Avaya Solution & Interoperability Test Lab

Application Notes for AudioCodes MediaPack 1288 with Avaya Aura® Communication Manager and Avaya Aura® Session Manager – Issue 1.0

Abstract

These Application Notes contain interoperability instructions for configuring AudioCodes MediaPack 1288 with Avaya Aura® Communication Manager and Avaya Aura® Session Manager. Compliance testing was conducted to verify interoperability.

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 in Thornton, CO.

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1. Introduction

AudioCodes MediaPack (MP) 1288 Analog VoIP gateways implement voice technology that connect analog telephones (FXS) to IP based PBX systems. In the compliance test, AudioCodes MP-1288 analog gateway was used to verify interoperability within an Avaya Aura® IP Telephony Environment.

The AudioCodes MP-1288 is a best-of-breed high density analog media gateway offering a cost-effective solution for organizations transitioning to all-IP that need to integrate large numbers of analog devices (such as legacy phones, fax machines and modems) into their new infrastructure.

- Ideal for VoIP deployments with large install base of analog devices
- Scalable solution with four capacity options: 288, 216, 144 and 72 ports
- Cost-effective - single management interface, single IP Address
- Reduced footprint - 3U chassis
- Designed for carrier-grade environments with 1+1 power supply modules and Ethernet redundancy
- Eliminate the need to stack and cable multiple small analog gateways

2. General Test Approach and Test Results

Interoperability compliance testing focused on verifying various inbound and outbound call flows between AudioCodes MP-1288, Communication Manager and Session Manager.

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

Analog lines on AudioCodes MP-1288 were configured to register as SIP users on Session Manager. SIP TLS and SRTP were utilized during this test effort. The following features and functionalities were covered during compliance testing:

- Incoming calls to AudioCodes MP-1288
- Outgoing calls from AudioCodes MP-1288
- Voice codecs G.711U, G.711A and G.729 using SRTP
- DTMF tone transmission with RFC2833
- Calls using various Avaya endpoints, including analog, H.323 and SIP.
- Basic features including Hold/Resume, DTMF transmission, Voicemail with Message Waiting Indicator (MWI).

2.2. Test Results

All test cases were successfully executed.

2.3. Support

Technical support for AudioCodes MP-1288 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

The reference configuration consists of Communication Manager, Session Manager, System Manager, Messaging, AudioCodes MP-1288, and a number of Avaya telephones. AudioCodes MP-1288 is used as an analog gateway that connects analog endpoints to Avaya IP telephony network. The Session Manager in the right block, managed through the System Manager in the same block, routes the calls between the different entities using SIP Trunks.

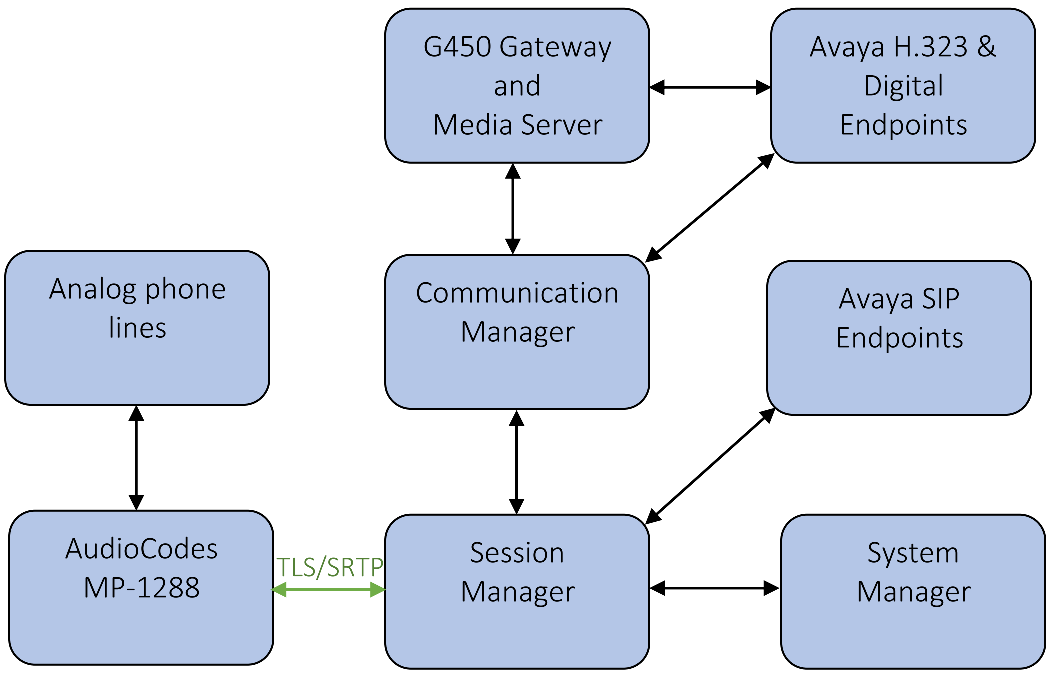


Figure 1: AudioCodes MP-1288 with Avaya Aura® Communication Manager and Avaya Aura® Session Manager

4. Equipment and Software Validated

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

Equipment/Software	Version
Avaya Aura® Communication Manager	8.1.1.0.0.890.25763
Avaya G450 Media Gateway	40.20.1
Avaya Aura® Session Manager	8.1.1.0.811021
Avaya Aura® System Manager	8.1.1.0.0310782
Avaya Aura® Media Server	8.0.2.61
Avaya 96x0 Series IP Telephones (H.323)	3.2.8
Avaya J100 Series Telephones (SIP)	4.0.4
Avaya 96x1 Series IP Telephones (H.323) Avaya J100 Series Telephones (H.323)	6.8.3
Avaya 96x1 Series IP Telephones (SIP)	7.1.7
Avaya one-X® Communicator (SIP)	6.2.10
Avaya Digital Telephone	-
AudioCodes MediaPack 1288	7.20A.256.511

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 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      4
         Maximum Administered Remote Office Trunks: 12000      0
Max 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     0
               Maximum Video Capable IP Softphones:   2400     0
                   Maximum Administered SIP Trunks: 12000     10
  Max Administered Ad-hoc Video Conferencing Ports: 12000      0
   Max Number of DS1 Boards with Echo Cancellation: 688    0
```

On **Page 5,** verify **ISDN/PRI** and **Media Encryption Over IP** fields are 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
```

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. Configure this network region as follows:

- Set **Location** to **1**
- Set **Codec Set** to **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? y
   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
```

5.3. Administer IP Codec Set

Use the **change ip-codec-set *n*** command to configure IP codec set, where ***n*** is an existing codec set number. Configure this codec set as follows, on **Page 1**:

- Set **Audio Codec 1, 2** to **G.711MU, G.711A,** respectively.
- Set **Media Encryption 1** to **1-srtp-aescm128-hmac80.**

Note: G.711MU, G.711A and G.729AB were tested 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:
 4:
 5:
 6:
 7:

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

5.4. Administer IP Node Names

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

```
change node-names ip                                            Page   1 of   2
                                  IP NODE NAMES
    Name              IP Address
aes81               10.64.110.215
aes811              10.64.110.209
ams81               10.64.110.214
cms19               10.64.110.225
default             0.0.0.0
procr               10.64.110.213
procr6              ::
sm81                10.64.110.212
```

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. Configure this signaling group as follows:

- Set **Group Type** to **sip**
- Set **Transport Method** to **TLS**
- Set **Near-end Node Name** to **procr**
- Set **Far-end Node Name** to the configured Session Manager in **Section 5.4**, i.e. **sm81**
- Set **Far-end Network region** to the configured region in **Section 5.2**, i.e. **1**
- Set **Direct IP-IP Audio Connections** to **y**

```
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? 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:
                                             Bypass If IP Threshold Exceeded? n
Incoming Dialog Loopbacks: eliminate                  RFC 3389 Comfort Noise? n
         DTMF over IP: rtp-payload            Direct IP-IP Audio Connections? y
Session Establishment Timer(min): 3                     IP Audio Hairpinning? n
         Enable Layer 3 Test? y                    Initial IP-IP Direct Media? n
H.323 Station Outgoing Direct Media? n             Alternate Route Timer(sec): 6
```

Note: The Signaling Group, Trunk Group and Route Pattern for PSTN calls via PRI were pre-configured and are not shown in this document

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

- Set **Group Type** to **sip**
- Enter a **Group Name**, e.g. **SM Trunk**
- 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**, i.e. **1**
- Enter a desired number in **Number of Members** field

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

Group Number: 1                    Group Type: sip          CDR Reports: y
  Group Name: SM Trunk                    COR: 1        TN: 1        TAC: 101
   Direction: two-way       Outgoing Display? n
 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 **Number Format** to **private**

```
trunk-group 1                                           Page   3 of   5
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

                                                 Hold/Unhold Notifications? y
                                Modify Tandem Calling Number: no
               Send UCID? y



 Show ANSWERED BY on Display? y
```

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 a name in **Pattern Name** field
- For line 1, set **Grp No** to the trunk group configured in **Section 5.6**, i.e. **1**
- For line 1, set **FRL** to **0**

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

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:

- Add entries for trunk group configured in **Section 5.6**

Note: For compliance testing, 5-digit extensions beginning with 7 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
 7  1              1                           5     Total Administered: 1
                                                        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 AudioCodes MP-1288.

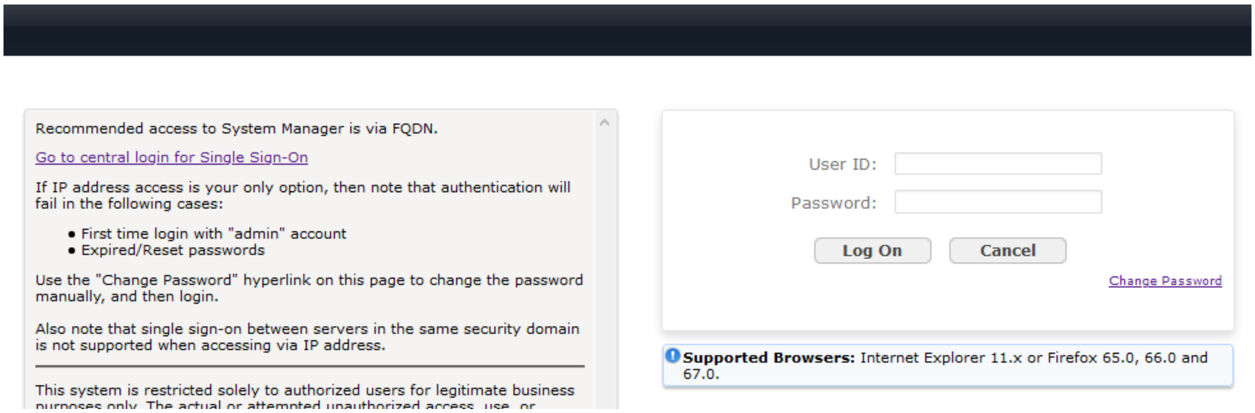
- Set **Dialed String** to starting digits of extensions that will be used, e.g. **701**
- Set **Min** and **Max** to **5** for 5 digit extensions
- Set **Route Pattern** to pattern configured in **Section 5.7**, i.e. **1**
- Set **Call Type** to **lev0**

Note: An entry to dial plan will need to be added for extension range used in this step.

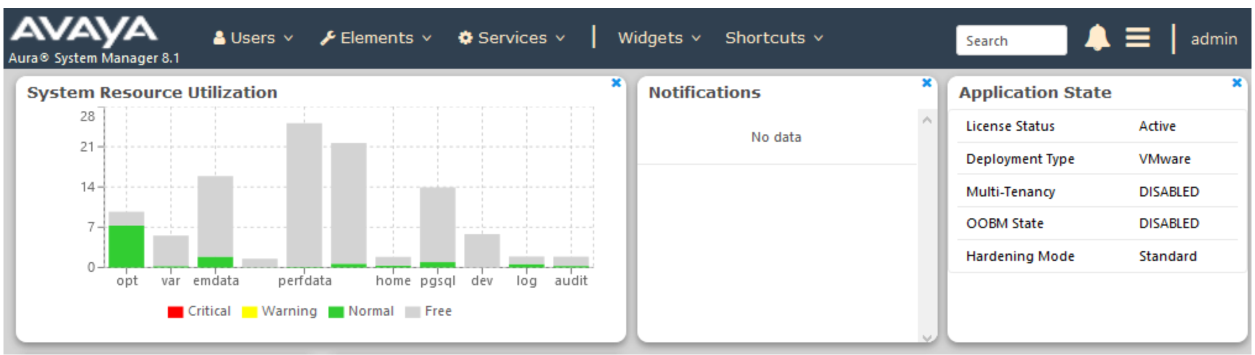
```
change aar analysis 701                                       Page   1 of   2
                             AAR DIGIT ANALYSIS TABLE
                                   Location: all           Percent Full: 0

          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

Access the Session Manager Administration web interface by entering https://<ip-address>/SMGR URL in a web browser, where <ip-address> is the IP address of System Manager.



Log in using appropriate credentials

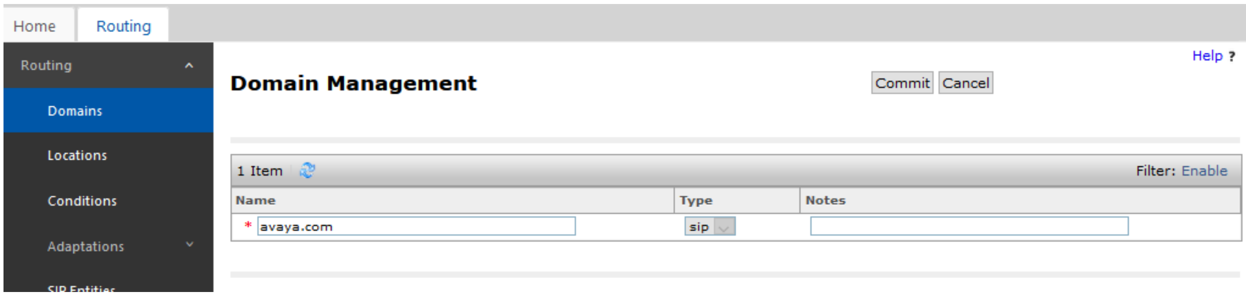


6.1. Add SIP Domain

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**) i.e. **avaya.com**
- Set **Type** to **sip**

Click **Commit** to save changes.



6.2. Add Location

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

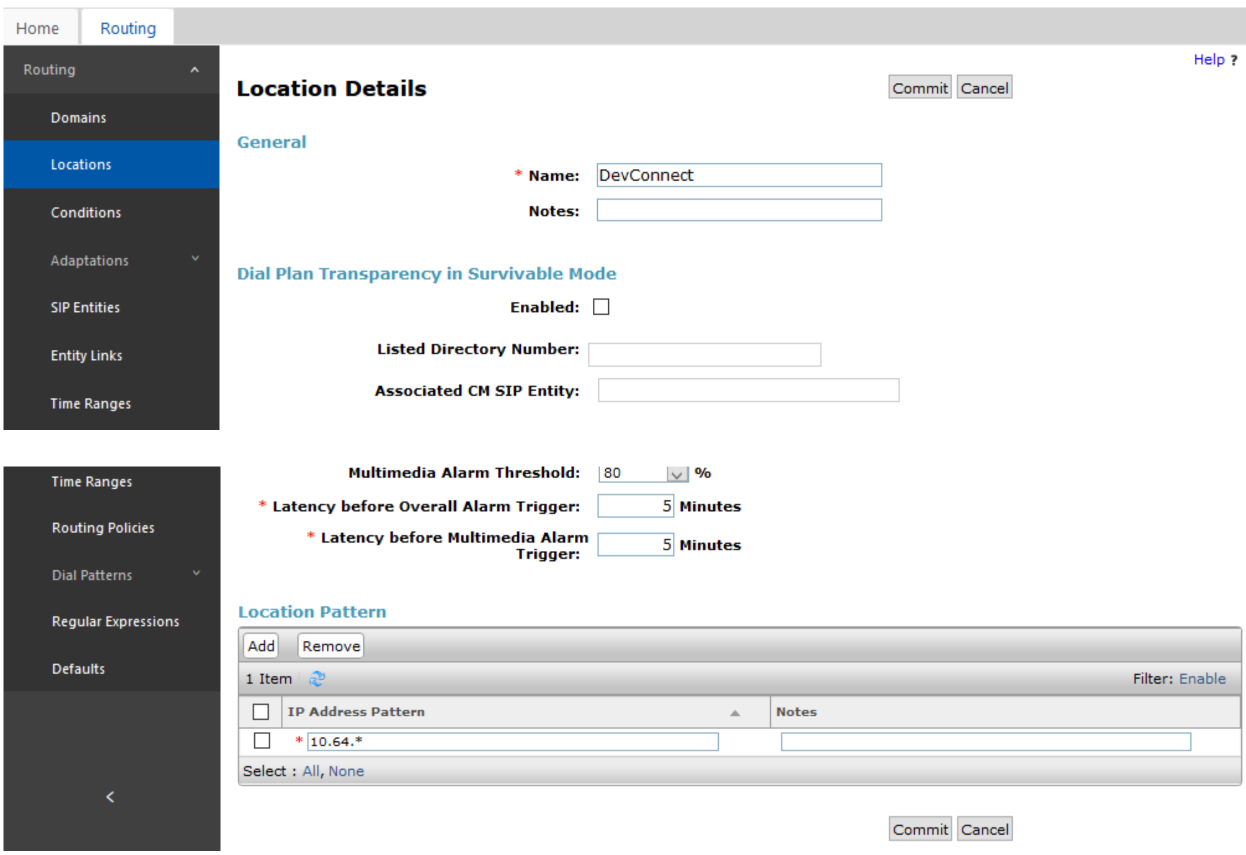
Under **General**:

- Type in a descriptive **Name**

At the bottom of the page, 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.



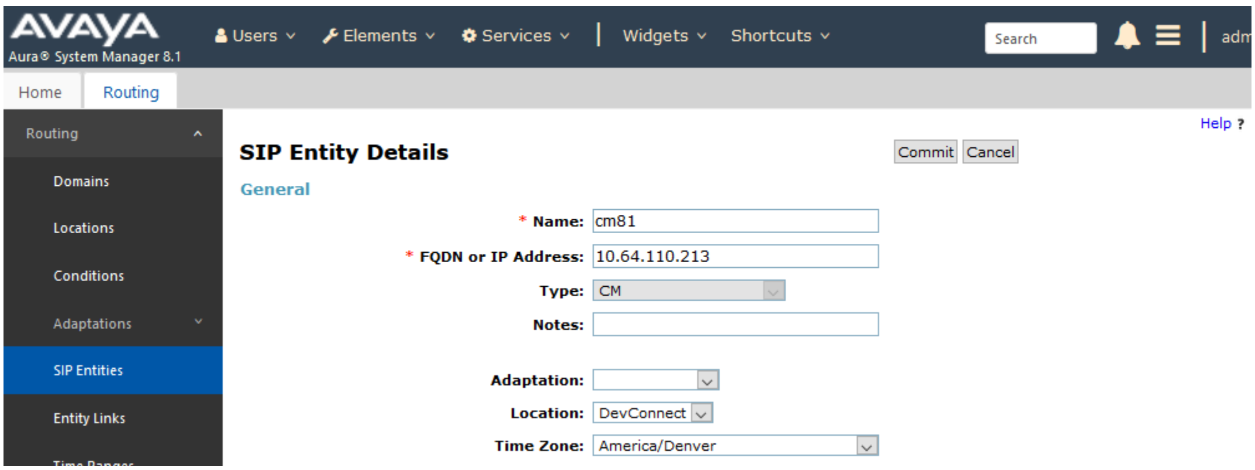
6.3. Add SIP Entity

Add Communication Manager as a SIP Entity. Navigate to **Elements → Routing → SIP Entities,** click on **New** (no shown) and configure as follows:

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

Click **Commit** to save changes

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

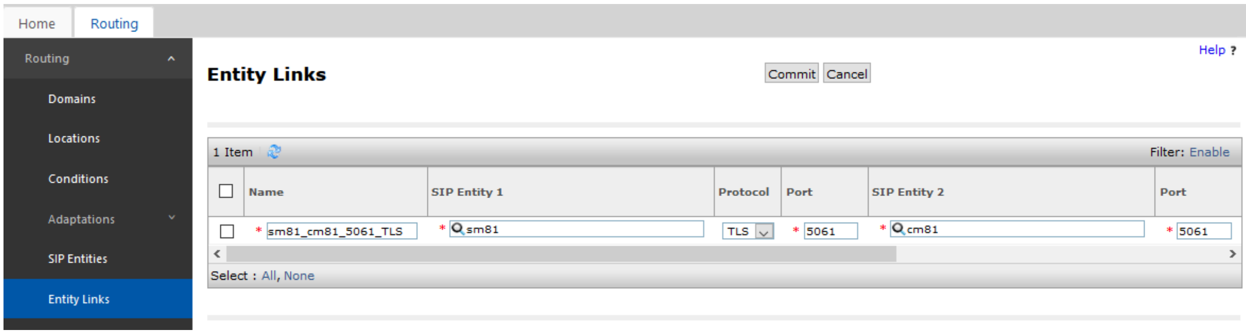


6.4. Add Entity Link

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
- Set **SIP Entity 2** to Communication Manager SIP Entity configured in **Section 6.3**

Click **Commit** to save changes.

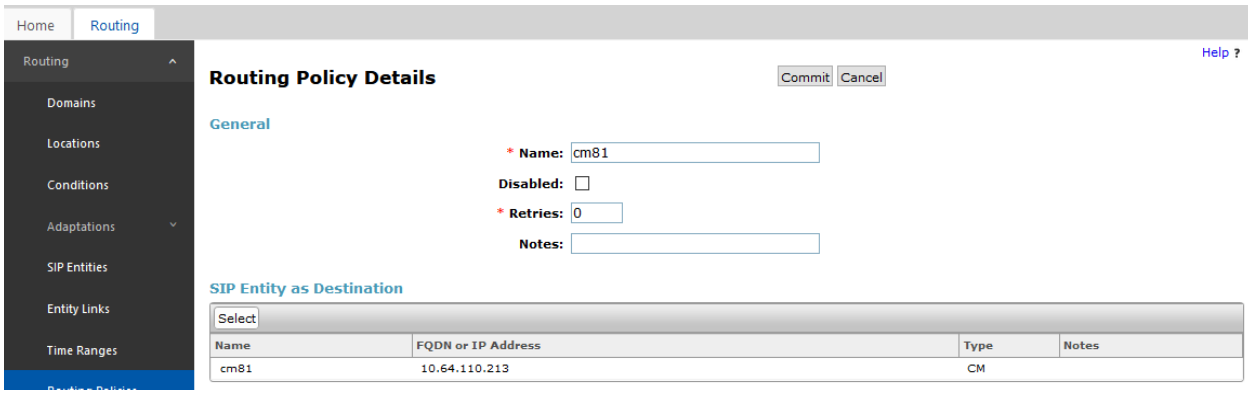


6.5. Add Routing Policy

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 6.3**

Click **Commit** to save changes.



6.6. Add Dial Patterns

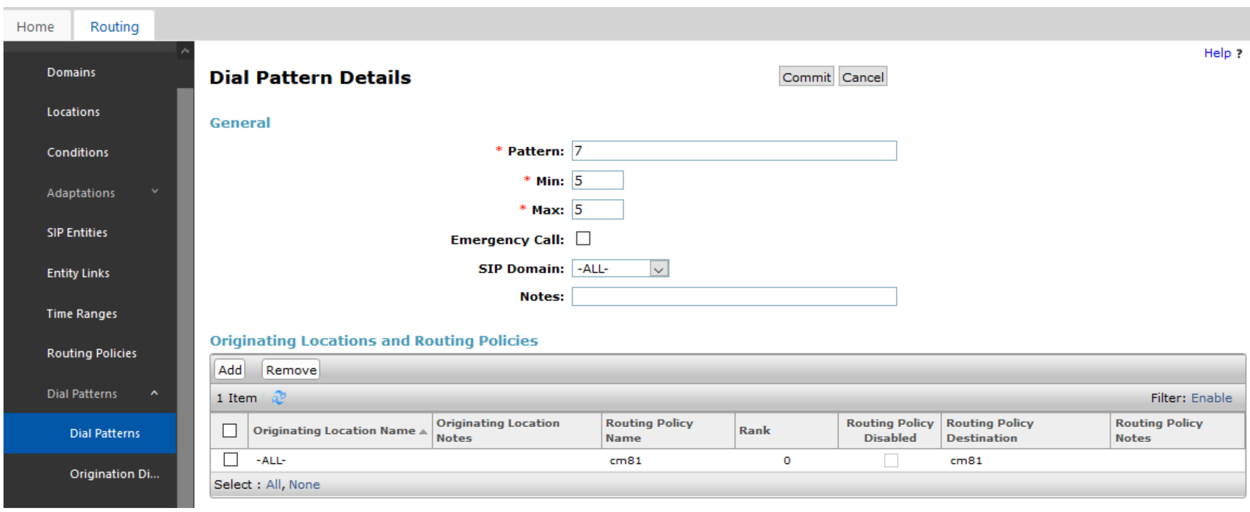
Navigate to **Elements → Routing → Dial Patterns**, click on **New** (not shown) and configure as follows:

Under **General:**

- Set **Pattern** to prefix of dialed number
- Set **Min** to minimum length of dialed number
- Set **Max** to maximum length of dialed number

Under **Originating Locations and Routing Policies:**
Click **Add** and select originating location and Communication Manager routing policy.
Click **Commit** to save changes (not shown).

Note: For Compliance testing, dialed number of 7xxxx were used.



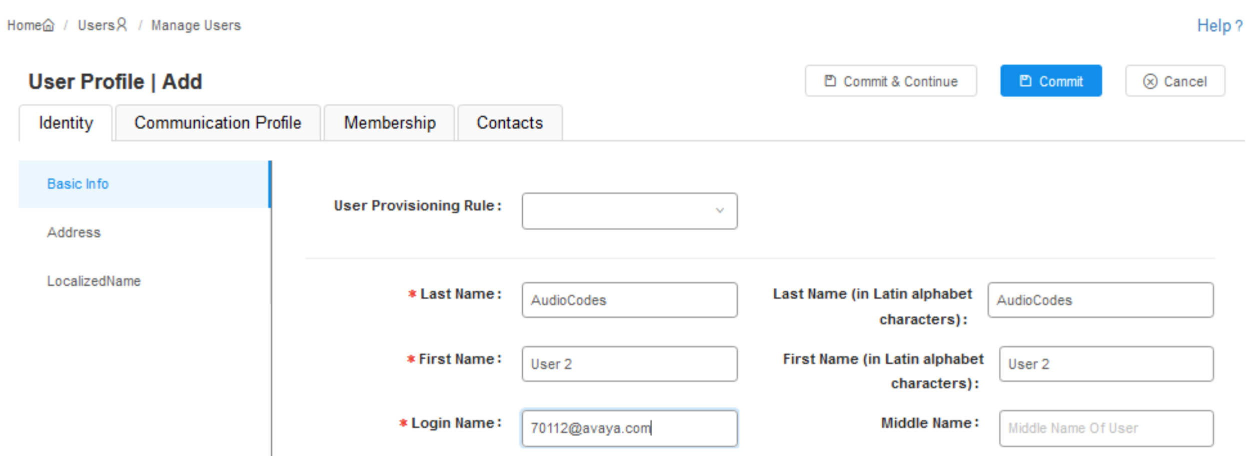
6.7. Add User

For each analog line on AudioCodes MP-1288, a user needs to be added on Session Manager. Information in this section will be used by AudioCodes MP-1288 for registering to Session Manager.

Navigate to **Users → User Management → Manage User**, click on **New** (not shown) and configure as follows:

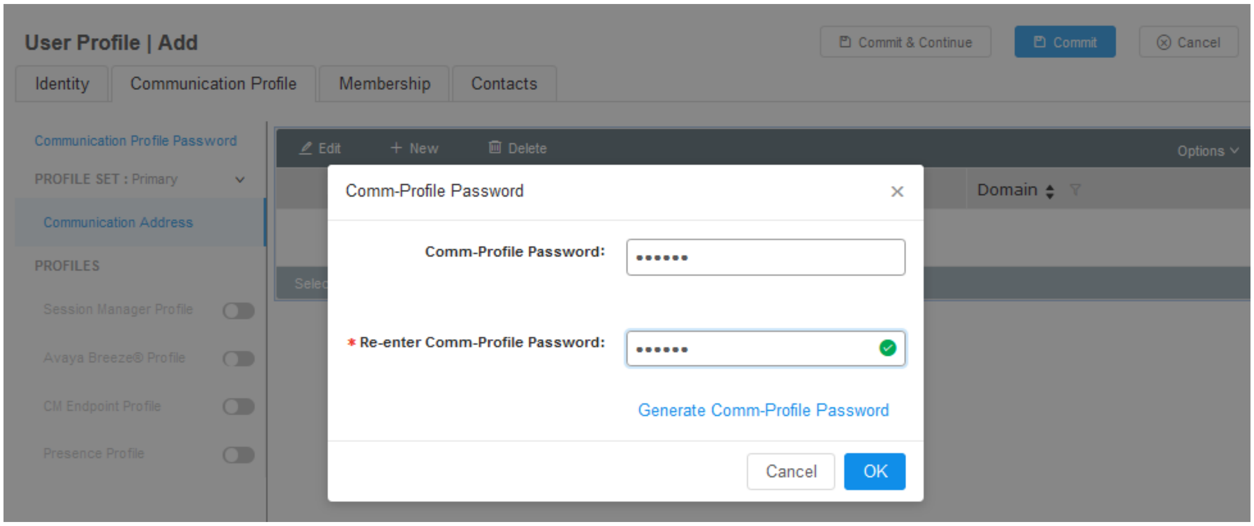
Under **Identity** tab:

- Type in **Last Name** and **First Name**
- In **Login Name** field type in <extension>@<domain>. <Extension> is an extension which will be configured on AudioCodes MP-1288 to receive and make calls. <domain> is as configured in **Section 6.1**

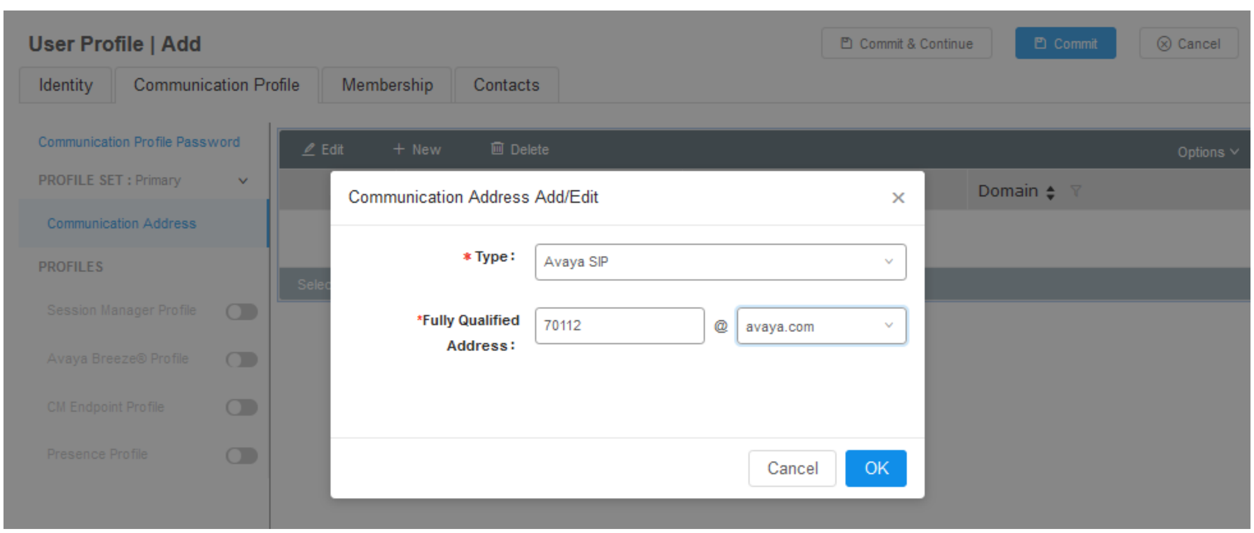


Under **Communication Profile** tab:

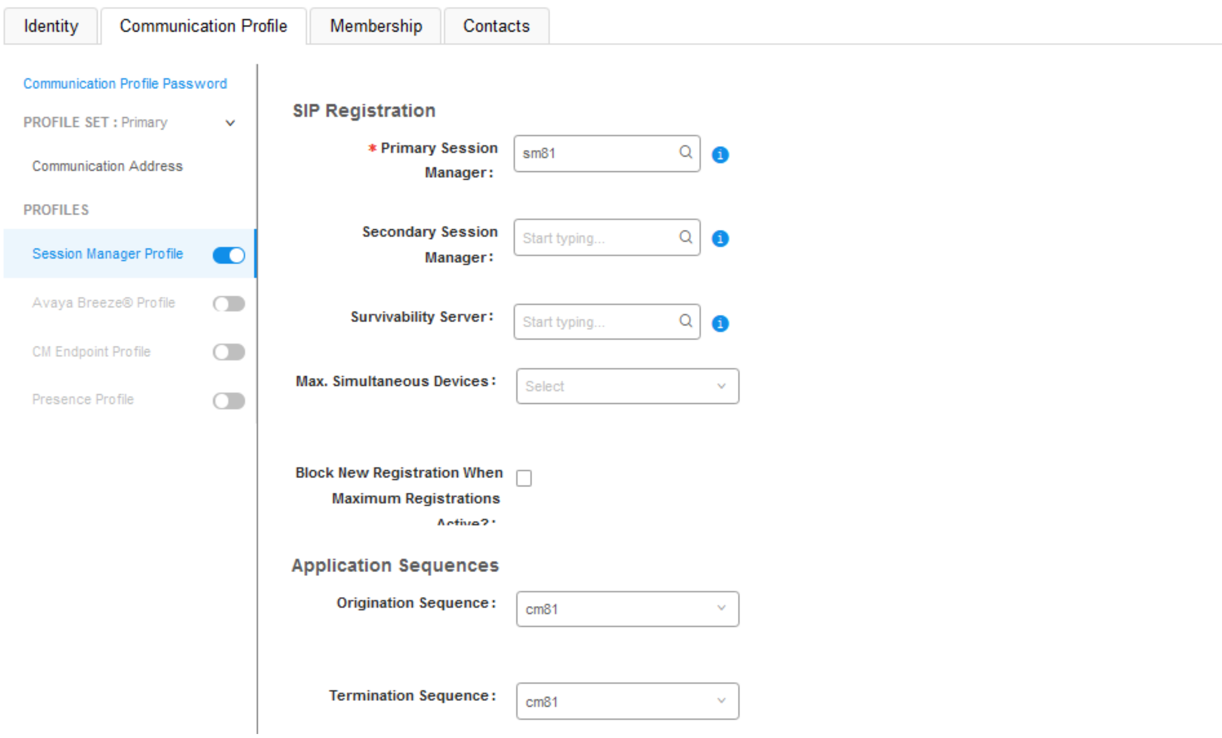
- Under **Communication Profile**, select **Communication Profile Password** and fill in **Password** details. This password will be used by AudioCodes MP-1288 to log in.



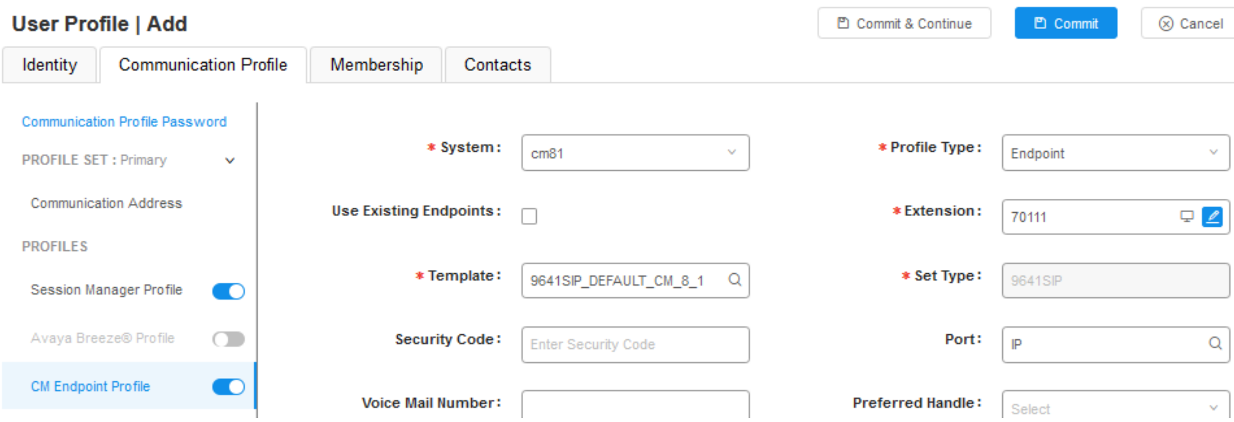
- Under **Communication Address** sub-section, click on **New**
  - Set **Type** to **Avaya SIP**
  - Type in <extension> in the text field, select domain configured in **Section 6.1** for **Fully Qualified Address**. <Extension> is the same extension configured for login name under Identity tab. Click on **Add**. Please note that AudioCodes MP-1288 will use this information has login name to register to Session Manager



- Toggle the **Session Manager Profile** button to turn it on:
  - Set **Primary Session Manager** to Session Manager. i.e. sm81
  - Set **Origination Application Sequence** and **Termination Application Sequence** to Communication Manager entity. Please note that configuration for Application Sequence is not shown in this document. Please refer to document [2] in reference section of this document for further details.
  - Set **Home Location** (not shown)

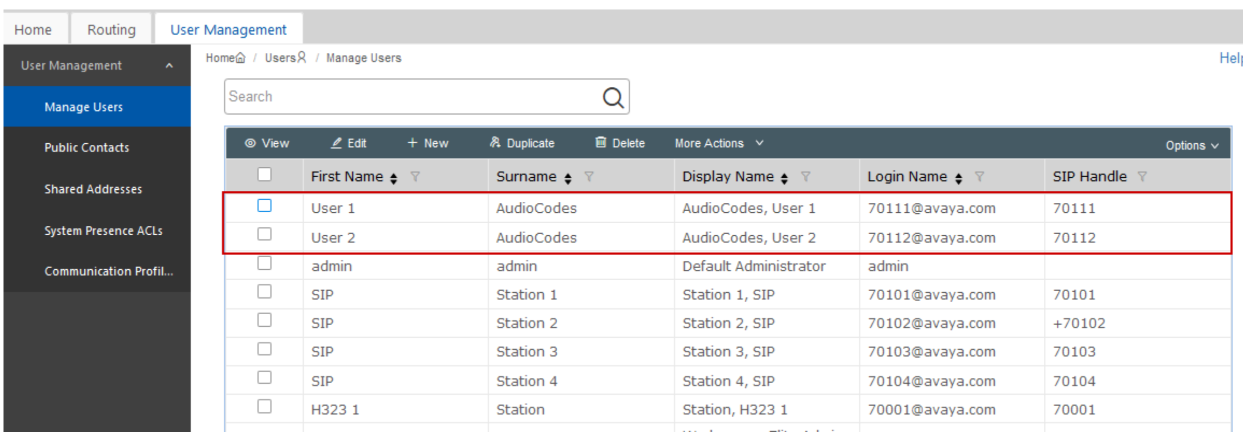


- Toggle **CM Endpoint Profile** button to turn it on:
    - Set **System** to Communication Manager entity
    - Set **Profile Type** to **Endpoint**
    - Type in extension number used in this section for **Extension** field
    - Set **Template** to **9641SIP_ DEFAULT_CM_8_1**



Click **Commit** to save changes.

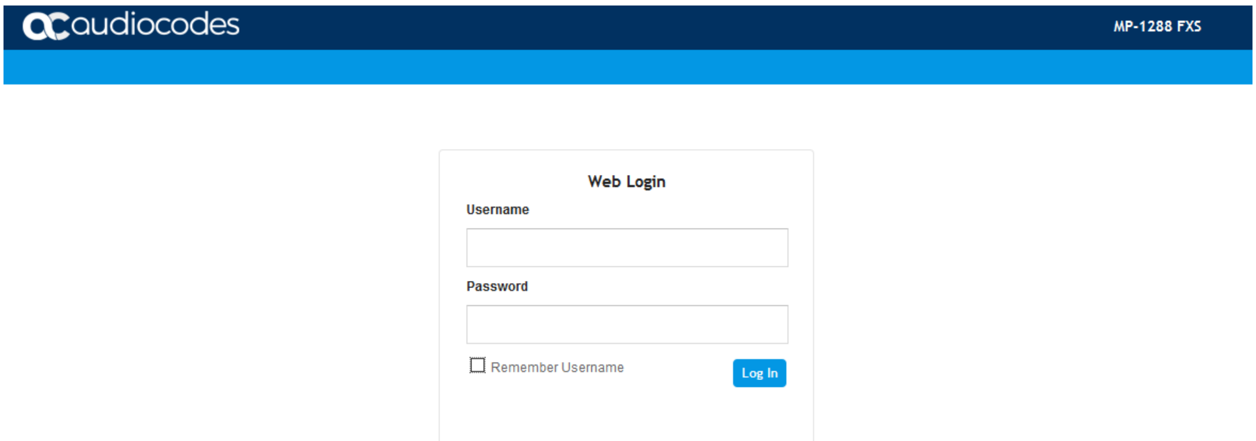
Two SIP Users for AudioCodes MP-1288 were configured during the Compliance test.



7. Configure AudioCodes MediaPack 1288

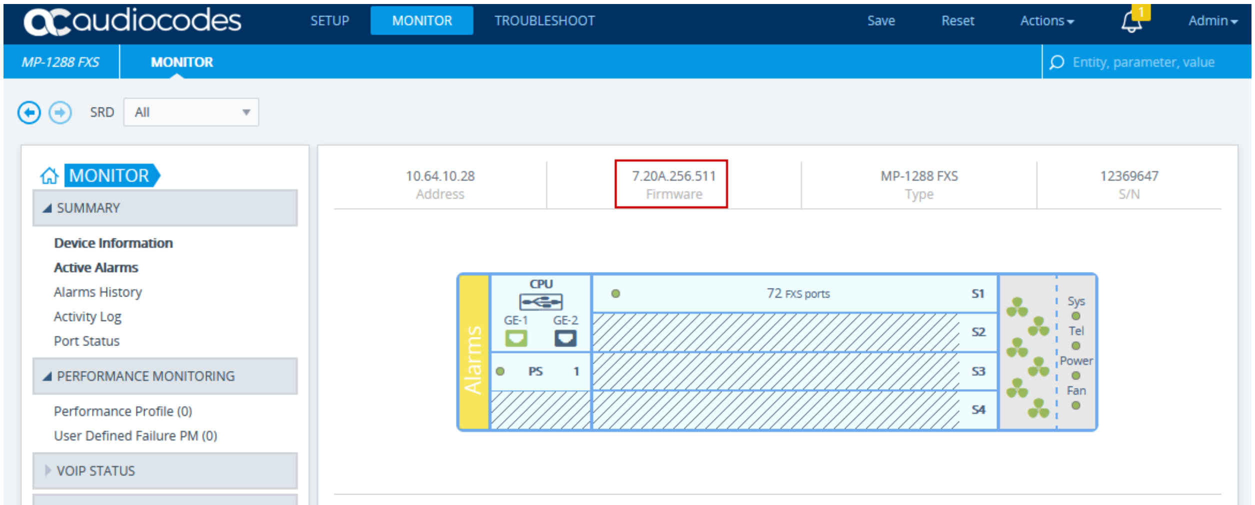
Administration for AudioCodes MP-1288 series is done via an administrative console. Type in http://<ip-address> URL in a web browser, where <ip-address> is the IP Address of AudioCodes MP-1288. It is assumed that AudioCodes MP-1288 has been assigned an IP Address and initial configuration has already been performed.

Log on to administrative console using appropriate credentials.



7.1. Verify/Upgrade Firmware Version

Once logged in, select **MONITOR** and verify that the firmware version is **7.20A.256.511** or higher. If firmware needs to be upgraded, navigate to **SETUP → ADMINISTRATION → MAINTENANCE → Software Upgrade** and select **Start Software Upgrade** (not shown).

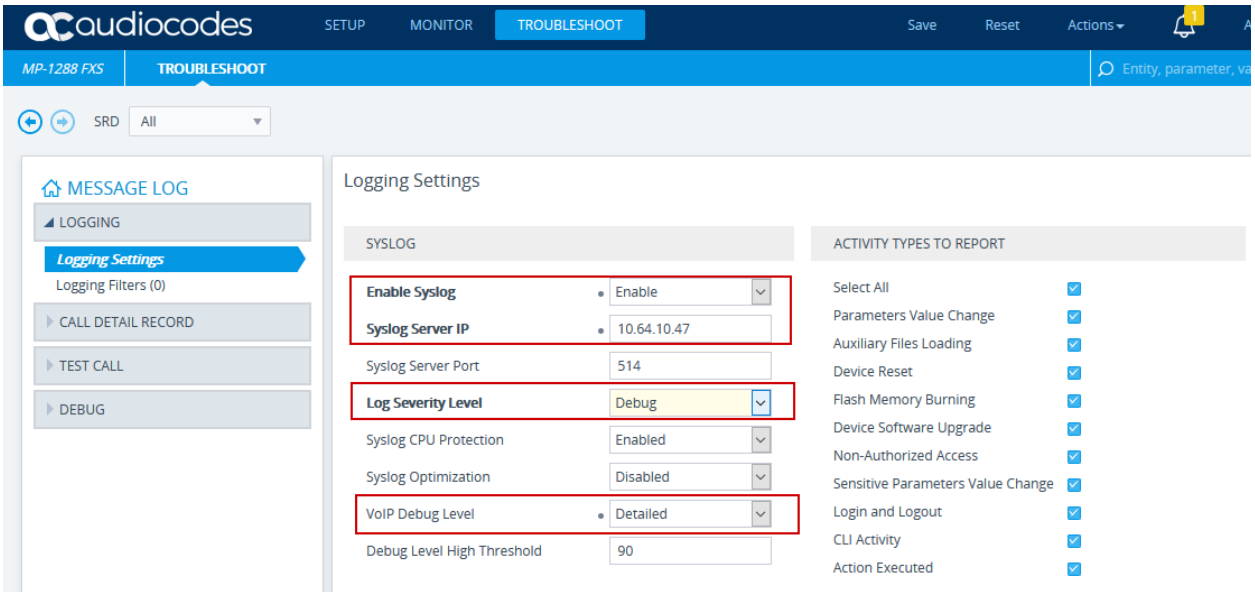


7.2. Administer Syslog Settings

Syslog can be enabled for troubleshooting purposes. Navigate to **System → Syslog Settings**

- 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. **ACSyslog**
- Set **Log Severity Level** to **Debug**
- Set **VoIP Debug Level** to **Detailed**

Click **Apply** to save changes (not shown).



7.3. IP Interfaces

From the configuration menu at the top, navigate to **IP NETWORK → CORE ENTITIES → IP Interfaces.** During the compliance test, the IP Interface was configured as follows:

#0[mp1288.avaya.com] Edit

GENERAL		
Name	• mp1288.avaya.com	
Application Type	OAMP + Media + Control	
Ethernet Device	• vlan 1	View

IP ADDRESS	
Interface Mode	IPv4 Manual
IP Address	• 10.64.10.28
Prefix Length	• 24
Default Gateway	• 10.64.10.1

DNS	
Primary DNS	• 10.64.110.100
Secondary DNS	• 75.75.75.75

7.4. Administer TLS Contexts

In order for TLS/SRTP to work correctly with Session Manager, a TLS context needs to be configured. Navigate to **SETUP → IP NETWORK → SECURITY → TLS Contexts** and select **+ New** to add a new TLS Context. The following was created during the compliance test to be used by System Manager to sign the AudioCodes MP-1288 certificate.

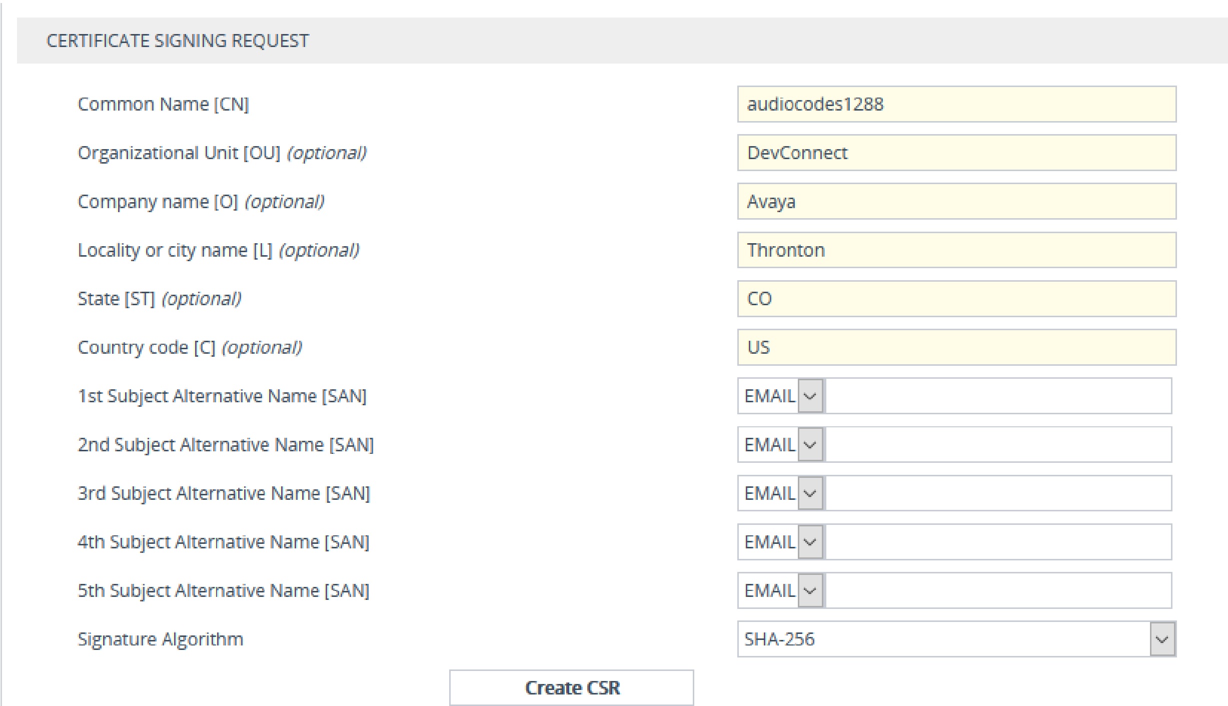
#1[audiocodes1288] Edit

GENERAL	
Name	• audiocodes1288
TLS Version	TLSv1.0 TLSv1.1 and TLSv1.2
DTLS Version	Any
Cipher Server	DEFAULT
Cipher Client	DEFAULT
Strict Certificate ...	Disable
DH key Size	• 2048
TLS Renegotiation	Enable

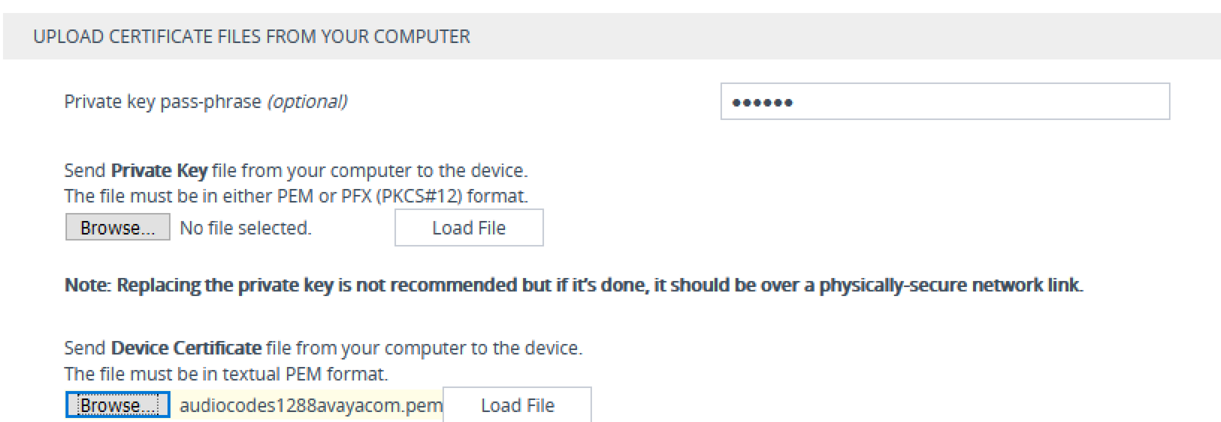
OCSP	
OCSP Server	Disable
Primary OCSP Se...	0.0.0.0
Secondary OCSP...	0.0.0.0
OCSP Port	2560
OCSP Default Re...	Reject

Certificate Information >> Change Certificate >> Trusted Root Certificates >>

Select **Change Certificate** and fill in the needed information and **Create CSR.**

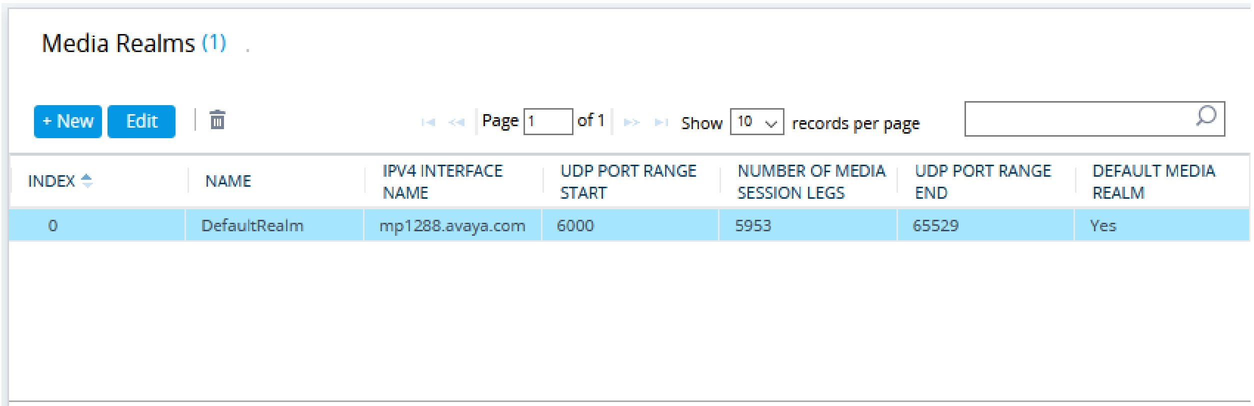


The generated CSR will be signed by System Manager and a **Device Certificate** will be generated (not shown). At the bottom of the page, **Browse** to the certificate generated by System Manager and type in the password used during the certificate generation in **Private key pass-phrase.** Select **Load File** to add the certificate to the device. Note that when the **Device Certificate** is loaded, the root certificate of System Manager is also added as a trusted certificate.



7.5. Media Realms

From the configuration menu at the top, navigate to **SIGNALING & MEDIA → CORE ENTITIES → Media Realms.** During the compliance tests, the default Media Realms were used as shown below.


Media Realms (1)

+ New Edit | Page 1 of 1 | Show 10 records per page

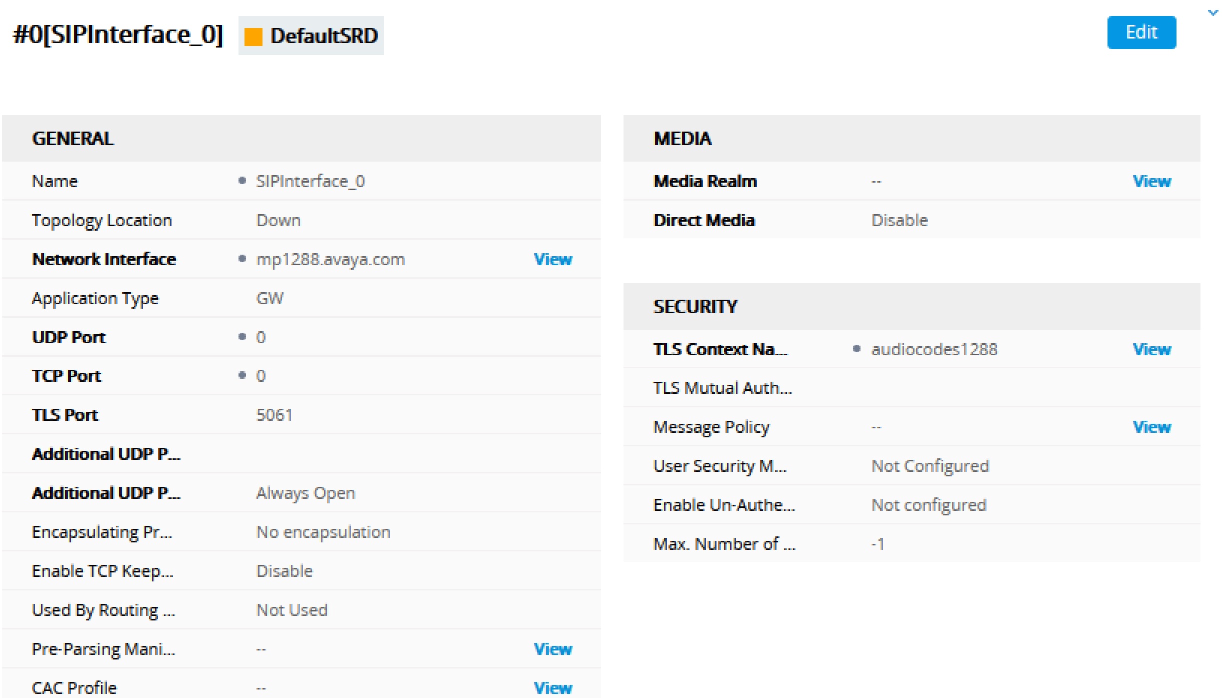
INDEX	NAME	IPV4 INTERFACE NAME	UDP PORT RANGE START	NUMBER OF MEDIA SESSION LEGS	UDP PORT RANGE END	DEFAULT MEDIA REALM
0	DefaultRealm	mp1288.avaya.com	6000	5953	65529	Yes

7.6. SIP Interfaces

Select **SIP Interfaces** on the left and add a SIP Interface for the Session Manager. Configure the SIP Interface for Session Manager as shown below:

- **Name:** Provide a descriptive name.
- **Network Interface:** Select the private interface from **Section 7.3.**
- **TLS Context:** Select the TLS Context configured in **Section 7.4.**

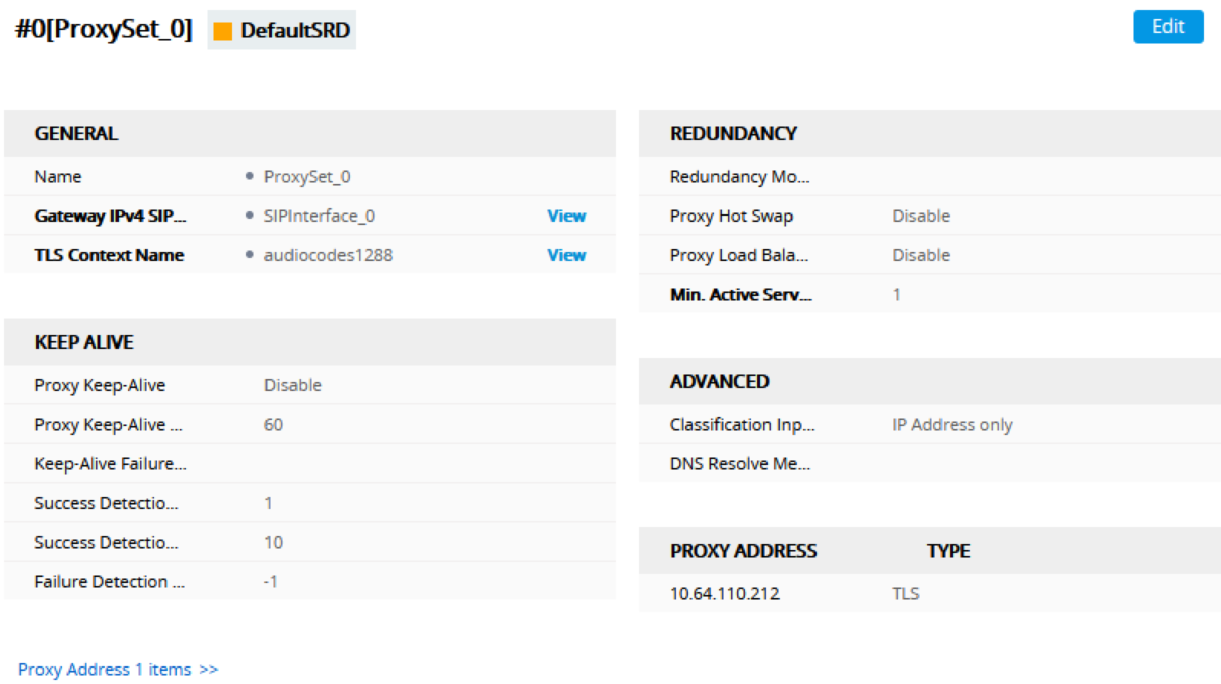
Retain default values for the rest of the fields.



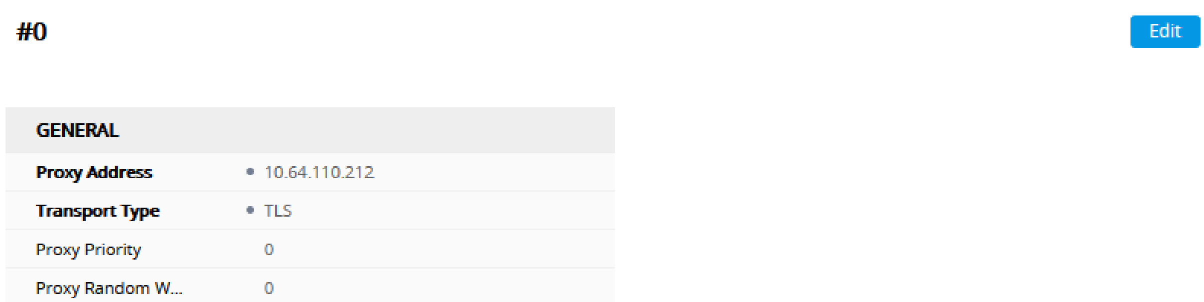
7.7. Proxy Sets

Select **Proxy Sets** on the left. Add a Proxy Sets for Session Manager and configure as shown below:

- **Name:** Provide a descriptive name.
- **Gateway IPv4 SIP Interface:** Select the SIP Interface from **Section 7.6.**
- **TLS Context Name:** Select the TLS Context configured in **Section 7.4.**

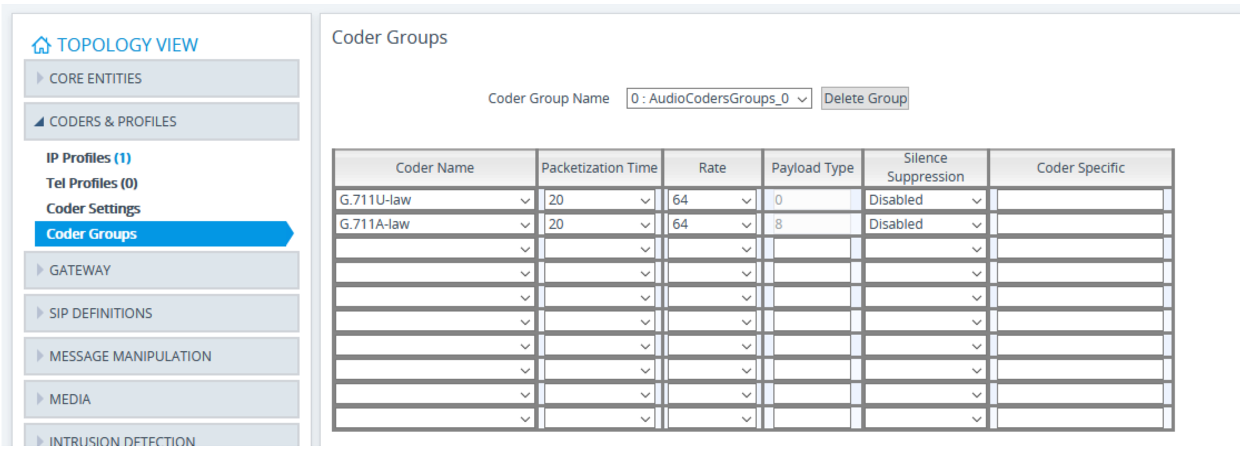


Continuing from above, select **Proxy Address items.** Select **+ New** to a new Proxy Address. For **Proxy Address** field, type the Session Manager SIP security module IP Address and select **Transport Type** of **TLS** from the drop-down menu. The following was configured during the compliance test.



7.8. Audio Coders Group

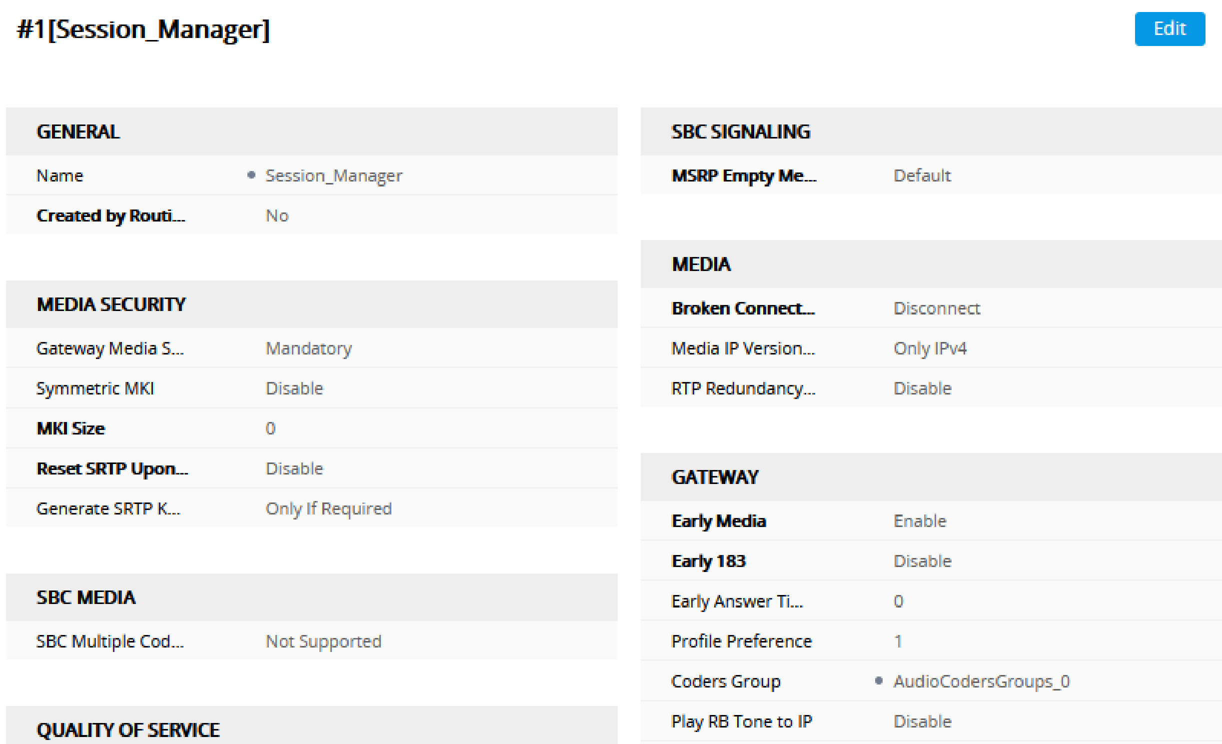
Select **CODERS & PROFILE → Coders Groups** on the left and configure the Coders as shown below:



7.9. IP Profile

Select **CODERS & PROFILE → IP Profiles** on the left. Add an IP Profile for Session Manager and configure as shown below:

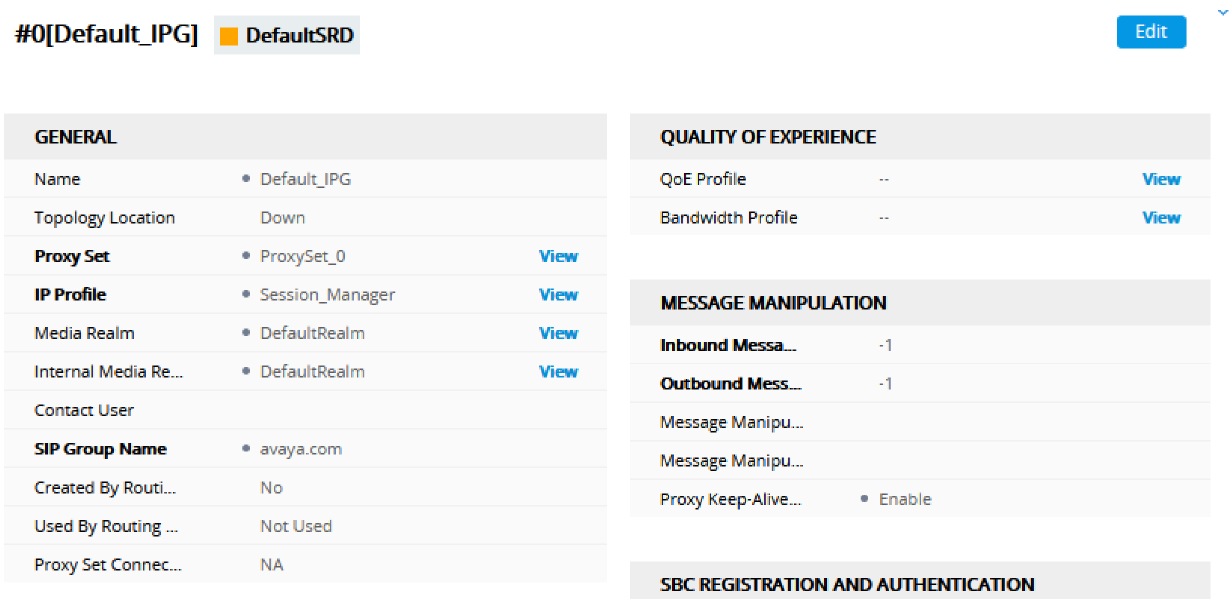
- **Name:** Provide a descriptive name (not shown).
- **Coders Group:** Select the Audio Coder Group from **Section 7.8.**



7.10. IP Groups

Select **CORE ENTITITES → IP Groups** on the left. During the compliance test, the default IP Group was used for Session Manager. It was configured as shown below:

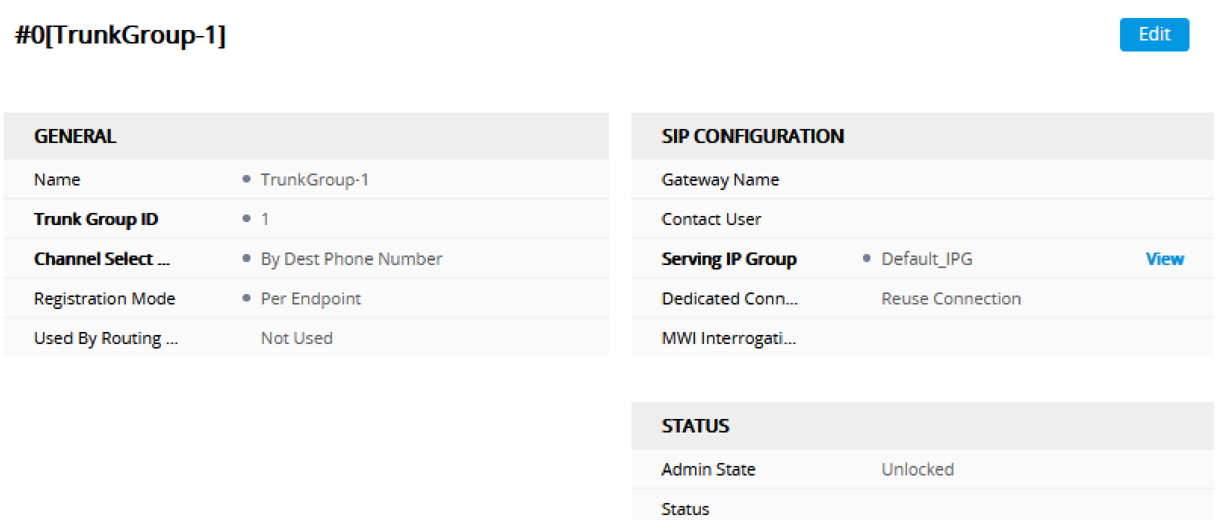
- **Name:** Provide a descriptive name.
- **Proxy Set:** Select the Proxy set from **Section 7.7.**
- **IP Profile:** Select the IP Profile from **Section 7.9.**
- **Media Realm:** Select the Media Realm from **Section 7.5.**
- **Internal Media Realm:** Select the Media Realm from **Section 7.5.**
- **SIP Group Name:** Type in the SIP Domain of Session Manager.
- **Proxy Keep-Alive:** Select **Enable** from the drop-down menu.



7.11. Trunk Group Settings

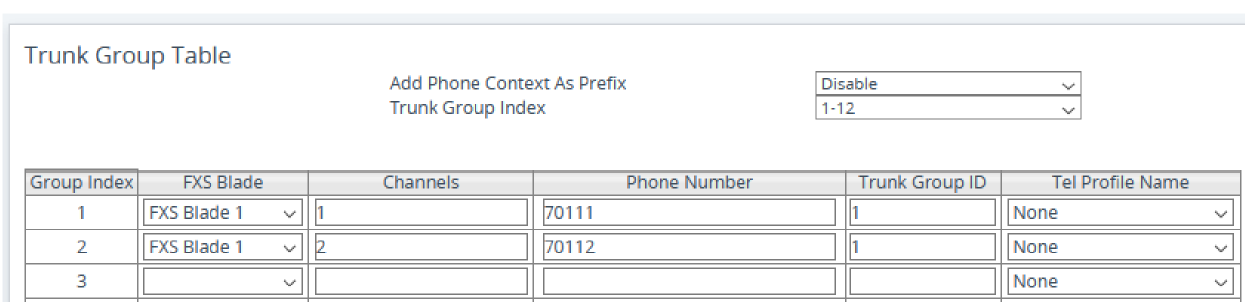
Select **Gateway → Trunk & Groups → Trunk Group Settings** on the left. Add a new Trunk Group as shown below. This trunk group will be assigned the analog channels.

- **Name:** Provide a descriptive name
- **Trunk Group ID:** Enter an available trunk group ID
- **Channel Select Mode:** Set to **By Dest Phone Number**
- **Registration Mode:** Set to **Per Endpoint**
- **Serving IP Group:** Set to the IP Group from **Section 7.10.**



7.12. Trunk Groups

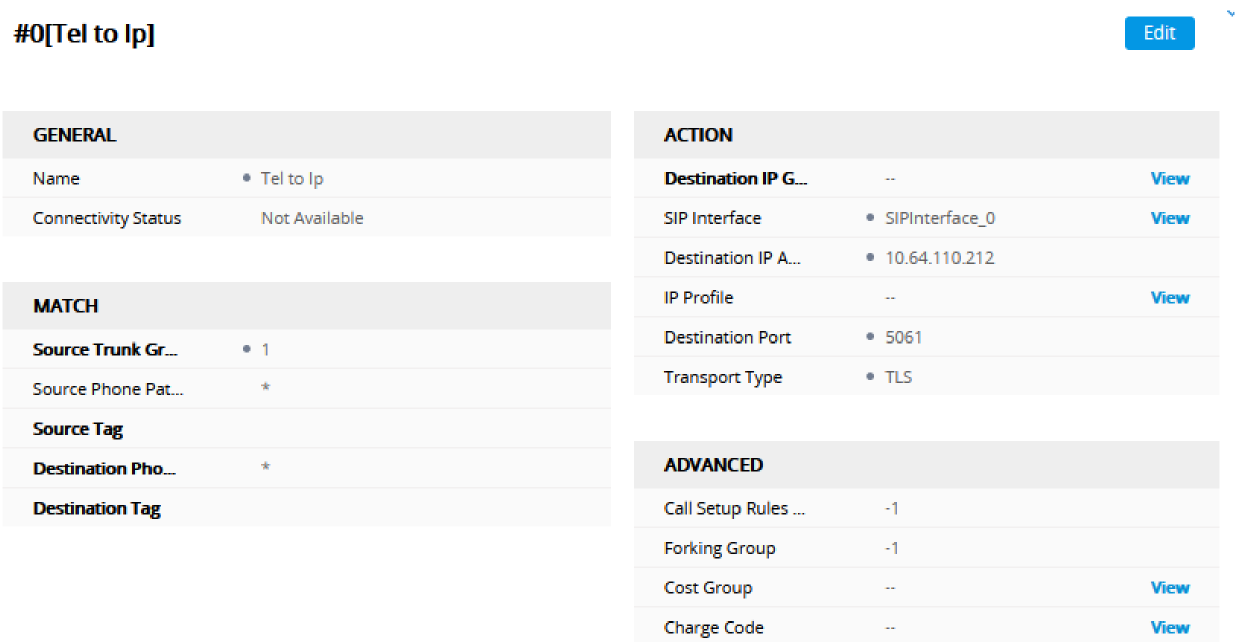
Select **Gateway → Trunk & Groups → Trunk Group** on the left. Assign the Trunk Group channels to the trunk group. Two channels were used during the compliance test and it was configured as shown below. Note that the SIP Users configured in **Section 6.7** were configured here.



7.13. Tel-to-IP Routing

Select **Gateway → Routing → Tel-to-IP Routing** on the left. Configure the routing from AudioCodes MP-1288 to Session Manager as shown below:

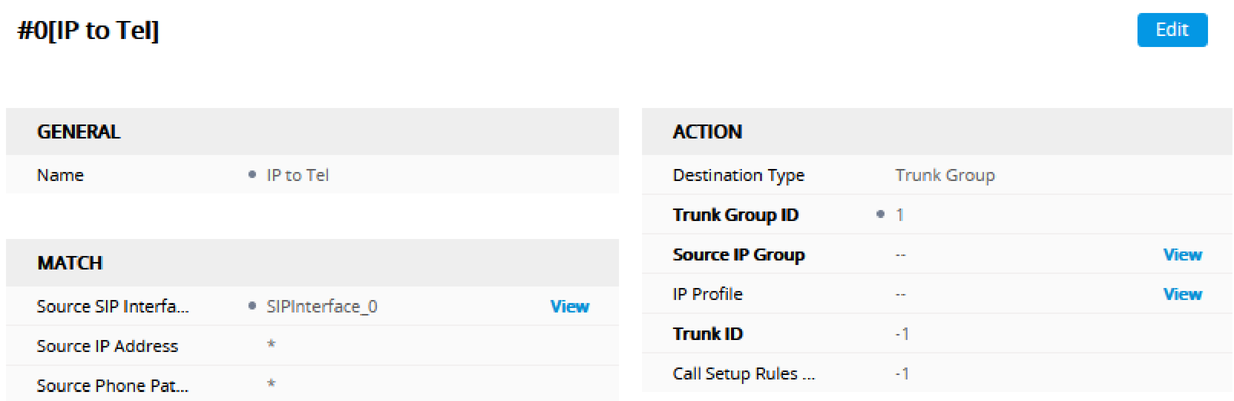
- **Name:** Provide a descriptive name
- **Source Trunk Group:** Trunk group from **Section 7.11**
- **SIP Interface:** SIP Interface from **Section 7.6.**
- **Destination IP Address:** Set to the SIP IP Address of Session Manager
- **Destination Port:** Set to the port configured in **Section 6.4**
- **Transport Type:** Set to **TLS**



7.14. IP-to-Tel Routing

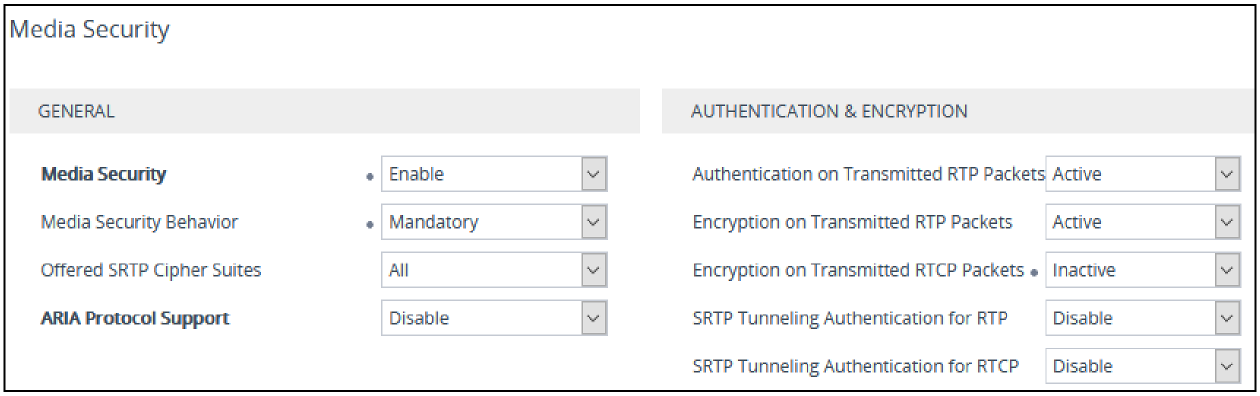
Select **Gateway → Routing → IP-to-Tel Routing** on the left. Configure the routing from Session Manager to AudioCodes MP-1288 as shown below:

- **Name:** Provide a descriptive name
- **Source SIP Interface:** SIP Interface from **Section 7.6**
- **Trunk Group ID:** Trunk Group IP from **Section 7.11**



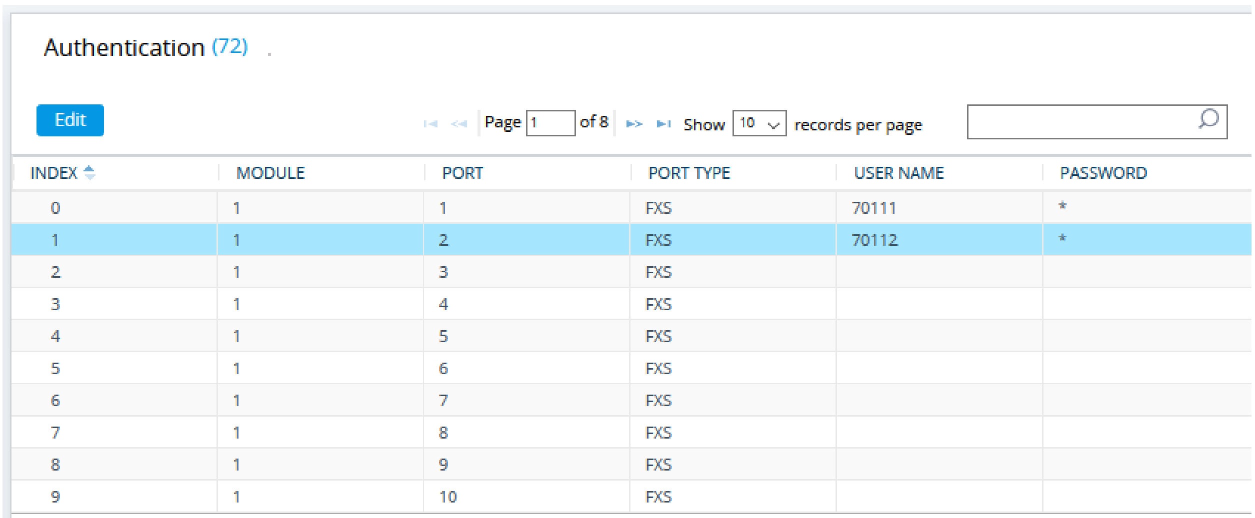
7.15. Media Security

Select **MEDIA → Media Security** on the left; set **Media Security** to **Enable** and **Media Security Behavior** to **Mandatory.**



7.16. Authentication

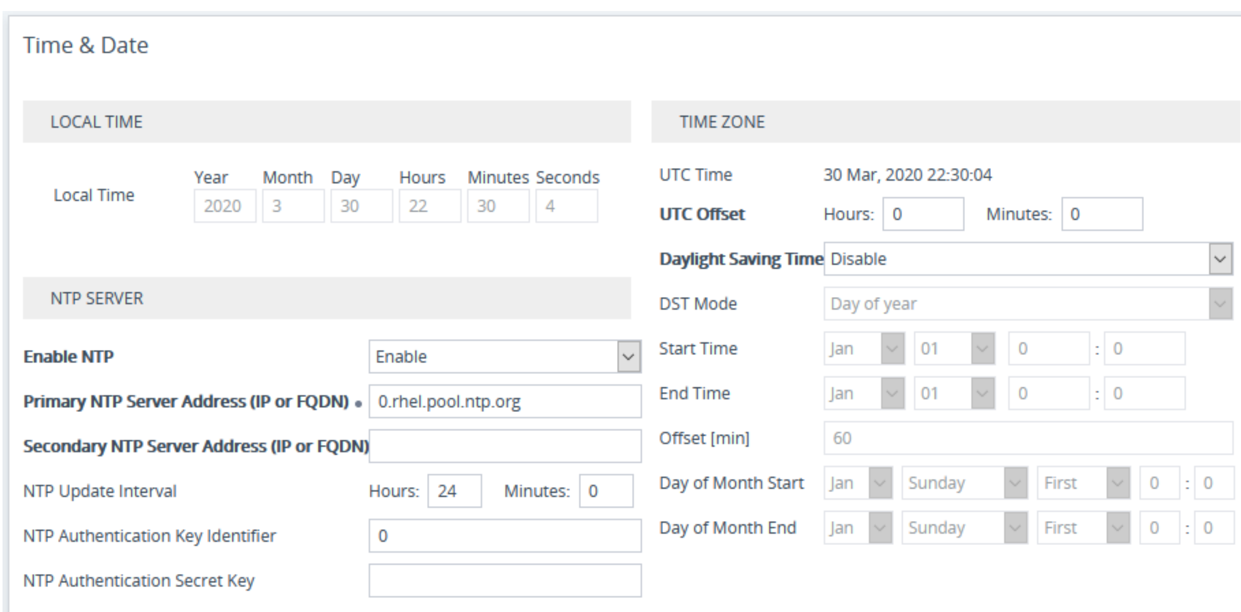
SIP Users added in Session Manager need to be added on AudioCodes MP-1288. Each SIP User will be assigned to a channel. Navigate to **Gateway → Analog Gateway → Authentication.** The two SIP Users and Passwords from **Section 6.7** are configure as shown below.



7.17. Time and Date

Navigate to **SETUP → ADMINISTRATION → TIME & DATE.**

- **Enable NTP:** Select **Enable** from the drop-down menu.
- **Primary NTP Server Address:** Type in a designated NTP Server.

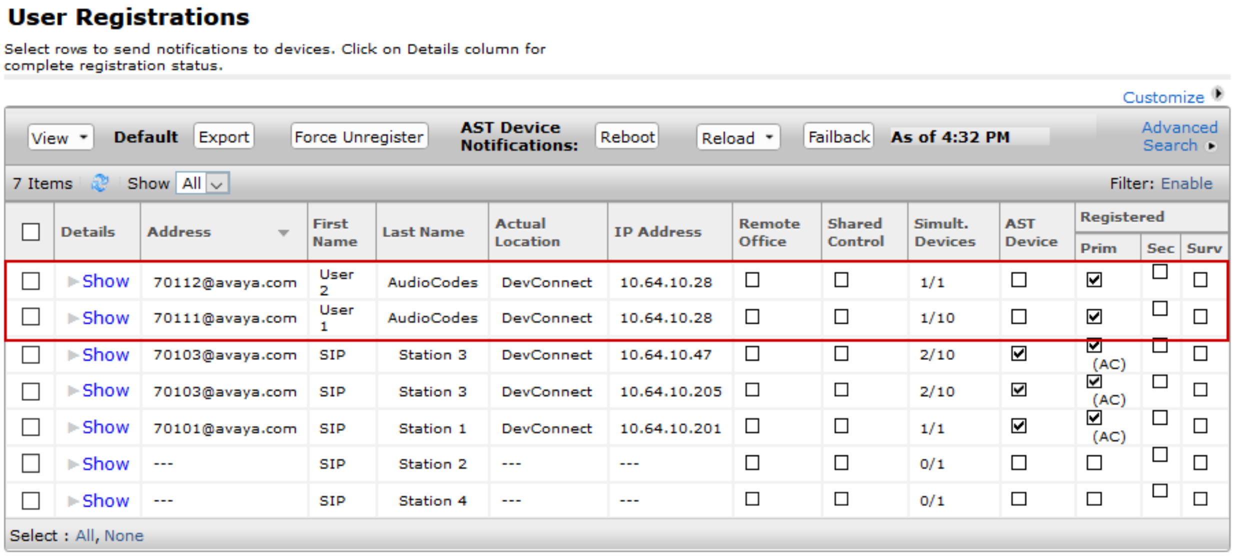


8. Verification Steps

- 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 6**. 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 SIP Users (from **Section 6.7)** registration from AudioCodes MP-1288 to Session Manager via System Manager console, http://<ip-address>/SMGR
- Navigate to **Home → Session Manager → System Status → User Registration**



	Details	Address	First Name	Last Name	Actual Location	IP Address	Remote Office	Shared Control	Simult. Devices	AST Device	Registered Prim	Registered Sec	Registered Surv
☐	Show	70112@avaya.com	User 2	AudioCodes	DevConnect	10.64.10.28	☐	☐	1/1	☐	☑	☐	☐
☐	Show	70111@avaya.com	User 1	AudioCodes	DevConnect	10.64.10.28	☐	☐	1/10	☐	☑	☐	☐
☐	Show	70103@avaya.com	SIP	Station 3	DevConnect	10.64.10.47	☐	☐	2/10	☑	☑ (AC)	☐	☐
☐	Show	70103@avaya.com	SIP	Station 3	DevConnect	10.64.10.205	☐	☐	2/10	☑	☑ (AC)	☐	☐
☐	Show	70101@avaya.com	SIP	Station 1	DevConnect	10.64.10.201	☐	☐	1/1	☑	☑ (AC)	☐	☐
☐	Show	---	SIP	Station 2	---	---	☐	☐	0/1	☐	☐	☐	☐
☐	Show	---	SIP	Station 4	---	---	☐	☐	0/1	☐	☐	☐	☐

9. Conclusion

These Application Notes describe the configuration steps required for AudioCodes MP-1288 to successfully interoperate with Avaya Aura® Communication Manager and Avaya Aura® Session Manager. All feature and serviceability test cases completed and pass with observations/exceptions noted in **Section 2.2**

10. Additional References

Avaya references are available at http://support.avaya.com

[1] "Administering Avaya Aura® Session Manager", Release 8.1.1, October 2019

[2] "Administering Avaya Aura® Communication Manager", Release 8.1.x, Issue 5, November 2019

AudioCodes MP-1288 references can directly be obtained from AudioCodes.

A. Appendix

AudioCodes MP-1288 .ini file generated during the compliance testing is as follows: Please use it only for reference purposes.

```
;**************
;** Ini File **
;**************

;Time & Date: 30/03/2020 22:37:39
;Device Up Time: 2d:22h:36m:53s
;Board: MP-1288 FXS
;Board Type: 86
;Serial Number: 12369647
;Product Key: DT3507405
;Software Version: 7.20A.256.511
;DSP Software Version: 5033AE3_R => 723.04
;Board IP Address: 10.64.10.28
;Board Subnet Mask: 255.255.255.0
;Board Default Gateway: 10.64.10.1
;CPU: Cavium Networks Octeon V0.1 @ 700Mhz, total 2 cores, 2 cpus, 1 sockets
;Cores mapping:
;core #0, on cpu #0, on socket #0
;core #1, on cpu #1, on socket #0
;Memory: 1024 MB
;Flash size: 128 MB
;Num of DSP Cores: 6
;Num of physical LAN ports: 2
;;;Key features:;Board Type: MP-1288 FXS ;Coders: G723 G729 G727 G722 ;DSP
Voice features: IpmDetector ;Security: IPSEC MediaEncryption StrongEncryption
EncryptControlProtocol ;System features: ProducrKey=DT3507405 ;PSTN
Protocols: CAS ;Channel Type: RTP DspCh=312 ;IP Media: VXML ;Control
Protocols: SIP MSFT EMS ;Default features:;Coders: G711 G726;

;-----  HW components -----
;
; Slot # : Module type : # of ports
;----------------------------------------------
;      1 : FXS         : 72
;      2 : Empty
;      3 : Empty
;      4 : Empty
;----------------------------------------------

;USB Port 1: Empty
;Power Supply 1: Exists
;Power Supply 2: Not Exists
;Fan Tray: Exists
;----------------------------------------------


[SYSTEM Params]

SyslogServerIP = 10.64.10.47
```

```
EnableSyslog = 1
ENABLEPARAMETERSMONITORING = 1
ActivityListToLog = 'pvc', 'afl', 'dr', 'fb', 'swu', 'naa', 'spc', 'll',
'cli', 'ae'
TLSPkeySize = 2048
TR069ACSPASSWORD = '$1$gQ=='
TR069CONNECTIONREQUESTPASSWORD = '$1$gQ=='
NTPServerIP = '0.rhel.pool.ntp.org'
SyslogLogLevel = 7
PM_VEDSPUtil = '1,280,312,15'

[BSP Params]

PCMLawSelect = 3
UdpPortSpacing = 10
EnterCpuOverloadPercent = 99
ExitCpuOverloadPercent = 95

[Analog Params]

PolarityReversalType = 1
MinFlashHookTime = 100

[ControlProtocols Params]

AdminStateLockControl = 0

[Voice Engine Params]

FaxTransportMode = 0
V22ModemTransportType = 0
V23ModemTransportType = 0
V32ModemTransportType = 0
V34ModemTransportType = 0
RFC2833TxPayloadType = 101
V34FAXTRANSPORTTYPE = 0
RTCPEncryptionDisableTx = 1
ENABLEMEDIASECURITY = 1
PLThresholdLevelsPerMille_0 = 5
PLThresholdLevelsPerMille_1 = 10
PLThresholdLevelsPerMille_2 = 20
PLThresholdLevelsPerMille_3 = 50
CallProgressTonesFilename = 'usa_tones_13.dat'

[WEB Params]

Languages = 'en-US', '', '', '', '', '', '', '', '', ''

[SIP Params]

ISREGISTERNEEDED = 1
SIPDESTINATIONPORT = 5061
GWDEBUGLEVEL = 5
ENABLEEARLYMEDIA = 1
SIPGATEWAYNAME = 'avaya.com'
ENABLEMWISUBSCRIPTION = 1
```

```
MWISERVERIP = '10.64.110.212'
MWIANALOGLAMP = 1
MWIDISPLAY = 1
ENABLEMWI = 1
ENABLESIPS = 1
MEDIASECURITYBEHAVIOUR = 1
MWISERVERTRANSPORTTYPE = 2
MSLDAPPRIMARYKEY = 'telephoneNumber'
ENERGYDETECTORCMD = 587202560
ANSWERDETECTORCMD = 10486144

[SNMP Params]


[ PhysicalPortsTable ]

FORMAT Index = Port, Mode, SpeedDuplex, PortDescription, GroupMember;
PhysicalPortsTable 0 = "GE_1", 1, 4, "User Port #0", "GROUP_1";
PhysicalPortsTable 1 = "GE_2", 1, 4, "User Port #1", "GROUP_1";

[ \PhysicalPortsTable ]


[ EtherGroupTable ]

FORMAT Index = Group, Mode, Member1, Member2;
EtherGroupTable 0 = "GROUP_1", 2, "GE_1", "GE_2";
EtherGroupTable 1 = "GROUP_2", 0, "", "";

[ \EtherGroupTable ]


[ DeviceTable ]

FORMAT Index = VlanID, UnderlyingInterface, DeviceName, Tagging, MTU;
DeviceTable 0 = 1, "GROUP_1", "vlan 1", 0, 1500;

[ \DeviceTable ]


[ InterfaceTable ]

FORMAT Index = ApplicationTypes, InterfaceMode, IPAddress, PrefixLength,
Gateway, InterfaceName, PrimaryDNSServerIPAddress,
SecondaryDNSServerIPAddress, UnderlyingDevice;
InterfaceTable 0 = 6, 10, 10.64.10.28, 24, 10.64.10.1, "mp1288.avaya.com",
10.64.110.100, 75.75.75.75, "vlan 1";

[ \InterfaceTable ]


[ WebUsers ]

FORMAT Index = Username, Password, Status, PwAgeInterval, SessionLimit,
CliSessionLimit, SessionTimeout, BlockTime, UserLevel, PwNonce, SSHPublicKey;
```

```
WebUsers 0 = "Admin",
"$1$VTEzY2o4aWNqaD86UlkAVQUDB14LDQ4JD1ReXUJBRhYRFxRGHh8YSk0YSE7iuOextrO/vuvtv
O/qu77rpajw8K0=", 1, 0, 5, -1, 15, 60, 200,
"e8dc63dcc641f32406dc4e8fb84918d9", "";
WebUsers 1 = "User",
"$1$4oCA0NXW0YuK3tiI297ZwZOTkZLCwsXKnZiezsTLyjA2OjUxMWM/MTE6ODw4a2x1dXQndnchd
X0ueyp/eX19RUA=", 1, 0, 5, -1, 15, 60, 50,
"745000ce182823aea635d7017e610170", "";

[ \WebUsers ]


[ TLSContexts ]

FORMAT Index = Name, TLSVersion, DTLSVersion, ServerCipherString,
ClientCipherString, RequireStrictCert, TlsRenegotiation, OcspEnable,
OcspServerPrimary, OcspServerSecondary, OcspServerPort, OcspDefaultResponse,
DHKeySize;
TLSContexts 0 = "default", 7, 0, "DEFAULT", "DEFAULT", 0, 1, 0, , , 2560, 0,
1024;
TLSContexts 1 = "audiocodes1288", 7, 0, "DEFAULT", "DEFAULT", 0, 1, 0,
0.0.0.0, 0.0.0.0, 2560, 0, 2048;

[ \TLSContexts ]


[ AudioCodersGroups ]

FORMAT Index = Name;
AudioCodersGroups 0 = "AudioCodersGroups_0";

[ \AudioCodersGroups ]


[ IpProfile ]

FORMAT Index = ProfileName, IpPreference, CodersGroupName, IsFaxUsed,
JitterBufMinDelay, JitterBufOptFactor, IPDiffServ, SigIPDiffServ,
RTPRedundancyDepth, CNGmode, VxxTransportType, NSEMode, IsDTMFUsed,
PlayRBTone2IP, EnableEarlyMedia, ProgressIndicator2IP, EnableEchoCanceller,
CopyDest2RedirectNumber, MediaSecurityBehaviour, CallLimit,
DisconnectOnBrokenConnection, FirstTxDtmfOption, SecondTxDtmfOption,
RxDTMFOption, EnableHold, InputGain, VoiceVolume, AddIEInSetup,
SBCExtensionCodersGroupName, MediaIPVersionPreference, TranscodingMode,
SBCAllowedMediaTypes, SBCAllowedAudioCodersGroupName,
SBCAllowedVideoCodersGroupName, SBCAllowedCodersMode,
SBCMediaSecurityBehaviour, SBCRFC2833Behavior, SBCAlternativeDTMFMethod,
SBCSendMultipleDTMFMethods, SBCAssertIdentity, AMDSensitivityParameterSuit,
AMDSensitivityLevel, AMDMaxGreetingTime, AMDMaxPostSilenceGreetingTime,
SBCDiversionMode, SBCHistoryInfoMode, EnableQSIGTunneling,
SBCFaxCodersGroupName, SBCFaxBehavior, SBCFaxOfferMode, SBCFaxAnswerMode,
SbcPrackMode, SBCSessionExpiresMode, SBCRemoteUpdateSupport,
SBCRemoteReinviteSupport, SBCRemoteDelayedOfferSupport,
SBCRemoteReferBehavior, SBCRemote3xxBehavior, SBCRemoteMultiple18xSupport,
SBCRemoteEarlyMediaResponseType, SBCRemoteEarlyMediaSupport,
EnableSymmetricMKI, MKISize, SBCEnforceMKISize, SBCRemoteEarlyMediaRTP,
```

```
SBCRemoteSupportsRFC3960, SBCRemoteCanPlayRingback, EnableEarly183,
EarlyAnswerTimeout, SBC2833DTMFPayloadType, SBCUserRegistrationTime,
ResetSRTPStateUponRekey, AmdMode, SBCReliableHeldToneSource,
GenerateSRTPKeys, SBCPlayHeldTone, SBCRemoteHoldFormat,
SBCRemoteReplacesBehavior, SBCSDPPtimeAnswer, SBCPreferredPTime,
SBCUseSilenceSupp, SBCRTPRedundancyBehavior, SBCPlayRBTToTransferee,
SBCRTCPMode, SBCJitterCompensation, SBCRemoteRenegotiateOnFaxDetection,
JitterBufMaxDelay, SBCUserBehindUdpNATRegistrationTime,
SBCUserBehindTcpNATRegistrationTime, SBCSDPHandleRTCPAttribute,
SBCRemoveCryptoLifetimeInSDP, SBCIceMode, SBCRTCPMux, SBCMediaSecurityMethod,
SBCHandleXDetect, SBCRTCPFeedback, SBCRemoteRepresentationMode,
SBCKeepVIAHeaders, SBCKeepRoutingHeaders, SBCKeepUserAgentHeader,
SBCRemoteMultipleEarlyDialogs, SBCRemoteMultipleAnswersMode,
SBCDirectMediaTag, SBCAdaptRFC2833BWToVoiceCoderBW, CreatedByRoutingServer,
SBCFaxReroutingMode, SBCMaxCallDuration, SBCGenerateRTP, SBCISUPBodyHandling,
SBCISUPVariant, SBCVoiceQualityEnhancement, SBCMaxOpusBW, SBCEnhancedPlc,
LocalRingbackTone, LocalHeldTone, SBCGenerateNoOp, SBCRemoveUnKnownCrypto,
SBCMultipleCoders, DataDiffServ, SBCMSRPReinviteUpdateSupport,
SBCMSRPOfferSetupRole, SBCMSRPEmpMsg;
IpProfile 1 = "Session_Manager", 1, "AudioCodersGroups_0", 0, 10, 10, 46, 24,
0, 0, 0, 0, 0, 0, 1, -1, 1, 0, 1, -1, 1, 4, -1, 1, 1, 0, 0, "", "", 0, 0, "",
"", "", 0, 0, 0, 0, 0, 0, 0, 8, 300, 400, 0, 0, 0, "", 0, 0, 1, 3, 0, 2, 2,
1, 0, 0, 1, 0, 1, 0, 0, 0, 0, 0, 1, 0, 0, 0, 0, 0, 0, 1, 0, 0, 0, 0, 0, 0, 0,
0, 0, 0, 0, 0, 300, -1, -1, 0, 0, 0, 0, 0, 0, 0, -1, -1, -1, -1, -1, 0, "",
0, 0, 0, 0, 0, 0, 0, 0, 0, 0, -1, -1, 0, 0, 0, 0, 1, 2, 0;

[ \IpProfile ]


[ CpMediaRealm ]

FORMAT Index = MediaRealmName, IPv4IF, IPv6IF, RemoteIPv4IF, RemoteIPv6IF,
PortRangeStart, MediaSessionLeg, PortRangeEnd, TCPPortRangeStart,
TCPPortRangeEnd, IsDefault, QoeProfile, BWProfile, TopologyLocation;
CpMediaRealm 0 = "DefaultRealm", "mp1288.avaya.com", "", "", "", 6000, 5953,
65529, 0, 0, 1, "", "", 0;

[ \CpMediaRealm ]


[ SBCRoutingPolicy ]

FORMAT Index = Name, LCREnable, LCRAverageCallLength, LCRDefaultCost,
LdapServerGroupName;
SBCRoutingPolicy 0 = "Default_SBCRoutingPolicy", 0, 1, 0, "";

[ \SBCRoutingPolicy ]


[ SRD ]

FORMAT Index = Name, BlockUnRegUsers, MaxNumOfRegUsers,
EnableUnAuthenticatedRegistrations, SharingPolicy, UsedByRoutingServer,
SBCOperationMode, SBCRoutingPolicyName, SBCDialPlanName, AdmissionProfile;
SRD 0 = "DefaultSRD", 0, -1, 1, 0, 0, 0, "Default_SBCRoutingPolicy", "", "";
```

```
[ \SRD ]


[ SIPInterface ]

FORMAT Index = InterfaceName, NetworkInterface,
SCTPSecondaryNetworkInterface, ApplicationType, UDPPort, TCPPort, TLSPort,
SCTPPort, AdditionalUDPPorts, AdditionalUDPPortsMode, SRDName,
MessagePolicyName, TLSContext, TLSMutualAuthentication, TCPKeepaliveEnable,
ClassificationFailureResponseType, PreClassificationManSet,
EncapsulatingProtocol, MediaRealm, SBCDirectMedia, BlockUnRegUsers,
MaxNumOfRegUsers, EnableUnAuthenticatedRegistrations, UsedByRoutingServer,
TopologyLocation, PreParsingManSetName, AdmissionProfile,
CallSetupRulesSetId;
SIPInterface 0 = "SIPInterface_0", "mp1288.avaya.com", "", 0, 0, 0, 5061, 0,
"", 0, "DefaultSRD", "", "audiocodes1288", -1, 0, 500, -1, 0, "", 0, -1, -1,
-1, 0, 0, "", "", -1;

[ \SIPInterface ]


[ ProxySet ]

FORMAT Index = ProxyName, EnableProxyKeepAlive, ProxyKeepAliveTime,
ProxyLoadBalancingMethod, IsProxyHotSwap, SRDName, ClassificationInput,
TLSContextName, ProxyRedundancyMode, DNSResolveMethod, KeepAliveFailureResp,
GWIPv4SIPInterfaceName, SBCIPv4SIPInterfaceName, GWIPv6SIPInterfaceName,
SBCIPv6SIPInterfaceName, MinActiveServersLB, SuccessDetectionRetries,
SuccessDetectionInterval, FailureDetectionRetransmissions;
ProxySet 0 = "ProxySet_0", 0, 60, 0, 0, "DefaultSRD", 0, "audiocodes1288", -
1, -1, "", "SIPInterface_0", "", "", "", 1, 1, 10, -1;

[ \ProxySet ]


[ IPGroup ]

FORMAT Index = Type, Name, ProxySetName, VoiceAIConnector, SIPGroupName,
ContactUser, SipReRoutingMode, AlwaysUseRouteTable, SRDName, MediaRealm,
InternalMediaRealm, ClassifyByProxySet, ProfileName, MaxNumOfRegUsers,
InboundManSet, OutboundManSet, RegistrationMode, AuthenticationMode,
MethodList, SBCServerAuthType, OAuthHTTPService, EnableSBCClientForking,
SourceUriInput, DestUriInput, ContactName, Username, Password, UUIFormat,
QOEProfile, BWProfile, AlwaysUseSourceAddr, MsgManUserDef1, MsgManUserDef2,
SIPConnect, SBCPSAPMode, DTLSContext, CreatedByRoutingServer,
UsedByRoutingServer, SBCOperationMode, SBCRouteUsingRequestURIPort,
SBCKeepOriginalCallID, TopologyLocation, SBCDialPlanName,
CallSetupRulesSetId, Tags, SBCUserStickiness, UserUDPPortAssignment,
AdmissionProfile, ProxyKeepAliveUsingIPG, SBCAltRouteReasonsSetName,
TeamsMediaOptimization, TeamsMOInitialBehavior, SIPSourceHostName;
IPGroup 0 = 0, "Default_IPG", "ProxySet_0", "", "avaya.com", "", -1, 0,
"DefaultSRD", "DefaultRealm", "DefaultRealm", 0, "Session_Manager", -1, -1, -
1, 0, 0, "", -1, "", 0, -1, -1, "", "", "$1$gQ==", 0, "", "", 0, "", "", 0,
0, "audiocodes1288", 0, 0, -1, 0, 0, 0, "", -1, "", 0, 0, "", 1, "", 0, 0,
"";
```

```
[ \IPGroup ]


[ PREFIX ]

FORMAT Index = RouteName, DestinationPrefix, DestAddress, SourcePrefix,
ProfileName, MeteringCodeName, DestPort, DestIPGroupName, TransportType,
SrcTrunkGroupID, DestSIPInterfaceName, CostGroup, ForkingGroup,
CallSetupRulesSetId, DestTags, SrcTags;
PREFIX 0 = "Tel to Ip", "*", "10.64.110.212", "*", "", "", 5061, "", 2, 1,
"SIPInterface_0", "", -1, -1, "", "";

[ \PREFIX ]


[ TrunkGroup ]

FORMAT Index = TrunkGroupNum, FirstTrunkId, FirstBChannel, LastBChannel,
FirstPhoneNumber, ProfileName, LastTrunkId, Module;
TrunkGroup 0 = 1, 255, 1, 1, "70111", "", 255, 1;
TrunkGroup 1 = 1, 255, 2, 2, "70112", "", 255, 1;

[ \TrunkGroup ]


[ PstnPrefix ]

FORMAT Index = RouteName, DestPrefix, TrunkGroupId, SourcePrefix,
SourceAddress, ProfileName, SrcIPGroupName, DestHostPrefix, SrcHostPrefix,
SrcSIPInterfaceName, TrunkId, CallSetupRulesSetId, DestType, DestTags,
SrcTags;
PstnPrefix 0 = "IP to Tel", "*", 1, "*", "*", "", "", "*", "*",
"SIPInterface_0", -1, -1, 0, "", "";

[ \PstnPrefix ]


[ ProxyIp ]

FORMAT Index = ProxySetId, ProxyIpIndex, IpAddress, TransportType, Priority,
Weight;
ProxyIp 0 = "0", 0, "10.64.110.212", 2, 0, 0;

[ \ProxyIp ]


[ TrunkGroupSettings ]

FORMAT Index = TrunkGroupId, ChannelSelectMode, RegistrationMode,
GatewayName, ContactUser, ServingIPGroupName, MWIInterrogationType,
TrunkGroupName, UsedByRoutingServer, AdminState, DedicatedConnectionMode;
TrunkGroupSettings 0 = 1, 0, 0, "", "", "Default_IPG", 255, "TrunkGroup-1",
0, 0, 0;

[ \TrunkGroupSettings ]
```

```
[ Authentication ]

FORMAT Index = UserId, UserPassword, Module, Port, PortType;
Authentication 0 = "70111", "$1$tIWHhYONjw==", 1, 1, "FXS";
Authentication 1 = "70112", "$1$tIWHhYONjw==", 1, 2, "FXS";

[ \Authentication ]


[ GwRoutingPolicy ]

FORMAT Index = Name, LCREnable, LCRAverageCallLength, LCRDefaultCost,
LdapServerGroupName;
GwRoutingPolicy 0 = "GwRoutingPolicy", 0, 1, 0, "";

[ \GwRoutingPolicy ]


[ ResourcePriorityNetworkDomains ]

FORMAT Index = Name, Ip2TelInterworking;
ResourcePriorityNetworkDomains 1 = "dsn", 1;
ResourcePriorityNetworkDomains 2 = "dod", 1;
ResourcePriorityNetworkDomains 3 = "drsn", 1;
ResourcePriorityNetworkDomains 5 = "uc", 1;
ResourcePriorityNetworkDomains 7 = "cuc", 1;

[ \ResourcePriorityNetworkDomains ]


[ AudioCoders ]

FORMAT Index = AudioCodersGroupId, AudioCodersIndex, Name, pTime, rate,
PayloadType, Sce, CoderSpecific;
AudioCoders 0 = "AudioCodersGroups_0", 0, 2, 2, 90, -1, 0, "";
AudioCoders 1 = "AudioCodersGroups_0", 1, 1, 2, 90, -1, 0, "";

[ \AudioCoders ]
```

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.