



Avaya Solution & Interoperability Test Lab

Application Notes for AudioCodes MediaPack 11x with Avaya Aura® Communication Manager 7.0 and Avaya Aura® Session Manager 7.0 – Issue 1.0

Abstract

These Application Notes contain interoperability instructions for configuring AudioCodes MediaPack 11x 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) 11x Analog VoIP gateways implement voice technology that connect analog telephones, fax machines (FXS) and landlines (FXO) to IP based PBX systems. In the compliance test, AudioCodes MP-118 VoIP gateway was used to verify interoperability within an Avaya Aura® IP Telephony Environment.

2. General Test Approach and Test Results

Interoperability compliance testing focused on verifying various inbound and outbound call flows between AudioCodes MP-11x, 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-11x 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-11x
- Outgoing calls from AudioCodes MP-11x
- Voice codecs G.711U, G.711A and G.729 using SRTP
- Incoming and outgoing faxes using T.38
- 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 executed and passed with the following exception/observations:

MWI was tested by sending MWI indication to the MP-11x and verifying that stutter tone was heard on the analog phone.

2.3. Support

Technical support for AudioCodes MP-11x 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: crm.audiocodes.com
- E-Mail: support@audiocodes.com

3. Reference Configuration

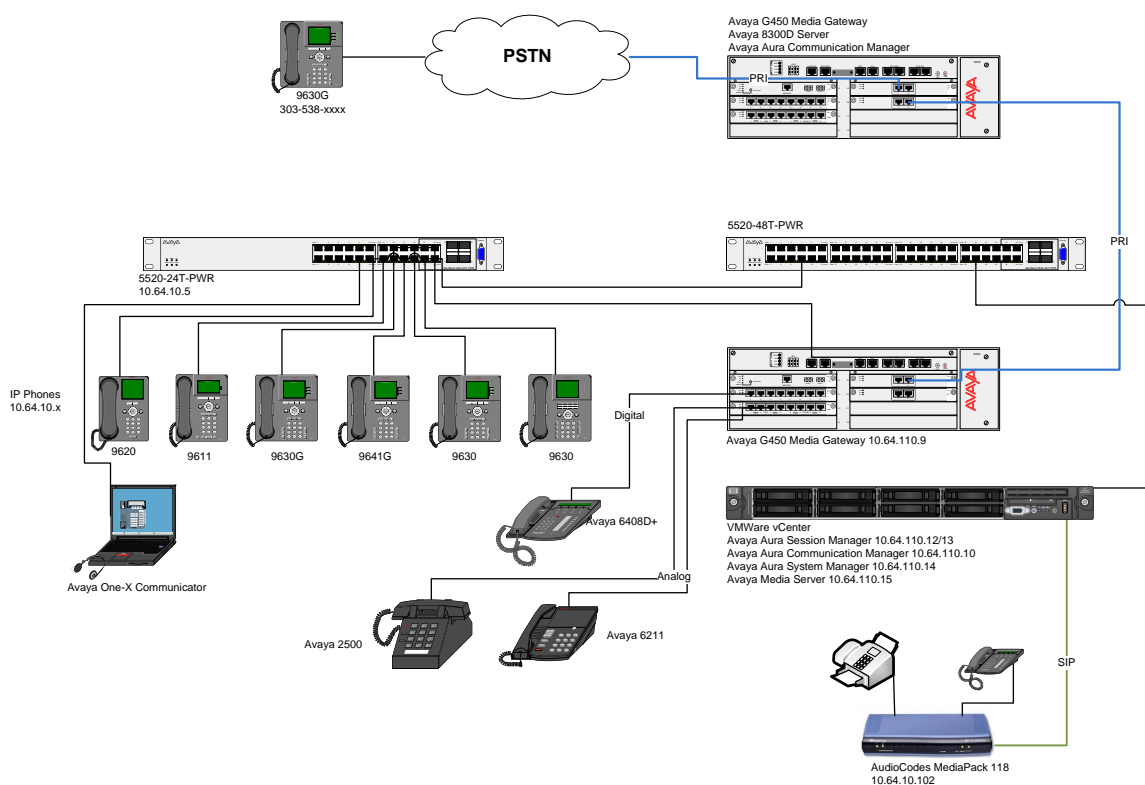


Figure 1: AudioCodes MP-118 with Avaya Aura® Communication Manager and Avaya Aura® Session Manager

4. Equipment and Software Validated

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

| Equipment/Software | Release/Version |
|--|--------------------------|
| Avaya Aura® Communication Manager running on Avaya S8300D Server | 7.0.1.1.1.441.23384 |
| Avaya Aura® Session Manager | 7.0.1.1.701114 |
| Avaya Aura® System Manager | 7.0.1.1.065378 |
| Avaya 9600 Series IP Deskphones | H.323 3.2.7 |
| | SIP 7.0.1.2, H.323 6.6.3 |
| Avaya 6408D+ Digital Deskphone | - |
| Avaya 2500 and 6221 Analog Phone | - |
| AudioCodes MediaPack 118 | 6.60A.328.004 |

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: 4000 | 20 | |
| Maximum Concurrently Registered IP Stations: 2400 | 2 | |
| Maximum Administered Remote Office Trunks: 4000 | 0 | |
| Maximum Concurrently Registered Remote Office Stations: 2400 | 0 | |
| Maximum Concurrently Registered IP eCons: 68 | 0 | |
| Max Concur Registered Unauthenticated H.323 Stations: 100 | 0 | |
| Maximum Video Capable Stations: 2400 | 0 | |
| Maximum Video Capable IP Softphones: 2400 | 15 | |
| Maximum Administered SIP Trunks: 4000 | 20 | |
| Maximum Administered Ad-hoc Video Conferencing Ports: 4000 | 0 | |
| Maximum Number of DS1 Boards with Echo Cancellation: 80 | 0 | |
| (NOTE: You must logoff & login to effect the permission changes.) | | |

On **Page 5**, verify **ISDN/PRI** and **Media Encryption Over IP** fields are set to **y**.

| | | |
|---|---|--------------|
| display system-parameters customer-options | | Page 4 of 11 |
| OPTIONAL FEATURES | | |
| Emergency Access to Attendant? y | IP Stations? y | |
| Enable 'dadmin' Login? y | | |
| Enhanced Conferencing? y | ISDN Feature Plus? n | |
| Enhanced EC500? y | ISDN/SIP Network Call Redirection? y | |
| Enterprise Survivable Server? n | ISDN-BRI Trunks? y | |
| Enterprise Wide Licensing? n | ISDN-PRI? y | |
| ESS Administration? y | Local Survivable Processor? n | |
| Extended Cvg/Fwd Admin? y | Malicious Call Trace? y | |
| External Device Alarm Admin? y | Media Encryption Over IP? 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.

5.2.1. IP Network Region for Voice and Fax Calls

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
H.323 IP ENDPOINTS    AUDIO RESOURCE RESERVATION PARAMETERS
  H.323 Link Bounce Recovery? y      RSVP Enabled? n
  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

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.729AB, 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 CODEC SET

Codec Set: 1

Audio      Silence      Frames      Packet
Codec      Suppression   Per Pkt    Size(ms)
1: G.711MU          n          2         20
2: G.729AB          n          2         20
3: G.711A          n          2         20
4:
5:
6:
7:

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

On **Page 2**:

- Set **Fax Mode** to **t.38-standard**

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

                                IP CODEC SET

                                Allow Direct-IP Multimedia? y
                                Maximum Call Rate for Direct-IP Multimedia: 2048:Kbits
                                Maximum Call Rate for Priority Direct-IP Multimedia: 2048:Kbits

                                Mode                                Redundancy                                Packet
                                t.38-standard                        0                                Size (ms)
FAX
Modem                                off                                0
TDD/TTY                              US                                3
H.323 Clear-channel                  n                                0
SIP 64K Data                          n                                0                                20

                                ECM: y
```

5.4. Administer IP Node Names

Use the **change node-names ip** command to add and entry for Session Manager. For compliance testing, **asm** and **10.64.110.13** entry was added.

| change node-names ip | | Page 1 of 2 |
|----------------------|---------------------|-------------|
| IP NODE NAMES | | |
| Name | IP Address | |
| acms | 10.64.110.18 | |
| aes | 10.64.110.15 | |
| ams | 10.64.110.16 | |
| asm | 10.64.110.13 | |
| biscom | 10.64.101.152 | |
| cms17 | 10.64.10.85 | |
| default | 0.0.0.0 | |
| egw1 | 10.64.110.200 | |
| egw2 | 10.64.110.201 | |
| hospitality | 10.64.10.47 | |
| kpc | 10.64.10.47 | |
| mx | 10.64.10.22 | |
| procr | 10.64.110.10 | |
| procr6 | :: | |

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 **Far-end Node Name** to the configured Session Manager in **Section 5.4**, i.e. **asm**
- Set **Far-end Network region** to the configured region in **Section 5.2.1**, i.e. **1**
- Enter a **Far-end Domain**, e.g. **avaya.com**

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

Group Number: 1                      Group Type: sip
IMS Enabled? n                      Transport Method: tls
  Q-SIP? n
  IP Video? n                      Enforce SIPS URI for SRTP? n
Peer Detection Enabled? y Peer Server: SM
Prepend '+' to Outgoing Calling/Alerting/Diverting/Connected Public Numbers? y
Remove '+' from Incoming Called/Calling/Alerting/Diverting/Connected Numbers? n
Alert Incoming SIP Crisis Calls? y
  Near-end Node Name: procr          Far-end Node Name: asm
  Near-end Listen Port: 5061        Far-end Listen Port: 5061
                                   Far-end Network Region: 1

Far-end Domain: avaya.com

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

Note: Signaling Group, Trunk Group and Route Pattern for PSTN calls via PRI was pre-configured and is 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.

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 a **Group Name**, e.g. **asm**
- Enter a valid **TAC**, e.g. **101**
- Set **Service Type** to **tie**
- Enter **Signaling Group** value to the signaling group configured in **Section 0**, i.e. **1**
- Enter a desired number in **Number of Member** field

```
change trunk-group 1                                     Page 1 of 21
                                     TRUNK GROUP
Group Number: 1                Group Type: sip            CDR Reports: y
  Group Name: asm              COR: 1                    TN: 1      TAC: 101
  Direction: two-way          Outgoing Display? y
  Dial Access? n
  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**

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

  Suppress # Outpulsing? n  Numbering Format: private
                                     UUI Treatment: service-provider
                                     Replace Restricted Numbers? n
                                     Replace Unavailable Numbers? n
```

5.7. Administer Route Pattern

Use the **change route-pattern *n*** command to configure a route pattern, where ***n*** is an available route patterns

5.7.1. Route Pattern for Voice and Fax Calls

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 5.6.1**, i.e. **1**
- For line 1, set **FRL** to **0**

| | | |
|-------------------------------------|------------------------------------|---------------|
| change route-pattern 1 | Page | 1 of 3 |
| Pattern Number: 1 | Pattern Name: Voice and Fax | |
| SCAN? n | Secure SIP? n | |
| Grp FRL NPA Pfx Hop Toll No. | DCS/ IXC | |
| No | QSIG | |
| Mrk Lmt List Del Digits | | |
| 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 1 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 Len | Ext Code | Trk Grp(s) | Private Prefix | Total Len | |
| 5 | 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**. Add two entries, one for voice and fax calls and another one for modem calls. For compliance testing, extensions starting with **111** were used for voice and fax calls.

- Set **Dialed String** to starting digits of extensions that will be used, e.g. **111**
- 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 11 | | | | | | | Page 1 of 2 |
| AAR DIGIT ANALYSIS TABLE | | | | | | | |
| Location: all | | | | | | | Percent Full: 0 |
| | Dialed String | Total Min | Total Max | Route Pattern | Call Type | Node Num | ANI Req'd |
| 111 | | 5 | 5 | 1 | lev0 | | n |

5.10. Administer ARS Analysis

Use the **change ars analysis *n*** command to configure routing for calling to PSTN, where *n* is an NPA or starting digit of PSTN numbers. This configuration is used to route PSTN calls to an FXO line on AudioCodes MP-11x. For compliance testing, PSTN calls were placed to phones with NPA of 303.

- Set **Dialed String** to **303**
- Set **Total Min** and **Total Max** to **10**
- Set **Route Pattern** to pattern configured in **Section 5.7**, i.e. **1**
- Set **Call Type** to **hnpa**

Note: This administration is only required if FXO line is being configured on AudioCodes MP-11x

| change ars analysis 3 | | | | | | Page 1 of 2 | |
|--------------------------|---------------|-------|-----|---------|------|-----------------|------|
| ARS DIGIT ANALYSIS TABLE | | | | | | | |
| Location: all | | | | | | Percent Full: 3 | |
| | Dialed String | Total | | Route | Call | Node | ANI |
| | | Min | Max | Pattern | Type | Num | Reqd |
| 303 | | 10 | 10 | 1 | hnpa | | n |
| 4 | | 7 | 7 | deny | hnpa | | n |
| 411 | | 3 | 3 | deny | svcl | | n |
| 444 | | 10 | 10 | 2 | fnpa | | n |
| 5 | | 7 | 7 | deny | hnpa | | n |
| 555 | | 7 | 7 | deny | hnpa | | n |
| 6 | | 7 | 7 | deny | hnpa | | n |
| 611 | | 3 | 3 | 1 | svcl | | n |

5.11. Administer Stations

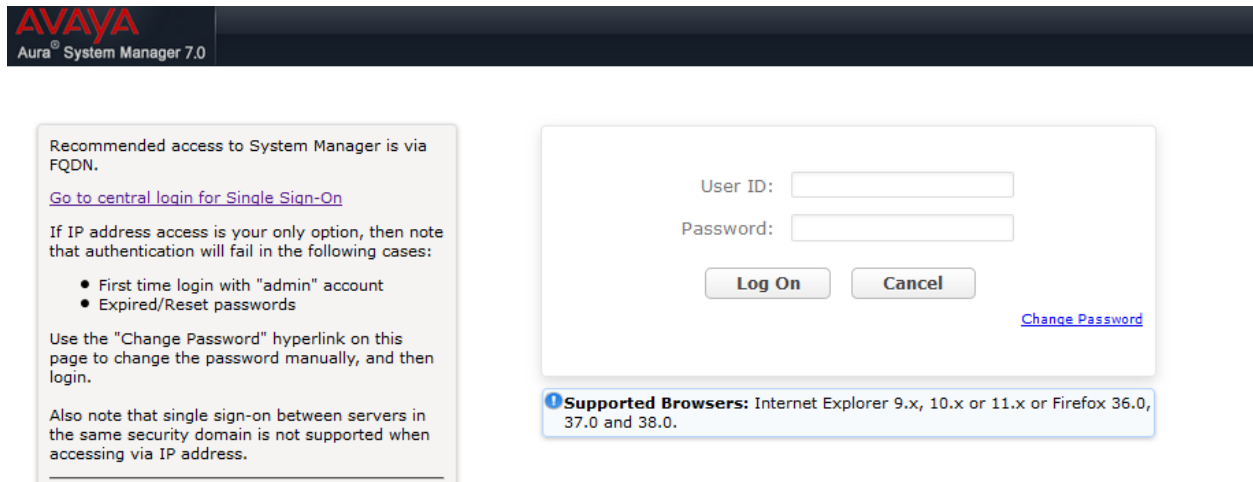
Administration of Avaya Stations/Extensions in Communication Manager and Session Manager is not shown in this document. Please refer to document [1] and/or [2] in reference section of this document.

Note: During compliance testing, Media Encryption settings were modified in the 46xxsettings.txt to enable SRTP for the Avaya SIP Deskphones. TLS registration for Avaya Deskphones is required for SRTP and is enabled by default.

```
## Media Encryption Support
## Specifies media encryption (SRTP) options supported by
## phone. Up to 2 options may be selected. Values are in
## comma-separated list. Options should match those
## specified in CM IP-codec-set form.
## 1 = aescm128-hmac80
## 2 = aescm128-hmac32
## 3 = aescm128-hmac80-unauth
## 4 = aescm128-hmac32-unauth
## 5 = aescm128-hmac80-unenc
## 6 = aescm128-hmac32-unenc
## 7 = aescm128-hmac80-unenc-unauth
## 8 = aescm128-hmac32-unenc-unauth
## 9 = none (default)
SET MEDIAENCRYPTION "1"
```

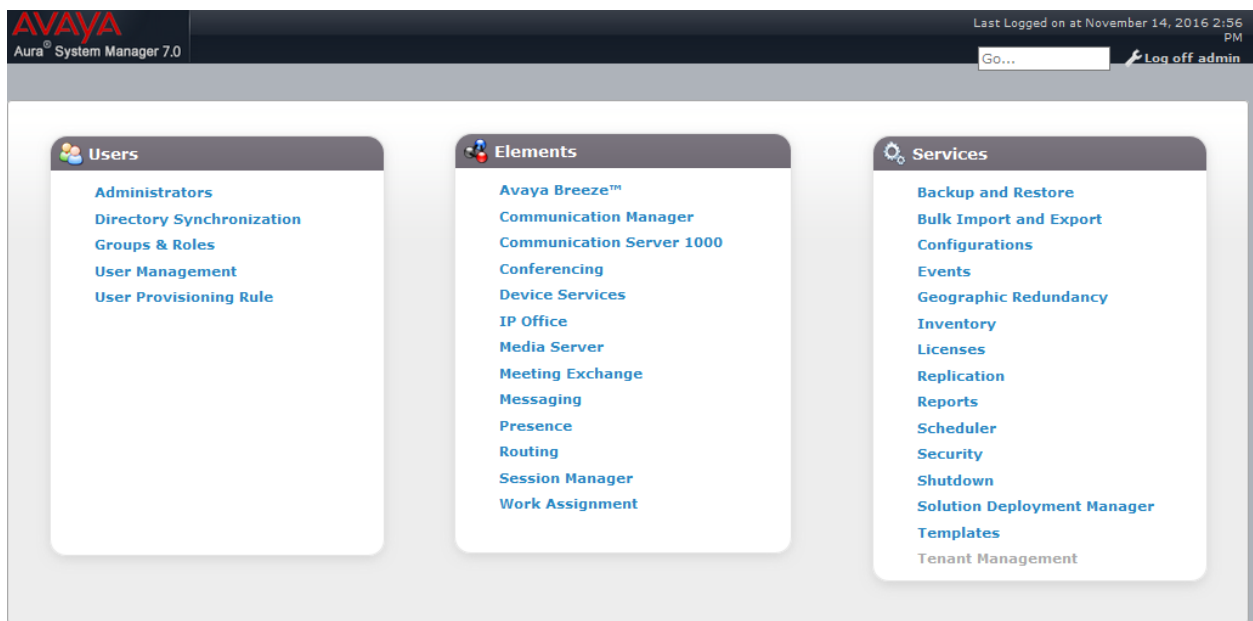
6. Configure Avaya Aura® Session Manager

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.



The image shows the Avaya Aura System Manager 7.0 login page. At the top left is the Avaya logo and "Aura® System Manager 7.0". The main content area is divided into two columns. The left column contains instructions: "Recommended access to System Manager is via FQDN. Go to central login for Single Sign-On. If IP address access is your only option, then note that authentication will fail in the following cases: • First time login with 'admin' account • Expired/Reset passwords. Use the 'Change Password' hyperlink on this page to change the password manually, and then login. Also note that single sign-on between servers in the same security domain is not supported when accessing via IP address." The right column contains a login form with "User ID:" and "Password:" labels, each followed by a text input field. Below the fields are "Log On" and "Cancel" buttons. A "Change Password" link is at the bottom right of the form. At the bottom of the right column is a blue box with a warning icon and text: "Supported Browsers: Internet Explorer 9.x, 10.x or 11.x or Firefox 36.0, 37.0 and 38.0."

Log in using appropriate credentials



6.1. Add SIP Domain

Navigate to **Home → 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.

AVAYA
Aura® System Manager 7.0

Last Logged on at November 14, 2016 2:56 PM
Go... Log off admin

Home Routing

Routing
Domains
Locations
Adaptations
SIP Entities
Entity Links
Time Ranges
Routing Policies
Dial Patterns
Regular Expressions
Defaults

Home / Elements / Routing / Domains

Domain Management

Commit Cancel

1 Item Filter: Enable

| Name | Type | Notes |
|-------------|------|-------|
| * avaya.com | sip | |

Commit Cancel

6.2. Add Location

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

Under **General**:

- Type in a descriptive **Name**

Under **Location Pattern** click on **New** (not shown):

- Type in **IP Address Pattern** for applicable subnets, e.g. **10.64.10.***

Click **Commit** to save changes.

Location Details [Commit] [Cancel]

General

* Name: DevConnect-Lab

Notes:

Dial Plan Transparency in Survivable Mode

Enabled: ☐

Listed Directory Number:

Associated CM SIP Entity:

Overall Managed Bandwidth

Managed Bandwidth Units: Kbit/sec

Total Bandwidth:

Multimedia Bandwidth:

Audio Calls Can Take Multimedia Bandwidth: ☒

Per-Call Bandwidth Parameters

Maximum Multimedia Bandwidth (Intra-Location): 2000 Kbit/Sec

Maximum Multimedia Bandwidth (Inter-Location): 2000 Kbit/Sec

* Minimum Multimedia Bandwidth: 64 Kbit/Sec

* Default Audio Bandwidth: 80 Kbit/sec

Alarm Threshold

Overall Alarm Threshold: 80 %

Multimedia Alarm Threshold: 80 %

* Latency before Overall Alarm Trigger: 5 Minutes

* Latency before Multimedia Alarm Trigger: 5 Minutes

Location Pattern

Add Remove

2 Items Filter: Enable

| | IP Address Pattern | Notes |
|--------------------------|--------------------|-------|
| <input type="checkbox"/> | * 10.64.10.* | |
| <input type="checkbox"/> | * 10.64.101.* | |

Select : All, None

6.3. Add SIP Entity

Add Communication Manager as a SIP Entity. Navigate to **Home → 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.

The screenshot displays the Avaya Aura System Manager 7.0 web interface. The top navigation bar includes the Avaya logo, the text 'Aura System Manager 7.0', and a 'Last Logged on at November 14, 2016 2:56 PM' status. A search bar with 'Go...' and a 'Log off admin' link are also present. The left sidebar shows a tree view with 'Routing' selected, and its sub-items: Domains, Locations, Adaptations, SIP Entities (highlighted), Entity Links, Time Ranges, Routing Policies, Dial Patterns, Regular Expressions, and Defaults. The main content area is titled 'SIP Entity Details' and includes 'Commit' and 'Cancel' buttons. Under the 'General' tab, the following fields are visible: 'Name' (text box with 'acm'), 'FQDN or IP Address' (text box with '10.64.110.10'), 'Type' (dropdown menu with 'CM' selected), 'Notes' (text box), 'Adaptation' (dropdown menu), 'Location' (dropdown menu with 'DevConnect-Lab' selected), 'Time Zone' (dropdown menu with 'America/Denver' selected), 'SIP Timer B/F (in seconds)' (text box with '4'), 'Credential name' (text box), and 'Securable' (checkbox).

6.4. Add Entity Link

Navigate to **Home → 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.

The screenshot shows the Avaya Aura System Manager 7.0 web interface. The top header includes the Avaya logo, the text "Aura System Manager 7.0", and a "Last Logged on at November 14, 2016 2:56 PM" status. A "Go..." search bar and a "Log off admin" link are also present. The left sidebar contains a navigation menu with options: Home, Routing, Domains, Locations, Adaptations, SIP Entities, Entity Links (selected), Time Ranges, Routing Policies, Dial Patterns, Regular Expressions, and Defaults. The main content area is titled "Entity Links" and includes "Commit" and "Cancel" buttons. Below the title is a table with the following structure:

| 1 Item | | | | | Filter: Enable |
|--------------------------|--------------------|--------------|----------|--------|----------------|
| <input type="checkbox"/> | Name | SIP Entity 1 | Protocol | Port | SIP Entity 2 |
| <input type="checkbox"/> | * asm_acm_5061_TLS | * Qasm | TLS | * 5061 | * Qacm |

Below the table, there is a "Select : All, None" option and a "Help ?" link in the top right corner of the main area.

6.5. Add Routing Policy

Navigate to **Home → 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.

AVAYA
Aura® System Manager 7.0

Last Logged on at November 14, 2016 2:56 PM
Go... Log off admin

Home Routing x

Home / Elements / Routing / Routing Policies

Routing Policy Details

Commit Cancel

Help ?

General

* Name: acm

Disabled: ☐

* Retries: 0

Notes:

SIP Entity as Destination

Select

| Name | FQDN or IP Address | Type | Notes |
|------|--------------------|------|-------|
| acm | 10.64.110.10 | CM | |

6.6. Add Dial Patterns

Navigate to **Home → 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.

Note: For Compliance testing, dialed number of 110xx were used.

AVAYA
Aura® System Manager 7.0

Last Logged on at November 14, 2016 2:56 PM
Go... Log off admin

Home Routing

Home / Elements / Routing / Dial Patterns

Dial Pattern Details

Commit Cancel

Help ?

General

* Pattern: 110

* Min: 4

* Max: 5

Emergency Call: ☐

Emergency Priority: 1

Emergency Type:

SIP Domain: -ALL-

Notes:

Originating Locations and Routing Policies

Add Remove

1 Item Filter: Enable

| <input type="checkbox"/> | Originating Location Name | Originating Location Notes | Routing Policy Name | Rank | Routing Policy Disabled | Routing Policy Destination | Routing Policy Notes |
|--------------------------|---------------------------|----------------------------|---------------------|------|--------------------------|----------------------------|----------------------|
| <input type="checkbox"/> | DevConnect-Lab | | acm | 3 | <input type="checkbox"/> | acm | |

Select : All, None

6.7. Add User

For each analog line on AudioCodes MP-11x, a user needs to be added on Session Manager. Information in this section will be used by AudioCodes MP-11x for registering to Session Manager.

Navigate to **Home → 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-11x to receive and make calls. <domain> is as configured in **Section 6.1**

The screenshot shows the Avaya Aura System Manager 7.0 interface. The top navigation bar includes 'Home', 'Routing', and 'User Management'. The left sidebar lists 'User Management' options: 'Manage Users', 'Public Contacts', 'Shared Addresses', 'System Presence', 'ACLs', 'Communication', 'Profile Password', and 'Policy'. The main content area is titled 'User Profile Edit: 11151@avaya.com' and includes buttons for 'Commit & Continue', 'Commit', and 'Cancel'. The 'Identity' tab is selected, showing fields for 'Last Name' (Audiocodes), 'First Name' (User 1), 'Middle Name', 'Description', 'Update Time' (November 1, 2016 2:54:18), 'Login Name' (11151@avaya.com), and 'User Type' (Basic). A 'User Provisioning Rule' dropdown is also visible.

Under **Communication Profile** tab:

- Under **Communication Profile**, type in desired **Communication Profile Password** and **Confirm Password**
- Under **Communication Address**, click on **New** (now shown)
 - 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-11x will use this information as login name to register to Session Manager
- Check the **Session Manager Profile** box:
 - Set **Primary Session Manager** to Session Manager. i.e. acm
 - 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**
- Check the **CM Endpoint Profile** box:
 - 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 **9630SIP_DEFAULT_CM_7_0**
 - Set **Security Code** to a desired value. Please note that AudioCodes MP-11x will use this security code as password to register to Session Manager

Click **Commit** to save changes.

Identity * Communication Profile Membership Contacts

Communication Profile *

Communication Profile Password:

Confirm Password: [Cancel](#)

[New](#) [Delete](#) [Done](#) [Cancel](#)

Name

☒ Primary

Select : None

* Name:

Default : ☒

Communication Address *

[New](#) [Edit](#) [Delete](#)

| Type | Handle | Domain |
|------------------------------------|--------|-----------|
| <input type="checkbox"/> Avaya SIP | 11151 | avaya.com |

Select : All, None

☒ Session Manager Profile *

SIP Registration

* Primary Session Manager:

| Primary | Secondary | Maximum |
|---------|-----------|---------|
| 12 | 0 | 12 |

Secondary Session Manager:

Survivability Server:

Max. Simultaneous Devices:

Block New Registration When Maximum Registrations Active? ☐

Application Sequences

Origination Sequence:

Termination Sequence:

Call Routing Settings

* Home Location:

Conference Factory Set:

Call History Settings

Enable Centralized Call History? ☐

☐ Avaya Breeze Profile *

☒ CM Endpoint Profile *

* System:

* Profile Type:

Use Existing Endpoints: ☐

* Extension: [Endpoint Editor](#)

Template:

Set Type:

7. Configure AudioCodes MediaPack 11x

Administration for AudioCodes MP-11x series is done via administrative console. Type in <http://<ip-address>> URL in a web browser, where <ip-address> is the IP Address of AudioCodes MP-11x.

Note: It is assumed that AudioCodes MP-11x has been assigned an IP address. If AudioCodes MP-11x is using the factory defaults, change the computer IP address to 10.1.10.1 with a subnet of 255.255.0.0 so that it can communicate with the device using its factory default IP address of 10.1.10.10. For further details, please see document [5] in additional references section of this document.

Note: Configuration mentioned in the section is performed for AudioCodes MP-118. AudioCodes MP-11x series devices' administrative console is similar to the one that is configured in this section.

Log on to administrative console using appropriate credentials.

7.1. Verify/Upgrade Firmware Version

Once logged in, the firmware version can be found on the Home page under **General Information** box. The firmware version should be **6.60A** or higher.

Please note that during the compliance testing, the version of AudioCodes MP-11x used, was **6.60A.328.003**.

If the version is different, contact AudioCodes support and get the correct version firmware. Once it has been obtained:

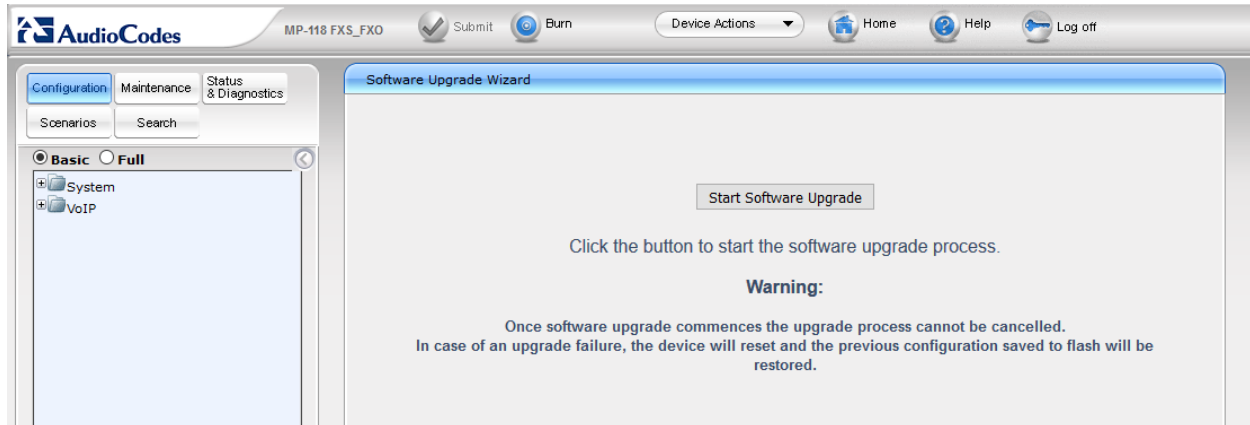
- Click on the drop down menu as shown and select **Software Upgrade Wizard**

The screenshot displays the AudioCodes MP-118 FXS_FXO Home Page. The interface includes a top navigation bar with the AudioCodes logo, the device name 'MP-118 FXS_FXO', and buttons for 'Submit', 'Burn', 'Device Actions', 'Home', and 'Log off'. On the left, there is a sidebar with tabs for 'Configuration', 'Maintenance', and 'Status & Diagnostics'. Under 'Configuration', there are sub-tabs for 'Scenarios' and 'Search'. The 'Basic' tab is selected, showing a tree view with 'System' and 'VoIP' options. The main content area features a 'MP-118 FXS_FXO Home Page' header, an 'Alarms' section with a row of 8 status indicators (all green), and a 'General Information' table. A 'Color-Code Key' legend is also present on the right.

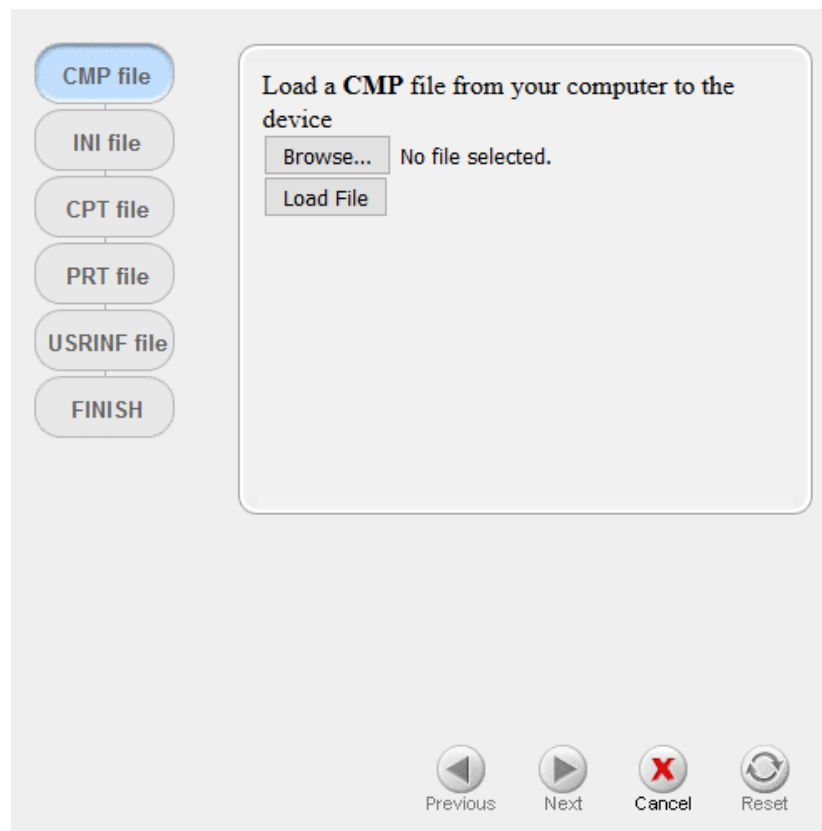
| General Information | |
|---------------------------|---------------|
| IP Address | 10.64.10.102 |
| Subnet Mask | 255.255.255.0 |
| Default Gateway Address | 10.64.10.1 |
| Firmware Version | 6.60A.328.003 |
| Protocol Type | SIP |
| Gateway Operational State | UNLOCKED |
| Analog Ports Number | 8 |

| Color-Code Key | |
|-------------------|--|
| ● Fail | |
| ○ Inactive | |
| ● Handset Offhook | |
| ● RTP Active | |

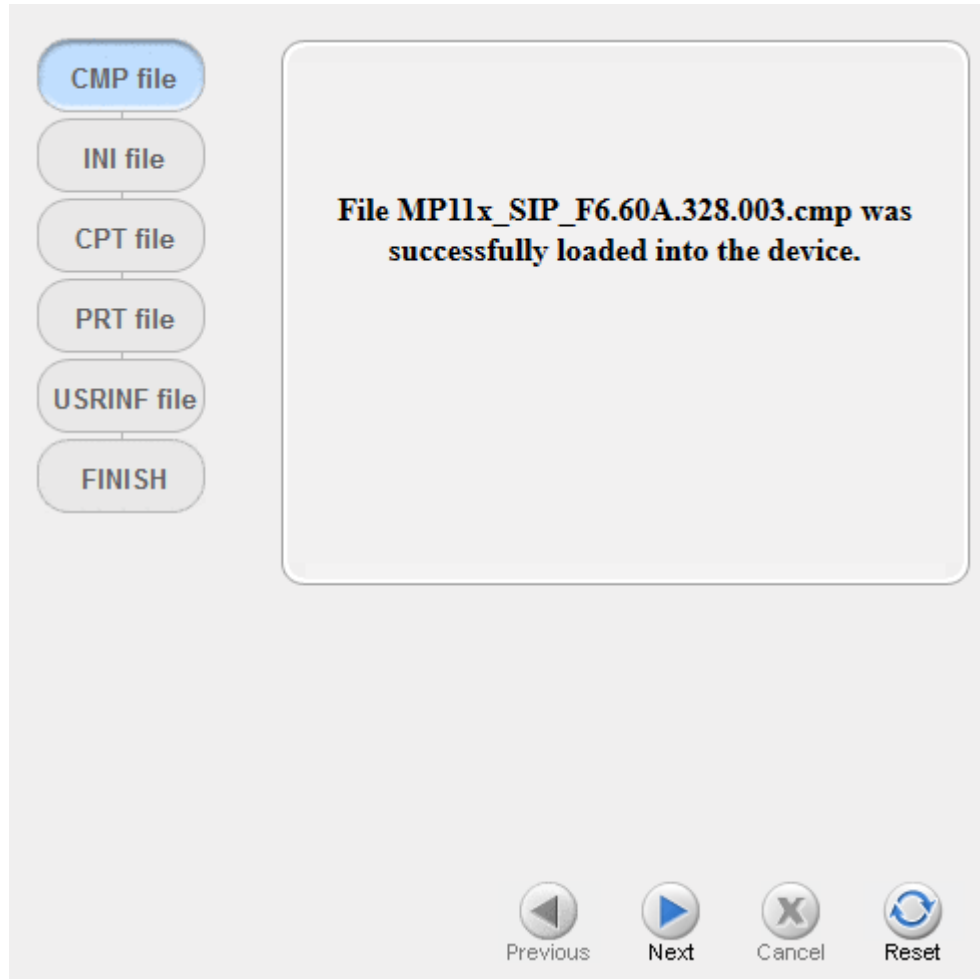
- Click on **Start Software Upgrade**



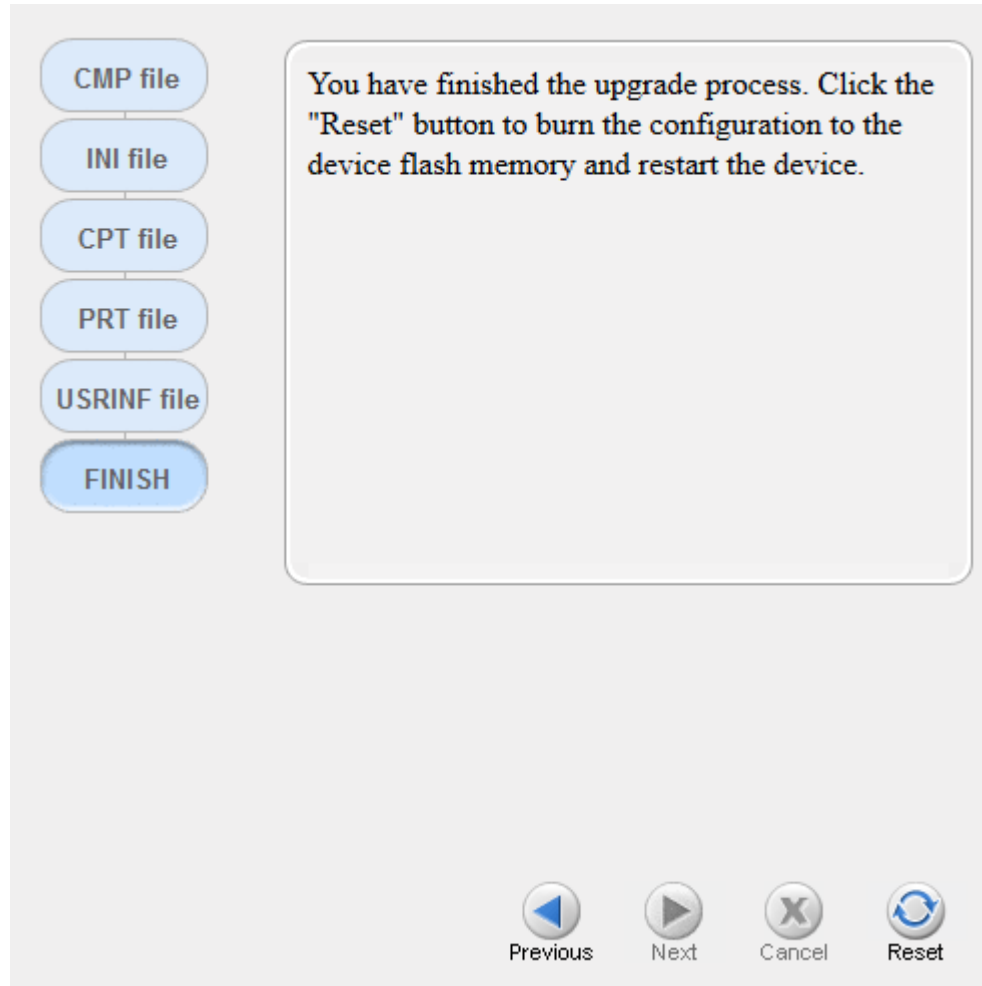
- A pop-up window will open, click on **CMP File**. Select **Browse** and locate the firmware (not shown)
- Click on **Load File**



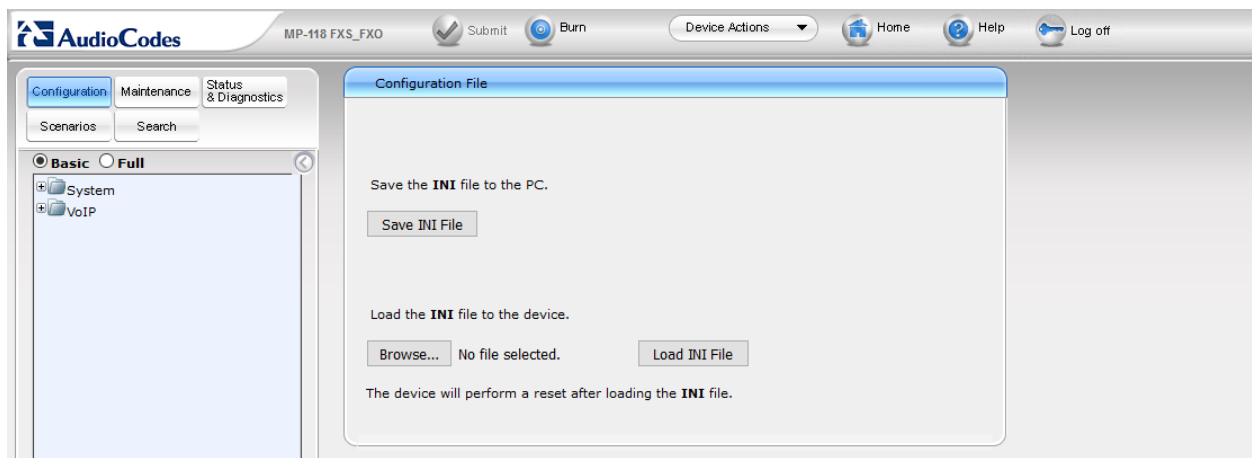
- Once the file is loaded on the device, following text will be displayed



- Click **Reset** to upgrade the device.



- Verify the version once the upgrade process is complete as mentioned earlier in this section
- To ensure that the configuration is default, in a case where the box was previously used, upload an **ini** file that is empty:
- Open Notepad on a Windows machine, and save as **empty.ini**
- From AudioCodes MP-11x console, click on the drop down menu **Device Actions** in the top center of the screen, and select **Load Configuration File** (not shown)
- Click on **Choose File** and navigate to the location of recently created **empty.ini** and select it.
- Click on **Send INI File**
- Click **OK** to confirm (not shown)



7.2. Administer Application Settings

All administration is done via the left pane on administrative console.

- Click on **Full** as shown on the screen shot
- Expand **System** folder and click on **Application Settings**
- Fill in **NTP Server IP Address**
- Fill in **NTP UTC Offset** for the local time zone
- Under **DHCP Setting**, ensure that **Enable DHCP** is set to **Disable** (not shown)

Note: For compliance testing a local NTP server was provided and an offset of -6 was used
Click on **Submit** to save changes.

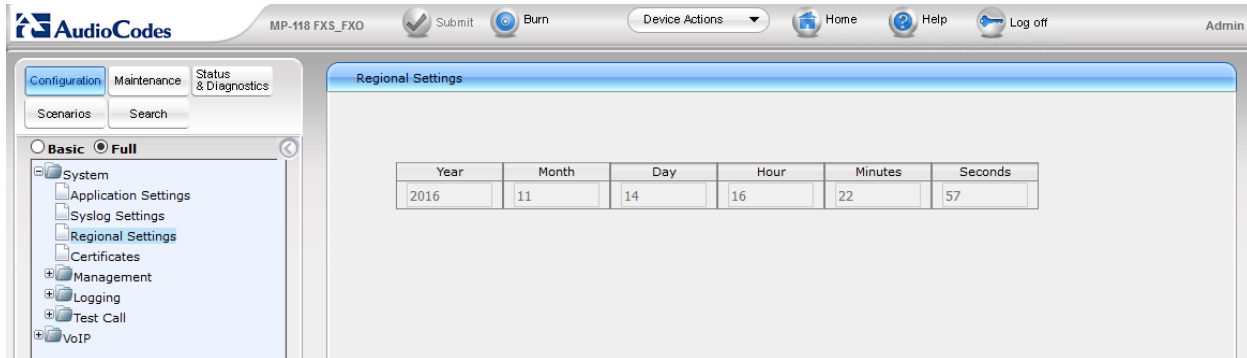
The screenshot displays the AudioCodes administrative console interface. The top navigation bar includes the AudioCodes logo, a device identifier 'MP-118 FXS_FXO', and buttons for 'Submit', 'Burn', 'Device Actions', 'Home', 'Help', and 'Log off'. The user 'Admin' is logged in.

The left-hand navigation pane is divided into 'Configuration', 'Maintenance', and 'Status & Diagnostics' tabs. Under 'Configuration', there are 'Scenarios' and 'Search' buttons. The 'System' folder is expanded, showing sub-items: 'Application Settings' (selected), 'Syslog Settings', 'Regional Settings', 'Certificates', 'Management', 'Logging', 'Test Call', and 'VoIP'. The 'Full' configuration mode is selected.

The main content area, titled 'Application Settings', contains several sections:

- NTP Settings:** Includes fields for 'NTP Server Address (IP or FQDN)' (128.138.140.44), 'NTP UTC Offset' (Hours: -6, Minutes: 0), 'NTP Updated Interval' (Hours: 24, Minutes: 0), and 'NTP Secondary Server IP'.
- Day Light Saving Time:** Includes a 'Day Light Saving Time' dropdown (Disable), 'DST Mode' (Day of year), 'Start Time' (Mar 09 2 : 0), 'End Time' (Nov 02 2 : 0), 'Offset [min]' (60), 'Day of Month Start' (Mar Sunday First 2 : 0), and 'Day of Month End' (Nov Sunday First 2 : 0).
- STUN Settings:** Includes 'Enable STUN' (Disable), 'STUN Server Primary IP' (0.0.0.0), and 'STUN Server Secondary IP' (0.0.0.0).
- NFS Settings:** Includes an 'NFS Table' button.
- DHCP Settings:** Includes 'Enable DHCP' (Disable).

- Click on **Regional Setting** on the left pane. It should now display the correct local time



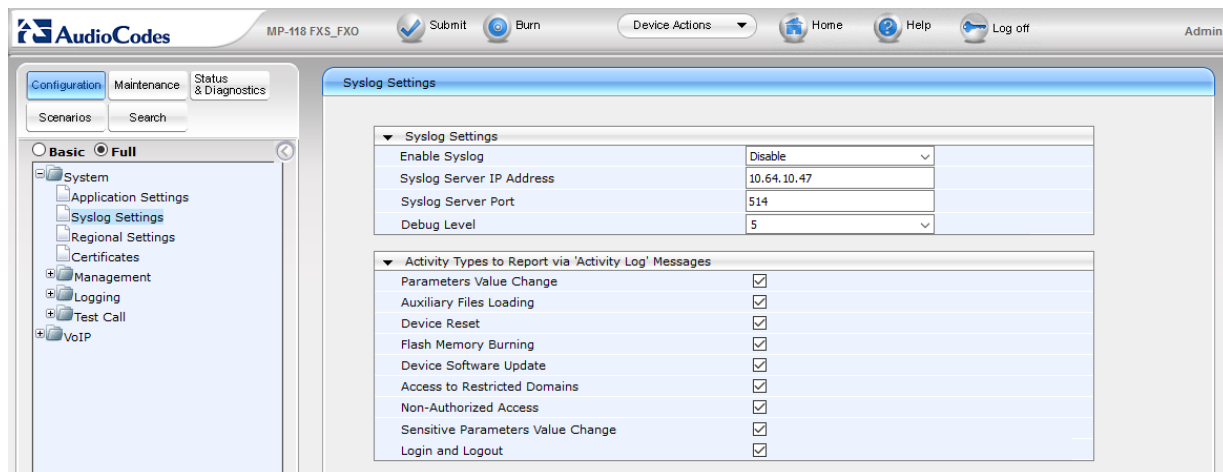
7.3. Administer Syslog Settings

If Syslog needs to be enabled, 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 **Debug Level** to **5**
- Under **Activity Types to Report via 'Activity Log' Messages**, check all boxes

Click **Submit** to save changes.

Reset the device to save changes to flash memory of AudioCodes MP-11x.



7.4. Administer Certificates

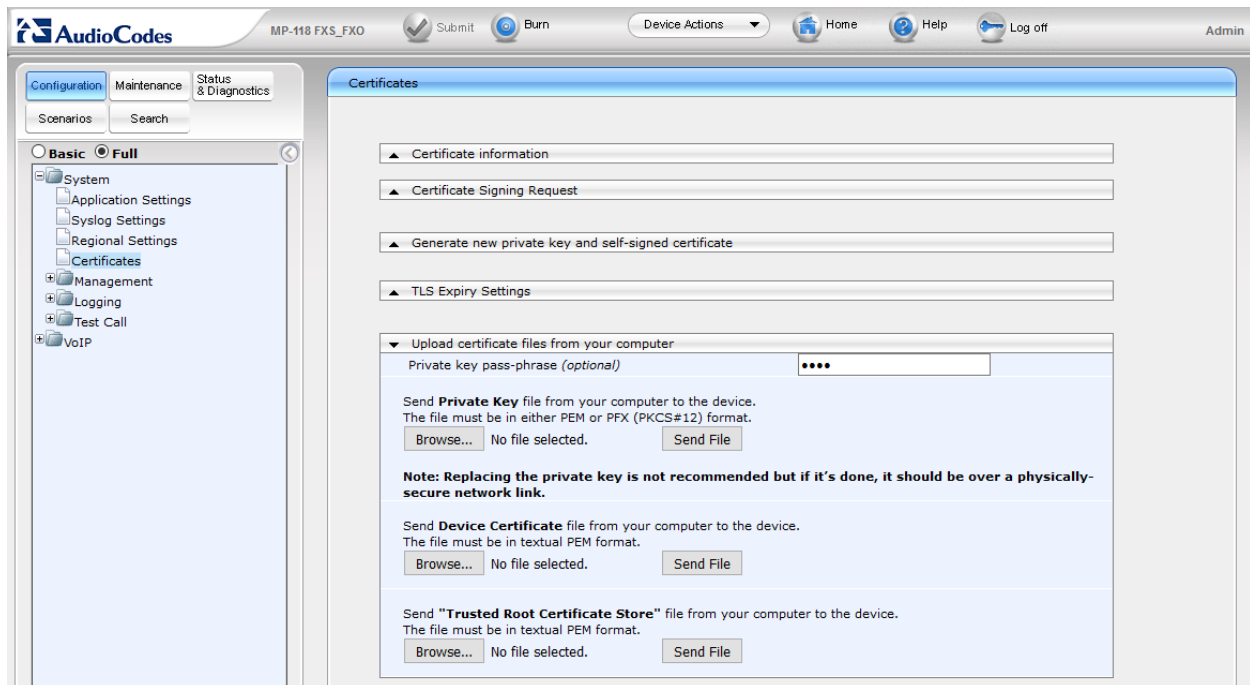
In order for TLS/SRTP to work, three certificates are needed

1. Private Key Certificate
2. Trusted Root Certificate
3. Server Certificate

Please note that without these certificates, TLS/SRTP will not work. Certificates were generated using System Manager as the Certificate Authority.

Once certificates have been obtained, configure as follows:

- Navigate to **System → Certificates** on the left pane
- For each certificate, click on **Upload certificate files from your computer**, next click **Browse** and locate the file. Click on **Send File**



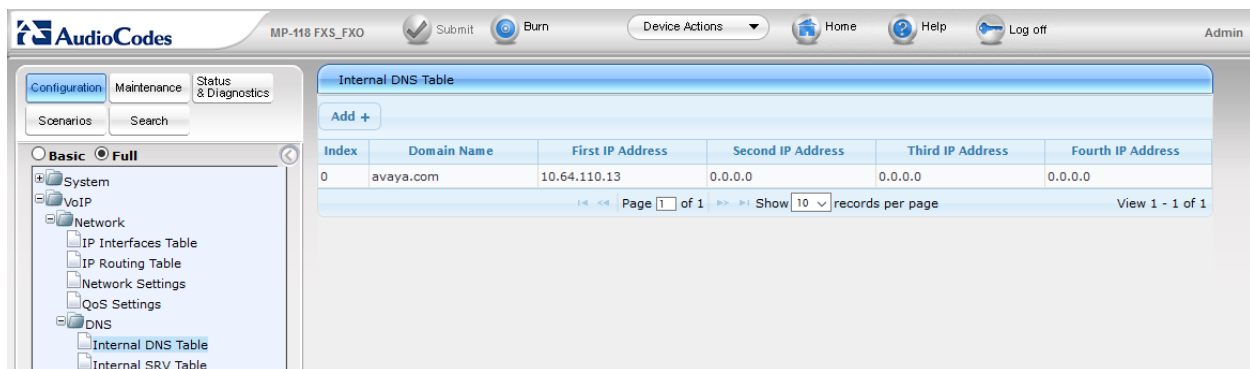
7.5. Administer DNS Setting

DNS entries can be entered manually by navigating to **VoIP → Network → DNS → Internal DNS Table**

- Fill in the information for **Domain** and **First IP Address** i.e. **SIP Domain and IP Address of Session Manager.**

Click **Submit** to save changes.

Note: It is good practice to click **Burn** on top of the page, after each change is made, to burn changes into flash memory of AudioCodes MP-11x.



The screenshot shows the AudioCodes MP-11x web interface. The top navigation bar includes the AudioCodes logo, the device name 'MP-11x FXS_FXO', and buttons for 'Submit', 'Burn', 'Device Actions', 'Home', 'Help', and 'Log off'. The user is logged in as 'Admin'. The left sidebar shows a tree view with 'Basic' and 'Full' tabs. Under 'Full', the 'DNS' section is expanded, showing 'Internal DNS Table' and 'Internal SRV Table'. The main content area displays the 'Internal DNS Table' with an 'Add +' button and a table with the following data:

| Index | Domain Name | First IP Address | Second IP Address | Third IP Address | Fourth IP Address |
|-------|-------------|------------------|-------------------|------------------|-------------------|
| 0 | avaya.com | 10.64.110.13 | 0.0.0.0 | 0.0.0.0 | 0.0.0.0 |

Below the table, there is a pagination control showing 'Page 1 of 1' and 'Show 10 records per page'. The status 'View 1 - 1 of 1' is also displayed.

7.6. Administer General Security

On the left pane, navigate to **VoIP → Security → General Security Settings**

- Set **TLS Version** to **Any – Including SSLv3**
- Set **TLS Remote Subject Name** to IP Address of Session Manager

Click **Submit** to save changes.

The screenshot displays the AudioCodes configuration web interface for an MP-118 FXS_FXO device. The left sidebar shows a navigation tree with 'Basic' and 'Full' tabs. Under 'Full', the 'Security' section is expanded, and 'General Security Settings' is selected. The main content area shows the configuration for 'General Security Settings' with several sections:

- IPSec Setting**
 - Enable IP Security: Disable
 - IKE Certificate Ext Validate: Disable
- TLS Settings**
 - TLS Version: Any - Including SSLv3
 - Strict Certificate Extension Validation: Disable
 - FIPS140 Mode: Disable
 - Client Cipher String: ALL
- SIP TLS Settings**
 - TLS Client Re-Handshake Interval: 0
 - TLS Mutual Authentication: Disable
 - Peer Host Name Verification Mode: Disable
 - TLS Client Verify Server Certificate: Disable
 - TLS Remote Subject Name: 10.64.110.13
- OCSP Settings**
 - Enable OCSP Server: Disable
 - Primary Server IP: ::
 - Secondary Server IP: ::
 - Server Port: 2560
 - Default Response When Server Unreachable: Reject
- Misc. Parameters**
 - Enable Management Two Factor Authentication: Disable

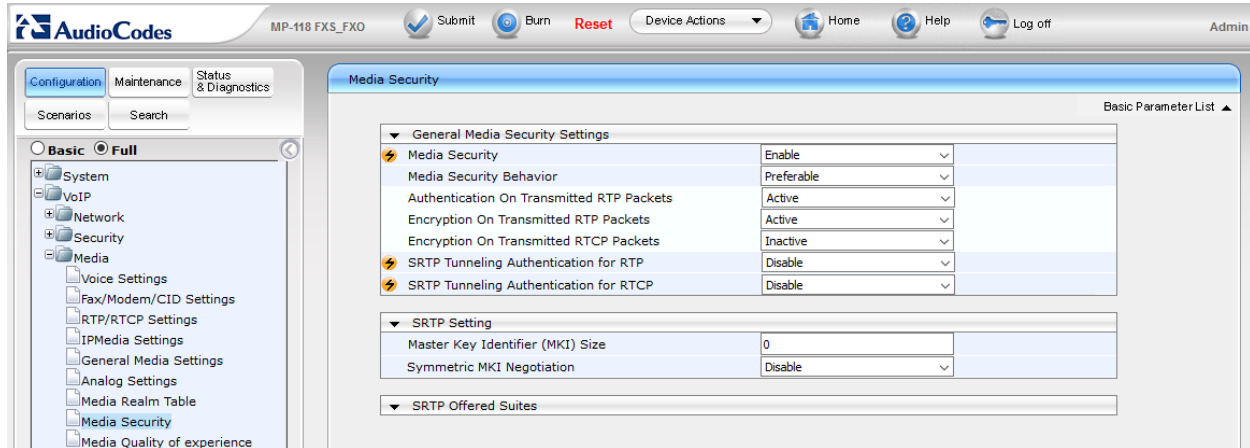
The interface includes a top navigation bar with 'Submit' and 'Burn' buttons, and a bottom status bar with 'Device Actions', 'Home', 'Help', 'Log off', and 'Admin' links.

7.7. Administer Media Security

On the left pane, navigate to **VoIP → Media → Media Security**

- Set **Media Security** to **Enable**
- Set **Media Security Behavior** to **Preferable**
- Set **Encryption on Transmitted RTCP Packets** to **Inactive**

Click **Submit** to save changes.



Note: This change requires a reset of AudioCodes MP-11x. Please burn the changes and reset the device before performing any further configuration.

7.8. Administer General Parameters

On the left pane, navigate to **VoIP → SIP Definitions → General Parameters**

- Set **Enable Early Media** to **Enable**
- Set **Fax Signaling Method** to **T.38 Relay**
- Set **SIP Transport Type** to **TLS**
- Set **Enable SIPS** to **Enable**
- Set **SIP Destination Port** to **5061**

Click **Submit** to save changes.

The screenshot shows the AudioCodes MP-118 FXS_FXO configuration interface. The left pane shows the navigation tree with 'SIP Definitions' expanded and 'General Parameters' selected. The main pane displays the 'SIP General Parameters' configuration table.

| SIP General | |
|--|---------------------------|
| NAT IP Address | 0.0.0.0 |
| PRACK Mode | Supported |
| Channel Select Mode | By Dest Phone Number |
| Enable Early Media | Enable |
| 183 Message Behavior | Progress |
| Session-Expires Time | 0 |
| Minimum Session-Expires | 90 |
| Session Expires Method | re-INVITE |
| Asserted Identity Mode | Disabled |
| Fax Signaling Method | T.38 Relay |
| Detect Fax on Answer Tone | Initiate T.38 on Preamble |
| SIP Transport Type | TLS |
| SIP UDP Local Port | 5060 |
| SIP TCP Local Port | 5060 |
| SIP TLS Local Port | 5061 |
| Enable SIPS | Enable |
| Enable TCP Connection Reuse | Enable |
| TCP Timeout | 0 |
| SIP Destination Port | 5061 |
| Use user=phone in SIP URL | Yes |
| Use user=phone in From Header | No |
| Use Tel URI for Asserted Identity | Disable |
| Tel to IP No Answer Timeout | 180 |
| Enable Remote Party ID | Disable |
| Add Number Plan and Type to RPI Header | Yes |
| Enable History-Info Header | Disable |
| Use Source Number as Display Name | No |
| Use Display Name as Source Number | No |
| Enable Contact Restriction | Disable |
| Play Dinchark Tone to IP | Don't Play |

7.9. Administer Advanced Parameter

On the left pane, navigate to **VoIP → SIP Definitions → Advanced Parameter**

- Set **Disconnect on Broken Connection** to **No**
- Set **CDR Server IP Address** to the same ip address configured for syslog ip address
- Set **CDR Report Level** to **Start & End & Connect Call**

Click **Submit** to save changes.

The screenshot shows the AudioCodes MP-118 FXS_FXO configuration interface. The left sidebar shows the navigation tree with 'SIP Definitions' expanded and 'Advanced Parameters' selected. The main area displays the 'Advanced Parameters' configuration page, which is divided into three sections: 'Disconnect and Answer Supervision', 'CDR and Debug', and 'Misc. Parameters'. Each section contains a list of parameters with their current values and dropdown menus for selection.

| Disconnect and Answer Supervision | |
|--------------------------------------|------------------------|
| Send Digit Pattern on Connect | |
| Polarity Reversal | Disable |
| Current Disconnect | Disable |
| Disconnect on Broken Connection | No |
| Broken Connection Timeout [100 msec] | 100 |
| Disconnect Call on Silence Detection | No |
| Silence Detection Period [sec] | 120 |
| Silence Detection Method | Voice/Energy Detectors |
| Enable Fax Re-Routing | Disable |

| CDR and Debug | |
|------------------------|----------------------------|
| CDR Server IP Address | 10.64.10.47 |
| CDR Report Level | Start & End & Connect Call |
| Media CDR Report Level | None |

| Misc. Parameters | |
|------------------------------------|----------------|
| Progress Indicator to IP | Not Configured |
| Enable Busy Out | Disable |
| Graceful Busy Out Timeout [sec] | 0 |
| Default Release Cause | 3 |
| Max Number of Active Calls | 16 |
| Max Call Duration [min] | 0 |
| LAN Watchdog | Disable |
| Enable Calls Cut Through | Disable |
| Enable User-Information Usage | Disable |
| Out Of Service Behavior | ! Reorder Tone |
| Delay After Reset [sec] | 7 |
| T.38 Fax Max Buffer | 1024 |
| Enable Microsoft Extension | Disable |
| Reliable Connection Resistant Mode | Disable |

7.10. Administer Proxy and Registration

On the left name, navigate to **VoIP → SIP Definitions → Proxy & Registration**

- Set **Use Default Proxy** to **Yes**
- Set **Proxy Name** to **avaya.com**
- Set **Always Use Proxy** to **Enable**
- Set **Redundant Routing Mode** to **Disable**
- Set **Enable Registration** to **Enable**
- Set **Registrar Name** to the domain name used for registration, e.g. **avaya.com**
- Set **Registrar Transport Type** to **TLS**
- Fill in **Registration Time** to a desired value, in seconds
- Set **Re-register On INVITE Failure** to **Enable**
- Set **ReRegister On Connection Failure** to **Enable**
- Fill in **Gateway Name**, e.g. **avaya.com**

Click **Submit** to save changes. Screen capture on next page.

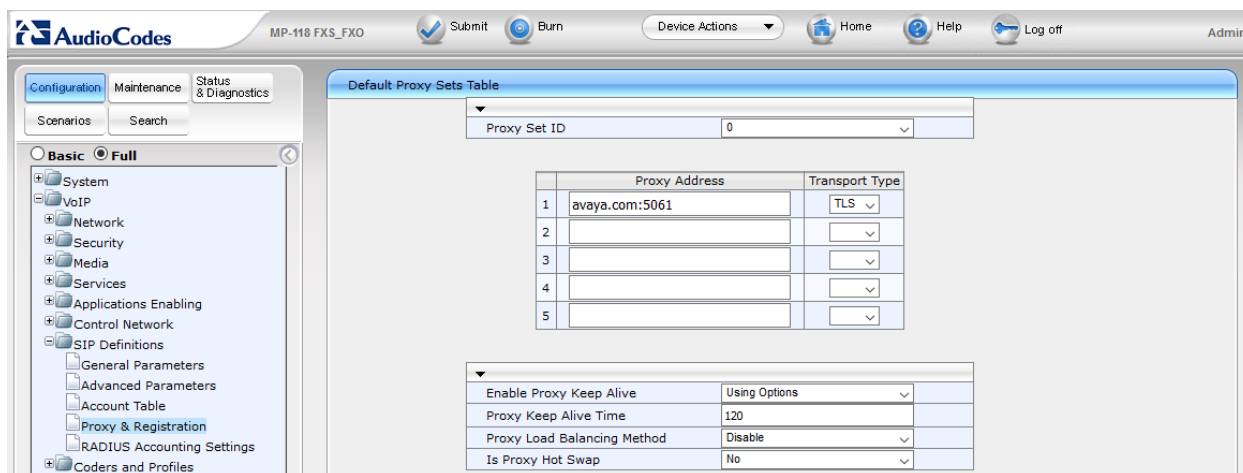
The screenshot shows the AudioCodes MP-118 FXS_FXO configuration interface. The left sidebar shows the navigation tree with 'SIP Definitions' expanded and 'Proxy & Registration' selected. The main area displays the 'Proxy & Registration' configuration form with various settings.

| Parameter | Value |
|---|---------------|
| Use Default Proxy | Yes |
| Proxy Set Table | + |
| Proxy Name | avaya.com |
| Redundancy Mode | Parking |
| Proxy IP List Refresh Time | 60 |
| Enable Fallback to Routing Table | Disable |
| Prefer Routing Table | No |
| Use Routing Table for Host Names and Profiles | Disable |
| Always Use Proxy | Enable |
| Redundant Routing Mode | Disable |
| SIP ReRouting Mode | Standard Mode |
| Enable Registration | Enable |
| Registrar Name | avaya.com |
| Registrar IP Address | |
| Registrar Transport Type | TLS |
| Registration Time | 600 |
| Re-registration Timing [%] | 50 |
| Registration Retry Time | 30 |
| Registration Time Threshold | 0 |
| Re-register On INVITE Failure | Enable |
| ReRegister On Connection Failure | Enable |
| Gateway Name | avaya.com |
| Gateway Registration Name | |
| DNS Query Type | A-Record |
| Proxy DNS Query Type | A-Record |
| Subscription Mode | Per Endpoint |
| Number of RTX Before Hot-Swap | 3 |
| Use Gateway Name for OPTIONS | No |
| User Name | |

On the **Proxy & Registration** page, shown in **Section 7.10** click on the Arrow icon, , next to **Proxy Set Table**

- Type in the **Proxy Address**, e.g. avaya.com:5061
- Set **Transport Type** to **TLS**
- Set **Enable Proxy Keep Alive** to **Using Options**
- Set **Proxy Keep Alive Time** to a desired value, in seconds.

Click **Submit** to save changes.



The screenshot shows the AudioCodes MP-118 FXS_FXO configuration interface. The left sidebar contains a tree view with categories like System, VoIP, Network, Security, Media, Services, Applications Enabling, Control Network, SIP Definitions, General Parameters, Advanced Parameters, Account Table, Proxy & Registration (selected), RADIUS Accounting Settings, and Coders and Profiles. The main area is titled 'Default Proxy Sets Table' and contains a table with 5 rows for Proxy Set ID, Proxy Address, and Transport Type. The first row is populated with 'avaya.com:5061' and 'TLS'. Below the table, there are settings for 'Enable Proxy Keep Alive' (Using Options), 'Proxy Keep Alive Time' (120), 'Proxy Load Balancing Method' (Disable), and 'Is Proxy Hot Swap' (No).

| Proxy Set ID | Proxy Address | Transport Type |
|--------------|----------------|----------------|
| 1 | avaya.com:5061 | TLS |
| 2 | | |
| 3 | | |
| 4 | | |
| 5 | | |

| | |
|-----------------------------|---------------|
| Enable Proxy Keep Alive | Using Options |
| Proxy Keep Alive Time | 120 |
| Proxy Load Balancing Method | Disable |
| Is Proxy Hot Swap | No |

7.11. Administer Coders

On the left pane, navigate to **VoIP → Coders and Profiles → Coders**

- Set coders as shown in the screen capture below

The screenshot shows the AudioCodes MP-118 FXS_FXO web interface. The top navigation bar includes the AudioCodes logo, the device model 'MP-118 FXS_FXO', and buttons for 'Submit', 'Burn', 'Device Actions', 'Home', 'Help', and 'Log off'. The user is logged in as 'Admin'. On the left, the 'Configuration' tab is active, with a tree view showing 'System', 'VoIP', 'Network', 'Security', 'Media', 'Services', 'Applications Enabling', 'Control Network', 'SIP Definitions', and 'Coders and Profiles'. Under 'Coders and Profiles', 'Coders' is selected. The main area displays the 'Coders Table' with the following data:

| Coder Name | Packetization Time | Rate | Payload Type | Silence Suppression |
|------------|--------------------|------|--------------|---------------------|
| G.711U-law | 20 | 64 | 0 | Disabled |
| G.711A-law | 20 | 64 | 8 | Disabled |
| G.729 | 20 | 8 | 18 | Disabled |
| | | | | |
| | | | | |
| | | | | |
| | | | | |
| | | | | |
| | | | | |
| | | | | |
| | | | | |

7.12. Administer End Point Phone Number

On the left pane, navigate to **VoIP → GW and IP to IP → Hunt Group → Endpoint Phone Number**, for each line

- Set **Channel** to a channel number, e.g. 1
- Set **Phone Number** to the extension configured in Session Manager, e.g. **11151**
- Set **Hunt Group ID** to a hunt group value (configured in next section), e.g. **1**
- Set **Tel Profile ID** to 0 for global

Click **Submit** to save changes.

Note: All phone numbers entered in this screen must be configured as users in Session Manager

Note: Channel 1 and 2 are FXS lines, Channel 3 is FXO line. When SRTP is configured, AudioCodes MP-11x DSP capacity is reduced by 25%. In this case, there are 8 analog ports on AudioCodes MP-118, but since SRTP is enabled, only 6 ports can be used.

Click on **Register** at the bottom of the screen to register with Session Manager.

The screenshot shows the AudioCodes MP-118 FXS_FXO configuration interface. The left pane shows the navigation tree with 'Endpoint Phone Number' selected under 'Hunt Group'. The main area displays the 'Endpoint Phone Number Table' with the following data:

| | Channel(s) | Phone Number | Hunt Group ID | Tel Profile ID |
|---|------------|--------------|---------------|----------------|
| 1 | 1 | 11151 | 1 | 0 |
| 2 | 2 | 11152 | 1 | 0 |
| 3 | | | | |
| 4 | | | | |
| 5 | | | | |
| 6 | | | | |

At the bottom of the table, there are three buttons: 'Register', 'Un-Register', and 'Submit'.

7.13. Administer Hunt Group Settings

On the left pane, navigate to **VoIP → GW and IP to IP → Hunt Group → Hunt Group Settings**

- Set **Hunt Group ID** to the ID used in previous step, i.e. **1**
- Set **Channel Select Mode** to **By Dest Phone Number** for FXS ports. If configuring FXO port, set it to **Cyclic Ascending**
- Set **Registration Mode** to **Per Endpoint**

Click **Submit** to save changes.

The screenshot shows the AudioCodes MP-118 FXS_FXO web interface. The left sidebar contains a navigation tree with the following structure:

- System
 - VoIP
 - Network
 - Security
 - Media
 - Services
 - Applications Enabling
 - Control Network
 - SIP Definitions
 - Coders and Profiles
 - GW and IP to IP
 - Hunt Group
 - Endpoint Phone Number
 - Hunt Group Settings**
 - Manipulations
 - Routing
 - DTMF and Supplementary
 - Analog Gateway
 - Advanced Applications
 - Charging

The main content area is titled 'Hunt Group Settings'. It features a 'Basic Parameter List' section with a table containing 12 rows. The first row is pre-filled with the following values:

| | Hunt Group ID | Channel Select Mode | Registration Mode | Serving IP Group ID | Gateway Name | Contact User |
|----|---------------|----------------------|-------------------|---------------------|--------------|--------------|
| 1 | 1 | By Dest Phone Number | Per Endpoint | | | |
| 2 | | | | | | |
| 3 | | | | | | |
| 4 | | | | | | |
| 5 | | | | | | |
| 6 | | | | | | |
| 7 | | | | | | |
| 8 | | | | | | |
| 9 | | | | | | |
| 10 | | | | | | |
| 11 | | | | | | |
| 12 | | | | | | |

7.14. Administer IP to Trunk Group Routing

On the left pane, navigate to **VoIP → GW and IP to IP → Routing → IP to Hunt Group Routing**, for each extension

- Set **Source Phone Prefix** to *
- Set **Dest. Phone Prefix** to the prefix of extensions, e.g. **1**
- Set **Hunt Group ID** to ID configured in previous step, i.e. **1**
- Set **IP Profile ID** to **0**

Click **Submit** to save changes

The screenshot shows the AudioCodes configuration interface. The left pane displays a tree view with the following structure:

- System
 - VoIP
 - Network
 - Security
 - Media
 - Services
 - Applications Enabling
 - Control Network
 - SIP Definitions
 - Coders and Profiles
 - GW and IP to IP
 - Hunt Group
 - Manipulations
 - Routing
 - Routing General Params
 - Tel to IP Routing
 - IP to Hunt Group Routing
 - Alternative Routing Reasons
 - Forward On Busy Hunt
 - DTMF and Supplementary
 - Analog Gateway

The right pane is titled "IP To Hunt Group Routing Table". It contains a "Basic Parameter List" with the following settings:

- Routing Index: 1-12
- IP To Tel Routing Mode: Route calls before manipulation

The table has the following columns: Dest. Host Prefix, Source Host Prefix, Dest. Phone Prefix, Source Phone Prefix, Source IP Address, and Hunt Group ID. The table is populated with 12 rows, with the first two rows showing the configuration for extension 1:

| | Dest. Host Prefix | Source Host Prefix | Dest. Phone Prefix | Source Phone Prefix | Source IP Address | Hunt Group ID |
|----|-------------------|--------------------|--------------------|---------------------|-------------------|---------------|
| 1 | | | 1 | * | | 1 |
| 2 | | | 1 | * | | 1 |
| 3 | | | | | | |
| 4 | | | | | | |
| 5 | | | | | | |
| 6 | | | | | | |
| 7 | | | | | | |
| 8 | | | | | | |
| 9 | | | | | | |
| 10 | | | | | | |
| 11 | | | | | | |
| 12 | | | | | | |

7.15. Administer DTMF and Dialing

On the left pane, navigate to **VoIP → GW and IP to IP → DTMF and Supplementary → DTMF & Dialing**

- Set **Max Digits In Phone Num** to **11**

The screenshot shows the AudioCodes MP-110 FXS_FXO configuration interface. The left pane shows the navigation tree with 'DTMF & Dialing' selected under 'GW and IP to IP'. The main pane displays the 'DTMF & Dialing' configuration table.

| Parameter | Value |
|----------------------------|----------|
| Max Digits In Phone Num | 11 |
| Inter Digit Timeout [sec] | 4 |
| Declare RFC 2833 in SDP | Yes |
| 1st Tx DTMF Option | RFC 2833 |
| 2nd Tx DTMF Option | |
| RFC 2833 Payload Type | 96 |
| Default Destination Number | 1000 |

Advanced Parameter List

Submit

7.16. Administer Supplementary Services

On the left pane, navigate to **VoIP → GW and IP to IP → DTMF and Supplementary → Supplementary Services**

- Set **Enable Caller ID** to **Enable**
- Under **Message Waiting Indication (MWI) Parameters**:
 - Set **Enable MWI** to **Enable**
 - Set **MWI Analog Lamp** to **Enable**
 - Set **MWI Display** to **Enable**
 - Set **Subscribe to MWI** to **Yes**
 - For **MWI Server IP Address**, type in Session Manager's IP Address
 - Set **MWI Server Transport Type** to **TLS**

Click **Submit** to save changes

The screenshot shows the AudioCodes MP-118 FXS_FXO configuration interface. The left pane shows the navigation tree with 'Supplementary Services' selected under 'DTMF and Supplementary'. The main pane displays the 'Supplementary Services' configuration page. The 'Basic Parameter List' is expanded, showing the following parameters:

| Parameter | Value |
|-----------------------------|------------|
| Enable Caller ID | Enable |
| Hook-Flash Code | |
| Flash Keys Sequence Style | Flash hook |
| Flash Keys Sequence Timeout | 2000 |
| Enable NRT Subscription | Disable |
| AS Subscribe IPGroupID | -1 |
| NRT Subscribe Retry Time | 120 |
| Call Forward Ring Tone ID | 1 |
| Send All Coders on Retrieve | Disable |

The 'Message Waiting Indication (MWI) Parameters' section is expanded, showing the following parameters:

| Parameter | Value |
|-------------------------------|--------------|
| Enable MWI | Enable |
| MWI Analog Lamp | Enable |
| MWI Display | Enable |
| Subscribe to MWI | Yes |
| MWI Server IP Address | 10.64.110.13 |
| MWI Server Transport Type | TLS |
| MWI Subscribe Expiration Time | 7200 |
| Stutter Tone Duration | 2000 |
| MWI Subscribe Retry Time | 120 |

The 'MLPP' section is expanded, showing the following parameters:

| Parameter | Value |
|---------------------------|---------|
| Call Priority Mode | Disable |
| Reminder Ring | Enable |
| MLPP Diffserv | 50 |
| Precedence Ringing Type | -1 |
| MLPP Default Namespace | DSN |
| Default Call Priority | 0 |
| Preemption tone Duration | 3 |
| RTP DSCP for MLPP Routine | -1 |

7.17. Administer FXO

On the left pane, navigate to **VoIP → GW and IP to IP → Analog Gateway → FXO Settings**

- Set **Dialing Mode** to **Two Stages**

Click **Submit** to save changes. Also, click **Burn** to save changes to the flash memory of AudioCodes MP-11x.

The screenshot shows the AudioCodes web interface for the MP-118 FXS_FXO device. The top navigation bar includes 'Submit' and 'Burn' buttons. The left sidebar shows a tree view with 'FXO Settings' selected under 'Analog Gateway'. The main content area displays the 'FXO Settings' configuration page with a table of settings.

| Setting | Value |
|--|------------|
| Dialing Mode | Two Stages |
| Waiting for Dial Tone | No |
| Time to Wait before Dialing [msec] | 1000 |
| Ring Detection Timeout [sec] | 8 |
| Reorder Tone Duration [sec] | 255 |
| Answer Supervision | No |
| Rings before Detecting Caller ID | 1 |
| Send Metering Message to IP | No |
| Disconnect Call on Busy Tone Detection (CAS) | Enable |
| Disconnect On Dial Tone | Disable |
| Guard Time Between Calls | 1 |
| FXO Double Answer | Disable |
| FXO AutoDial Play BusyTone | Disable |
| FXO Ring Timeout [100 msec] | 0 |

7.18. Administer Authentication for FXS Lines

On the left pane, navigate to **VoIP → GW and IP to IP → Analog Gateway → Authentication**

- Add the **User Name** and **Password** for the users configured on Session Manager.

The screenshot shows the AudioCodes MP-118 FXS_FXO configuration interface. The left pane shows the navigation tree with 'Authentication' selected under 'Analog Gateway'. The main pane displays the 'Authentication' configuration table.

| Gateway Port | User Name | Password |
|--------------|-----------|----------|
| Port 1 FXS | 11151 | ***** |
| Port 2 FXS | 11152 | ***** |
| Port 3 FXS | | |
| Port 4 FXS | | |
| Port 5 FXO | | |
| Port 6 FXO | | |

Click **Submit** to save changes. Also, click **Burn** to save changes to the flash memory of AudioCodes MP-11x.

7.19. Administer other parameters

Type in <http://<ip-address>/AdminPage> URL in a web browser, where <ip-address> is the ip address of AudioCodes MP-11x, click on **ini Parameters** on the left pane

The screenshot shows a web interface for administering parameters. On the left is a dark sidebar with three buttons: 'Image Load to Device', 'ini Parameters', and 'Back to Main'. The 'ini Parameters' button is highlighted. The main content area has a light blue background. At the top, there are two input fields: 'Parameter Name:' and 'Enter Value:', followed by an 'Apply New Value' button. Below these is a large white rectangular area labeled 'Output Window'.

- In the table below type in each parameter and associated value, click **Apply New Value**
- Verify the output shows updated value

| Parameter Name | Value |
|-------------------------|----------|
| HTTPSCIPHERSTRING | ALL:!ADH |
| RTCPENCRYPTIONDISABLERX | 1 |

The screen capture below shows output after applying a new value of **HTTPSCIPHERSTRING**.

The screenshot displays a web application interface with a dark blue sidebar on the left and a light blue main content area. The sidebar contains three buttons: "Image Load to Device", "ini Parameters", and "Back to Main". The main content area has a header section with two input fields: "Parameter Name:" containing "HTTPSCIPHERSTRING" and "Enter Value:" containing "ALL:!ADH". To the right of these fields is a button labeled "Apply New Value". Below the header is a section titled "Output Window" which contains a text box with the following text: "Parameter Name: HTTPSCIPHERSTRING", "Parameter New Value: ALL:!ADH", and "Parameter Description: Cipher string for HTTPS (in OpenSSL cipher list format)." A small cursor icon is visible at the bottom right of the text box.

| Parameter Name: | Enter Value: |
|-------------------|--------------|
| HTTPSCIPHERSTRING | ALL:!ADH |

Apply New Value

Output Window

```
Parameter Name: HTTPSCIPHERSTRING
Parameter New Value: ALL:!ADH
Parameter Description: Cipher string for HTTPS (in OpenSSL cipher list format).
```

8. Verification Steps

8.1. 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 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 registration from AudioCodes MP-11x to Session Manager via System Manager console, <http://<ip-address/>>
- Navigate to **Home → Session Manager → System Status → User Registration**

AVAYA
Aura® System Manager 7.0

Last Logged on at November 14, 2016 2:56 PM

GO... Log off admin

Home Routing * User Management * Licenses * **Session Manager** *

Home / Elements / Session Manager / System Status / User Registrations

User Registrations

Select rows to send notifications to devices. Click on Details column for complete registration status.

View: Default Force Unregister AST Device Notifications: Reboot Reload Failback As of 3:51 PM Customize Advanced Search

12 Items Show All Filter: Enable

| <input type="checkbox"/> | Details | Address | First Name | Last Name | Actual Location | IP Address | Remote Office | Shared Control | Simult. Devices | AST Device | Registered | Prim | Sec | Su |
|--------------------------|---------|-----------------|-------------|-------------|-----------------|---------------|--------------------------|--------------------------|-----------------|-------------------------------------|-------------------------------------|--------------------------|--------------------------|--------------------------|
| <input type="checkbox"/> | Show | 11151@avaya.com | User 1 | Audiocodes | DevConnect-Lab | 10.64.10.102 | <input type="checkbox"/> | <input type="checkbox"/> | 1/1 | <input type="checkbox"/> | <input checked="" type="checkbox"/> | (AC) | <input type="checkbox"/> | <input type="checkbox"/> |
| <input type="checkbox"/> | Show | --- | User 7 | SIP | --- | --- | <input type="checkbox"/> | <input type="checkbox"/> | 0/1 | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> |
| <input type="checkbox"/> | Show | 11105@avaya.com | User 5 | SIP | DevConnect-Lab | 10.64.10.211 | <input type="checkbox"/> | <input type="checkbox"/> | 1/1 | <input checked="" type="checkbox"/> | <input checked="" type="checkbox"/> | (AC) | <input type="checkbox"/> | <input type="checkbox"/> |
| <input type="checkbox"/> | Show | --- | User 3 | SIP | --- | --- | <input type="checkbox"/> | <input type="checkbox"/> | 0/1 | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> |
| <input type="checkbox"/> | Show | 11111@avaya.com | iPhone User | CounterPath | --- | 10.80.130.150 | <input type="checkbox"/> | <input type="checkbox"/> | 1/3 | <input type="checkbox"/> | <input checked="" type="checkbox"/> | (AC) | <input type="checkbox"/> | <input type="checkbox"/> |
| <input type="checkbox"/> | Show | --- | Station 6 | SIP | --- | --- | <input type="checkbox"/> | <input type="checkbox"/> | 0/1 | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> |
| <input type="checkbox"/> | Show | --- | iPad User | CounterPath | --- | --- | <input type="checkbox"/> | <input type="checkbox"/> | 0/3 | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> |
| <input type="checkbox"/> | Show | --- | Nexus User | CounterPath | --- | --- | <input type="checkbox"/> | <input type="checkbox"/> | 0/3 | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> |
| <input type="checkbox"/> | Show | 11152@avaya.com | User 2 | AudioCodes | DevConnect-Lab | 10.64.10.102 | <input type="checkbox"/> | <input type="checkbox"/> | 1/1 | <input type="checkbox"/> | <input checked="" type="checkbox"/> | (AC) | <input type="checkbox"/> | <input type="checkbox"/> |
| <input type="checkbox"/> | Show | --- | User 4 | SIP | --- | --- | <input type="checkbox"/> | <input type="checkbox"/> | 0/1 | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> |
| <input type="checkbox"/> | Show | --- | User 1 | SIP | --- | --- | <input type="checkbox"/> | <input type="checkbox"/> | 0/1 | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> |
| <input type="checkbox"/> | Show | --- | User 2 | SIP | --- | --- | <input type="checkbox"/> | <input type="checkbox"/> | 0/1 | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> |

Select : All, None

9. Conclusion

These Application Notes describe the configuration steps required for AudioCodes MP-11x 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

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

- [1] Administering Avaya Aura® Communication Manager, Release 7.0
- [2] Administering Avaya Aura® Session Manager, Release 7.0
- [3] AudioCodes Transport Layer Security Note, Document LTRT-31600, January 2012
- [4] MP-11x and MP-124 SIP User's Manual, Version 6.6, Document LTRT-65422, January 2014
- [5] MP-11x and MP-124 SIP Installation Manual, Version 6.6, Document LTRT-59820, December 2014
- [6] SIP CPE Release Notes, Version 6.6, Document LTRT-26916

A. Appendix

AudioCodes MP-118 .ini file generated after following the instruction is as follows: Please use it only for reference purposes.

Note: Please note that the password for registration will need to be changed if this ini file is loaded on an AudioCodes MP-11x device.

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

;Board: MP-118 FXS_FXO
;Board Type: 56
;Serial Number: 547031
;Slot Number: 1
;Software Version: 6.60A.328.003
;DSP Software Version: 204IM3=> 660.14
;Board IP Address: 10.64.10.102
;Board Subnet Mask: 255.255.255.0
;Board Default Gateway: 10.64.10.1
;Ram size: 32M   Flash size: 8M
;Num of DSP Cores: 2   Num DSP Channels: 8
;Profile: NONE
;License Key limits aren't active full features capabilities are available !;
;-----
```

[SYSTEM Params]

```
SyslogServerIP = 10.64.10.47
;NTPServerIP_abs is hidden but has non-default value
NTPServerUTCOffset = -21600
ENABLEPARAMETERSMONITORING = 1
ActivityListToLog = 'pvc', 'afl', 'dr', 'fb', 'swu', 'ard', 'naa', 'spc',
'11'
;VpFileLastUpdateTime is hidden but has non-default value
DayLightSavingTimeStart = '03:09:02:00'
DayLightSavingTimeEnd = '11:02:02:00'
TLSClientCipherString = 'ALL'
TLSPkeySize = 2048
NTPServerIP = '128.138.140.44'
LDAPSEARCHDNSINPARALLEL = 0
```

[BSP Params]

```
PCMLawSelect = 3
```

[Analog Params]

[ControlProtocols Params]

AdminStateLockControl = 0

[MGCP Params]

[MEGACO Params]

EP_Num_0 = 0
EP_Num_1 = 1
EP_Num_2 = 1
EP_Num_3 = 0
EP_Num_4 = 0
DIGITMAPPING = '1xxxx|91xxxxxxxxxxxx'

[Voice Engine Params]

CallProgressTonesFilename = 'usa_tones_13.dat'
RTCPEncryptionDisableTx = 1

[WEB Params]

LogoWidth = '145'
HTTPSCipherString = 'ALL:!ADH'
;HTTPSCertFileName is hidden but has non-default value
;HTTPSRootFileName is hidden but has non-default value

[SIP Params]

ENABLECALLERID = 1
MAXDIGITS = 19
REGISTRATIONTIME = 600
ISPROXYUSED = 1
ISREGISTERNEEDED = 1
CDRREPORTLEVEL = 4
;ENABLECDR is hidden but has non-default value
GWDEBUGLEVEL = 5
;ISPRACKREQUIRED is hidden but has non-default value
ENABLEEARLYMEDIA = 1
PROXYNAME = 'avaya.com'
SIPGATEWAYNAME = 'avaya.com'
ALWAYSSENDDTOPROXY = 1
;SHOULDREGISTER is hidden but has non-default value
DISCONNECTONBROKENCONNECTION = 0
CDRSYSLOGSERVERIP = 10.64.10.47
ENABLEMWISUBSCRIPTION = 1
MWISERVERIP = '10.64.110.13'
;SHOULDSSUBSCRIBE is hidden but has non-default value
MWIANALOGLAMP = 1
MWIDISPLAY = 1
ENABLEMWI = 1
ISFAXUSED = 1
SIPTRANSPORTTYPE = 1
REGISTRARNAME = 'avaya.com'
ENABLE3WAYCONFERENCE = 1
REGISTERONINVITEFAILURE = 1
REGISTRARTRANSPORTTYPE = 2

```

MWISERVERTRANSPORTTYPE = 1
TLSREMOTESUBJECTNAME = '10.64.50.31'
REREGISTERONCONNECTIONFAILURE = 1
MSLDAPPRIMARYKEY = 'telephoneNumber'

```

```
[IPsec Params]
```

```
[SNMP Params]
```

```
[ InterfaceTable ]
```

```

FORMAT InterfaceTable_Index = InterfaceTable_ApplicationTypes,
InterfaceTable_InterfaceMode, InterfaceTable_IPAddress,
InterfaceTable_PrefixLength, InterfaceTable_Gateway, InterfaceTable_VlanID,
InterfaceTable_InterfaceName, InterfaceTable_PrimaryDNSServerIPAddress,
InterfaceTable_SecondaryDNSServerIPAddress;
InterfaceTable 0 = 6, 10, 10.64.10.102, 24, 10.64.10.1, 1, "O+M+C",
75.75.75.75, 0.0.0.0;

```

```
[ \InterfaceTable ]
```

```
[ DspTemplates ]
```

```

;
; *** TABLE DspTemplates ***
; This table contains hidden elements and will not be exposed.
; This table exists on board and will be saved during restarts.
;

```

```
[ \DspTemplates ]
```

```
[ TrunkGroup ]
```

```

FORMAT TrunkGroup_Index = TrunkGroup_TrunkGroupNum, TrunkGroup_FirstTrunkId,
TrunkGroup_FirstBChannel, TrunkGroup_LastBChannel,
TrunkGroup_FirstPhoneNumber, TrunkGroup_ProfileId, TrunkGroup_LastTrunkId,
TrunkGroup_Module;
TrunkGroup 0 = 1, 255, 1, 1, "11151", 0, 255, 255;
TrunkGroup 1 = 1, 255, 2, 2, "11152", 0, 255, 255;

```

```
[ \TrunkGroup ]
```

```
[ PstnPrefix ]
```

```

; ** NOTE: Changes were made to active configuration.
; **      The data below is different from current values.
FORMAT PstnPrefix_Index = PstnPrefix_DestPrefix, PstnPrefix_TrunkGroupId,
PstnPrefix_SourcePrefix, PstnPrefix_SourceAddress, PstnPrefix_ProfileId,
PstnPrefix_SrcIPGroupId, PstnPrefix_DestHostPrefix, PstnPrefix_SrcHostPrefix,
PstnPrefix_SrcSRDID, PstnPrefix_TrunkId, PstnPrefix_CallSetupRulesSetId;
PstnPrefix 0 = "1", 1, "*", "", 0, -1, "", "", , -1, -1;

```

```

PstnPrefix 1 = "1", 1, "*", "", 0, -1, "", "", , -1, -1;

[ \PstnPrefix ]

[ Dns2Ip ]

FORMAT Dns2Ip_Index = Dns2Ip_DomainName, Dns2Ip_FirstIpAddress,
Dns2Ip_SecondIpAddress, Dns2Ip_ThirdIpAddress, Dns2Ip_FourthIpAddress;
Dns2Ip 0 = "avaya.com", 10.64.110.13, 0.0.0.0, 0.0.0.0, 0.0.0.0;

[ \Dns2Ip ]

[ ProxyIp ]

FORMAT ProxyIp_Index = ProxyIp_IpAddress, ProxyIp_TransportType,
ProxyIp_ProxySetId;
ProxyIp 0 = "avaya.com:5061", 2, 0;

[ \ProxyIp ]

[ TxDtmfOption ]

FORMAT TxDtmfOption_Index = TxDtmfOption_Type;
TxDtmfOption 0 = 4;

[ \TxDtmfOption ]

[ TrunkGroupSettings ]

FORMAT TrunkGroupSettings_Index = TrunkGroupSettings_TrunkGroupId,
TrunkGroupSettings_ChannelSelectMode, TrunkGroupSettings_RegistrationMode,
TrunkGroupSettings_GatewayName, TrunkGroupSettings_ContactUser,
TrunkGroupSettings_ServingIPGroup, TrunkGroupSettings_MWIInterrogationType,
TrunkGroupSettings_TrunkGroupName;
TrunkGroupSettings 0 = 1, 0, 0, "", "", -1, 255, "";

[ \TrunkGroupSettings ]

[ Authentication ]

FORMAT Authentication_Index = Authentication_UserId,
Authentication_UserPassword, Authentication_Port, Authentication_PortType;
Authentication 0 = "11151", *, 1, "FXS";
Authentication 1 = "11152", *, 2, "FXS";

[ \Authentication ]

[ ProxySet ]

```

```

FORMAT ProxySet_Index = ProxySet_EnableProxyKeepAlive,
ProxySet_ProxyKeepAliveTime, ProxySet_ProxyLoadBalancingMethod,
ProxySet_IsProxyHotSwap, ProxySet_SRD, ProxySet_ClassificationInput,
ProxySet_ProxyRedundancyMode, ProxySet_KeepAliveFailureResp;
ProxySet 0 = 1, 120, 0, 0, 0, 0, -1, "";

[ \ProxySet ]

[ CodersGroup0 ]

FORMAT CodersGroup0_Index = CodersGroup0_Name, CodersGroup0_pTime,
CodersGroup0_rate, CodersGroup0_PayloadType, CodersGroup0_Sce;
CodersGroup0 0 = "g711Ulaw64k", 20, 0, -1, 0;

[ \CodersGroup0 ]

[ RoutingRuleGroups ]

FORMAT RoutingRuleGroups_Index = RoutingRuleGroups_LCREnable,
RoutingRuleGroups_LCRAverageCallLength, RoutingRuleGroups_LCRDefaultCost;
RoutingRuleGroups 0 = 0, 0, 1;

[ \RoutingRuleGroups ]

[ ResourcePriorityNetworkDomains ]

FORMAT ResourcePriorityNetworkDomains_Index =
ResourcePriorityNetworkDomains_Name,
ResourcePriorityNetworkDomains_Ip2TelInterworking;
ResourcePriorityNetworkDomains 1 = "dsn", 0;
ResourcePriorityNetworkDomains 2 = "dod", 0;
ResourcePriorityNetworkDomains 3 = "drsn", 0;
ResourcePriorityNetworkDomains 5 = "uc", 1;
ResourcePriorityNetworkDomains 7 = "cuc", 0;

[ \ResourcePriorityNetworkDomains ]

```

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