accept default values for the next screen and click **Finish**. The default TLS version is **TLS 1.2.**

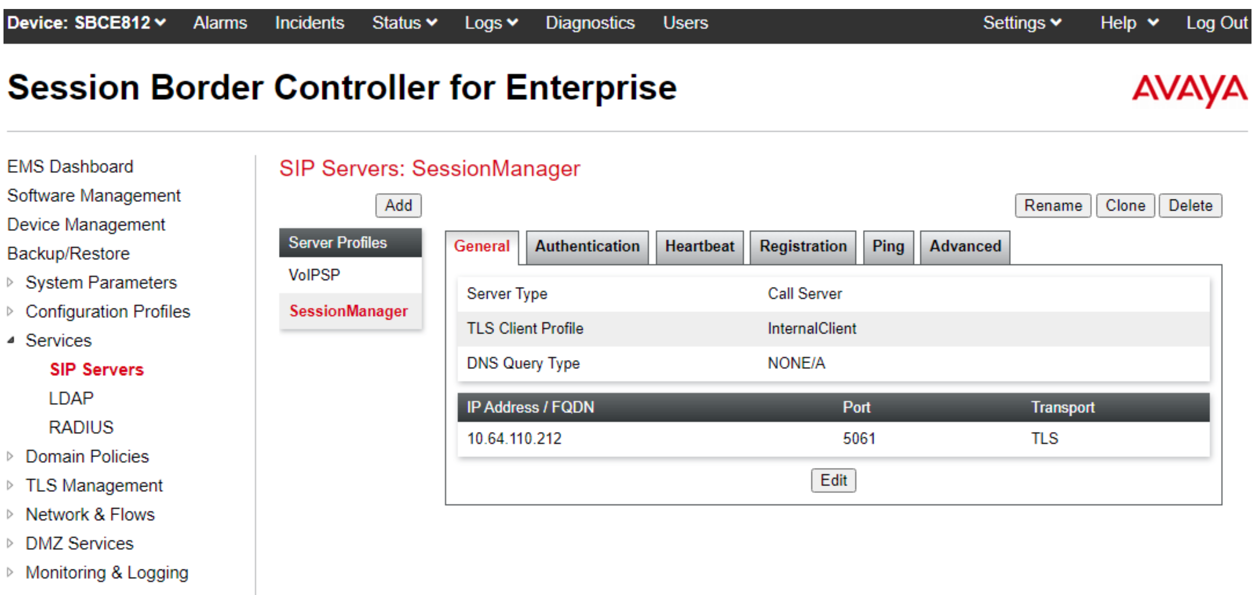


7.4. Administer SIP Servers

A SIP server must be defined for each server in the SBCE environment.
Note: TLS profiles are defined in **Section 7.3**. Certificate generation is not covered in these application notes. Certificate installation steps for Mediant 500Li is shown in **Section 8.6.1**. All TLS certificates used for the compliance test were signed by System Manager.

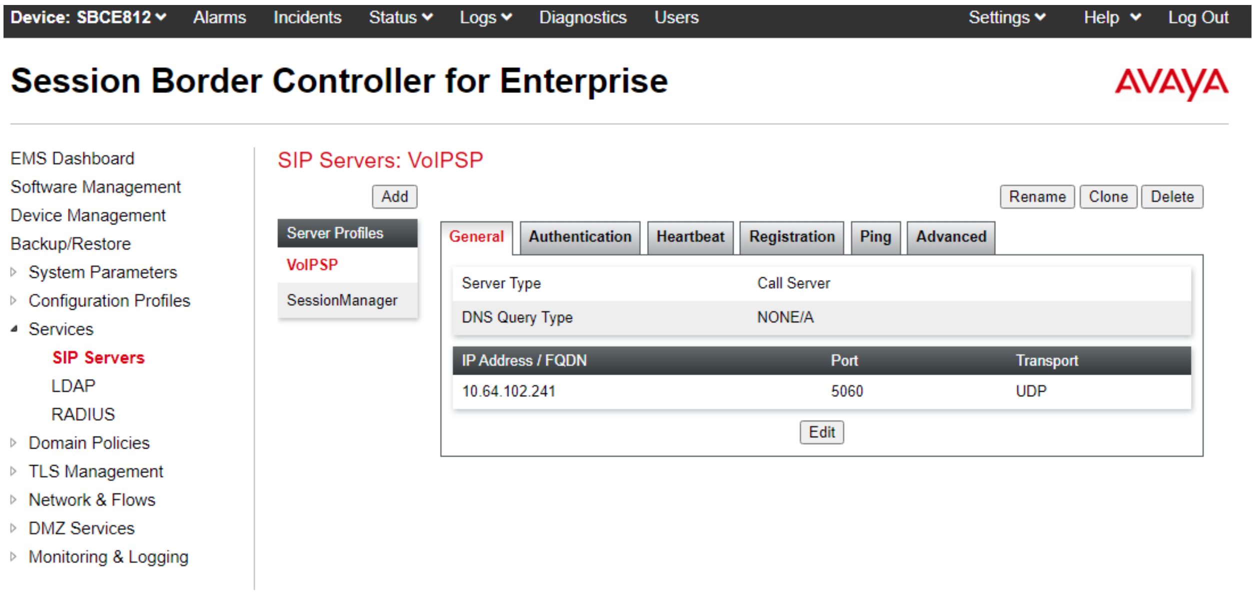
7.4.1. SIP Server for Session Manager

To define a SIP server, navigate to **Services → SIP Servers** from the left pane to display the existing SIP server profiles. Click **Add** to create a new SIP server or select a pre-configured SIP server to view its settings. The **General** tab of the Session Manager SIP Server was configured as follows. TLS transport was used for the Session Manager SIP trunk. The TLS profile from **Section 7.3.1** was used. All other tabs were left with their default values.



7.4.2. SIP Server for VoIPSP

The **General** tab of the VoIPSP SIP Server is shown for illustrative purposes. The SIP server was configured as follows. UDP transport was used for the VoIPSP SIP trunk. All other tabs were left with their default values.

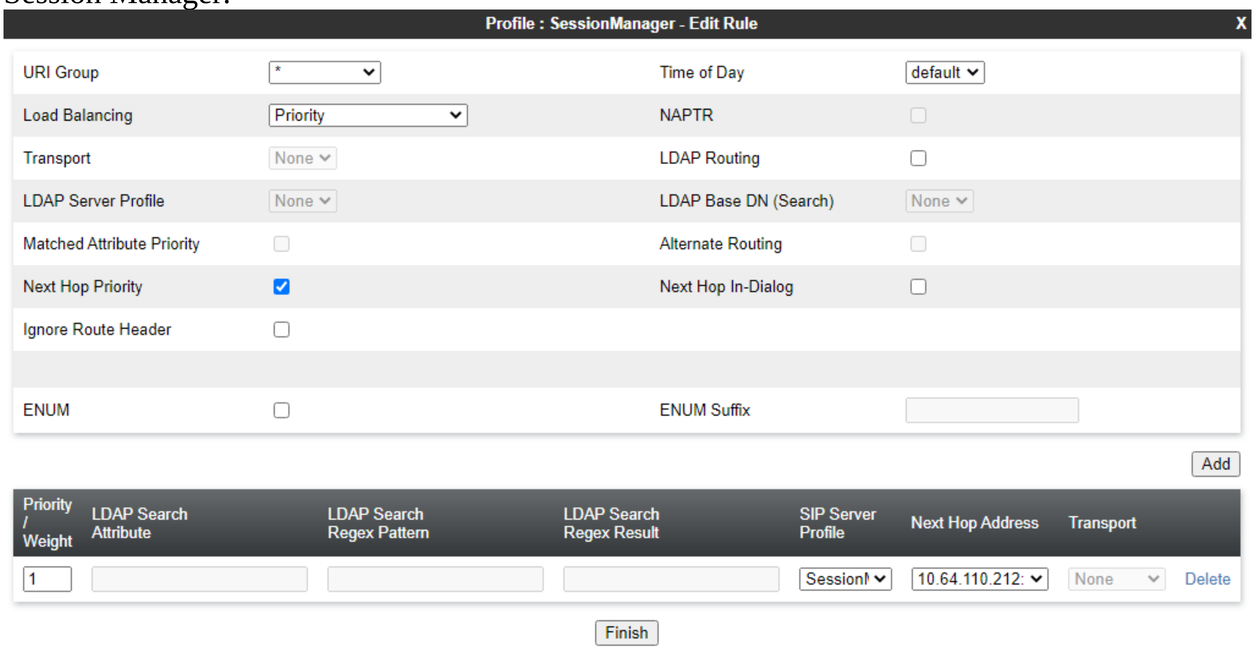


7.5. Administer Routing Profiles

A routing profile defines where traffic will be directed. Administer a routing profile for Session Manager and the VoIPSP if needed.

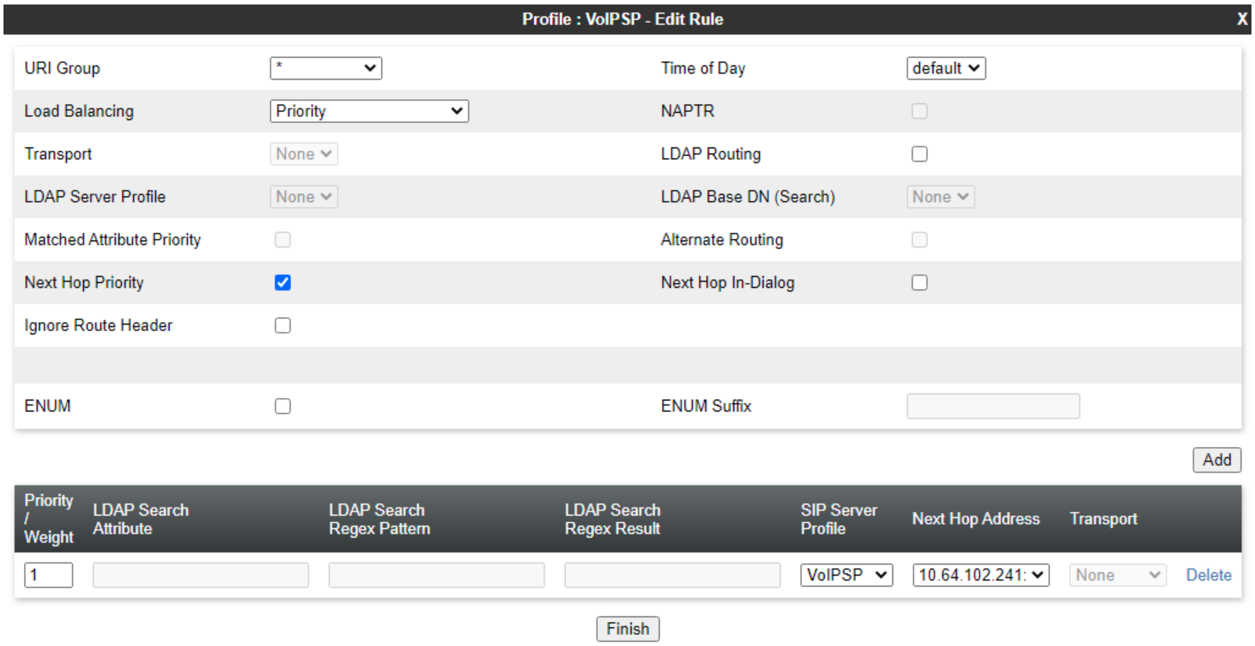
7.5.1. Routing Profile for Session Manager

To create a new profile, navigate to **Configuration Profiles → Routing** in the left pane. In the center pane, select **Add**. A pop-up window (not shown) will appear requesting the name of the new profile, followed by series of pop-up windows in which the profile parameters can be configured. To view the settings of an existing profile, select the profile from the center pane. The routing profile for calls to Session Manager is shown below. The routing profile was named **SessionManager**. This routing profile contains the IP address of the signaling interface of Session Manager.



7.5.2. Routing Profile for VoIPSP

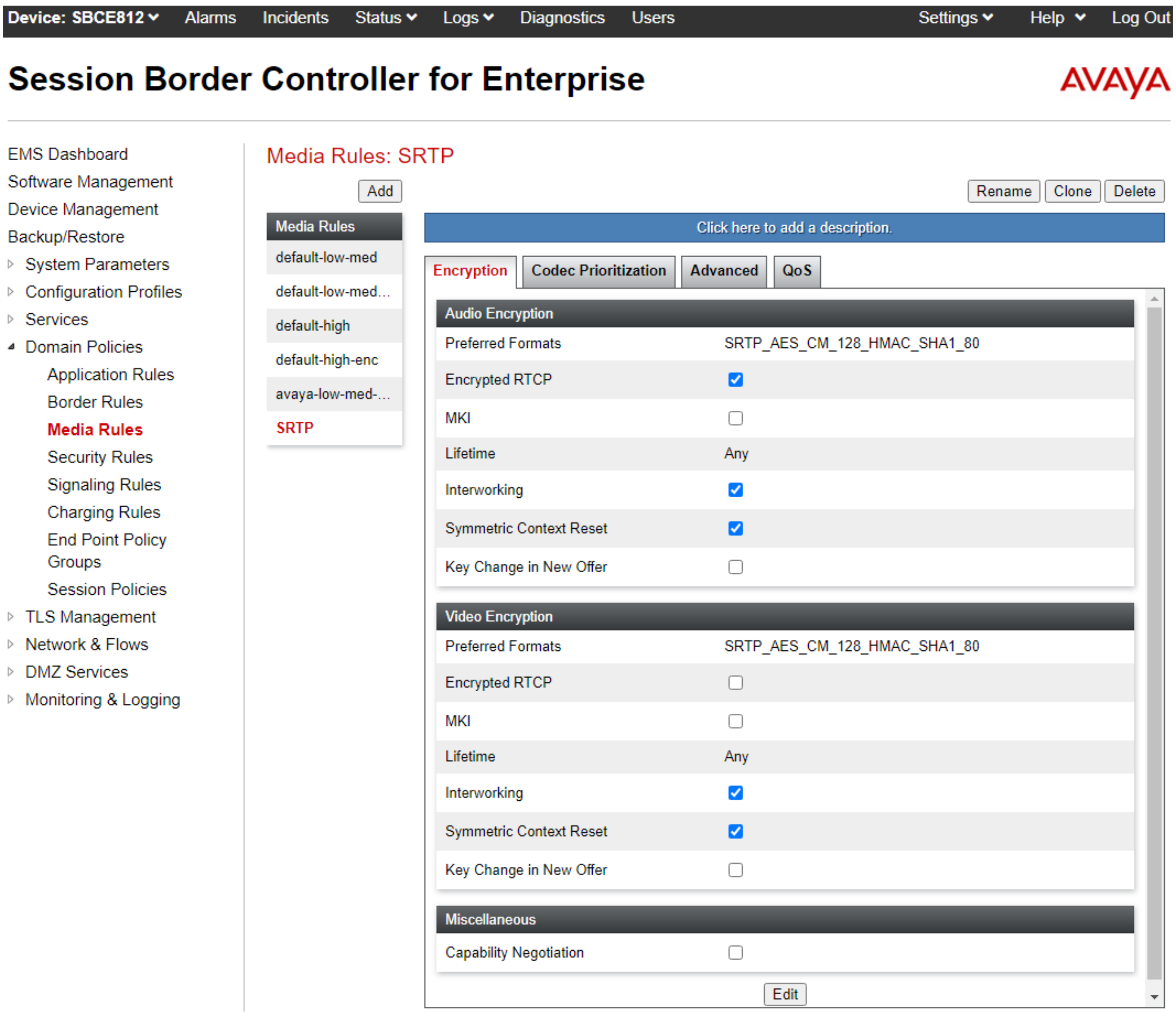
A routing profile for the VoIPSP trunk must exist. The routing profile for VoIPSP is shown below. This routing profile contains the IP address of the external SIP trunk interface of the VoIPSP Manager



7.6. Administer Media Rules

Media Rules define RTP media packet parameters such as codec prioritization and packet encryption techniques. These rules will be applied to the End Point Policy Groups configured in **Section 7.7**.

Navigate to **Domain Policies** → **Media Rules** in the left pane. In the center pane, select the rule **avaya-low-med-enc** and click the **Clone** button. Input an appropriate name, e.g., **SRTP.** Click Finish. The **Encryption** tab for the SRTP media rule is configured as seen below.

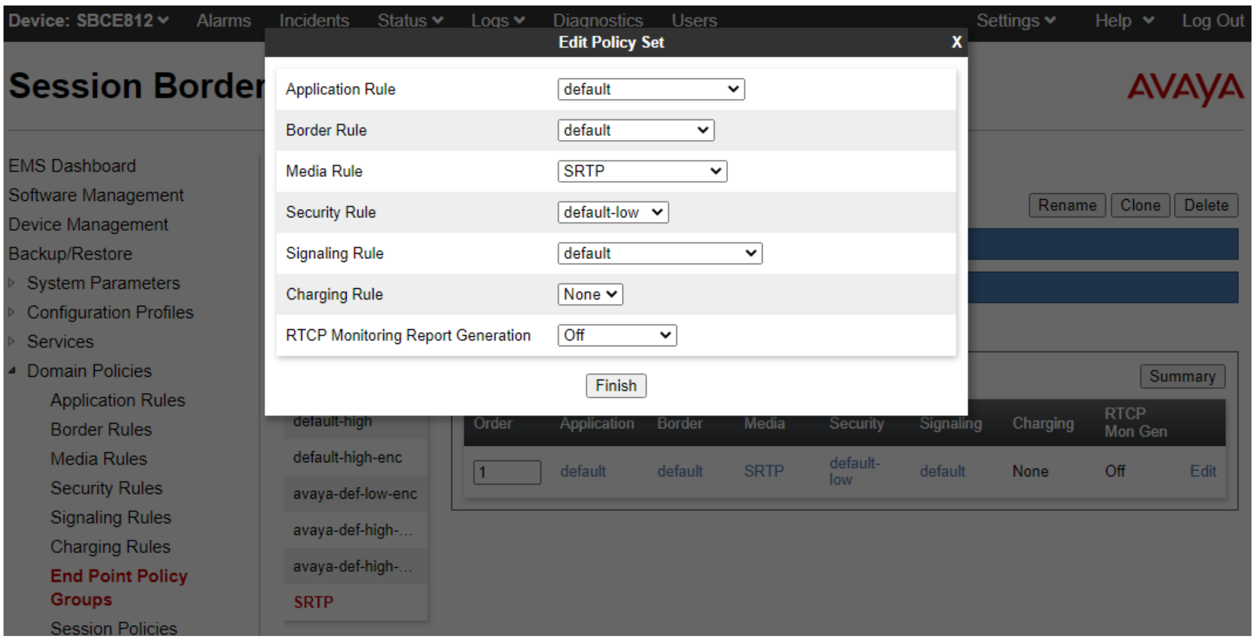


7.7. Administer End Point Policy Groups

End Point Policy Groups associate the different sets of rules (Media, Signaling, Security, etc.) to be applied to specific SIP messages traversing through the Avaya SBCE. For these application notes, a media rule will be configured to the policy group which is applied in **Section 7.10 End Point Flows**.

To create a new group, navigate to **Domain Policies→ End Point Policy Groups** in the left pane. In the right pane, select **Add**. A pop-up window (not shown) will appear requesting the name of the new group, e.g., **SRTP**, followed by the **Policy Group** window. Select **SRTP**, the media rule from **Section 7.6** for **Media Rule**. Click **Finish**.

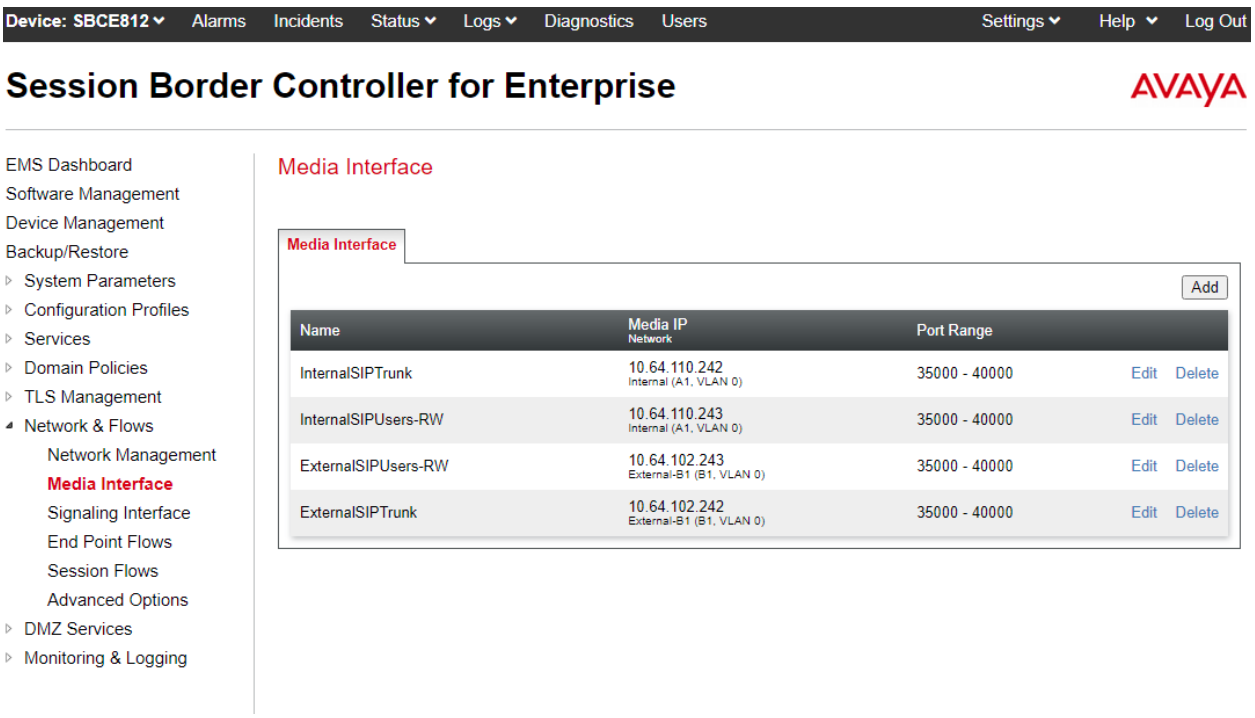
The new endpoint policy group, named **SRTP**, is shown below and is assigned the **SRTP** media rule configured in **Section 7.6**.



7.8. Administer Media Interfaces

Media interfaces are created to specify IP addresses and port range which SBCE will accept media streams. Separate media interfaces are needed for public and private interfaces. Navigate to **Networks & Flows → Media Interface** to define a new Media Interface. During the Compliance Testing the following interfaces were defined.

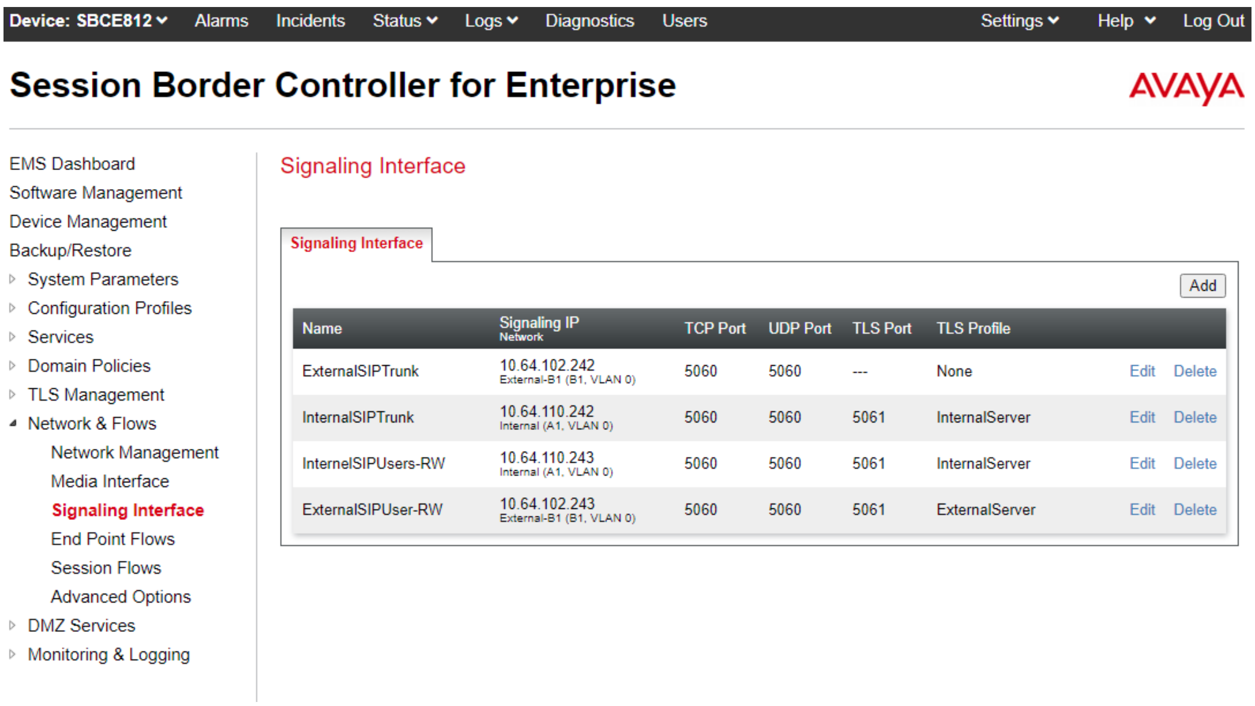
- **InternalSIPTrunk:** Interface used by Session Manager to send and receive media.
- **InternalSIPUsers-RW:** Interface used by Session Manager to send and receive media for remote workers.
- **ExternalSIPTrunk:** Interface used by the VoIPSP to send and receive media
- **ExternalSIPUsers-RW**: External interface used by remote workers.



7.9. Administer Signaling Interfaces

A signaling interface defines an IP address, protocols and listen ports that the SBCE can use for signaling. Create a signaling interface for both the internal and external sides of the SBCE.
Navigate to **Networks & Flows →Signaling Interface** to define a new **Signaling Interface**. During the Compliance Testing the following interfaces were defined. The signaling interfaces used for this solution are listed below.

- **InternalSIPTrunk:** Interface used by Session Manager to send and receive signaling.
- **InternalSIPUsers-RW:** Interface used by Session Manager to send and receive signaling to remote workers.
- **ExternalSIPTrunk:** Interface used by the service provider e.g., **VoIPSP** to send and receive signaling.
- **ExternalSIPUsers-RW:** Interface used by remote workers to send and receive signaling.



7.10. Administer End Point Flows

End Point Flows determine the path to be followed by the packets traversing through the Avaya SBCE. These flows combine the different sets of rules and profiles previously configured, to be applied to the SIP traffic traveling in each direction.
In the navigation pane, click **Network & Flows → End Point Flows**. Select **Subscriber Flows** or **End Point Flows** depending on the type of flow created. Click **Add**.

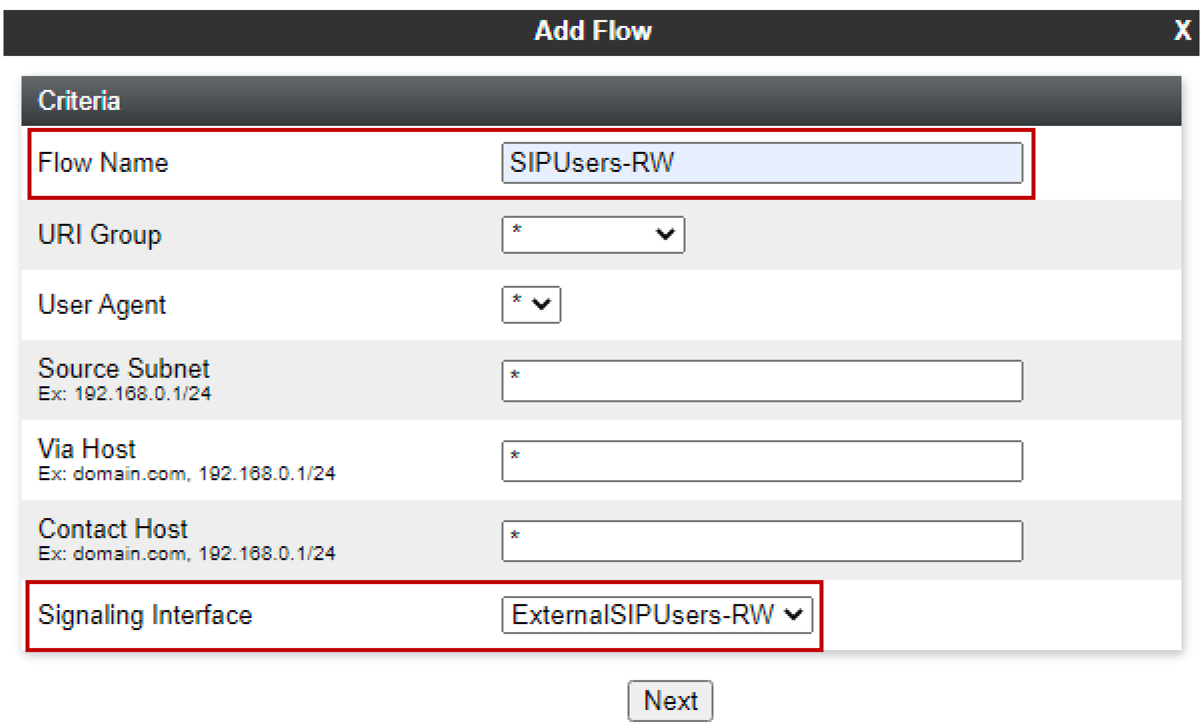
7.10.1. Remote Worker Subscriber Flow

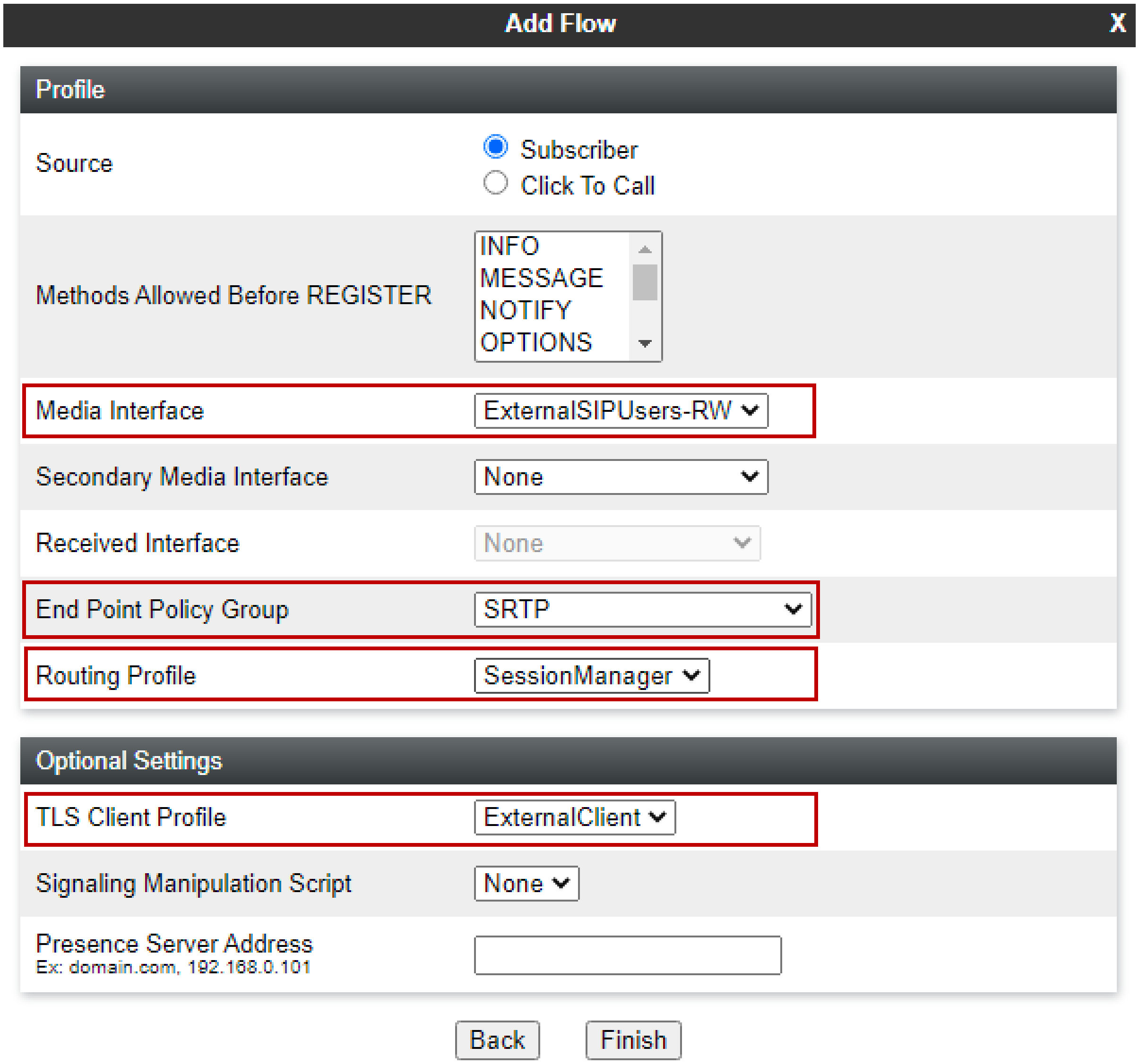
Subscriber End Point Flows refer to the actual endpoint devices, from which SIP messages originate and to which they are destined. End point devices may include hard phones, soft phone clients, and wireless handsets.

Create the SIP Users Remote Worker subscriber flow with the following inputs:

- **Flow Name:** Input an appropriate name e.g., **SIPUsers-RW**
- **Signaling Interface:** Select the external Remote Worker interface from **Section 7.9** e.g., **ExternalSIPUsers-RW**
- **Media Interface:** Select the external Remote Worker media interface from **Section 7.8** e.g., **ExternalSIPUsers-RW**
- **End Point Policy Group:** Select the external policy group (Incorporating the media rule) from **Section 7.6** e.g., **SRTP**
- **Routing Profile**: Select the Routing Profile from **Section 7.5.1** for Session Manager e.g., **Session Manager**
- **TLS Client Profile**: Select the client profile from **Section 7.3.1** e.g., **ExternalClient**

Click **Finish**.



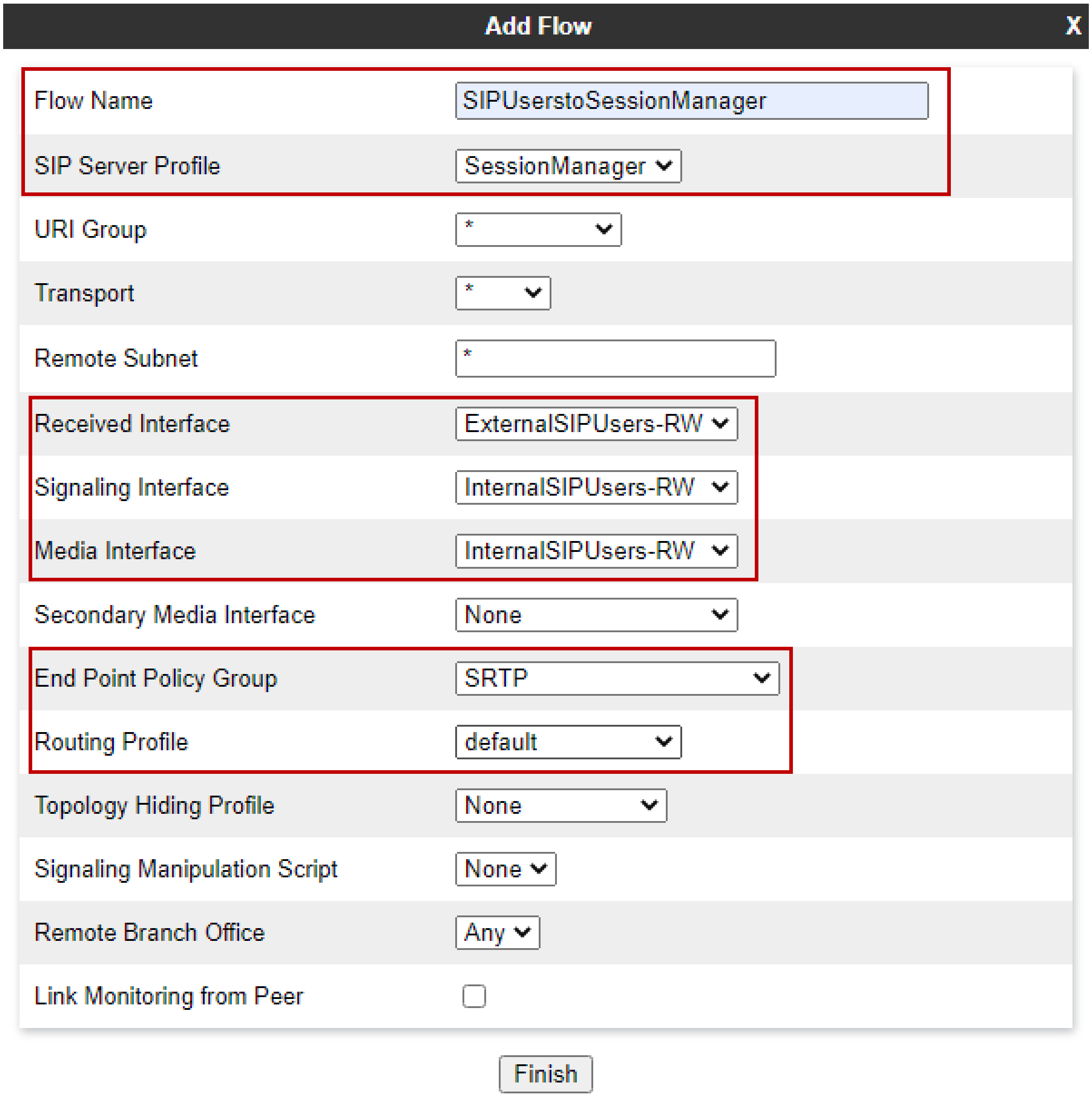

Add Flow
X
Profile
Source
Subscriber
Click To Call
Methods Allowed Before REGISTER
INFO
MESSAGE
NOTIFY
OPTIONS
Media Interface
ExternalSIPUsers-RW
Secondary Media Interface
None
Received Interface
None
End Point Policy Group
SRTP
Routing Profile
SessionManager
Optional Settings
TLS Client Profile
ExternalClient
Signaling Manipulation Script
None
Presence Server Address
Ex: domain.com, 192.168.0.101
Back
Finish

7.10.2. Session Manager Server Flows

Server End Point Flows refer to the IP call servers that connect to SIP trunk services.
Create the Session Manager server flow to Remote Workers with the following inputs:

- **Flow Name:** Input an appropriate name e.g., **SIPUsers-Session Manager**
- **SIP Server Profile:** Select the Server Profile for Session Manager from **Section 7.4.1** e.g., **SessionManager**
- **Received Interface:** Select the external Remote Worker signaling interface from **Section 7.9 e.g., ExternalSIPUsers-RW**
- **Signaling Interface:** Select the internal Remote Worker signaling interface from **Section Section 7.9** e.g., **InternalSIPUsers-RW**
- **Media Interface:** Select the internal Remote Worker signaling interface from **Section 7.8** e.g., **InternalSIPUsers-RW**
- **End Point Policy Group:** Select the end point policy group (Incorporating the media rule) from **Section 7.7** e.g., **SRTP**
- **Routing Profile**: Select the **default** Routing Profile

Click **Finish**.


Add Flow
Flow Name
SIPUserstoSessionManager
SIP Server Profile
SessionManager
URI Group
*
Transport
*
Remote Subnet
*
Received Interface
ExternalSIPUsers-RW
Signaling Interface
InternalSIPUsers-RW
Media Interface
InternalSIPUsers-RW
Secondary Media Interface
None
End Point Policy Group
SRTP
Routing Profile
default
Topology Hiding Profile
None
Signaling Manipulation Script
None
Remote Branch Office
Any
Link Monitoring from Peer
Finish

7.10.3. VoIPSP Server Flows

For illustrative purposes, the VoIPSP flows were created as such:

View Flow: VoIPSP-SessionManager

Criteria

Flow Name	VoIPSP-SessionManager
Server Configuration	SessionManager
URI Group	*
Transport	*
Remote Subnet	*
Received Interface	ExternalSIPTrunk

Profile

Signaling Interface	InternalSIPTrunk
Media Interface	InternalSIPTrunk
Secondary Media Interface	None
End Point Policy Group	SRTP
Routing Profile	VoIPSP
Topology Hiding Profile	To SMGR
Signaling Manipulation Script	None
Remote Branch Office	Any
Link Monitoring from Peer	☑

View Flow: SessionManager-VoIPSP

Criteria

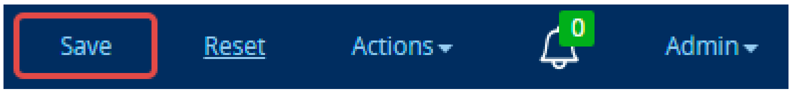
Flow Name	SessionManager-VoIPSP
Server Configuration	VoIPSP
URI Group	*
Transport	*
Remote Subnet	*
Received Interface	InternalSIPTrunk

Profile

Signaling Interface	ExternalSIPTrunk
Media Interface	ExternalSIPTrunk
Secondary Media Interface	None
End Point Policy Group	default-low
Routing Profile	SessionManager
Topology Hiding Profile	None
Signaling Manipulation Script	None
Remote Branch Office	Any
Link Monitoring from Peer	☐

8. Configure AudioCodes Mediant 500Li Analog Gateway

This section describes Mediant 500Li configuration for the test environment.
Note: It is **recommended** to save any configuration changes to flash memory after they have been applied by pressing the **SAVE** button in the top left of corner of the management console. If the configuration has been changed, the **SAVE** button is outlined in red. Click the **SAVE** button to burn the new configuration to flash memory. If the configuration change requires a reset, the **RESET** button next to the **SAVE** button will be outlined in red and should be initiated.



8.1. Initial Network Setup

Initial network administration is done via the command line interface. See **document [4]** in **Section 11 Additional References** for further information.
The scope of Mediant 500Li Interoperability tests only includes analog line functionality. As such, the Interoperability test initial configuration includes only that needed for analog functionality and is documented here.

8.1.1. Assumptions

Mediant 500Li is located outside of the enterprise in the remote worker configuration. If the device is connected to the open internet, it is **highly** recommended to modify the access-list to follow your network security guidelines. Consult references **[5]** and **[6]** for access-list configuration information.

8.1.2. Changes to the configuration example below

Change the following parameters to reflect the numbering scheme of your network:

- **a.a.a.a** = IP address of Mediant 500Li
- **b.b.b.b** = subnet mask (ex. 255.255.255.0)
- **c.c.c.c** = primary DNS address
- **d.d.d.d** = secondary DNS address
- **e.e.e.e** = default gateway

8.1.3. CLI Configuration

Follow the process to initially configure Mediant 500Li IP address. CLI commands below change the factory default network settings. Once complete, the web interface can be accessible through the network.

- Set the ethernet port on your laptop to DHCP

- Connect an ethernet cable from your laptop into the first yellow port on Mediant 500Li. It should be labeled Sx/GE LAN 1. Mediant 500Li is set to provide a DHCP address in the range of 192.168.0.3 – 192.168.0.8.

- SSH into the MSBR via the following commands:

ssh Admin@192.168.0.1
Password: **as appropriate**

M500Li> **en**
Password: **as appropriate**

- Enter the following commands:

```
configure data
access-list voip permit ip any any
interface gigabitethernet 0/0
ip address a.a.a.a b.b.b.b
ip name-server c.c.c.c d.d.d.d
ip access-group voip in
exit
ip route 0.0.0.0 0.0.0.0 e.e.e.e gigabitethernet 0/0 10
exit
```

- Verify you can log into the GUI via the IP address assigned to **gigabitethernet 0/0**

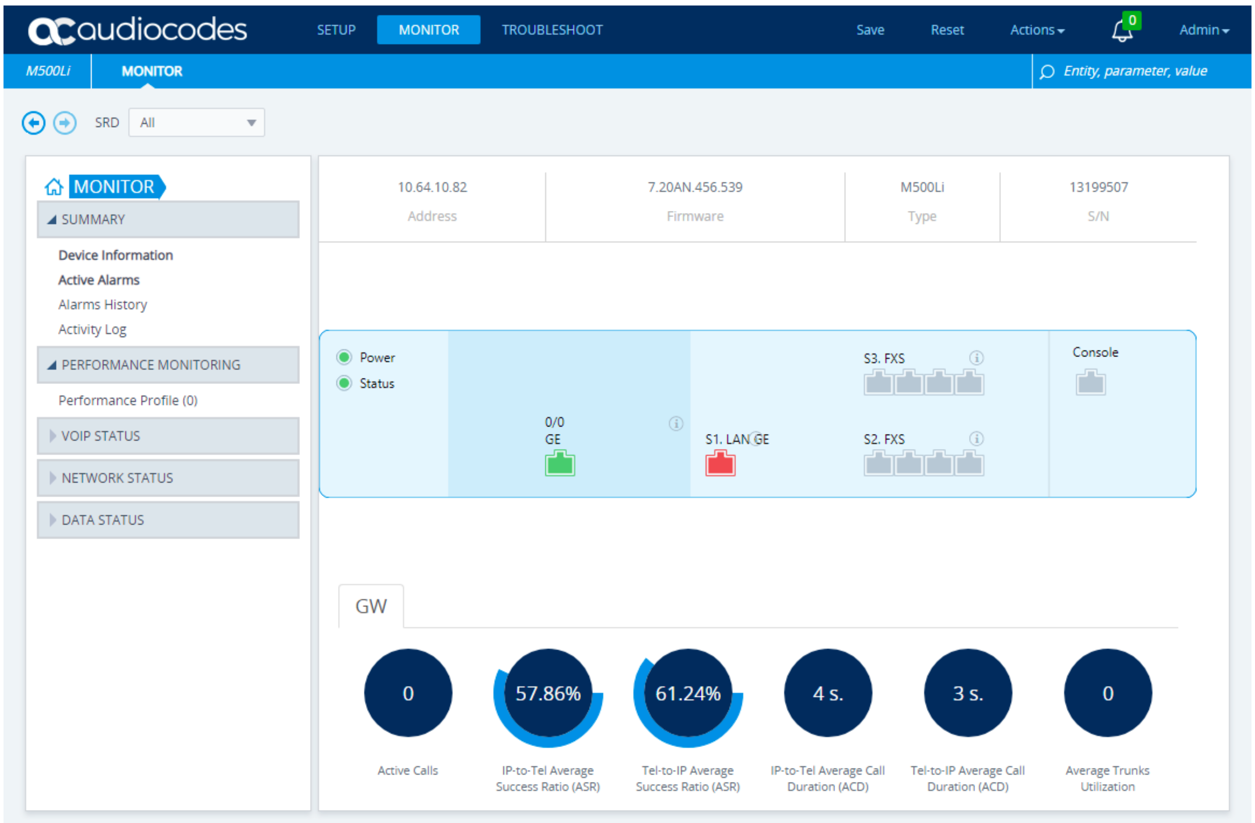
- Open a new terminal session and ssh into the CLI via the IP address assigned to **gigabitethernet 0/0**

```
configure data
interface GigabitEthernet 1/1
shutdown
exit
exit
write
```

NOTE: Shutting down the LAN port after IP configuration is optional
NOTE: The **write** command writes the current configuration to flash.

8.2. Verify/Upgrade Firmware Version

Configuration of the AudioCodes Mediant 500Li is done via web administrative console. Type in http://<**IP-address**> in a web browser, where <**IP-address**> is the IP Address of AudioCodes Mediant 500Li. Enter **Admin** as the username and the appropriate password. Click **Log In**. Once logged in, click on the **MONITOR** button in the title bar. The firmware version is displayed.



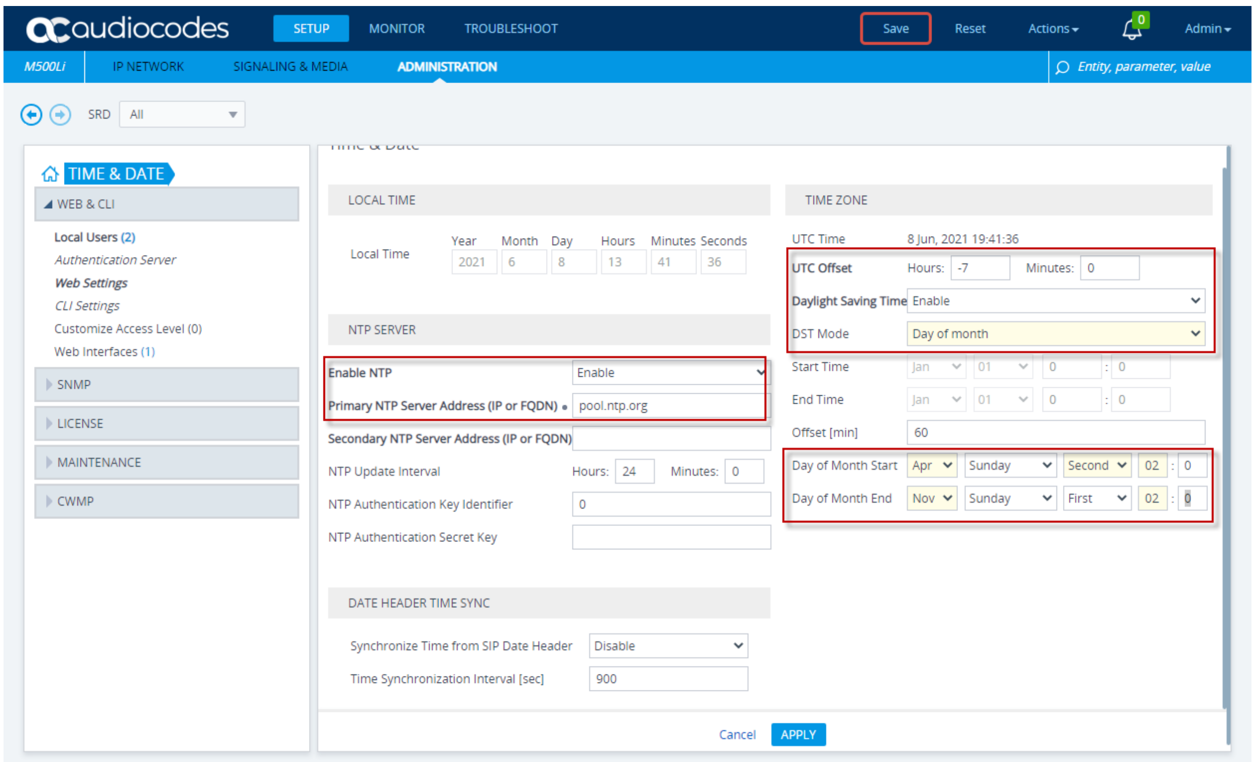
The firmware version used in Interoperability testing is **7.20AN.456.539**. To upgrade Mediant 500Li software, consult **document [4]** in **Section 11 Additional References** for instructions.

8.3. Set System Time and Date

Click on the **SETUP** button in the title bar. Select **ADMINISTRATION** in the toolbar. The time and date panel displays.

- Select **Enable** for **Enable NTP**
- Select an appropriate **Primary NTP Server Address**
- Select the appropriate **UTC Offset** and **DST Mode**
- Select **Day of month** for the **DST mode**
- Input the appropriate **Day of Month Start** and **Day of Month End**

Click **APPLY.** The local time should display correctly

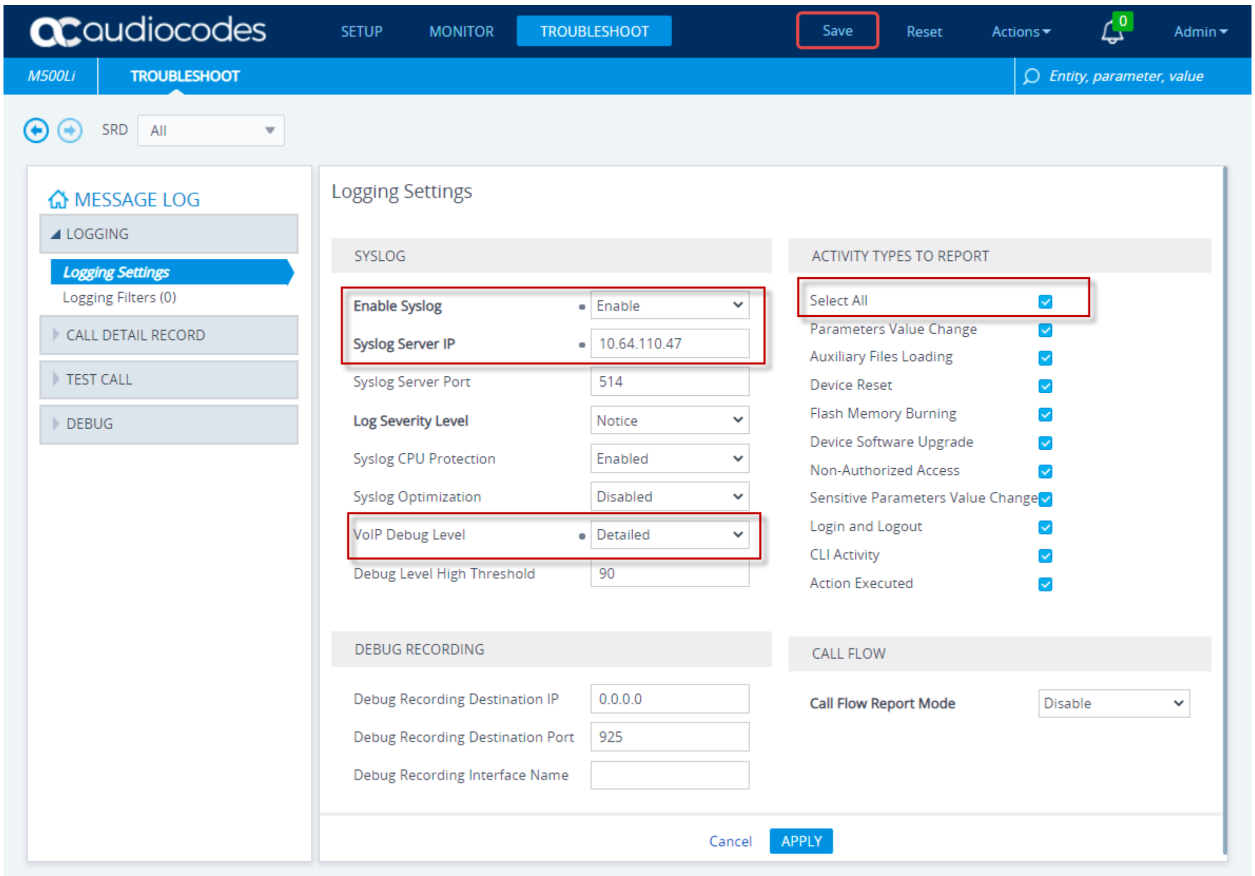


8.4. Administer Syslog Settings

Click on **TROUBLESHOOT** in the title bar**.** Select **LOGGING → Logging Settings** in the left pane.

- Set **Enable Syslog** to **Enable**
- For **Syslog Server IP** Address, type in the IP address of a workstation that is running a syslog application, e.g., **AudioCodes Syslog Viewer**
- Set **VoIP Debug Level** to **Detailed**
- Under the **ACTIVITY TYPES TO REPORT** subsection, check **Select All**

Click **APPLY.**



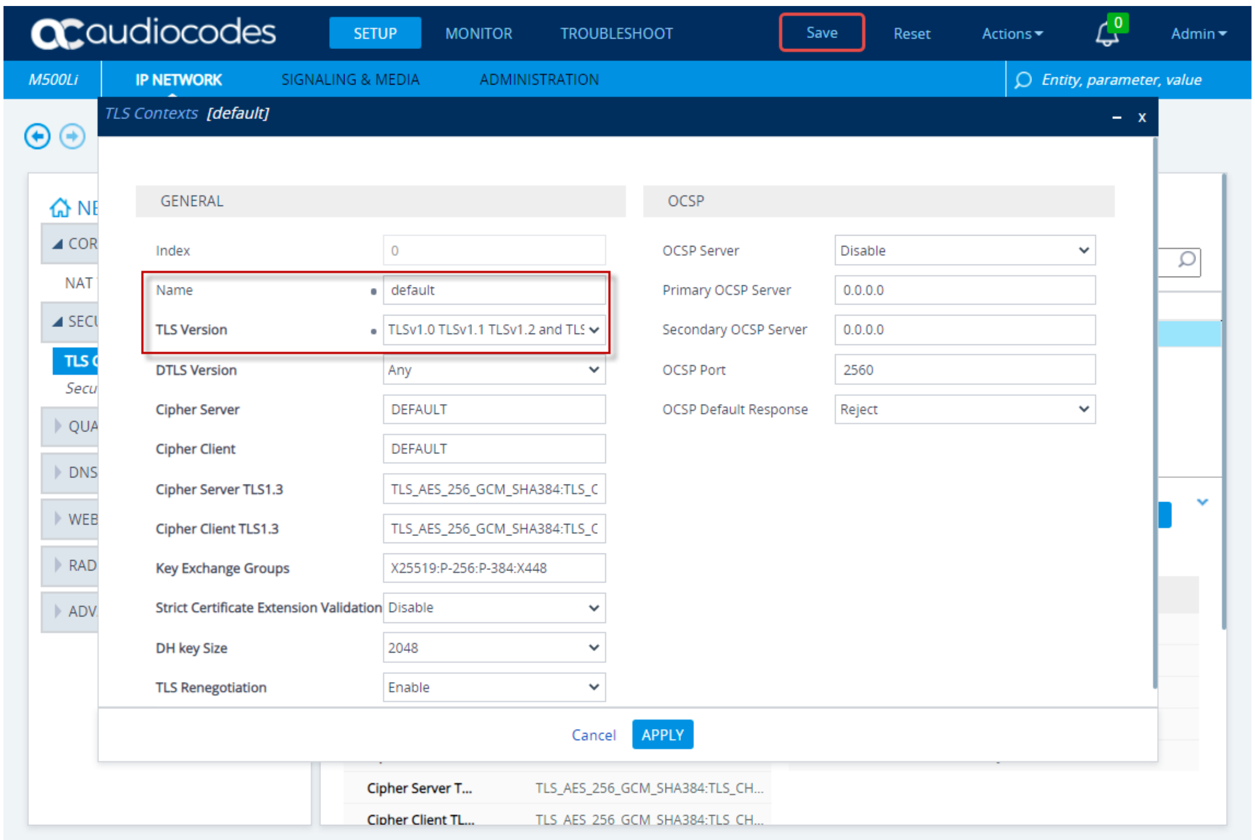
8.5. Administer Security

8.5.1. TLS Contexts

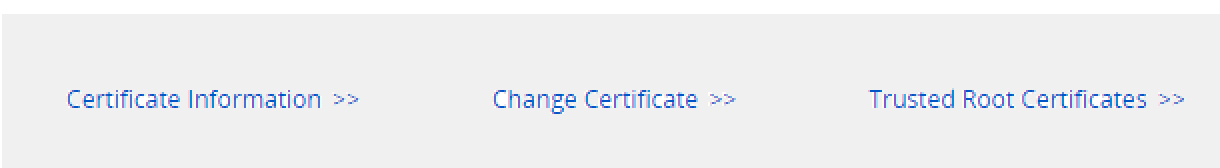
Click on **SETUP** in the title bar. Select **IP NETWORK** in the toolbar. Select **SECURITY → TLS Contexts** in the left pane. The default context displays.

- Click the **Edit** button (not shown)
- Select **default** for the **NAME**
- Select **TLSv1.0 TLSv1.1 TLSv1.2 and TLSv1.3** for the **TLS Version**

Click **APPLY.**



Note: Links at the bottom of the page are used for certificate administration as detailed in **Section 8.6.1.**

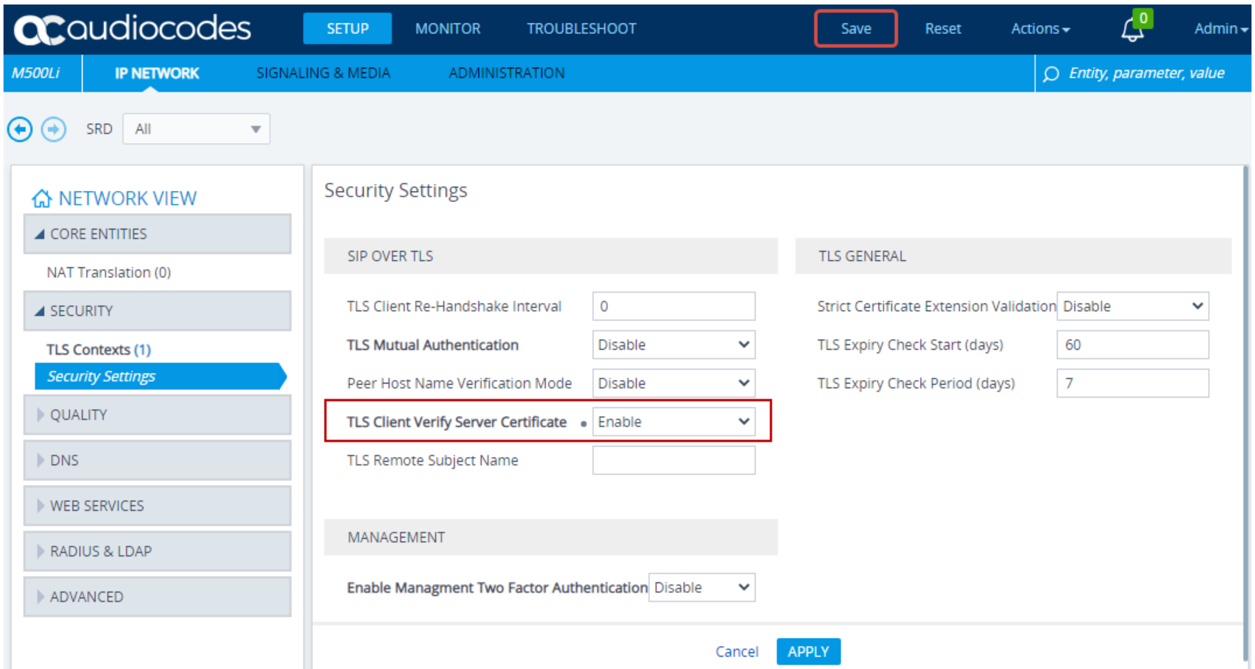


8.5.2. Administer Security Settings

Click on **SETUP** in the title bar. Select **IP NETWORK** in the toolbar. Select **SECURITY → Security Settings** in the left pane.

- Select **Enable** for **TLS Client Verify Server Certificate**

Click **APPLY.**



8.6. Administer Media

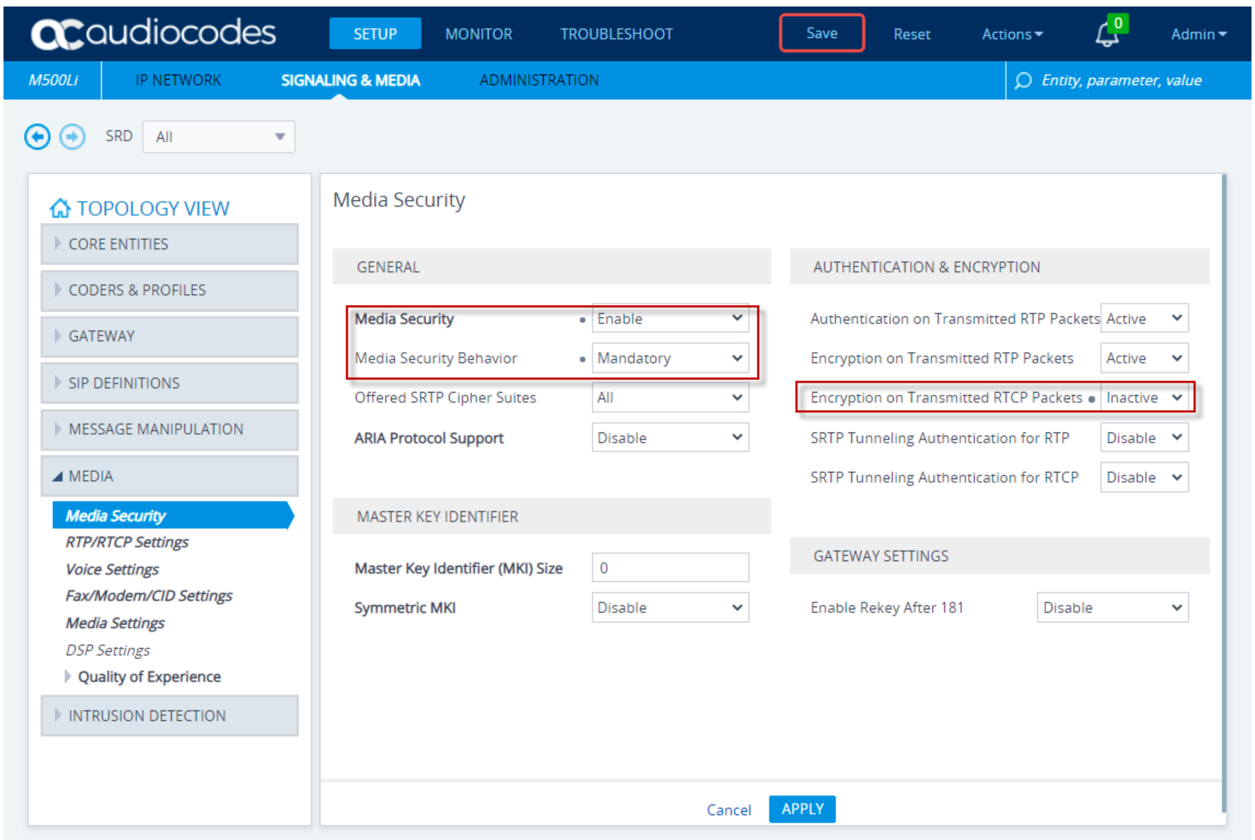
Media configuration administers SRTP connection parameters to the proxy server.

8.6.1. Administer Media Security

Select **SETUP** in the title bar. Select **SIGNALING & MEDIA** in the toolbar. Select **MEDIA → Media Security** in the left pane.

- Select **Enable** for **Media Security**
- Select **Mandatory**for **Media Security Behavior**.
- Select **Inactive** for **Encryption** on **Transmitted RTCP Packets**

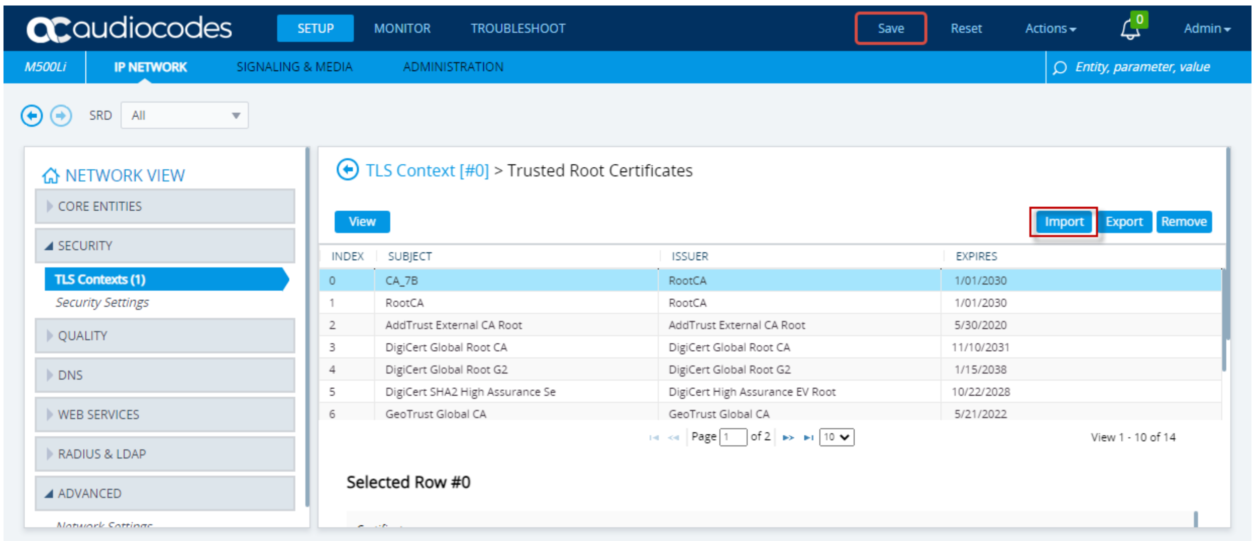
Click **APPLY.**



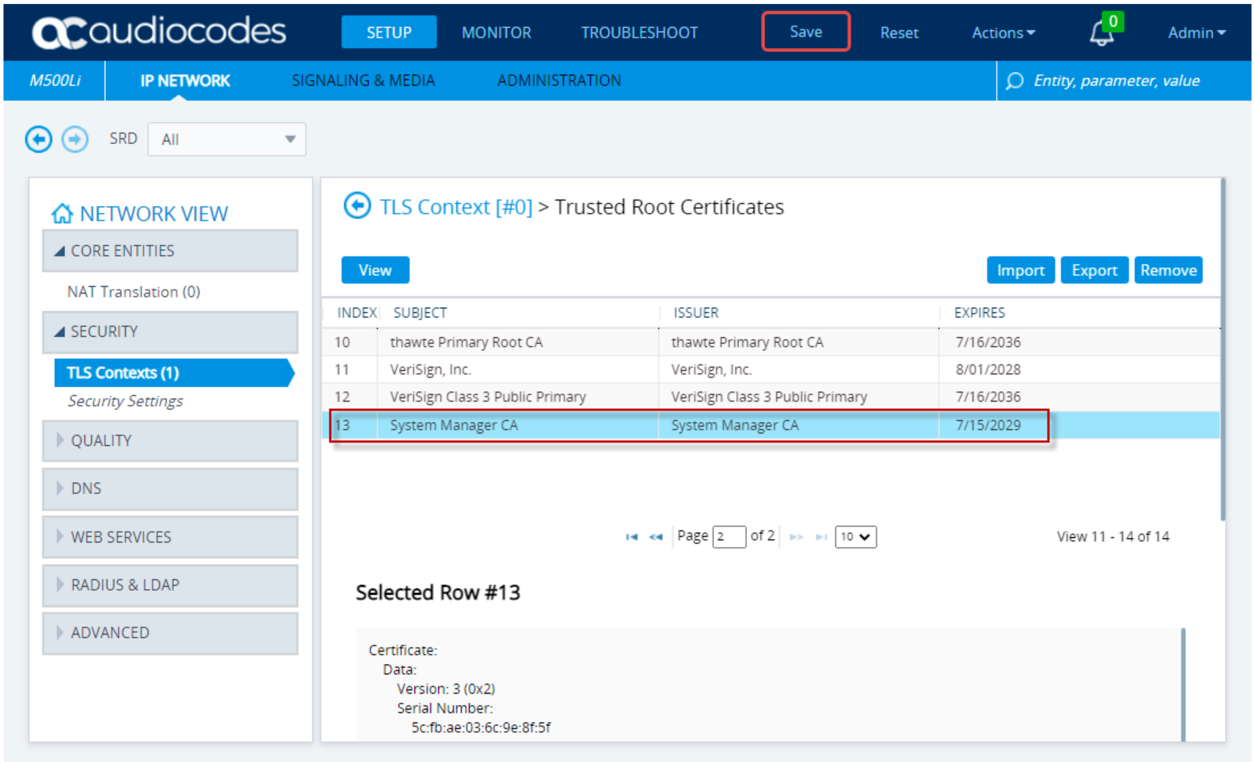
8.6.2. Install Certificates

Note: The certificate configuration tested used one-way authentication.
Note: The Certificate Authority used for Interoperability tests is Avaya Session Manager. Session Manager CA certificate file generation is not covered here. Consult reference **[2]** in **Section 11**. Please note that without this certificate, TLS/SRTP will not work.

Click on **SETUP** in the title bar. Select **IP NETWORK** in the toolbar. Select **SECURITY → TLS Contexts** pane, and select **Trusted Root Certificates** in the bottom of the pane (not shown).
Click the **Import** button and select the CA certificate file.



The imported certificate should display in the Trusted Root Certificates list.

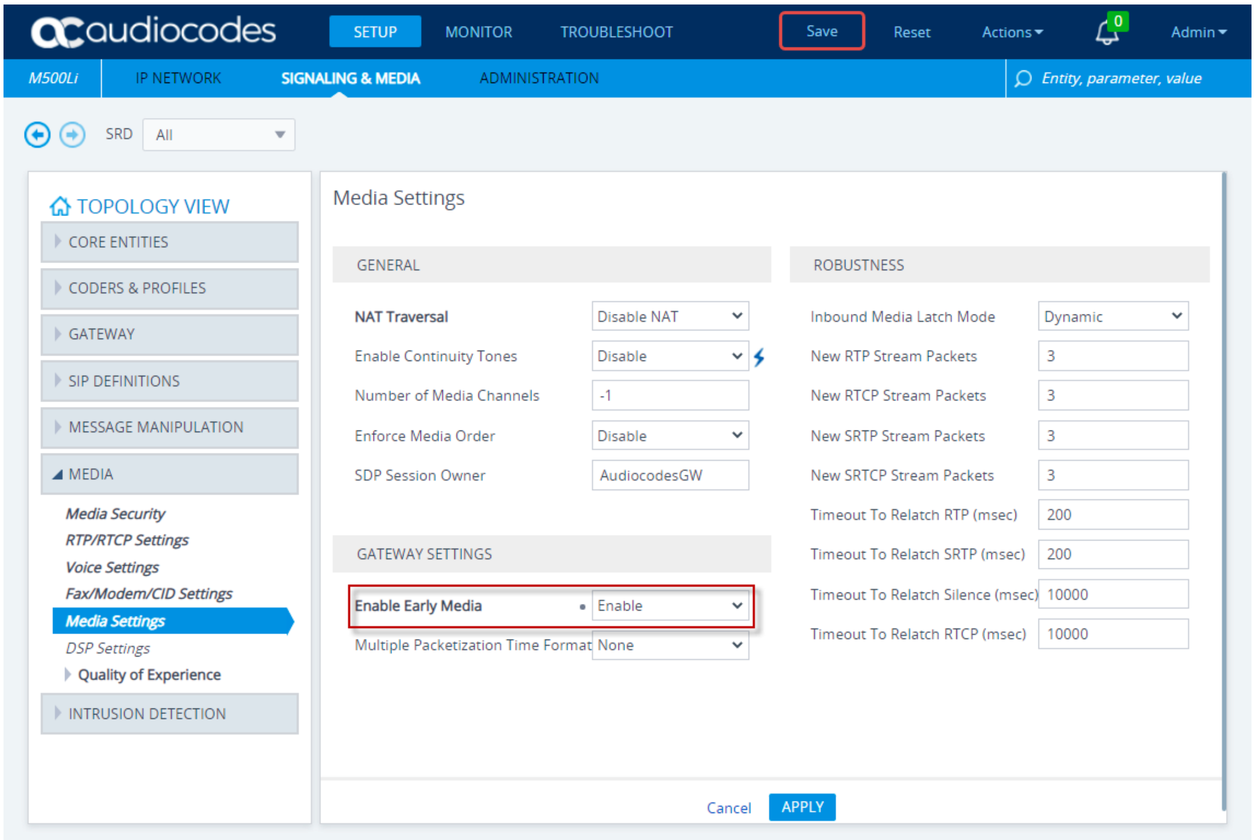

audiocodes
SETUP
MONITOR
TROUBLESHOOT
Save
Reset
Actions
0
Admin
M500Li
IP NETWORK
SIGNALING & MEDIA
ADMINISTRATION
Entity, parameter, value
SRD
All
NETWORK VIEW
CORE ENTITIES
NAT Translation (0)
SECURITY
TLS Contexts (1)
Security Settings
QUALITY
DNS
WEB SERVICES
RADIUS & LDAP
ADVANCED
TLS Context [#0] > Trusted Root Certificates
View
Import
Export
Remove
INDEX
SUBJECT
ISSUER
EXPIRES
10
thawte Primary Root CA
thawte Primary Root CA
7/16/2036
11
VeriSign, Inc.
VeriSign, Inc.
8/01/2028
12
VeriSign Class 3 Public Primary
VeriSign Class 3 Public Primary
7/16/2036
13
System Manager CA
System Manager CA
7/15/2029
Page 2 of 2
10
View 11 - 14 of 14
Selected Row #13
Certificate:
Data:
Version: 3 (0x2)
Serial Number:
5c:fb:ae:03:6c:9e:8f:5f

8.6.3. Administer Media Settings

Click on **SETUP** in the title bar. Select **Signaling & Media** in the toolbar. Select **MEDIA → Media Settings** in the left pane.

- Select **Enable** for **Enable Early Media**

Click **APPLY.**



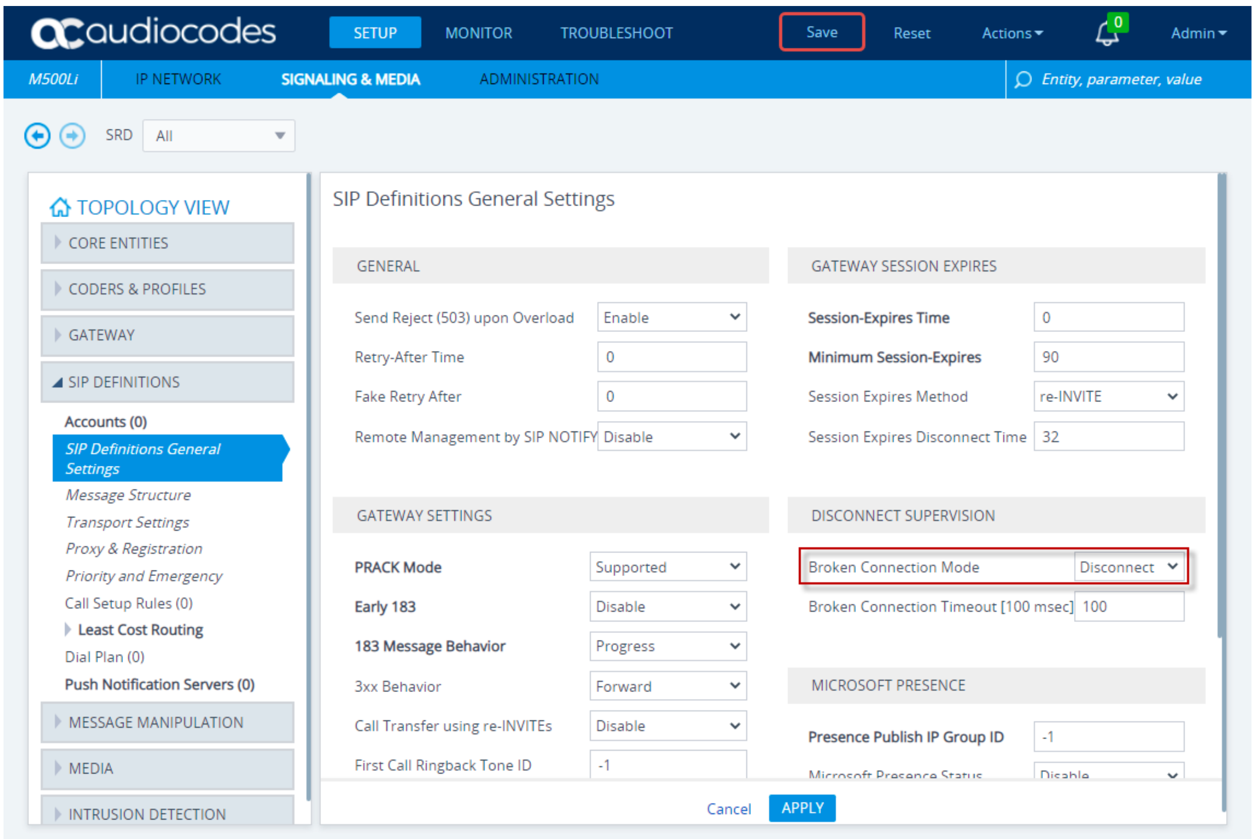
8.7. Administer SIP Definitions

8.7.1. General settings

Click **SETUP** in the title bar. Select **Signaling & Media** in the toolbar. Select **SIP DEFINITIONS → SIP Definitions General Settings** in the left pane.

- Select **Ignore** for **Broken Connection Mode.** Interoperability Testing used **Disconnect** which will cause Mediant 500Li to disconnect the call if RTP is not detected within the time specified in the Broken Connection Timeout. **Note:** The Default Broken Connection Timeout is 10 seconds (100 x 100ms). This does not affect the results of the interoperability testing.

Click **APPLY.**

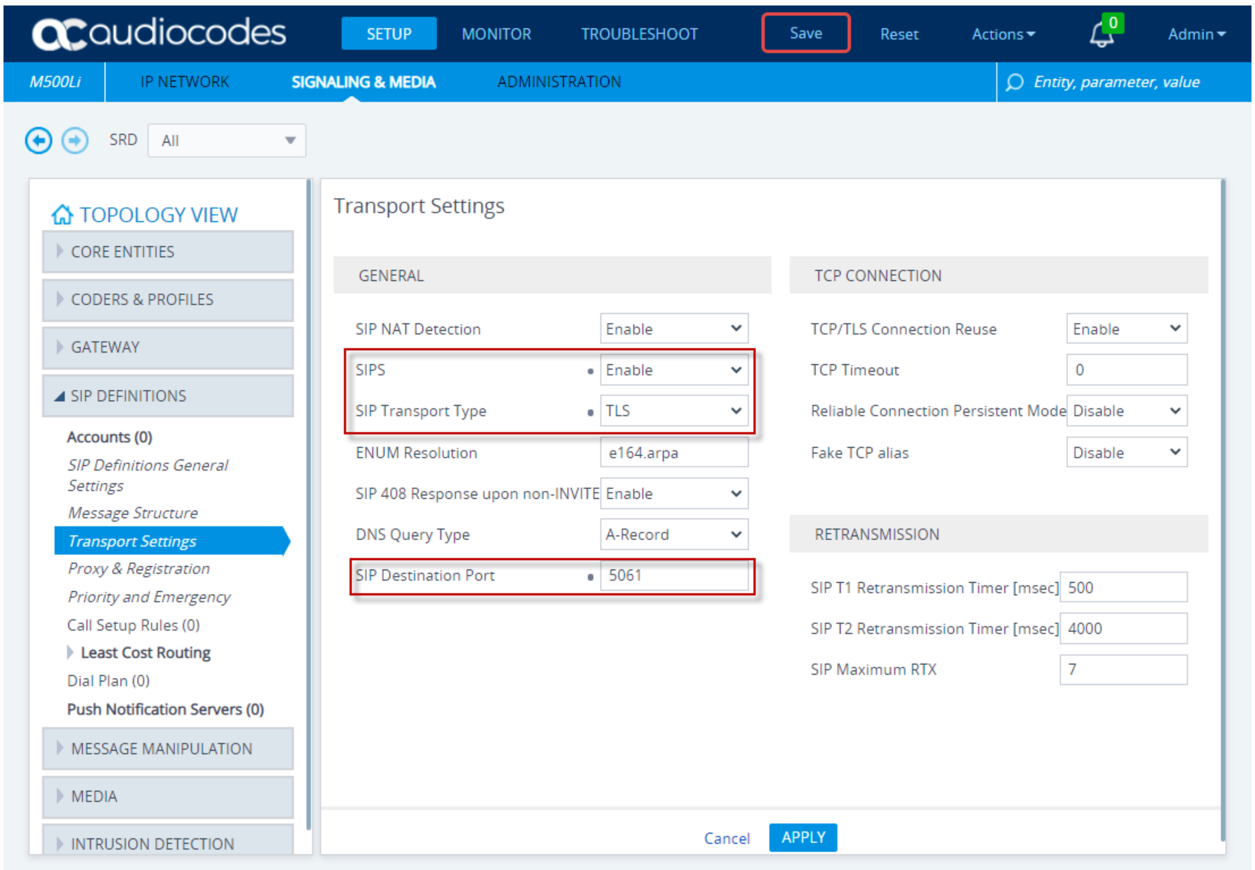


8.7.2. Transport Settings

Click on **SETUP** in the title bar. Select **Signaling & Media** in the toolbar. Select **SIP DEFINITIONS → Transport Settings** in the left pane.

- Select **Enable** for **SIPS**
- Select **TLS** for **SIP Transport Type**
- Select **5061** for the **SIP Destination Port**

Click **APPLY.**

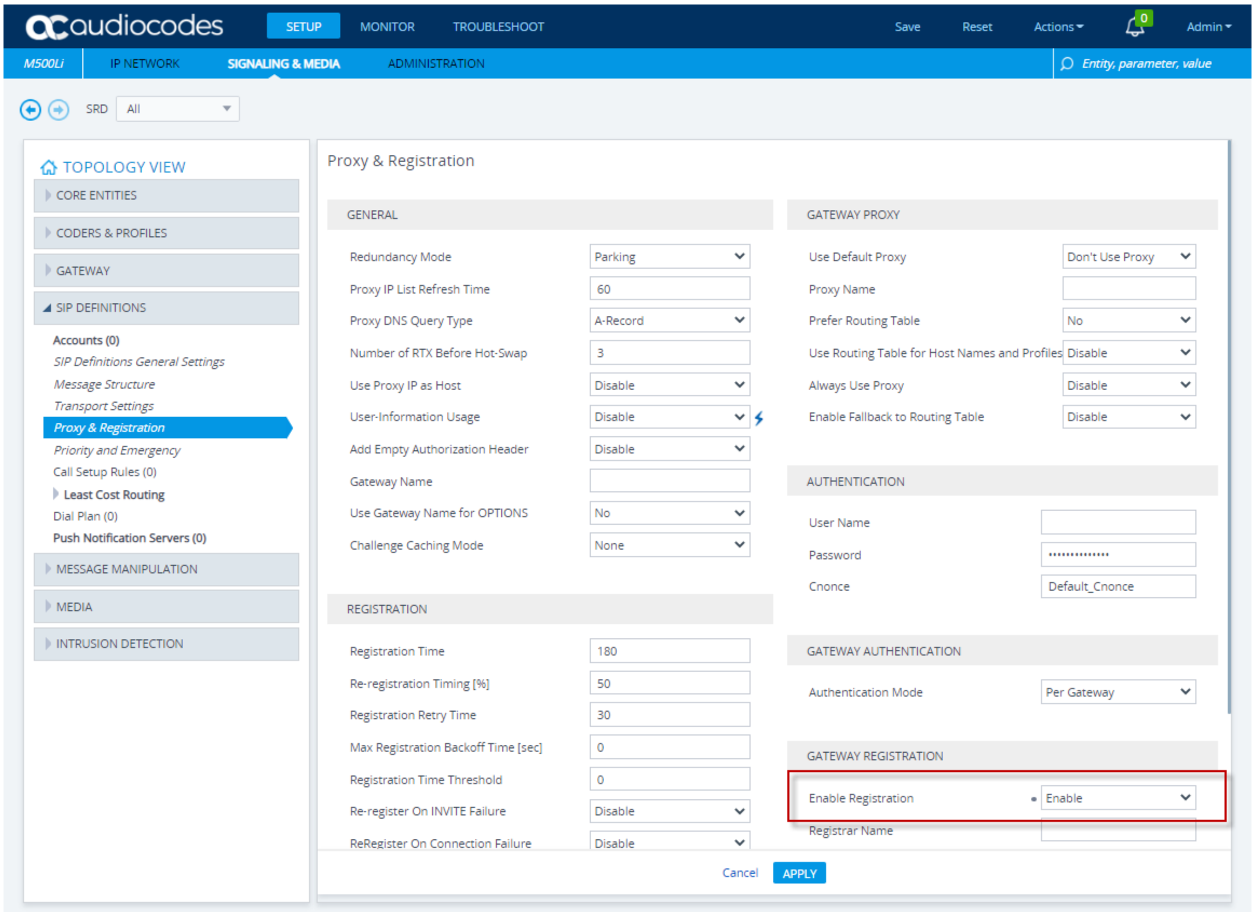


8.7.3. Proxy Registration

Click on **SETUP** in the title bar. Select **Signaling & Media** in the toolbar. Select **SIP DEFINITIONS → Proxy & Registration** in the left pane.

- Select **Enable** for **Enable Registration**

Click **Apply.**



8.7.4. Administer Call Detail Records

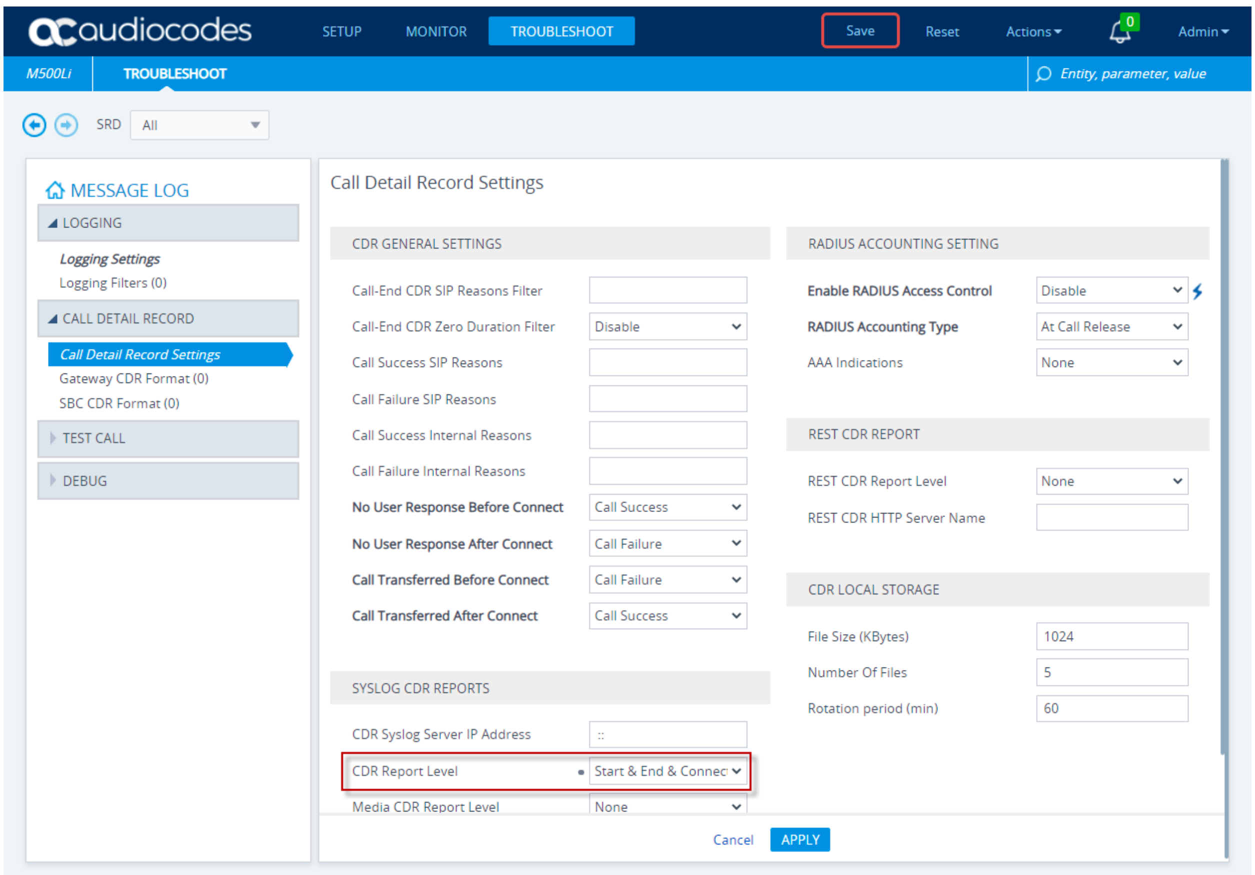
Call Detail Records (CDR) contains vital statistic information on calls made from the device. Configure call detail records for any needed troubleshooting. Enable the Syslog feature as in **Section 8.4** and configure a collecting server address. Refer to **Reference [4]** in **Section 11** for details.

.

Click on **TROUBLESHOOT** in the title bar. Select **CALL DETAIL RECORD → Call Detail Record Settings** in the left pane.

- Select **Start & End & Connect** for the **CDR Report Level**

Click **APPLY.**



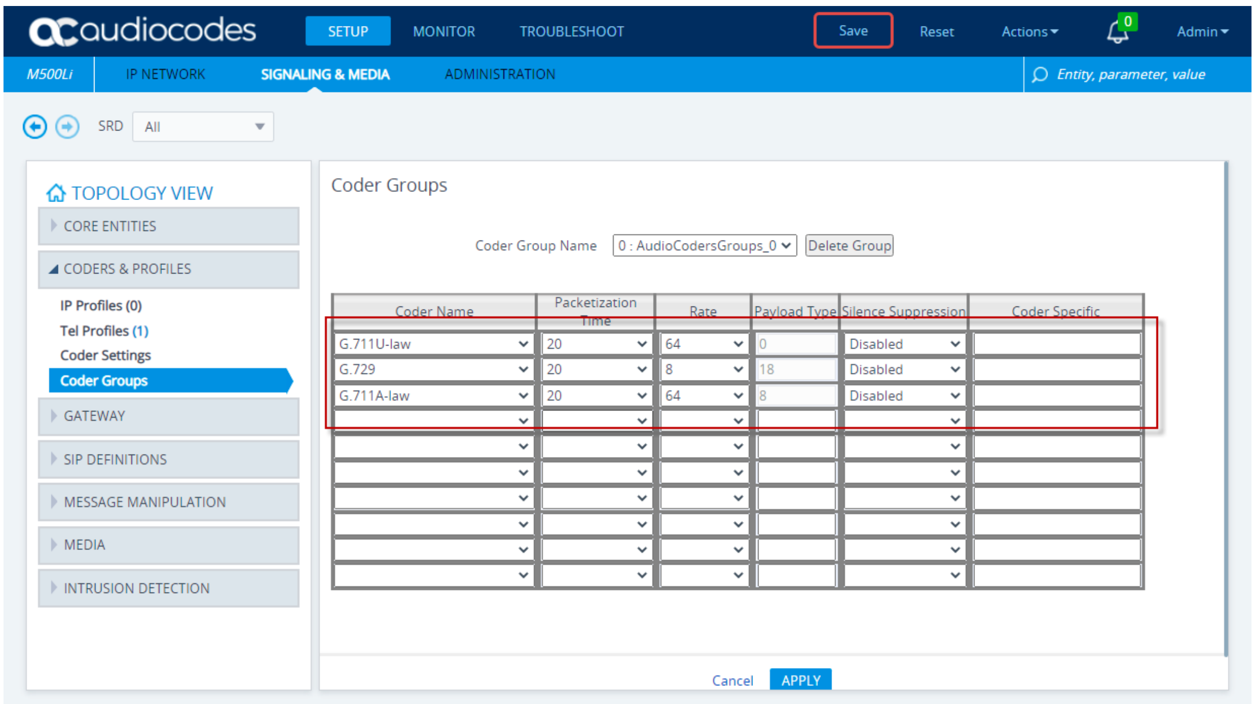
8.8. Administer Coder Groups

The default Coder Group can have all needed coders assigned and used. Interoperability testing used multiple Coder Groups with different coders to assign to the Tel Profile administered in **Section 8.8.1**.

Select **SETUP** from the title bar. Select **Signaling & Media** from the toolbar. Select **CODERS & PROFILES → Coder Groups.**

- Select the desired coder group in the **Coder Group Name.** The default Coder Group is **0:AudioCodersGroups_0**
- Select the **Coder Name** in the dropdown for each row. For the default Coder Group, **G.711U-law, G.729, and G.711A-law**.

Click **APPLY.**

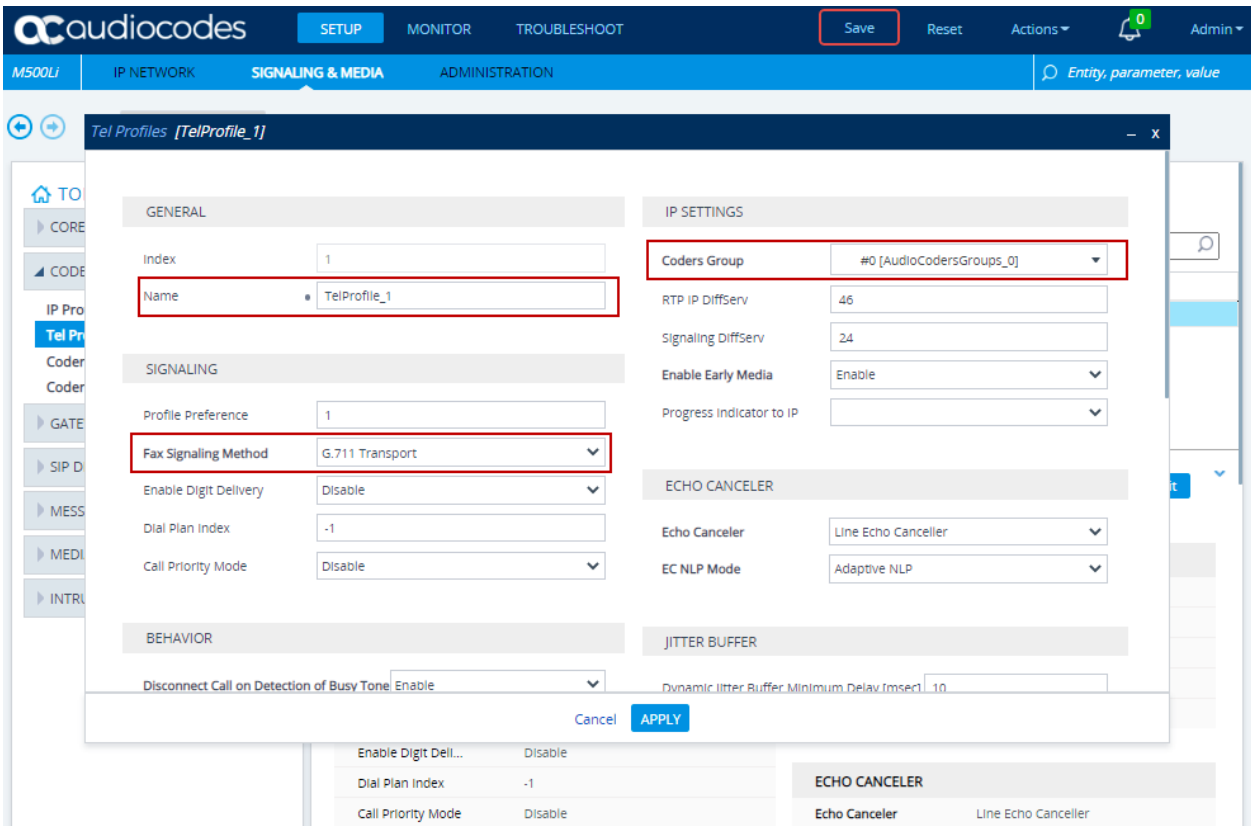


8.8.1. Administer Tel Profiles

Tel profiles are used to assign different Coder Groups and specify Fax signalling method.
NOTE: The Tel Profile is not needed if using the default Coder Group only.
Select **SETUP** from the title bar. Select **Signaling & Media** from the toolbar. Select **CODERS & PROFILES → Tel Profiles**

- Click the **+ NEW** button (not shown)
- Select an appropriate **Name** e.g., **TelProfile_1**
- Select the desired **Coders Group**
- Select **G.711 Transport** for **Fax Signaling Method**

Click **APPLY.**



8.9. Configure Core Administration

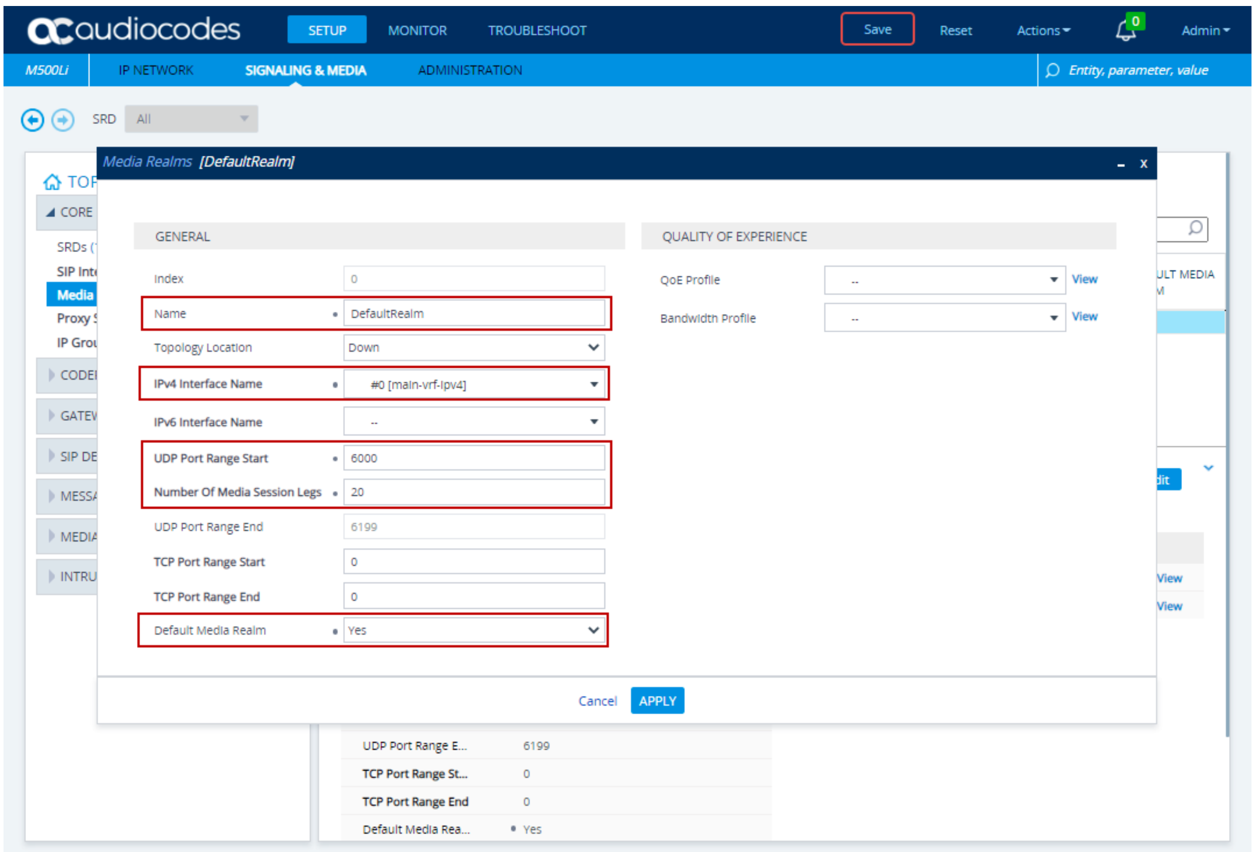
Core administration will configure Media and signaling parameters

8.9.1. Media Realm

Click on **SETUP** in the title bar. Select **Signaling & Media** in the toolbar. Select **CORE ENTITIES → Media Realms** in the left pane. Click **Edit** (not shown)

- Input an appropriate **Name**
- Select the default **Ipv4 Interface Name**
- Input **6000** for the **UDP Port Range Start**
- Input **20** for **Number Of Media Session Legs**
- Select **Yes** for the **Default Media Realm**

Click **Apply.**

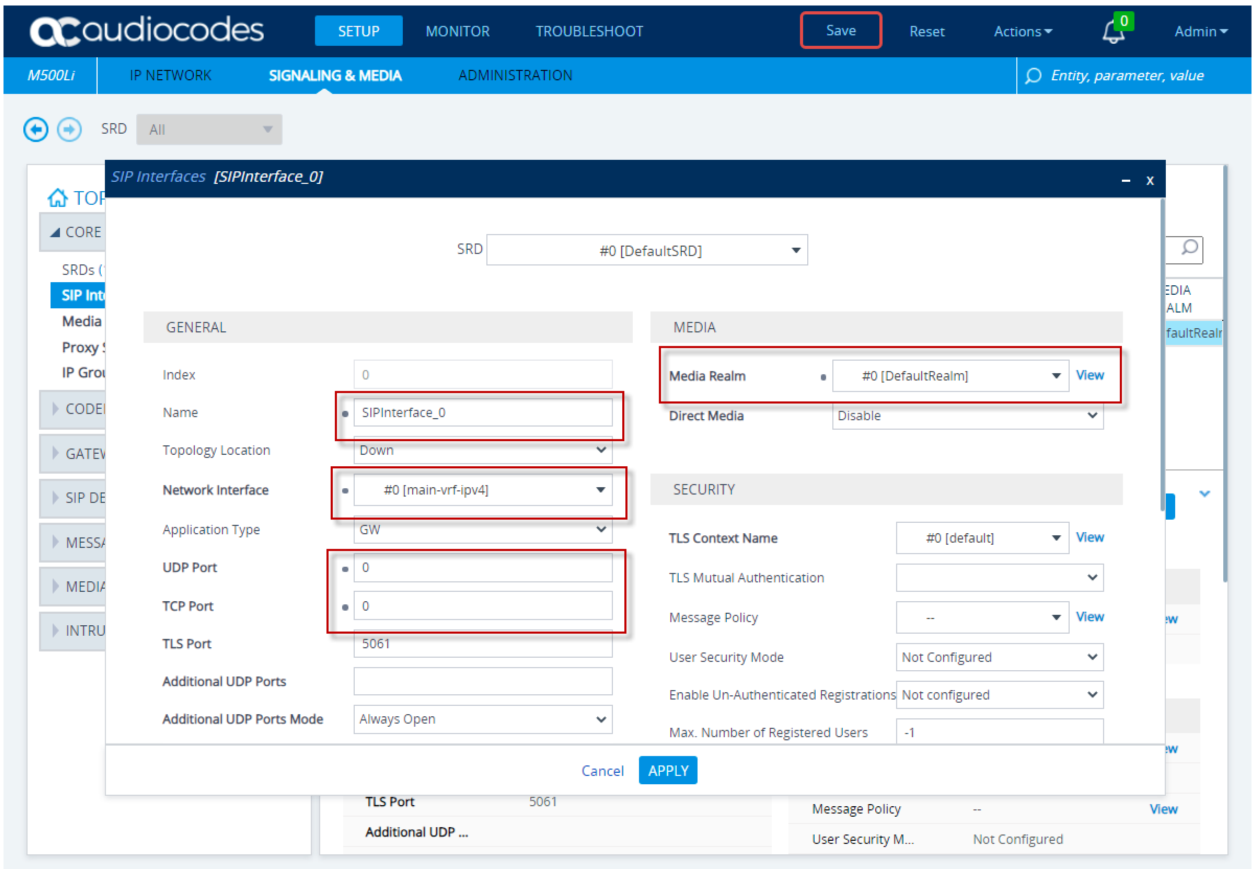


8.9.2. SIP Interface

Click on **SETUP** in the title bar. Select **Signaling & Media** in the toolbar. Select **CORE ENTITIES → SIP Interfaces** in the left pane. Click **Edit** (not shown)

- Input an appropriate **Name**. Interoperability testing used **SIPInterface_0**
- Select **#0 [DefaultRealm]** for the **Media Realm**
- Select **#0 [main-vrf-ipv4]** for **Network Interface**
- Input **0** for the **UDP Port** and **TCP Port** as interoperability testing used TLS

Click **APPLY.**

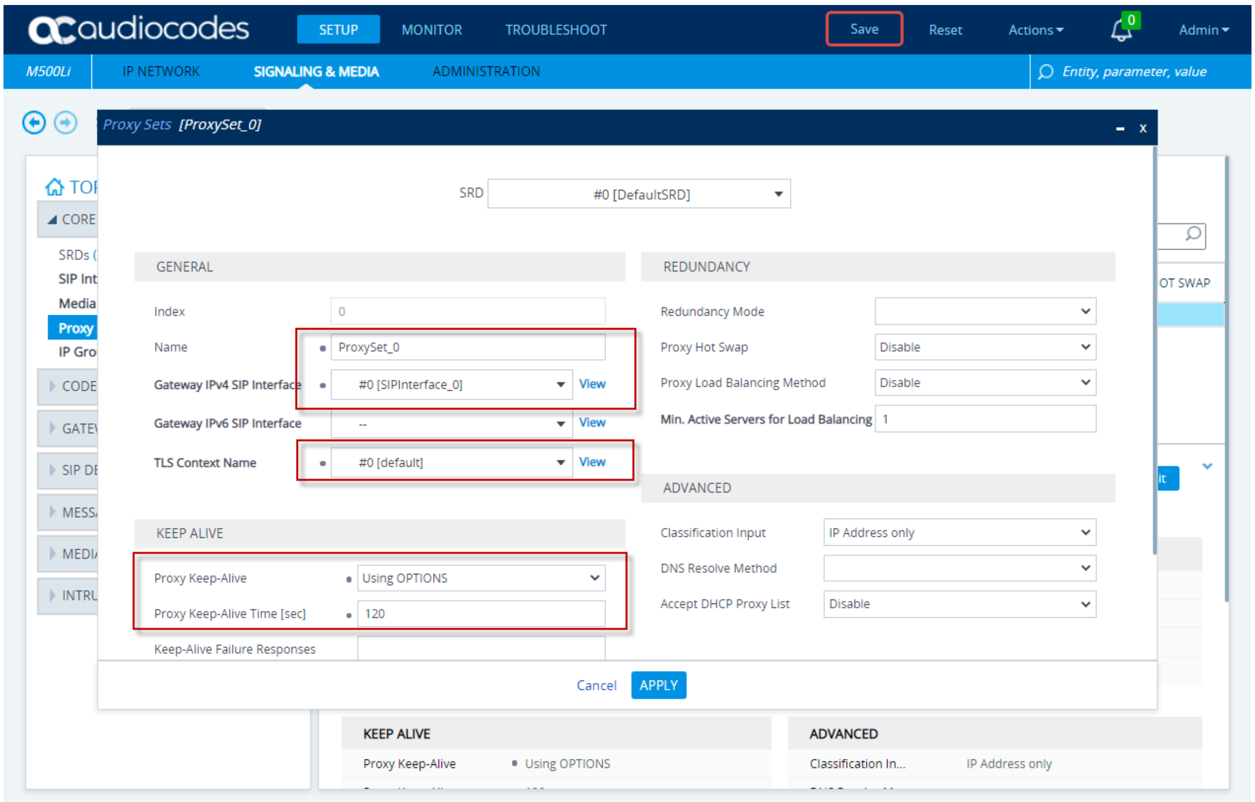


8.9.3. Proxy Set

Click on **SETUP** in the title bar. Select **Signaling & Media** in the toolbar. Select **CORE ENTITIES → Proxy Sets** in the left pane. Click **Edit** (not shown).

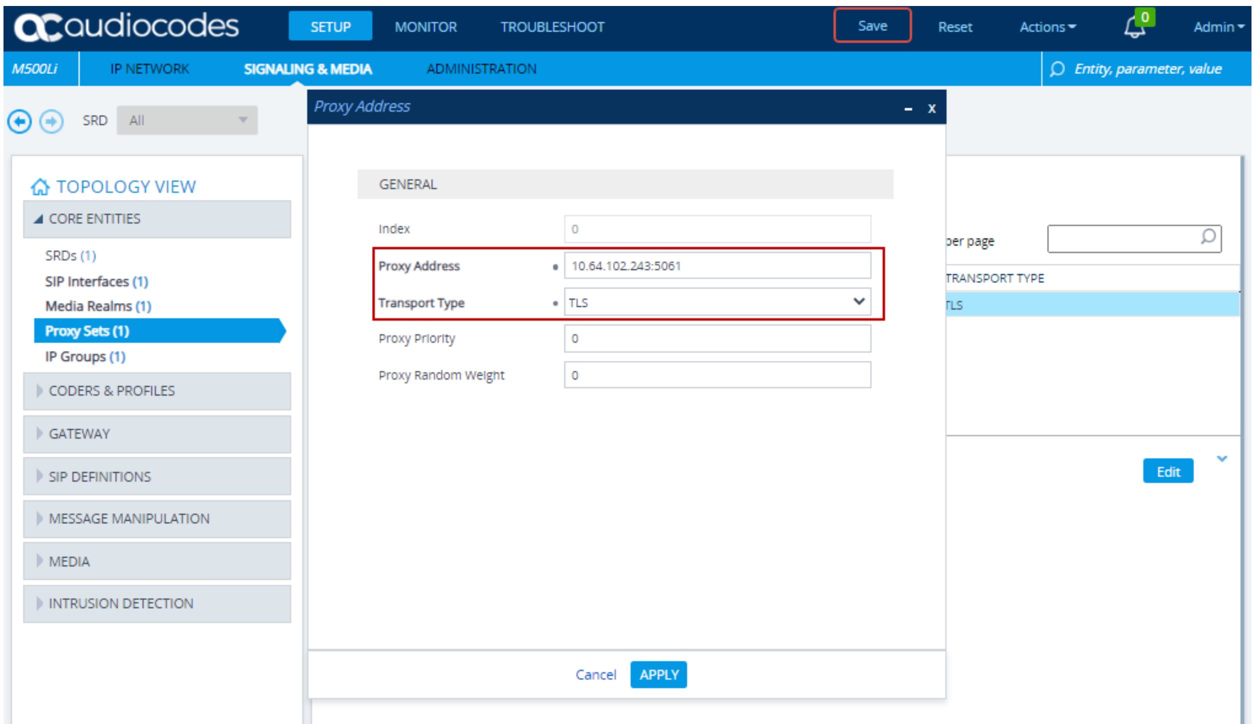
- Input an appropriate **Name**. Interoperability testing used **ProxySet_0**
- Select **#0 [SIPInterface_0]** for **Gateway Ipv4 SIP Interface**
- Select **#0 [default]** for **TLS Context Name**
- Select **Using OPTIONS** for **Proxy Keep-Alive**
- Input **120** for **Proxy Keep-Alive Time (sec)**

Click **APPLY.**



In the **Proxy Sets** pane, select the **Proxy Address Items** link at the bottom of the pane. Click **Edit** (not shown):

- If the Mediant 500Li is located within the enterprise, input Session Manager IP address used in **Section 5.4** for **Proxy Address** e.g., **10.64.110.212** (not shown)
- If the Mediant 500Li is located in a remote location, input the SBCE external interface B1 IP address used by remote workers in **Section 7.2** for **Proxy Address** e.g., **10.64.102.243**
- Select **TLS** for the **Transport Type**
  Click **APPLY.**

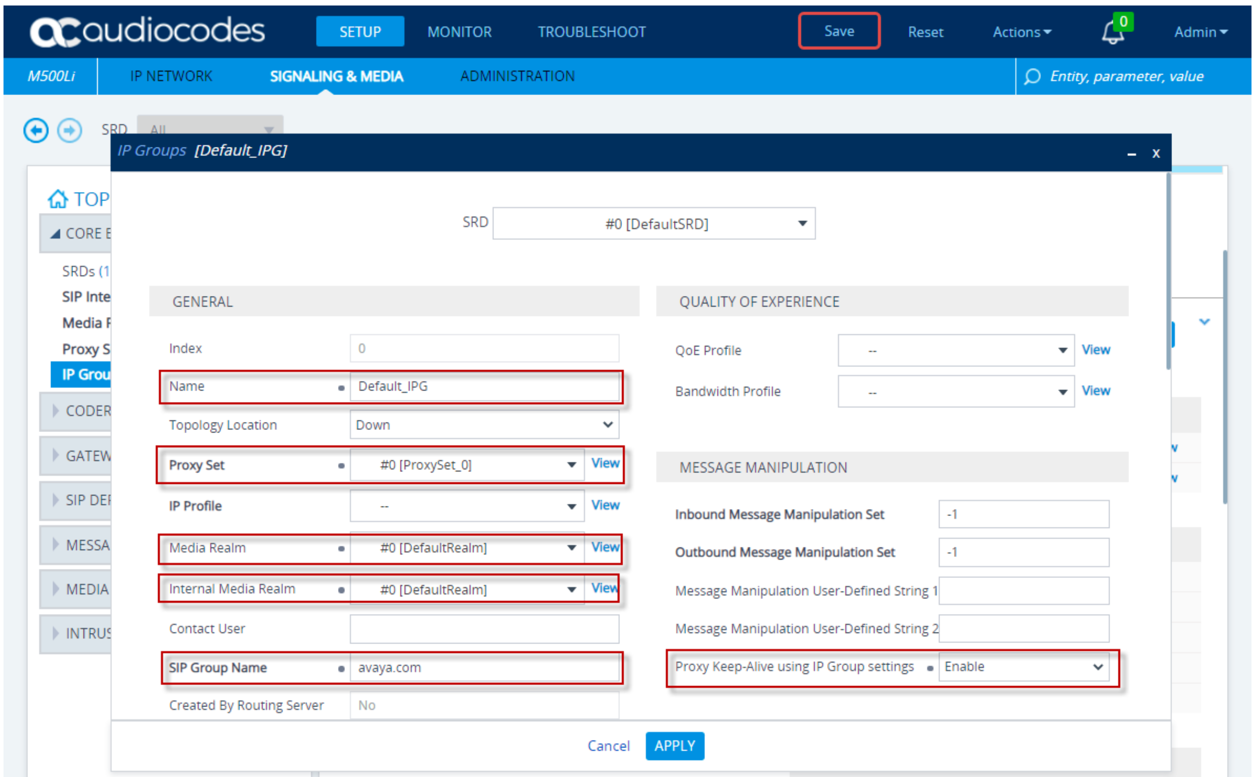


8.9.4. IP Groups

Click on **SETUP** in the title bar. Select **Signaling & Media** in the toolbar. Select **CORE ENTITIES → IP Groups** in the left pane. Click **Edit** (not shown)

- Input an appropriate **Name**. Interoperability testing used **Default_IPG**
- Select **#0 [ProxySet_0]** for **Proxy Set**
- Select **#0 [DefaultRealm]** for **Media Realm**
- Select **#0 [DefaultRealm]** for **Internal Media Realm**
- Input **avaya.com** for **SIP Group Name**
- Select **Enable** for **Proxy Keep-Alive using IP Group settings**

Click **APPLY.**



8.10. Administer Gateway

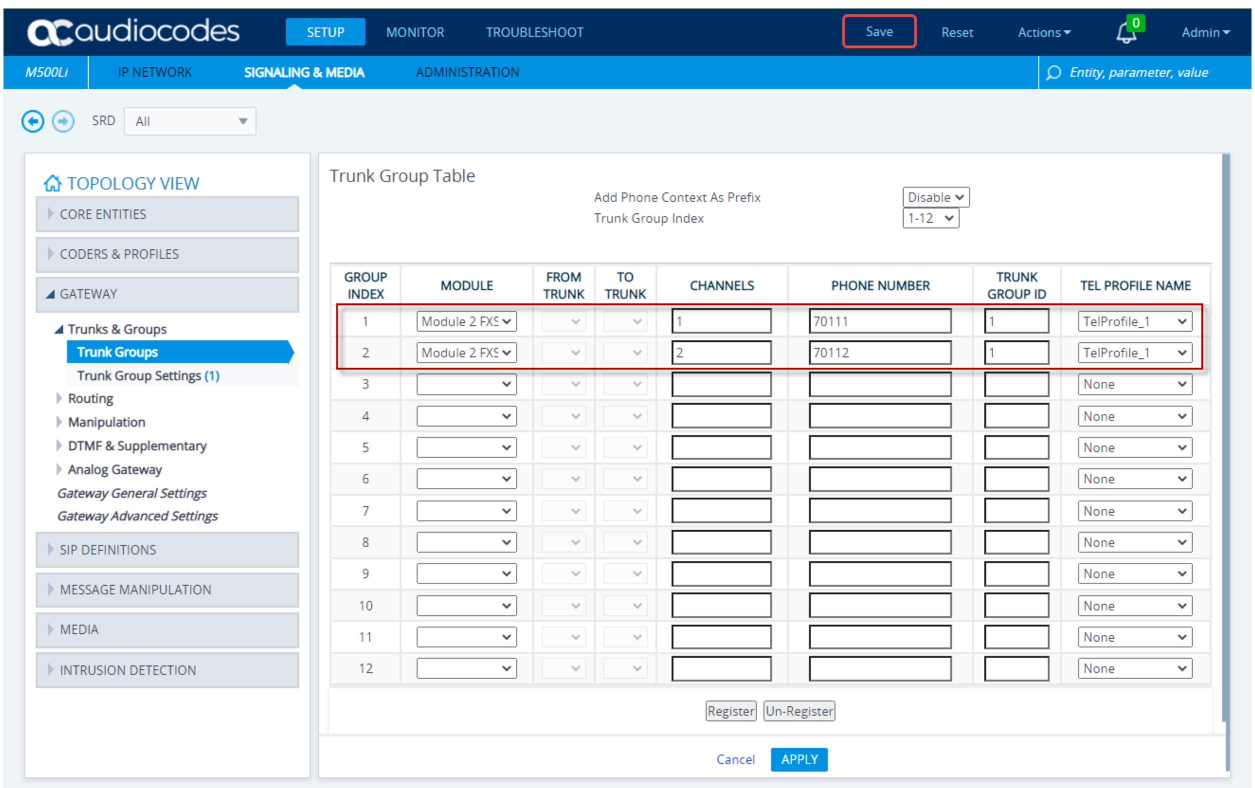
This section covers FXS port administration

8.10.1. Trunk Groups

Click on **SETUP** in the title bar. Select **Signaling & Media** in the toolbar. Select **Gateway → Trunks & Groups → Trunk Groups** in the left pane. Administer two analog extensions here. **Note:** The module selection refers to specific FXS ports in the back of the Mediant 500Li hardware.

- Select **Module 2 FXS** for the **MODULE**
- Select **1** for the **CHANNELS**
- Input the extension from the SIP users administered in **Section 6.8.5** for the **PHONE NUMBER**
- Select **1** for the **TRUNK GROUP ID**
- Select **TelProfile_1** for the **TEL PROFILE NAME**
- Repeat input using the second user extension for **PHONE NUMBER** and **2** for **CHANNELS** to create the second extension

Click **APPLY.**

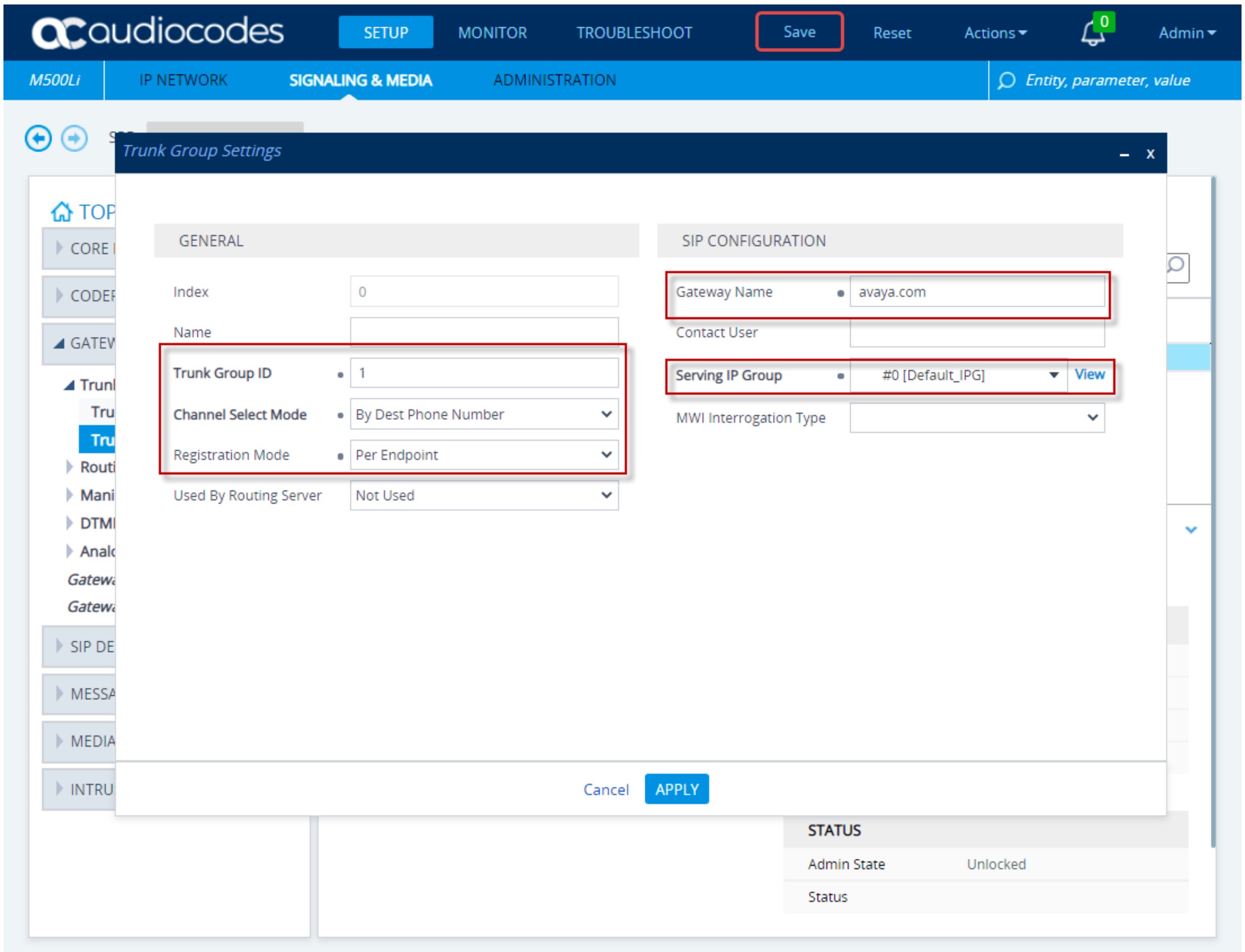


8.10.2. Trunk Group Settings

Click on **SETUP** in the title bar. Select **Signaling & Media** in the toolbar. Select **Gateway → Trunks & Groups → Trunk Group Settings** in the left pane. Click **Edit** (not shown)

- Input **1** for the **Trunk Group ID**
- Select **By Dest Phone Number** for **Channel Select Mode**
- Select **Per Endpoint** for **Registration Mode**
- Enter **avaya.com** for the Gateway
- Select **#0 [Default_IPG]** for the **Serving IP Group**

Click **APPLY.**



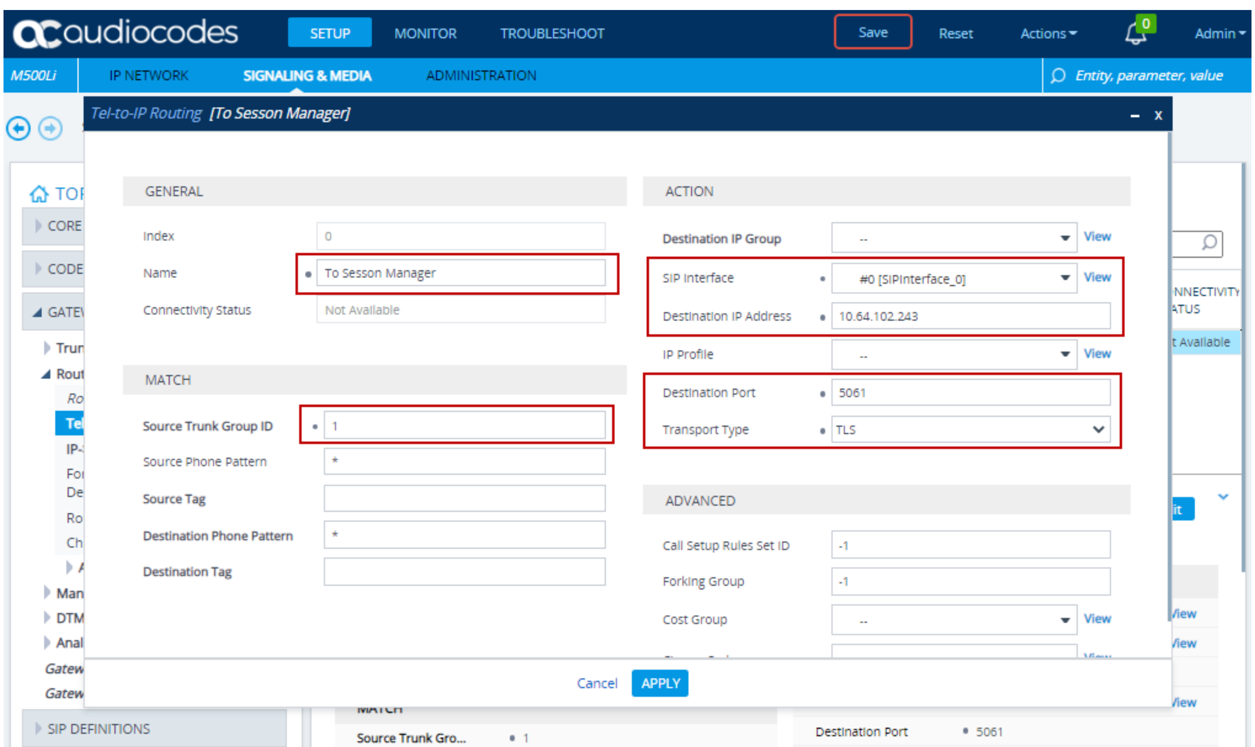
8.10.3. Tel-to-IP Routing

Click on **SETUP** in the title bar. Select **Signaling & Media** in the toolbar. Select **Gateway → Routing → Tel-to-IP Routing** in the left pane. Click **Edit** (not shown)

- Enter an appropriate value for **Name**
- Input **1** for the **Source Trunk Group ID**
- Select **#0 [SIPInterface_0]** for the **SIP Interface**
- If the Mediant 500Li is located within the enterprise, input Session Manager IP address used in **Section 5.4** for **Destination IP Address** e.g., **10.64.110.212** (not shown)
- If the Mediant 500Li is located in a remote location, input the SBCE external interface B1 IP address used by remote workers in **Section 7.2** for **Destination IP Address** e.g., **10.64.102.243**
- Input **5061** for the **Destination Port**
- Select **TLS** for the **Transport type**

Click **APPLY.**

Note: Tel-IP-Routing is set up to use the destination address. Mediant 500Li can be configured to use the default proxy which negates the need for the Tel-to-IP Routing configuration. Refer to **reference [4]** for more information

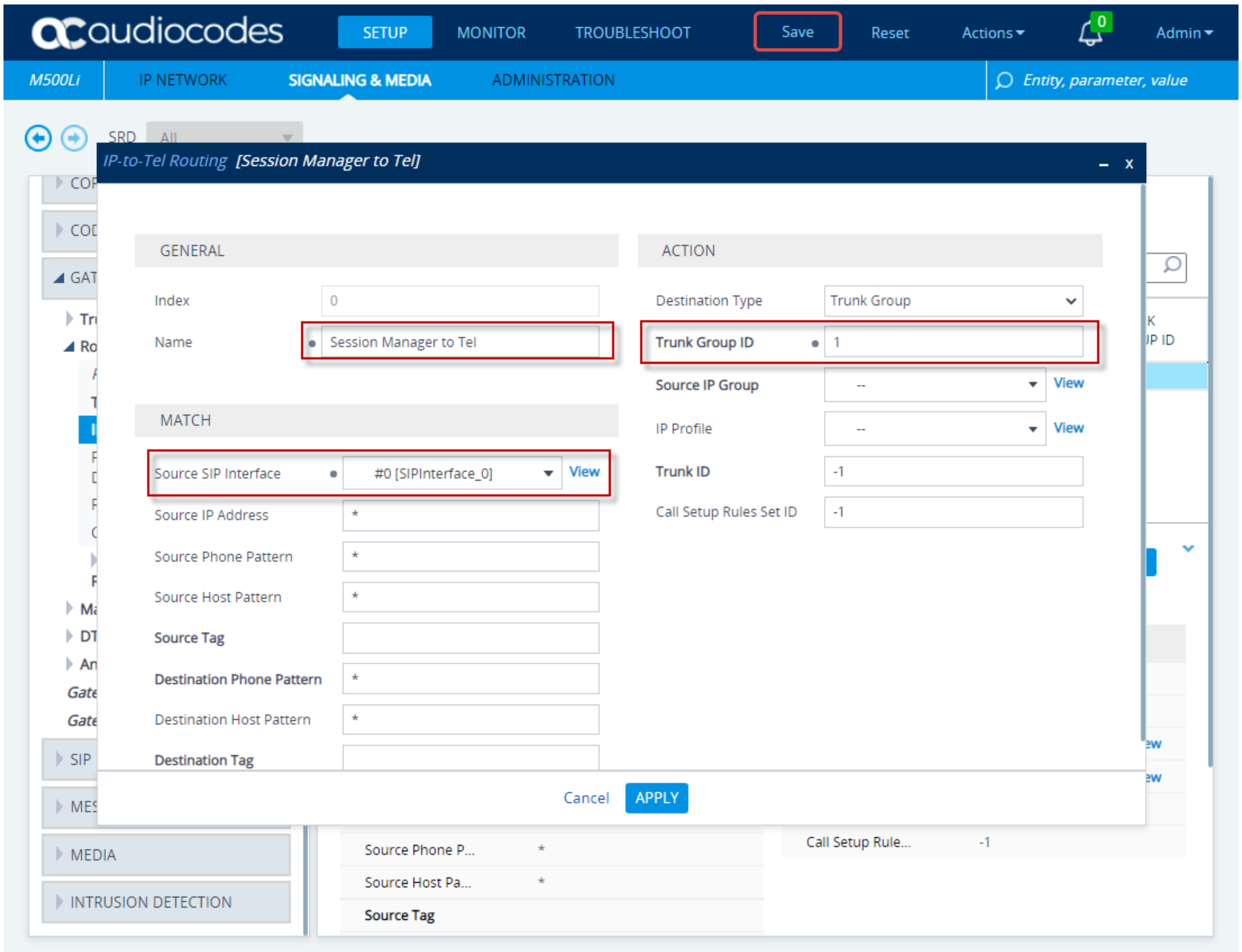


8.10.4. IP-to-Tel Routing

Click on **SETUP** in the title bar. Select **Signaling & Media** in the toolbar. Select **Gateway → Routing → IP-to-Tel Routing** in the left pane. Click **Edit** (not shown)

- Enter an appropriate value for **Name** e.g., **Session Manager to Tel**
- Select **#0 [SIPInterface_0]** for the **Source SIP Interface**
- Select **1** for the **Trunk Group ID**

Click **Apply.**



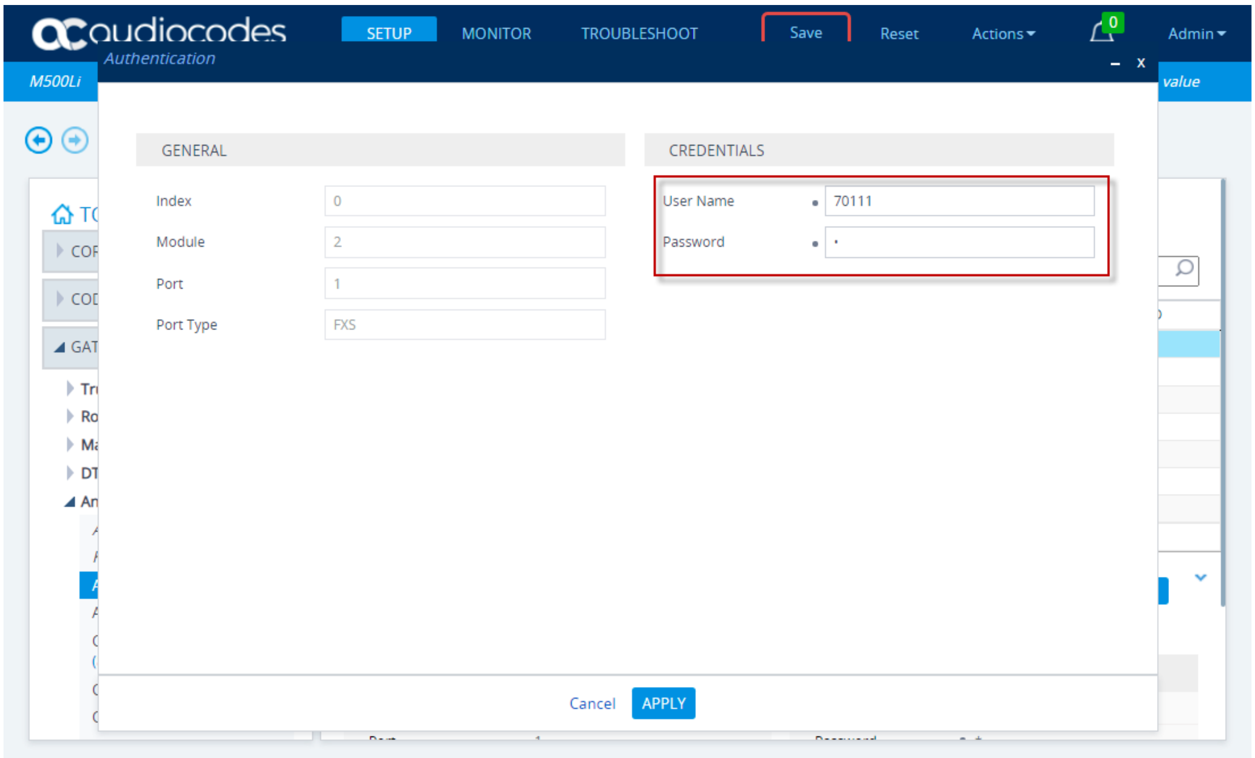
8.10.5. Authentication

Click on **SETUP** in the title bar. Select **Signaling & Media** in the toolbar. Select **Gateway → Analog Gateway → Authentication** in the left pane. Select the first entry created from **Section 8.10.1**. Click **Edit** (not shown)

- Input the SIP user created in **Section 6.8** above for **User Name**
- Input the analogous SIP user password for **Password**

Click **APPLY.**

Repeat administration for the second extension.

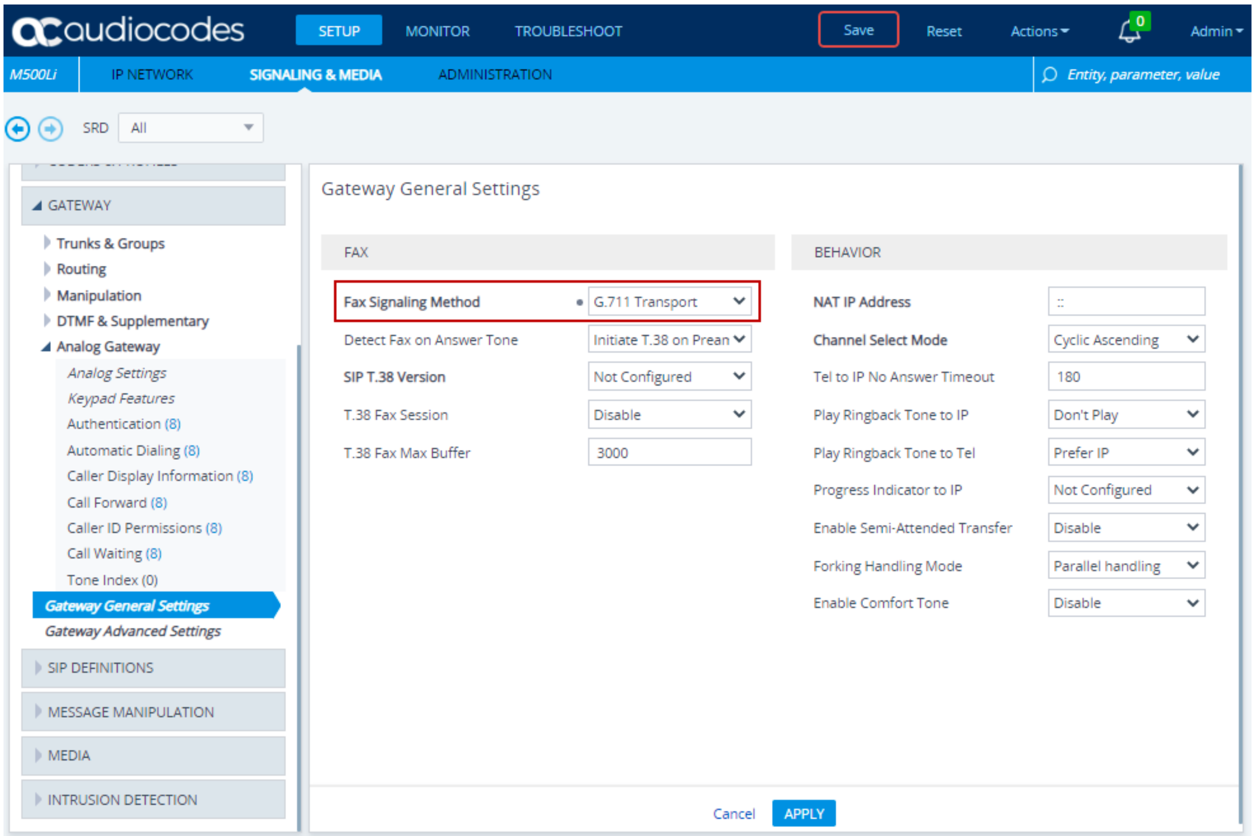


8.10.6. Gateway General Settings

Click on **SETUP** in the title bar. Select **Signaling & Media** in the toolbar. Select **Gateway → Gateway General Settings** in the left pane.

- Select **G.711 Transport** for the **Fax Signaling Method**

Click **APPLY.**

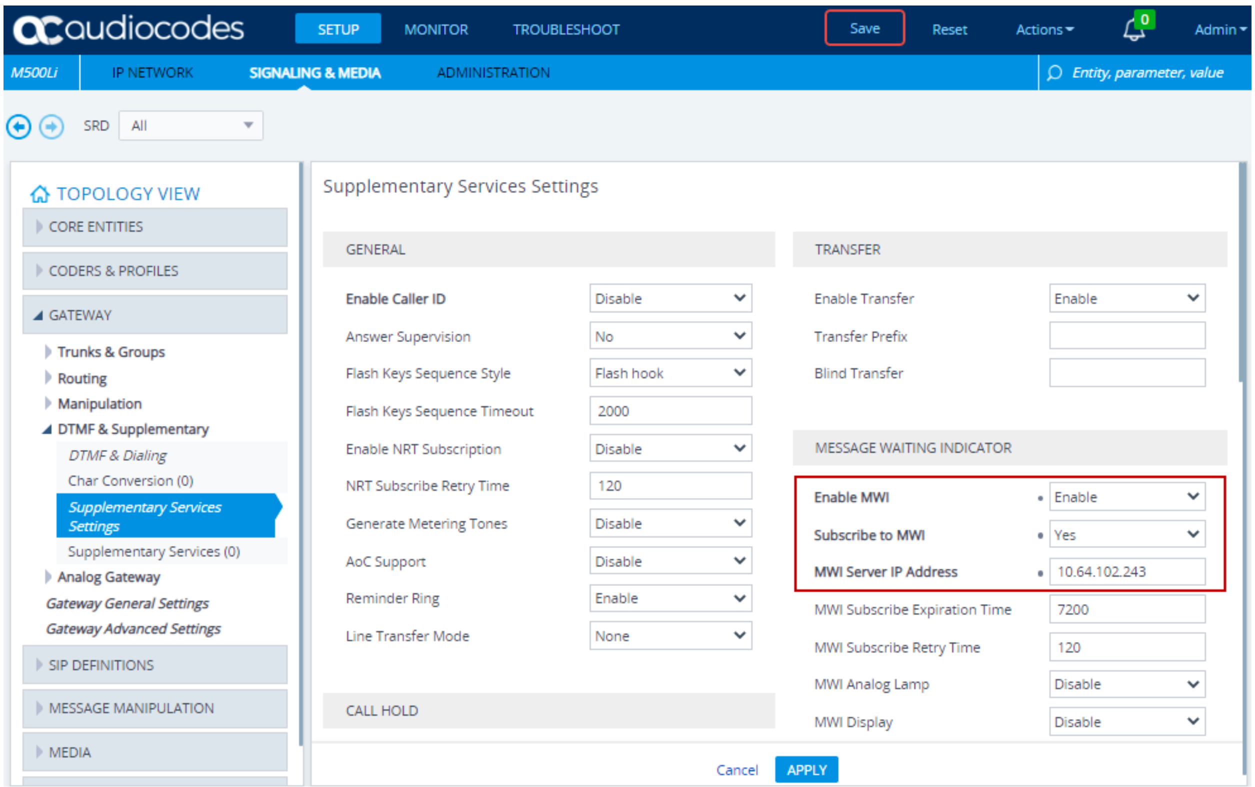


8.10.7. Supplementary Services Settings

Click on **SETUP** in the title bar. Select **Signaling & Media** in the toolbar. Select **Gateway → DTMF & Supplementary → Supplementary Services Settings** in the left pane.

- Select **Enable** for **Enable MWI**
- Select **Yes** for **Subscribe to MWI**
- If the Mediant 500Li is located within the enterprise, input Session Manager IP address used in **Section 5.4** for **MWI Server IP Address** e.g., **10.64.110.212** (not shown)
- If the Mediant 500Li is located in a remote location, input the SBCE external interface B1 IP address used by remote workers in **Section 7.2** for **MWI Server IP Address** e.g., **10.64.102.243**

Click **APPLY.**



9. Verification Steps

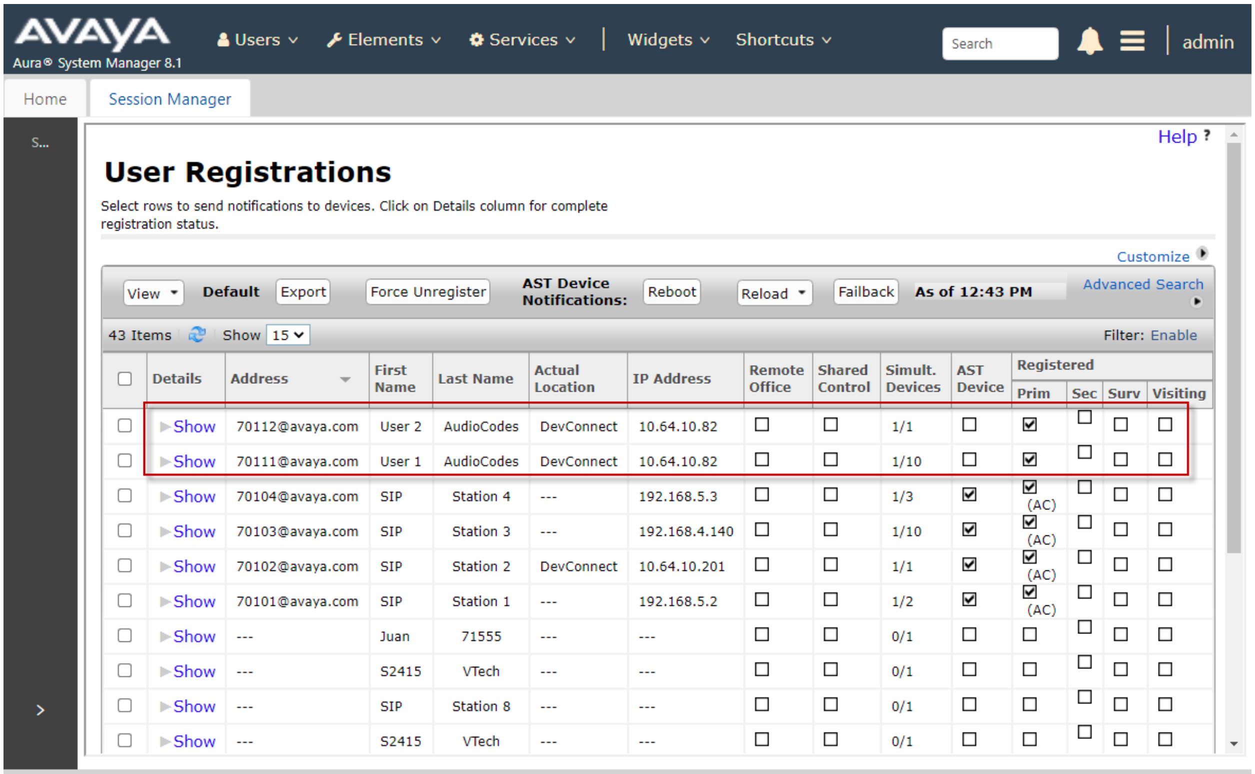
This section provides the tests that can be performed to verify proper configuration of Communication Manager, Session Manager, SBCE, and Mediant 500Li.

9.1. Avaya Aura® Communication Manager and Avaya Aura® Session Manager

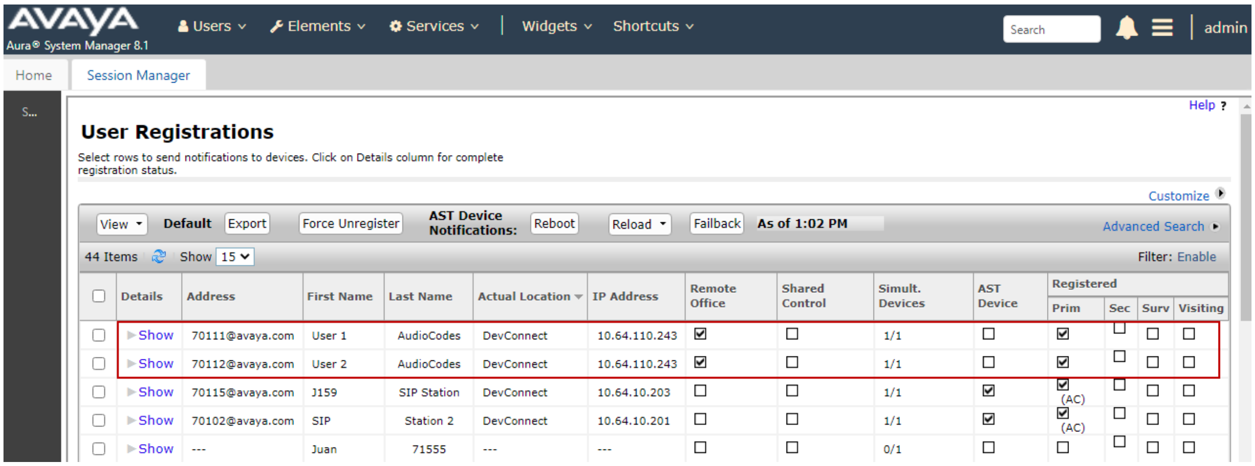
Verify SIP trunks to Session Manager are in service via SAT, using **status trunk *n***, where ***n*** is the number of the trunk configured in **Section 5.6.1**. The **Service State** column should show **in-service/idle.**

```
status trunk 1

                             TRUNK GROUP STATUS

Member   Port     Service State      Mtce Connected Ports
                                     Busy

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

Verify successful registration from AudioCodes Mediant 500Li to Session Manager via the System Manager console. Navigate to **Home → Session Manager → System Status → User Registration.** The SIP users administered in **Section 6.8** should appear. When Mediant 500Li is located in the enterprise, the IP Address should be that of Mediant 500Li administered in **Section 8.1.1.**

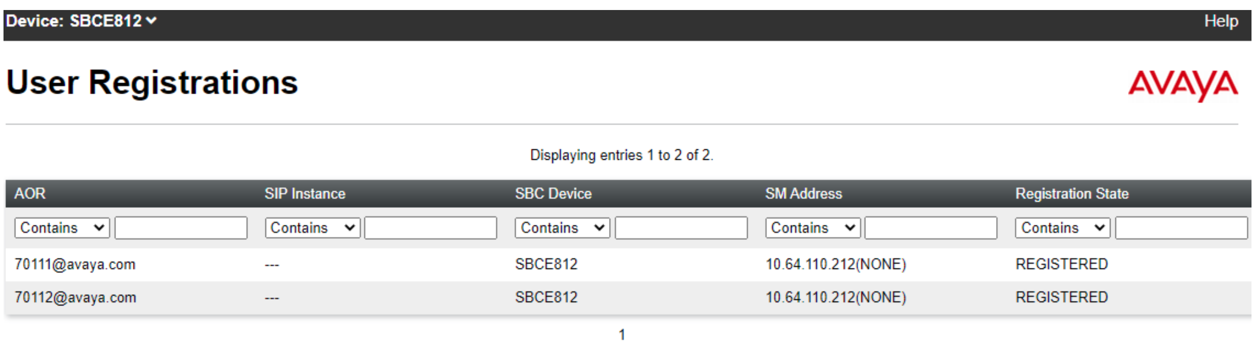


When Mediant 500Li is configured as Remote Worker, the IP address should be that of the internal Remote Worker interface IP shown in **Section 7.5**, e.g., **10.64.110.243**. The **Remote Office** column will be checked (representative image below if Session Manager Remote Access is configured). See **Reference [2]** in **Section 11** for administration details.



9.2. Avaya Session Border Controller for Enterprise

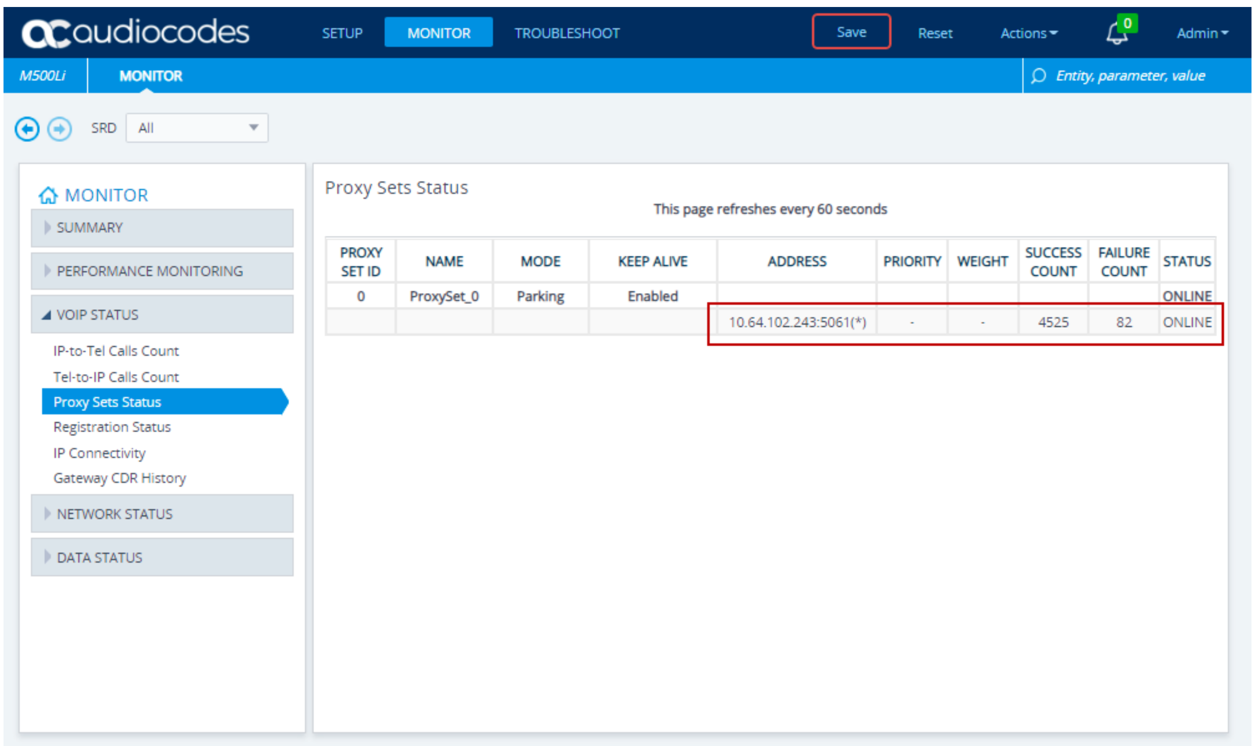
On SBCE, navigate to **Status → User Registrations**. Mediant 500Li SIP users should display as registered to Session Manager.



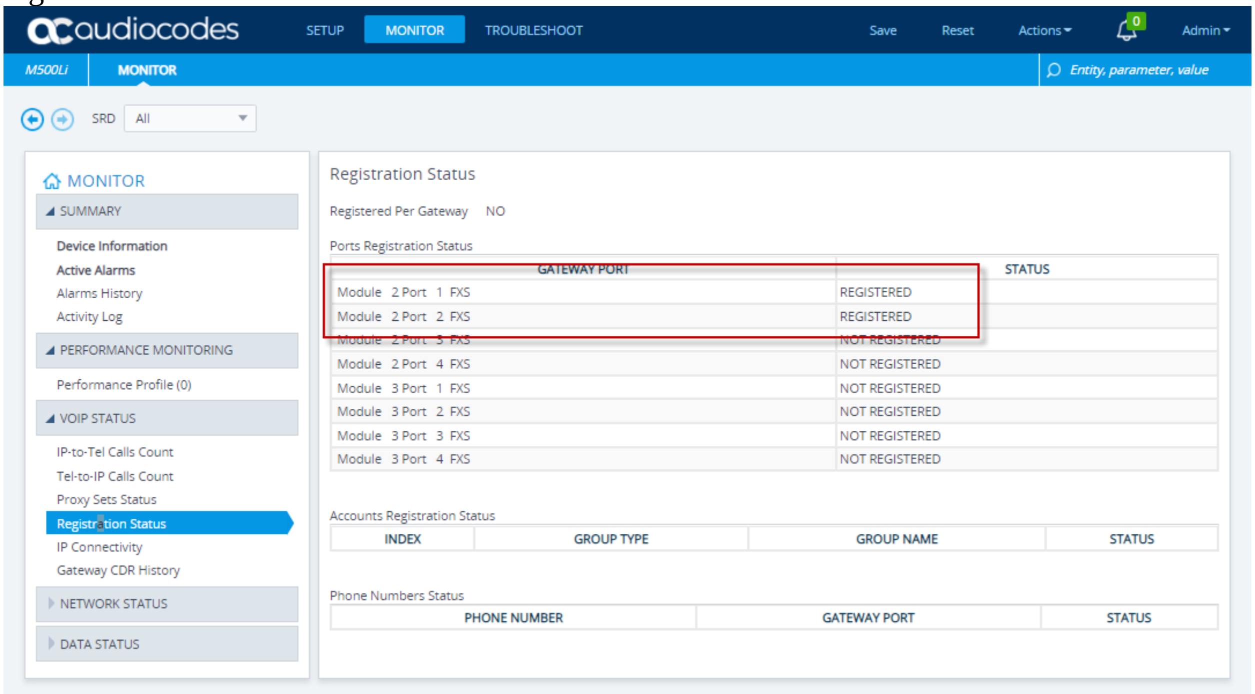
9.3. AudioCodes Mediant 500Li

Click on **MONITOR** in the title bar. Select **MONITOR** in the toolbar. Select **VOIP STATUS → Proxy Sets Status** in the left pane.

- If the Mediant 500Li is located within the enterprise, verify the **ADDRESS** is Session Manager's IP address configured in **Section 5.4** (not shown). Verify the **STATUS** is **ONLINE**.
- If the Mediant 500Li is located in a remote location, verify the **ADDRESS** is the SBCE external interface B1 IP address used by remote workers from **Section 7.2**. Verify the **STATUS** is **ONLINE**.



Click on **MONITOR** in the title bar. Select **MONITOR** in the toolbar. Select **VOIP STATUS → Registration Status** in the left pane. Verify the configured FXS lines from **Section 8.10.1** are registered.



Make calls to and from the analog lines to Avaya endpoints located in the enterprise and verify two-way audio path.

10. Conclusion

These Application Notes describe the configuration steps required for AudioCodes Mediant 500Li to successfully interoperate with Avaya Aura® Communication Manager and Avaya Aura® Session Manager in the enterprise configuration or a remote worker configuration employing Avaya Session Border Controller for Enterprise. All feature and serviceability test cases completed and pass with observations/exceptions noted in **Section 2.2**

11. Additional References

This section references the product documentation relevant for these Application Notes.

[1] *Administering Avaya Aura® Communication Manager*, Issue 12, Release 8.1.x, July 2021
[2] *Administering Avaya Aura® Session Manager*, Issue 10, Release 8.1.x, September 2021
[3] *Administering Avaya Session Border Controller for Enterprise*, Issue 5, Release 8.1.x, August 2021
[4] *AudioCodes Mediant 500Li MSBR Users Manual*, Version 7.2, March 18,2021:
[5] *AudioCodes Mediant 500Li MSBR CLI Reference Guide*
[6] *AudioCodes Mediant 500Li Security Setup CLI Configuration Guide*

AudioCodes Mediant 500Li General references:
https://www.audiocodes.com/library/technical-documents?productFamilyGroup=1647&productGroup=27483&versionGroup=Version+7.2

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.