Avaya Solution & Interoperability Test Lab

Application Notes for Configuring Avaya Aura® Session Manager and Avaya Aura® Communication Manager with AudioCodes Mediant 3000 Gateway for T1 access – Issue 1.0

Abstract

These Application Notes describe the procedure for configuring the AudioCodes Mediant 3000 Gateway to interoperate with Avaya Aura® Session Manager and Avaya Aura® Communication Manager using SIP trunking along with T1 access to a simulated PSTN.

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

Information in these Application Notes has been obtained through DevConnect compliance testing and additional technical discussions. Testing was conducted via the DevConnect Program at the Avaya Solution and Interoperability Test Lab.

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1. Introduction

These Application Notes describe the procedure for configuring the AudioCodes Mediant 3000 Gateway to interoperate with Avaya Aura® Session Manager (Session Manager) and Avaya Aura® Communication Manager (Communication Manager) using SIP trunking along with T1 access to a simulated PSTN.

These Application Notes present a sample configuration for an enterprise network consisting of Session Manager and Communication Manager, integrated with an AudioCodes Mediant 3000 Gateway using SIP and providing T1 access to a simulated PSTN. The AudioCodes Mediant 3000 is a feature-rich, highly available VoIP gateway supporting low to medium channel densities. The AudioCodes Mediant 3000 compact footprint (2U) allows high capacity and High Availability (HA) when business critical contact centers require such resilience. The AudioCodes Mediant 3000 has comprehensive PSTN access capabilities as well as SIP to SIP interworking features that enable the interconnection between enterprises and service providers. In addition to E1/T1 interfaces, the AudioCodes Mediant 3000 supports high-density PSTN interfaces, such as T3, STM-1 and OC3 to provide the enterprise with lower PSTN lease costs. The proven interoperability of the AudioCodes Mediant 3000 with different PBXs and PSTN switches facilitates smooth deployment. Even though the Mediant 3000 supports a variety of different protocols and features, only SIP and T1 access were verified in this compliance test. Note that AudioCodes Media 3000, at places, is referred as M3K in this document.

2. General Test Approach and Test Results

The general test approach was to make calls, verify codecs, and exercise common PBX features, between endpoints located in the enterprise and the simulated PSTN.

DevConnect Compliance Testing is conducted jointly by Avaya and DevConnect members. The jointly-defined test plan focuses on exercising APIs and/or standards-based interfaces pertinent to the interoperability of the tested products and their functionalities. DevConnect Compliance Testing is not intended to substitute full product performance or feature testing performed by DevConnect members, nor is it to be construed as an endorsement by Avaya of the suitability or completeness of a DevConnect member's solution.

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

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

For the testing associated with these Application Notes, the interface between Avaya systems and AudioCodes Mediant 3000 used TLS and SRTP.

2.1. Interoperability Compliance Testing

The interoperability compliance test included feature and serviceability. The feature testing focused on verifying the following:

- Simulated PSTN calls from and to Avaya endpoints
- Calling with various Avaya Deskphone models
- Support for G.711A, G.711MU and G.729 codecs
- SIP transport using UDP and TCP
- Codec negotiation
- Telephony supplementary features, such as Hold, Call Transfer, Conference Calling and Call Forwarding
- DTMF Tone Support
- Voicemail Coverage and Retrieval
- Direct IP-to-IP Media (also known as "Shuffling") over SIP Trunk. Direct IP-to-IP media allows compatible phones to reconfigure the RTP path after call establishment directly between the Avaya phones and the AudioCodes Mediant 3000 Gateway and release media processing resources on the Avaya Media Gateway

2.2. Test Results

The AudioCodes Mediant 3000 passed compliance testing.

2.3. Support

For technical support, contact AudioCodes via the support link at www.audiocodes.com.

3. Reference Configuration

As shown in **Figure 1**, the Avaya enterprise network uses SIP trunking for call signaling internally, and with the Mediant 3000 Gateway in order to access the simulated PSTN. The Mediant 3000 is managed by using the web interface. Session Manager, with its SM-100 (Security Module) network interface, routes calls between the different entities using SIP Trunks. All inter-system calls are carried over these SIP trunks. Session Manager supports flexible inter-system call routing based on the dialed number, the calling number and the system location; it can also provide protocol adaptation to allow multi-vendor systems to interoperate. Session Manager is managed by Avaya Aura® System Manager via the management network interface.

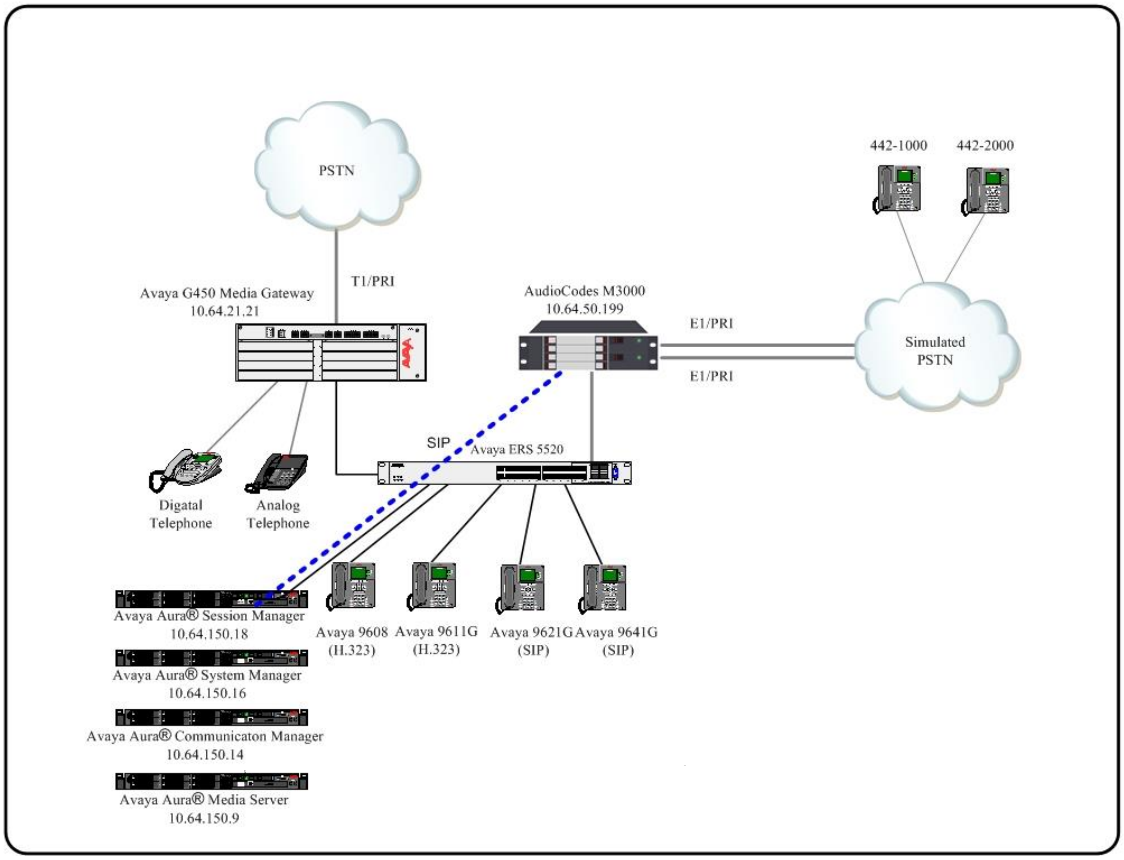


Figure 1: Compliance Test Reference Configuration

For the sample configuration shown in **Figure 1**, Session Manager, System Manager, Communication Manager, and Media Server all run in a virtual environment . These Application Notes focus on the configuration of the SIP trunks and call routing.

4. Equipment and Software Validated

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

Equipment/Software	Release/Version
Avaya Aura® Communication Manager in a Virtual Environment	7.1.2
Avaya Aura® Session Manager in a Virtual Environment	7.1.2
Avaya Aura® System Manager in a Virtual Environment	7.1.2
Avaya Aura® Media Server in a Virtual Environment	7.8.0.226
Avaya 96x1 Deskphone	SIP 7.1, H.323 6.6
Avaya 6211 and 6221 Analog Phone	-
AudioCodes Mediant 3000	7.00A.125.004

5. Configure Avaya Aura® Communication Manager

This section shows the configuration in Communication Manager. All configurations in this section are administered using the System Access Terminal (SAT). These Application Notes assumed that the basic configuration has already been administered. For further information on Communication Manager, please consult with **Reference [1]**. The procedures include the following areas:

- Verify Communication Manager License
- Administer System Parameters Features
- Administer IP Node Names
- Administer IP Network Region and Codec set
- Administer SIP Signaling Group and Trunk Group
- Administer Route Pattern
- Administer Private Numbering
- Administer Dial Plan and AAR analysis
- Administer ARS analysis
- Administer Feature Access Codes
- Save Changes

5.1. Verify Avaya Aura® Communication Manager License

Use the **display system-parameter customer options** command to verify, on **Page 2**, whether the **Maximum Administered SIP Trunks** field value with the corresponding value in the **USED** column. The difference between the two values needs to be greater than or equal to the desired number of simultaneous SIP trunk connections.

Note: The license file installed on the system controls the maximum features permitted. If there is insufficient capacity or a required feature is not enabled, contact an authorized Avaya sales representative to make the appropriate changes.

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

IP PORT CAPACITIES                                            USED
                     Maximum Administered H.323 Trunks: 12000 0
           Maximum Concurrently Registered IP Stations: 18000 6
             Maximum Administered Remote Office Trunks: 12000 0
Maximum Concurrently Registered Remote Office Stations: 18000 0
              Maximum Concurrently Registered IP eCons: 128   0
  Max Concur Registered Unauthenticated H.323 Stations: 100   0
                        Maximum Video Capable Stations: 36000 0
                   Maximum Video Capable IP Softphones: 18000 2
                       Maximum Administered SIP Trunks: 12000 10
  Maximum Administered Ad-hoc Video Conferencing Ports: 12000 0
   Maximum Number of DS1 Boards with Echo Cancellation: 522   0
```

5.2. Administer System Parameters Features

Use the **change system-parameters features** command to allow for trunk-to-trunk transfers. This feature is needed to allow for transferring an incoming/outgoing call from/to a remote switch back out to the same or different switch. For simplicity, the **Trunk-to-Trunk Transfer** field was set to **all** to enable all trunk-to-trunk transfers on a system wide basis.

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

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



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

5.3. Administer IP Node Names

Use the **change node-names ip** command to add entries for Communication Manager and Session Manager that will be used for connectivity. In the sample network, the processor Ethernet interface **procr** and **10.64.110.10** are entered as **Name** and **IP Address** for the signaling in Communication Manager running in a virtual environment. In addition, **asm** and **10.64.110.12** are entered for Session Manager.

```
change node-names ip                                            Page   1 of   2
                                  IP NODE NAMES
    Name              IP Address
aes                 10.64.110.17
ams                 10.64.110.13
asm                 10.64.110.12
cms                 10.64.110.18
default             0.0.0.0
procr               10.64.110.10
procr6              ::









( 7  of 7    administered node-names were displayed )
Use 'list node-names' command to see all the administered node-names
Use 'change node-names ip xxx' to change a node-name 'xxx' or add a node-name
```

5.4. Administer IP Network Region and Codec Set

Use the **change ip-network-region n** command, where **n** is the network region number, to configure the network region being used. In the sample network ip-network-region **1** is used. For the **Authoritative Domain** field, enter the SIP domain name configured for this enterprise and a descriptive **Name** for this ip-network-region. Set **Intra-region IP-IP Direct Audio** and **Inter-region IP-IP Direct Audio** to **yes** to allow for direct media between endpoints. Set the **Codec Set** to **1** to use ip-codec-set 1.

```
change ip-network-region 1                                      Page   1 of  20
                               IP NETWORK REGION
  Region: 1       NR Group: 1
Location: 1       Authoritative Domain: avaya.com
    Name:                           Stub Network Region: n
MEDIA PARAMETERS                Intra-region IP-IP Direct Audio: yes
      Codec Set: 1              Inter-region IP-IP Direct Audio: yes
   UDP Port Min: 2048                      IP Audio Hairpinning? n
   UDP Port Max: 3329
DIFFSERV/TOS PARAMETERS
 Call Control PHB Value: 46
        Audio PHB Value: 46
        Video PHB Value: 26
802.1P/Q PARAMETERS
 Call Control 802.1p Priority: 6
        Audio 802.1p Priority: 6
        Video 802.1p Priority: 5      AUDIO RESOURCE RESERVATION PARAMETERS
H.323 IP ENDPOINTS                                       RSVP Enabled? n
  H.323 Link Bounce Recovery? y
 Idle Traffic Interval (sec): 20
   Keep-Alive Interval (sec): 5
            Keep-Alive Count: 5
```

Use the **change ip-codec-set n** command where **n** is codec set used in the configuration. The codecs used in the compliance test are shown here. Configure the IP Codec Set as shown in the screen below. Note that in order to configure SRTP, "Media Encryption" will need to be enabled. Please refer to documentation in **Section 10** for additional information.

Retain the default values for the remaining fields.

```
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: 3-srtp-aescm128-hmac80-unauth
 4: 4-srtp-aescm128-hmac32-unauth
```

5.5. Administer SIP Trunks with Avaya Aura® Session Manager

In the test configuration, a SIP trunk was configured between Communication Manager and Session Manager for enterprise calling between Communication Manager and Session Manager registered endpoints. Additionally, a SIP trunk was configured between Session Manager and the Mediant 3000 in order to communicate between the enterprise and the simulated PSTN. To administer a SIP Trunk on Communication Manger, two steps are required: the creation of a signaling group and a trunk group.

5.5.1. Add SIP Signaling Group

Use the **add signaling-group n** command, where **n** is an available signaling group number, for one of the SIP trunks to the Session Manager, and fill in the indicated fields. Default values can be used for the remaining fields:

- **Group Type:** **sip**
- **Transport Method:** **tls**
- **Near-end Node Name:** **procr**
- **Far-end Node Name:** Session Manager node name from **Section 5.3** i.e., **asm**
- **Near-end Listen Port:** **5061**
- **Far-end Listen Port:** **5061**
- **Far-end Network Region:** **1**
- **DTMF over IP:** **rtp-payload**
- **Direct IP-IP Audio Connections:** **y**

```
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? y
  Peer Detection Enabled? y  Peer Server: SM
 Prepend '+' to Outgoing Calling/Alerting/Diverting/Connected Public Numbers? y
Remove '+' from Incoming Called/Calling/Alerting/Diverting/Connected Numbers? n
Alert Incoming SIP Crisis Calls? n
   Near-end Node Name: procr                  Far-end Node Name: asm
 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): 65                    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
```

5.5.2. Add Trunk Group

Add the corresponding trunk group controlled by this signaling group via the **add trunk-group n** command, where **n** is an available trunk group number and fill in the indicated fields.

- **Group Type:** **sip**
- **Group Name:** A descriptive name (i.e., **asm**)
- **TAC:** An available trunk access code (i.e., **101**)
- **Service Type:** **public-ntwrk**
- **Signaling Group:** The number of the signaling group associated (i.e., **1**)
- **Number of Members:** The number of SIP trunks to be allocated to calls routed to **Session Manager** (must be within the limits of the total trunks available from license verified in **Section 5.1**)

```
add 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? n
 Dial Access? n                                    Night Service:
Queue Length: 0
Service Type: public-ntwrk          Auth Code? n
                                             Member Assignment Method: auto
                                                      Signaling Group: 1
                                                    Number of Members: 10
```

Navigate to **Page 3** and change **Numbering Format** to **private**. Use default values for all other fields.

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



                     Numbering Format: private
                                                UUI Treatment: service-provider

                                                 Replace Restricted Numbers? n
                                                Replace Unavailable Numbers? n

                                                  Hold/Unhold Notifications? Y
                                Modify Tandem Calling Number: no




 Show ANSWERED BY on Display? y
```

5.6. Configure Route Patterns

Configure route patterns to correspond to the newly added SIP trunk group. Use the **change route pattern n** command, where **n** is an available route pattern.

The route pattern, as shown below, was configured to route calls to Session Manager and simulated PSTN endpoints.

5.6.1. Route Pattern for reaching Session Manager and Simulated PSTN Endpoints

When changing the route pattern, enter the following values for the specified fields and retain the default values for the remaining fields.

- **Pattern Name:** A descriptive name (i.e., **asm**)
- **Grp No:** The trunk group number from **Section 5.5.2**
- **FRL:** Enter a level that allows access to this trunk, with **0** being least restrictive
- **Numbering Format:** This was set to **lev0-pvt** in the tested configuration

```
change route-pattern 1                                          Page   1 of   3
                    Pattern Number: 1       Pattern Name: asm
    SCCAN? n    Secure SIP? n     Used for SIP stations? n

    Grp FRL NPA Pfx Hop Toll No.  Inserted                           DCS/ IXC
    No          Mrk Lmt List Del  Digits                             QSIG
                             Dgts                                    Intw
 1: 1    0                                                            n   user
 2:                                                                   n   user
 3:                                                                   n   user
 4:                                                                   n   user
 5:                                                                   n   user
 6:                                                                   n   user

     BCC VALUE  TSC CA-TSC    ITC BCIE Service/Feature PARM Sub  Numbering LAR
    0 1 2 M 4 W     Request                                 Dgts Format
 1: y y y y y n  n            rest                               lev0-pvt  none
 2: y y y y y n  n            rest                                         none
 3: y y y y y n  n            rest                                         none
 4: y y y y y n  n            rest                                         none
 5: y y y y y n  n            rest                                         none
 6: y y y y y n  n            rest                                         none
```

5.7. Administer Private Numbering

Use the **change private-numbering** command to define the calling party number to be sent out through the SIP trunk. In the sample network configuration below, all calls originating from a **5**-digit extension (**Ext Len**) beginning with **5** (**Ext Code**) and routed through any trunk will result in a **5**-digit calling number (**Total Len**). The calling party number will be in the SIP “From” header.

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

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

5.8. Administer Dial Plan and AAR analysis

Configure the dial plan for dialing 5-digit extensions beginning with **5** to stations registered with Session Manager.

Use the **change aar analysis n** command, where **n** is the dial string pattern to configure an **aar** entry for **Dialed String 5** (Extensions on Session Manager) to use **Route Pattern 1** (defined in **Section 5.6**). The **Call Type** was set to **lev0**.

```
change aar analysis 5                                        Page   1 of   2
                             AAR DIGIT ANALYSIS TABLE
                                   Location: all          Percent Full: 2

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

5.9. Administer AAR Analysis

For simulated calls, call dialed to a 5 digit number starting with 667 was routed to PSTN via AudioCodes Mediant 3000. Use the **change aar analysis 667** command and add an entry to specify how to route calls. Enter the following values for the specified fields and retain the default values for the remaining fields.

- **Dialed String:** Dialed digits to match on
- **Total Min:** Minimum number of digits, in this case **5**
- **Total Max:** Maximum number of digits, in this case **5**
- **Route Pattern:** The route pattern number from **Section 5.6**, i.e., **1**
- **Call Type:** **aar**

Note: The additional entries may be added for different number destinations.

```
change aar analysis 667                                      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
           667            5    5      1       aar           n
```

5.10. Administer Feature Access Code

Configure a feature access code to use for AAR and ARS routing. Use the **change feature access code** command to define **Access Code** for **Auto Alternate Routing (AAR)** and for **Auto Route Selection (ARS)**. In the test configuration, **8** and **9** were used respectively.

```
change feature-access-codes                                     Page   1 of  10
                               FEATURE ACCESS CODE (FAC)
         Abbreviated Dialing List1 Access Code:
         Abbreviated Dialing List2 Access Code:
         Abbreviated Dialing List3 Access Code:
Abbreviated Dial - Prgm Group List Access Code:
                      Announcement Access Code:
                       Answer Back Access Code:
                         Attendant Access Code:
      Auto Alternate Routing (AAR) Access Code: 8
    Auto Route Selection (ARS) - Access Code 1: 9     Access Code 2:
                 Automatic Callback Activation:        Deactivation:
```

5.11. Save Changes

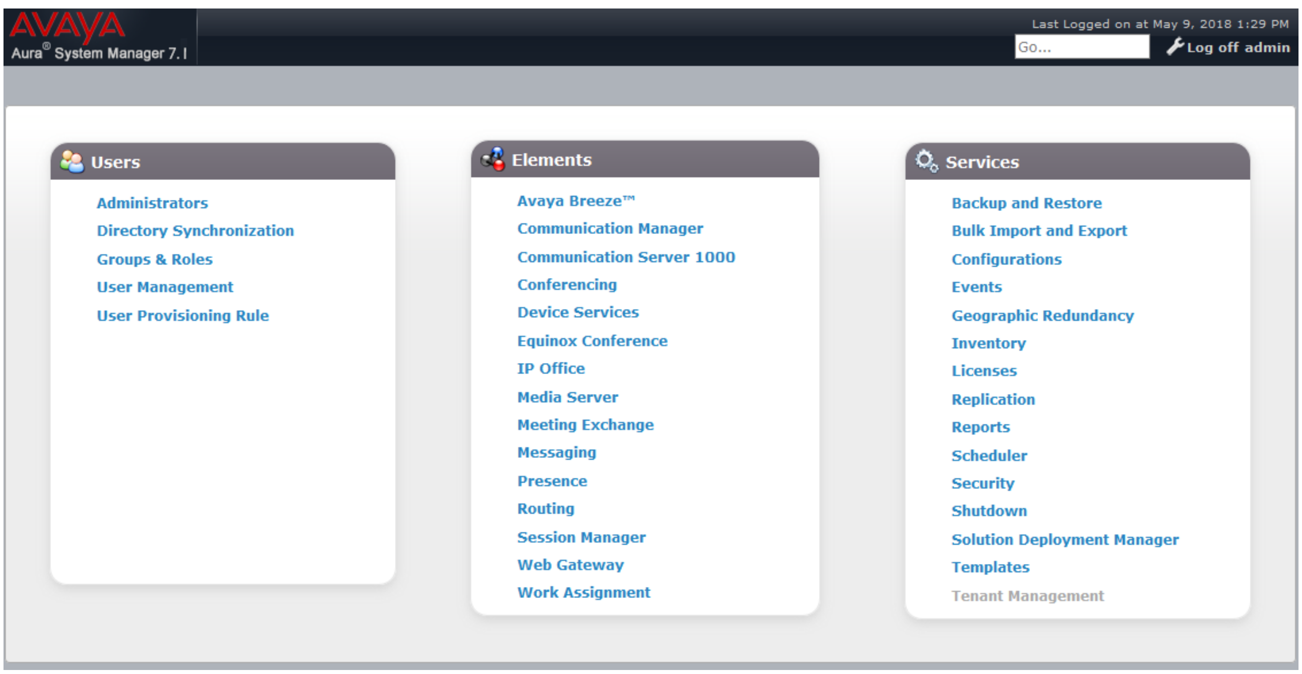
Use the **save translation** command to save all changes.

6. Configure Avaya Aura® Session Manager

This section provides the procedures for configuring Session Manager, assuming it has been installed and licensed as described in **Reference [2]**. The procedures include adding the following items:

- Specify SIP Domain
- Add Locations
- Add Adaptations
- Add SIP Entities and Entity Links
- Add Routing Policies
- Add Dial Patterns
- Add Users for SIP Phones

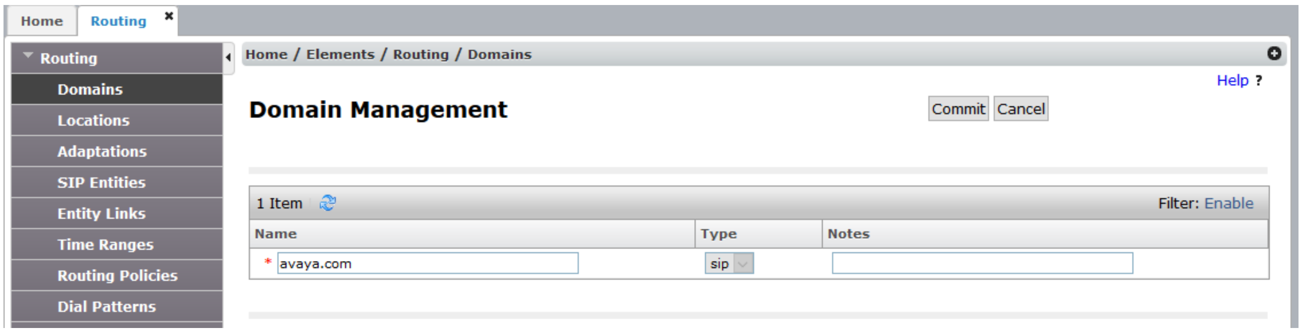
Configuration is accomplished by accessing the browser-based GUI of System Manager, using the URL **http://<ip-address>/SMGR**, where **<ip-address>** is the IP address of System Manager. Log in with the appropriate credentials. The home screen as shown below is displayed. Expand the **Routing** Link under **Elements**.



6.1. Specify SIP Domain

Add the SIP domain for which the communications infrastructure will be authoritative. Do this by selecting **Domains** on the left and clicking the **New** button on the right (not shown). The following screen will then be shown. Fill in the following fields and click **Commit**.

- **Name:** The authoritative domain name (e.g., **avaya.com**)
- **Type** Select **sip**
- **Notes:** Descriptive text (optional)



6.2. Add Locations

Locations can be used to identify logical and/or physical locations where SIP Entities reside, for purposes of bandwidth management. A single location is added to the configuration for Communication Manager and the Mediant 3000 Gateway. To add a location, select **Locations** on the left and click on the **New** button on the right (not shown). The following screen will then be shown. Fill in the following:

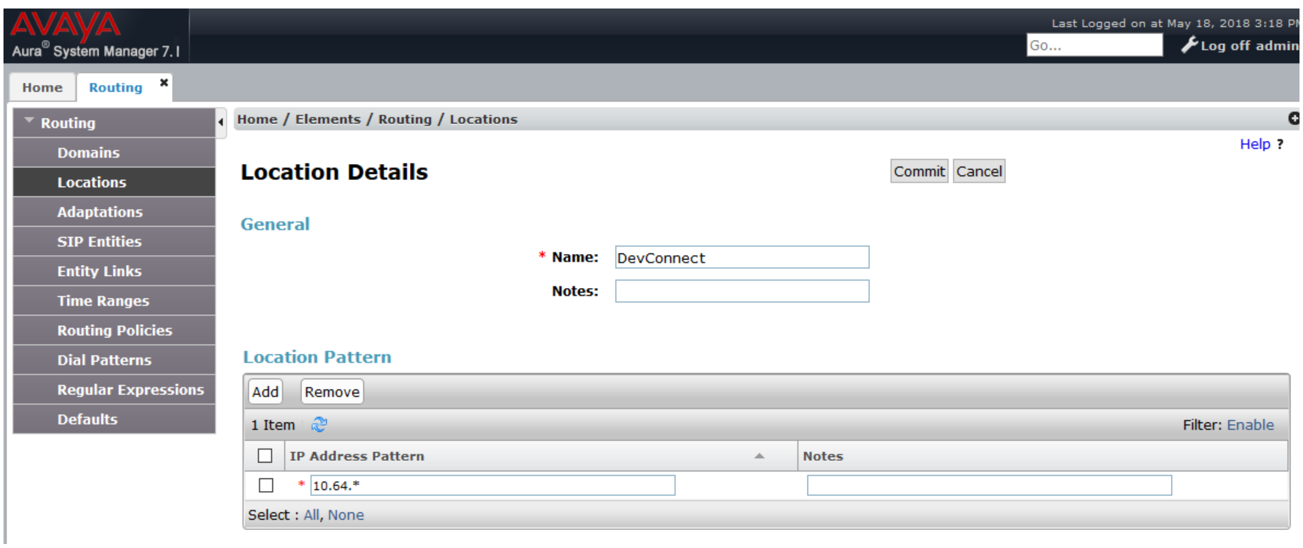
Under **General**:

- **Name:** A descriptive name

Under **Location Pattern**:

- **IP Address Pattern:** A pattern used to logically identify the location (optional). In these Application Notes, no pattern was defined.

Defaults can be used for the remaining fields. The screen below shows addition of the **DevConnect** location, which includes all the components of the compliance test environment. Click **Commit** to save.



6.3. Add SIP Entities and SIP Entity Links

A SIP Entity is required for each SIP-based telephony system wishing to communicate with Session Manager for call routing. In the sample configuration, a SIP Entity and SIP Entity Link is added for Communication Manager, and the Mediant 3000.

6.3.1. Adding Avaya Aura® Communication Manager SIP Entity and SIP Entity Link

Navigate to **Network Routing Policy → SIP Entities** on the left and click on the **New** button on the right (not shown).

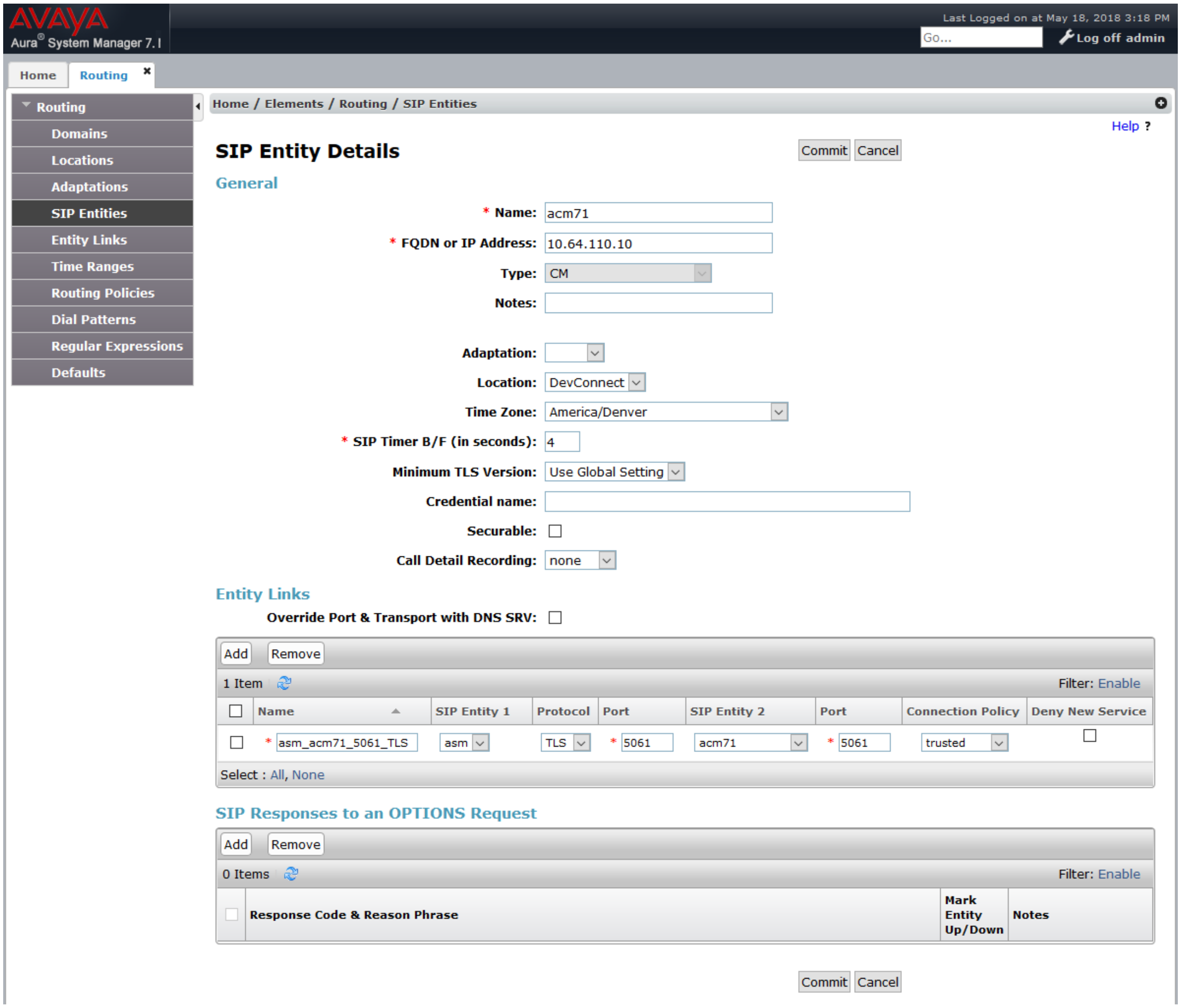
Under **General**:

- **Name:** A descriptive name, i.e., **acm71**
- **FQDN or IP Address:** IP address of the Communication Manager i.e., **10.64.110.10**
- **Type:** Select **CM**
- **Location:** Select one of the locations defined previously
- **Time Zone:** Time zone for this entity

Add Entity Links. Under **Entity Links**, click **Add**, and then edit the fields in the resulting new row as shown below:

- **Name:** Will be populated automatically
- **SIP Entity 2:** Will be populated automatically with the name of this SIP Entity.
- **SIP Entity 1:** Select Session Manager from the pull down box
- **Protocol:** Select the desired Protocol from the pull down box
- **Port:** Enter the desired port number for the Entity Link
- **Policy:** Select the appropriate Connection Policy from the pull down box

Defaults can be used for the remaining fields. Click **Commit** to save each SIP Entity definition. The following screen shows the addition of the SIP Entity for Communication Manager.


AVAYA
Aura® System Manager 7.1
Last Logged on at May 18, 2018 3:18 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 / SIP Entities
Help ?
SIP Entity Details
Commit
Cancel
General
* Name: acm71
* FQDN or IP Address: 10.64.110.10
Type: CM
Notes:
Adaptation:
Location: DevConnect
Time Zone: America/Denver
* SIP Timer B/F (in seconds): 4
Minimum TLS Version: Use Global Setting
Credential name:
Securable:
Call Detail Recording: none
Entity Links
Override Port & Transport with DNS SRV:
Add
Remove
1 Item
Filter: Enable
Name
SIP Entity 1
Protocol
Port
SIP Entity 2
Port
Connection Policy
Deny New Service
* asm_acm71_5061_TLS
asm
TLS
* 5061
acm71
* 5061
trusted
Select : All, None
SIP Responses to an OPTIONS Request
Add
Remove
0 Items
Filter: Enable
Response Code & Reason Phrase
Mark Entity Up/Down
Notes
Commit
Cancel

6.3.2. Adding AudioCodes Mediant 3000 Gateway SIP Entity

Navigate to **Network Routing Policy → SIP Entities** on the left and click on the **New** button on the right (not shown).

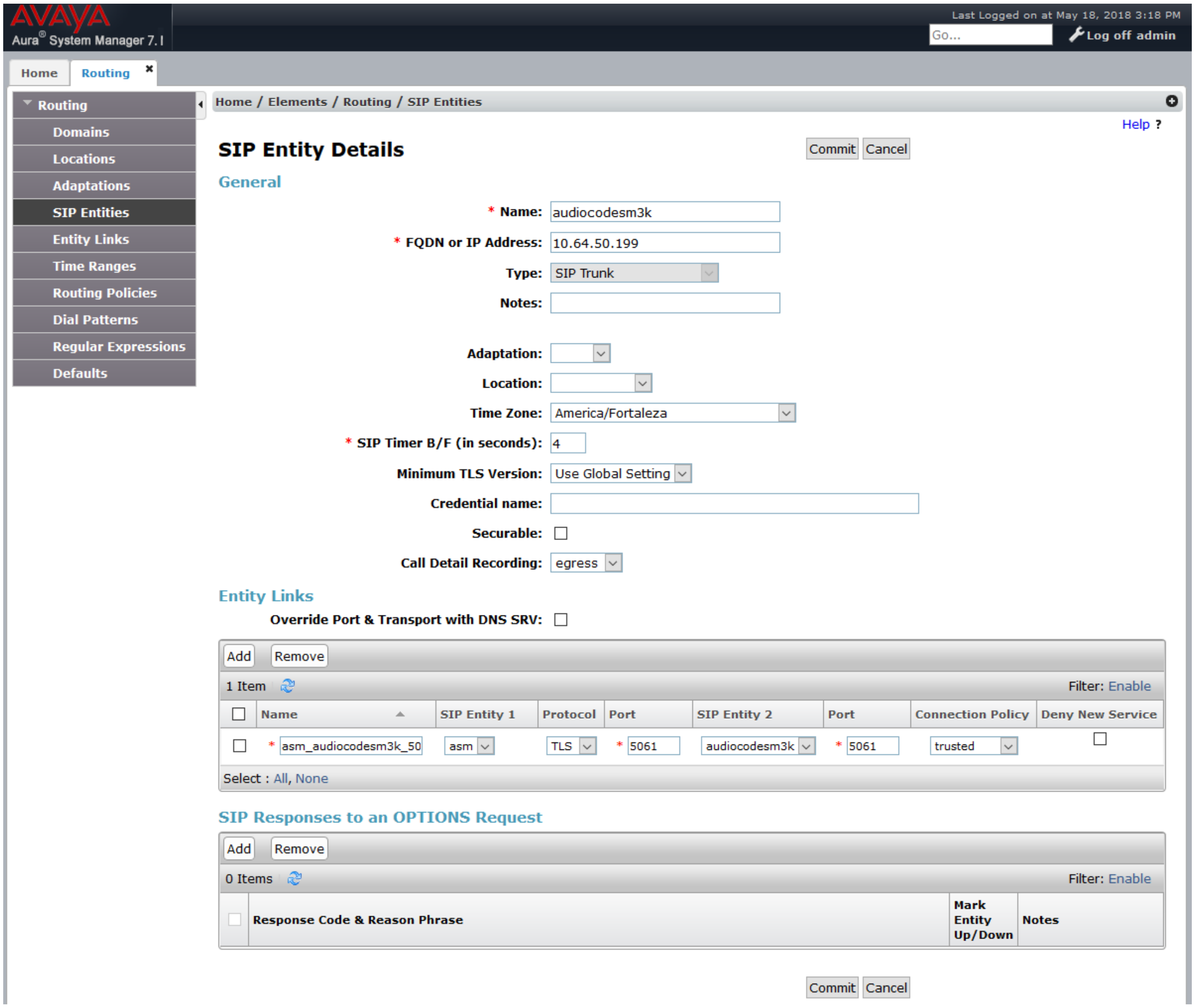
Under **General**:

- **Name:** A descriptive name, i.e., **audiocodesm3k**
- **FQDN or IP Address:** IP address of the Mediant 3000 i.e., **10.64.50.199**
- **Type:** Select **SIP Trunk**

Add Entity Links. Under **Entity Links**, click **Add**, and then edit the fields in the resulting new row as shown below:

- **Name:** Will be populated automatically
- **SIP Entity 2:** Will be populated automatically with the name of this SIP Entity.
- **SIP Entity 1:** Select Session Manager from the pull down box
- **Protocol:** Select the desired Protocol from the pull down box
- **Port:** Enter the desired port number for the Entity Link
- **Policy:** Select the appropriate Connection Policy from the pull down box

Defaults can be used for the remaining fields. Click **Commit** to save each SIP Entity definition. The following screen shows the addition of the SIP Entity for Mediant 3000.


AVAYA
Aura® System Manager 7.1
Last Logged on at May 18, 2018 3:18 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 / SIP Entities
Help ?
SIP Entity Details
Commit
Cancel
General
* Name: audiocodesm3k
* FQDN or IP Address: 10.64.50.199
Type: SIP Trunk
Notes:
Adaptation:
Location:
Time Zone: America/Fortaleza
* SIP Timer B/F (in seconds): 4
Minimum TLS Version: Use Global Setting
Credential name:
Securable:
Call Detail Recording: egress
Entity Links
Override Port & Transport with DNS SRV:
Add
Remove
1 Item
Filter: Enable
Name
SIP Entity 1
Protocol
Port
SIP Entity 2
Port
Connection Policy
Deny New Service
asm_audiocodesm3k_50
asm
TLS
5061
audiocodesm3k
5061
trusted
Select : All, None
SIP Responses to an OPTIONS Request
Add
Remove
0 Items
Filter: Enable
Response Code & Reason Phrase
Mark Entity Up/Down
Notes
Commit
Cancel

6.4. Add Routing Policies

Routing policies describe the condition under which calls will be routed to the SIP Entities specified in **Section 6.3**. A routing policy must be added for Communication Manager and the Mediant 3000 Gateway. To add a routing policy, select **Routing Policies** on the left and click on the **New** button on the right (not shown). The following screen is displayed. Fill in the following:

Under **General**

- Enter a descriptive **Name**

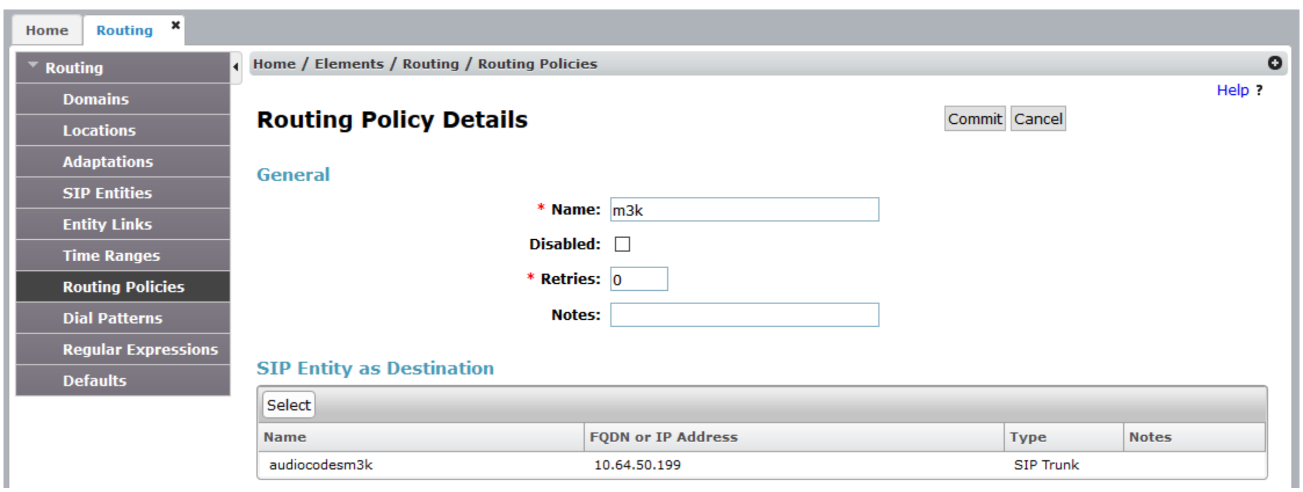
Under **SIP Entity as Destination**

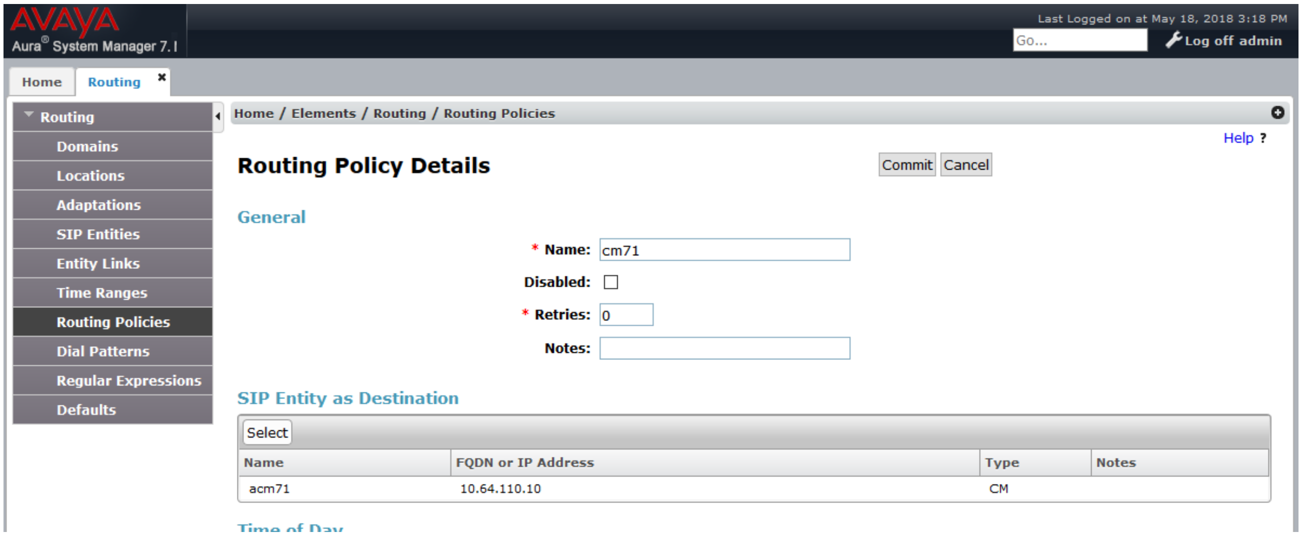
- Click **Select**, and then select the appropriate SIP entity to which this routing policy applies

Under **Time of Day:**

- Click **Add**, and select the time range configured. In these Application Notes, the predefined **24/7** Time Range is used

Defaults can be used for the remaining fields. Click **Commit** to save each Routing Policy definition. The following screens show the Routing Policies for Communication Manager and the Mediant 3000.




AVAYA
Aura® System Manager 7.1
Last Logged on at May 18, 2018 3:18 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 / Routing Policies
Help ?
Routing Policy Details
Commit
Cancel
General
* Name: cm71
Disabled:
* Retries: 0
Notes:
SIP Entity as Destination
Select
Name
FQDN or IP Address
Type
Notes
acm71
10.64.110.10
CM
Time of Day

6.5. Add Dial Patterns

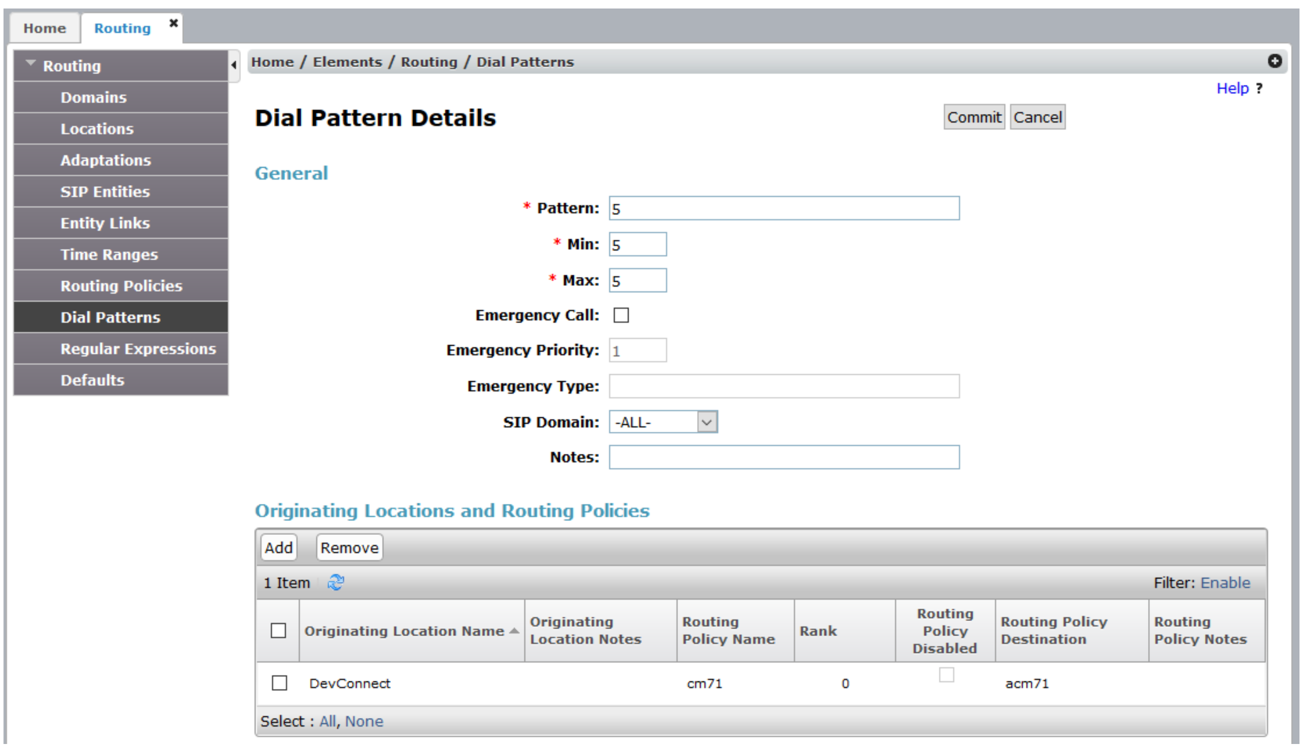
Dial patterns must be defined that will direct calls to the appropriate SIP Entity. In the sample configuration numbers beginning with **5** with 5-digit length reside in the Enterprise network. To add a dial pattern, select **Dial Patterns** on the left and click on the **New** button on the right (not shown). Fill in the following, as shown in the screen below, which corresponds to the dial pattern for routing calls to Communication Manager.

Under **General**:

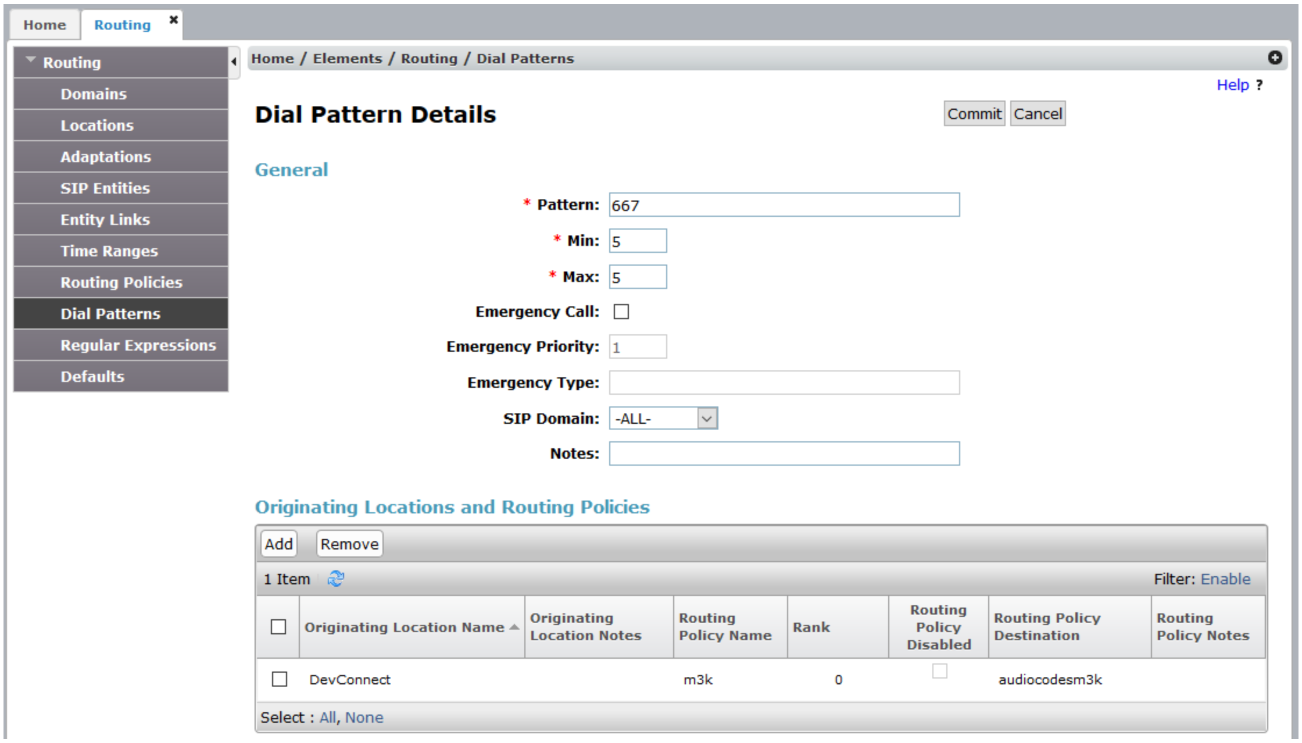
- **Pattern:** Dialed number or prefix i.e., **5**
- **Min:** Minimum length of dialed number i.e., **5**
- **Max:** Maximum length of dialed number i.e., **5**
- **SIP Domain:** Select **ALL**

Under **Originating Locations and Routing Policies**, click **Add**, and then select the appropriate location and routing policy from the list. Default values can be used for the remaining fields. Click **Commit** to save this dial pattern.

The following screen shows the dial pattern definition for calls within the Enterprise.



The following screen shows the dial pattern definition for calls destined for the Mediant 3000.



7. AudioCodes Mediant 3000 Configuration

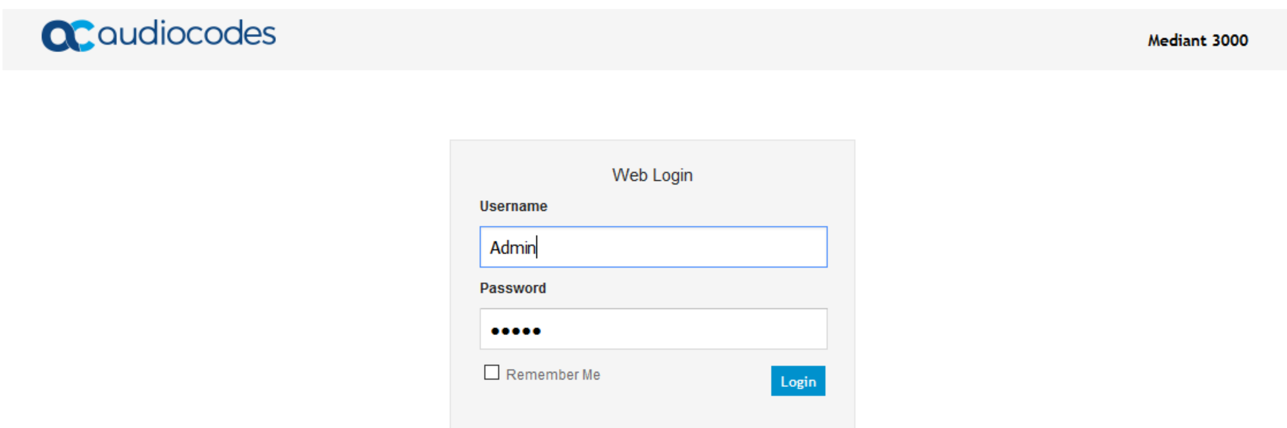
This section describes the configuration for enabling the Mediant 3000 to interoperate with Session Manager and Simulated PSTN.

The Mediant 3000 can be administered using the Native Web Interface or AudioCodes Element Management System (EMS) as described in **Reference [3]**. Note that this section displays the provisioning that was utilized for this sample configuration, and does not show exhaustive procedures for administering an initial configuration. In these Application Notes, configuration was accomplished with the web interface.

7.1. Log Into Mediant 3000

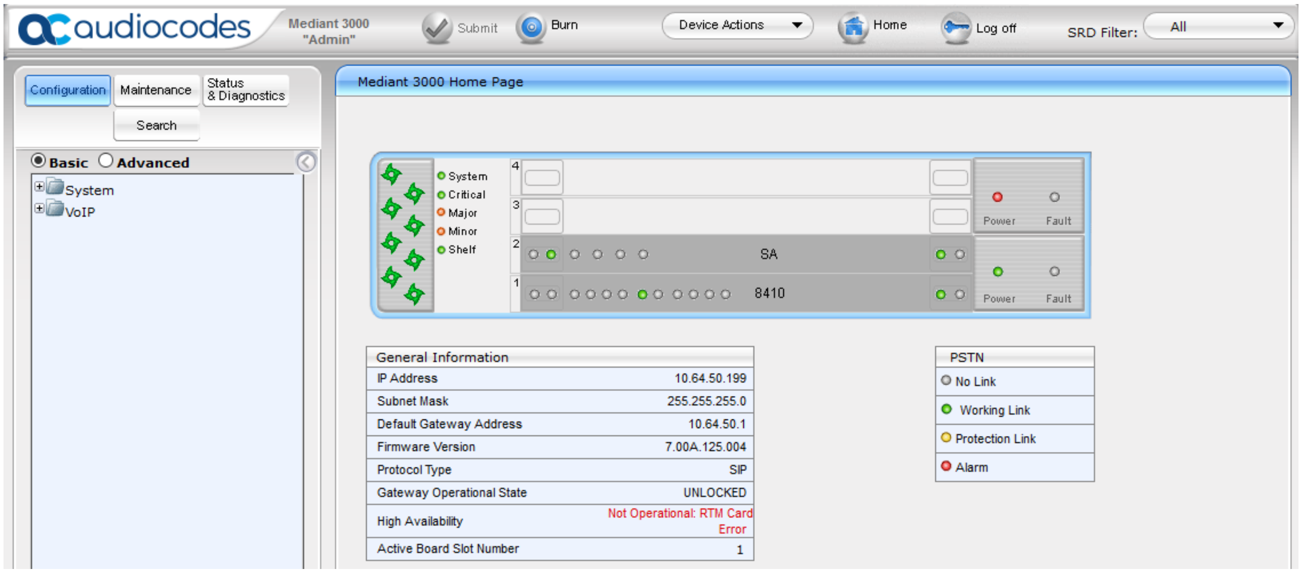
The configuration of the Mediant 3000 Gateway is done via a Web browser. To access the device, enter the **IP address** of the Mediant 3000 in the **Address** field of the web browser. The IP address was provisioned during initial installation.

Login credentials
The following pop-up window will appear. Log in with the proper credentials.



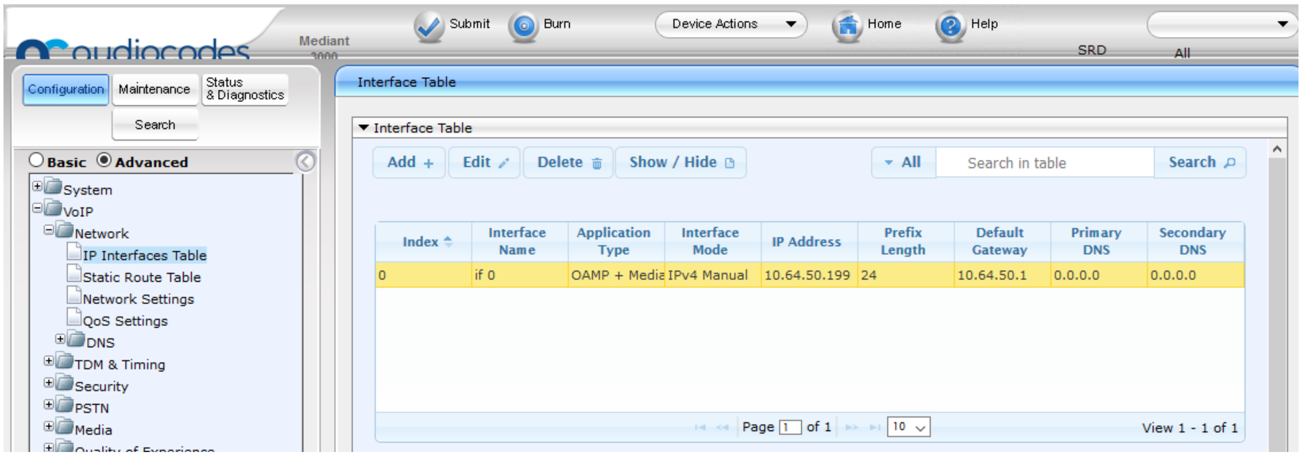
Mediant 3000 Home Page

The **Mediant 3000 Home Page** will appear as shown below.



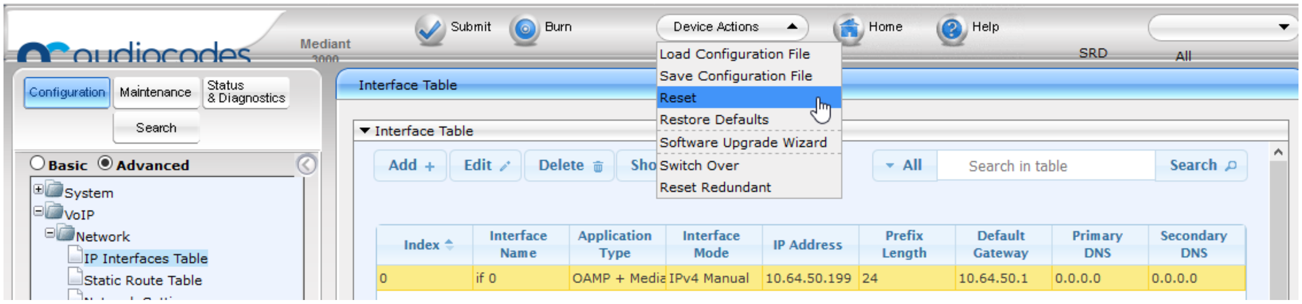
7.2. Configure Media Gateway IP Network Parameters

To configure the network parameters, navigate to **VoIP → Network → IP Interfaces Table** and click on the **Add** button to add an index with **Application Type** of **OAMP + Media + Control** and ensure the **Interface Mode** is set to **IPv4 Manual** and that **IP Address** (i.e., **10.64.50.199**), **Prefix Length** (i.e., **24**), and **Default Gateway** (i.e., **10.64.50.1**) are set according to the expected values.

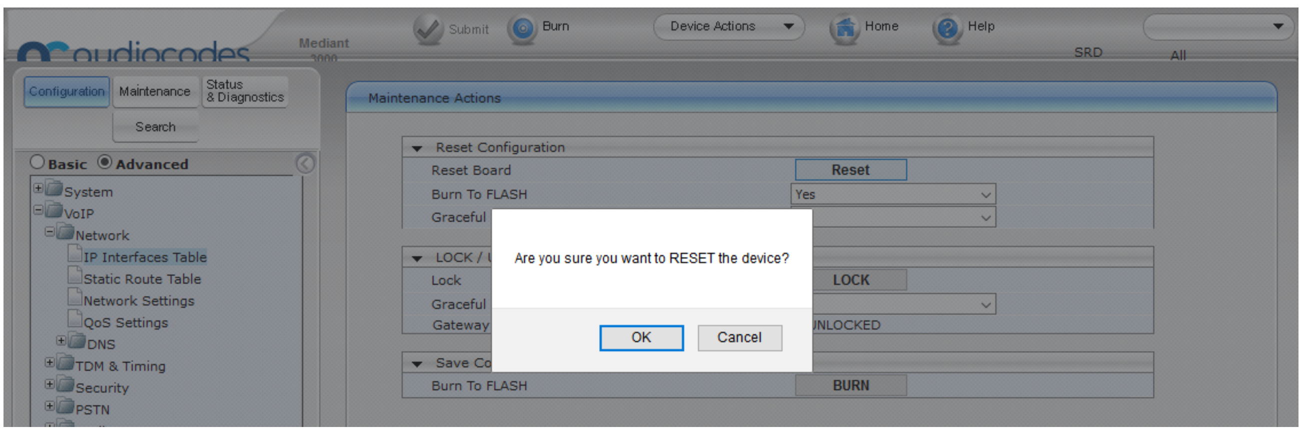


7.3. Saving Configuration and Resetting Mediant 3000

Save settings to the device's flash memory and reset the device by performing the following:
From the **Device Actions** pull-down menu, click **Reset** to display the **Maintenance Actions** screen.



Make sure **Burn To FLASH** is set to **Yes**, and then click the **Reset** button then click **OK** for confirmation. The device's new configuration is saved (burned) to the flash memory and the device resets.

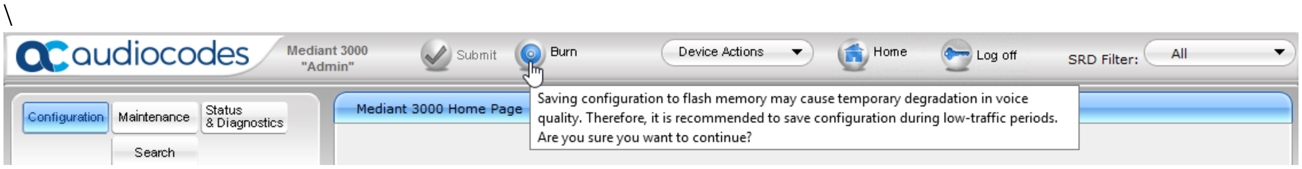


Note: if any parameter with the lightening symbol beside it (see the screenshot is **Section 7.6.1** for example) is changed, a Reset with Burn To Flash is required. The reset does not have to be done until all configuration is completed, and there will be a red reset notification at the top of the page (not shown).

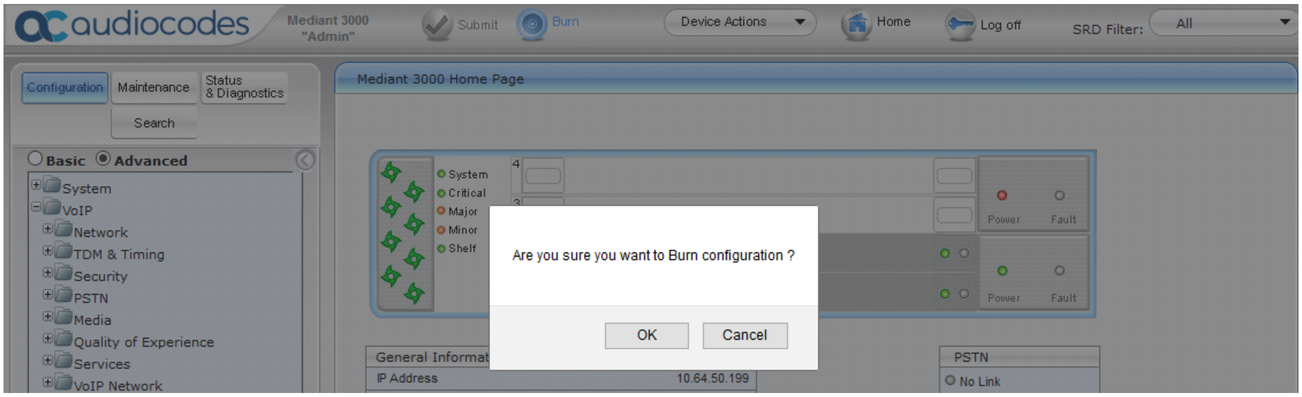
7.4. Saving Configuration

To permanently save settings to the device's flash memory, activate the **Maintenance Actions** page (**Maintenance** tab → **Maintenance** → **Maintenance Actions**) and click the **BURN** button under **Save Configuration** as shown below.

\



Also note the **Burn** button at the top of the screen. This is the shortest path to do a burn and can be used at any time. When clicked, it will present a pop-up similar to the one shown below.

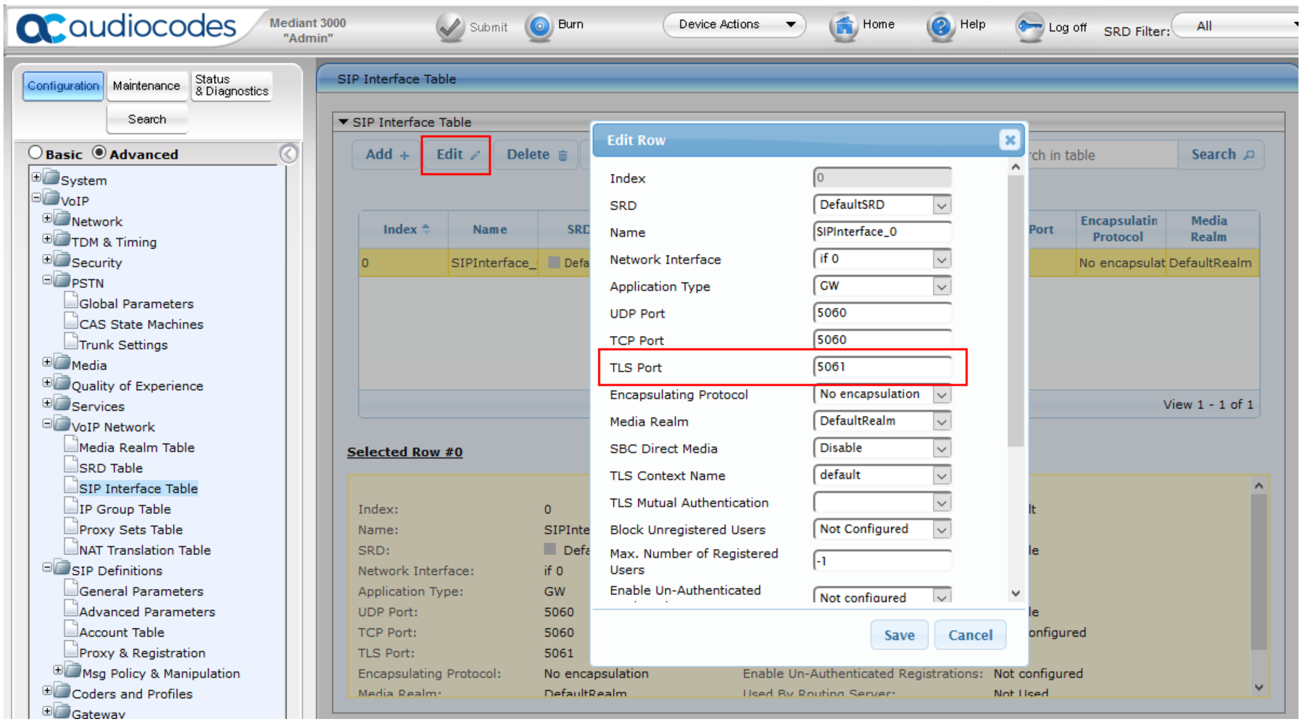


7.5. Configure SIP Interface to Avaya Aura® Session Manager

This section provides instructions to configure SIP Interface to Session Manager

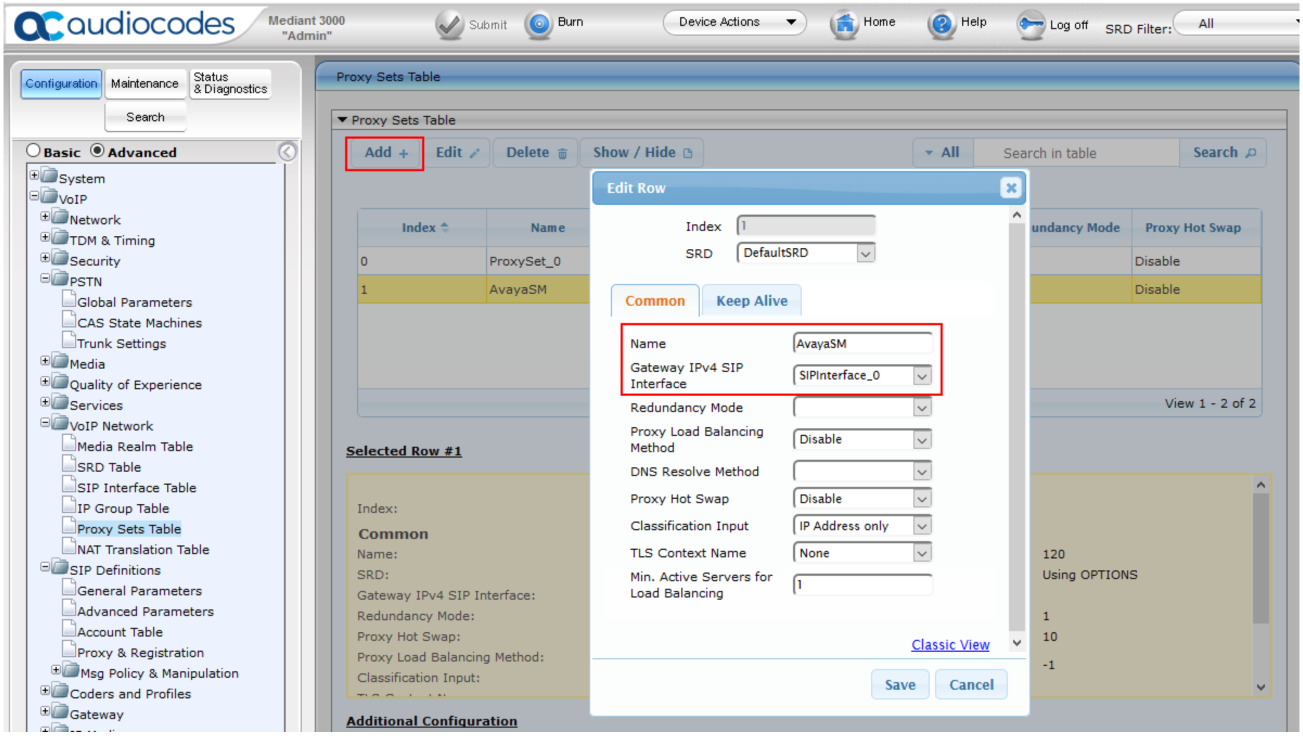
7.5.1. Configure SIP Interface Table

Configure the TLS port **5061,** which will be used to send SIP/TLS signaling between Session Manager and Mediant M3K. To configure SIP Interface Table, on the left pane navigate to **VoIP → VoIP Network → SIP Interface Table**. Edit the existing table, and set **TLS Port** to **5061.**

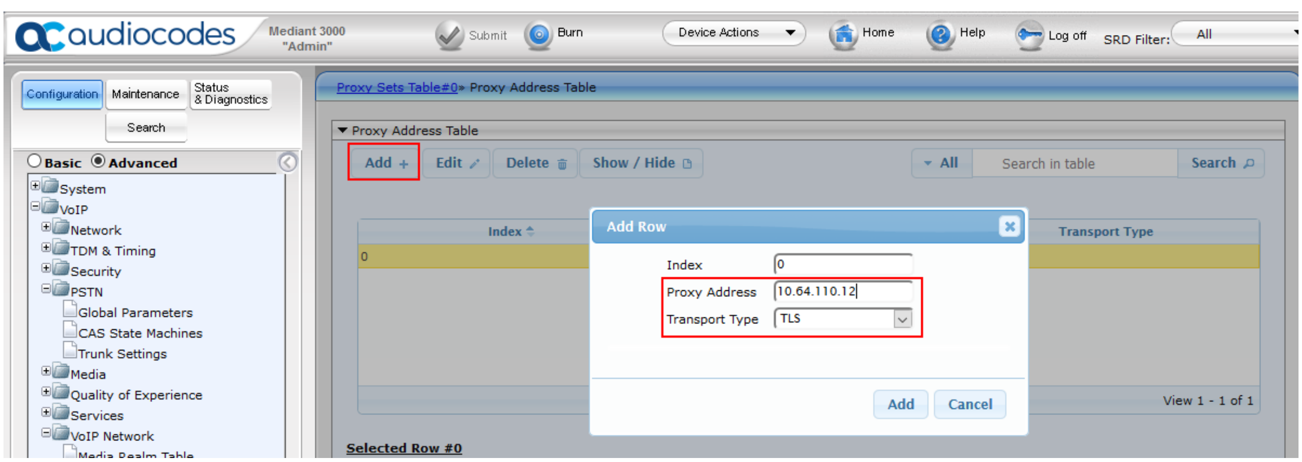


7.5.2. Configure Proxy Sets Table

Configure an Proxy Sets Table for Session Manager. To configure Proxy Sets Table, on the left pane navigate to **VoIP → VoIP Network → Proxy Sets Table.** Click **Add**, provide a **Name** and select the SIP Interface from previous section for **Gateway IPv4 SIP Interface.**



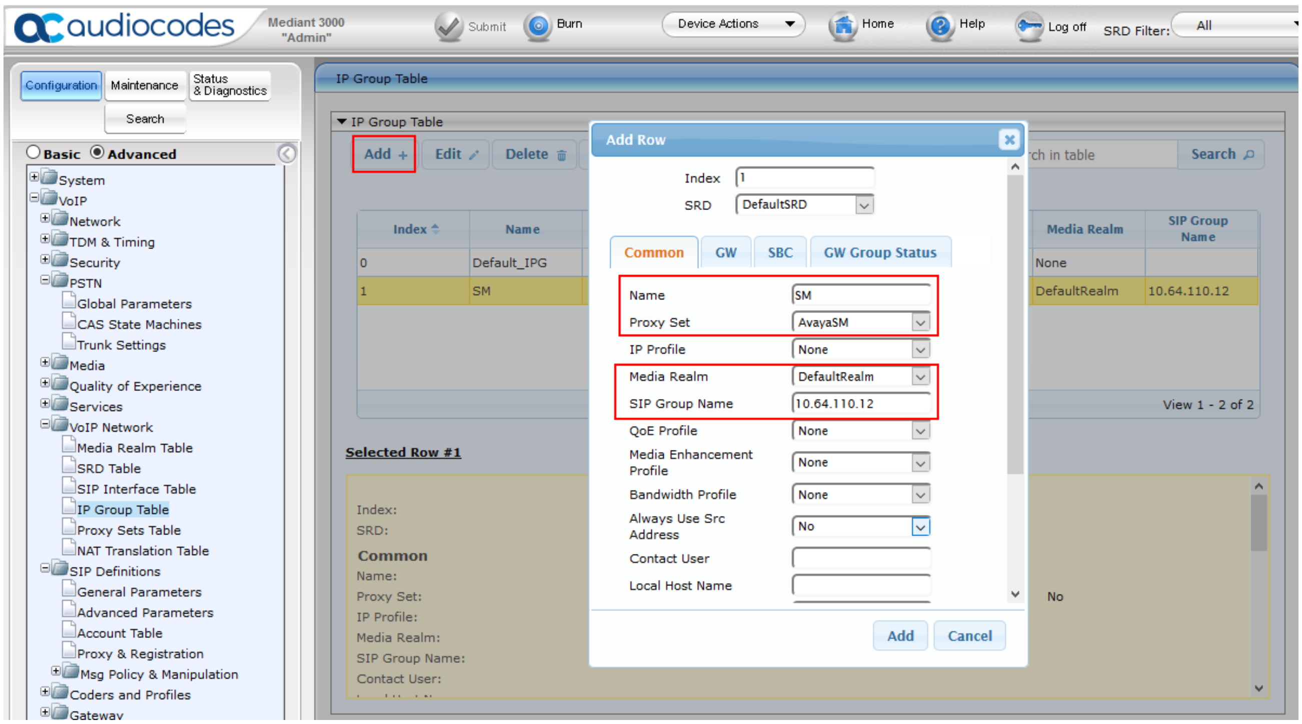
At the bottom of the page select **Proxy Address Table** (not shown). Select **Add** an entry for Session Manager. Configure **Proxy Address** to the SIP Signaling IP Address of Session Manager and **Transport Type** to **TLS.**



7.5.3. Configure IP Group Table

Configure an IP Group Table for Session Manager. To configure SIP Interface Table, on the left pane navigate to **VoIP → VoIP Network → IP Group Table.** Configure as follows:

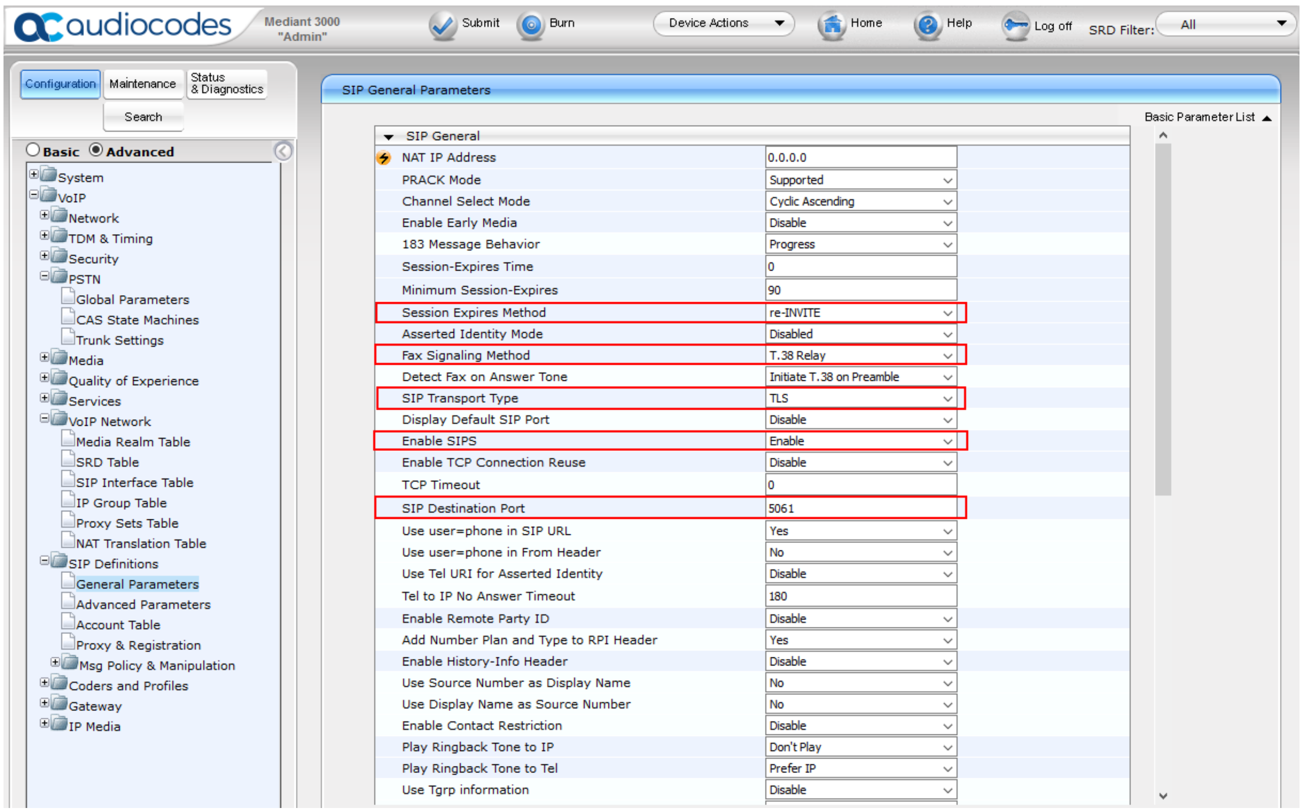
- **Name:** A desired name
- **Proxy Set:** Proxy Set configured in previous section
- **Media Realm:** **DefaultRealm**
- **SIP Group Name:** SIP Signaling IP Address of Session Manager



7.5.4. Configure General Parameters

To configure SIP General Parameters, navigate to **VoIP → SIP Definitions → General Parameters.** Configure as follows:

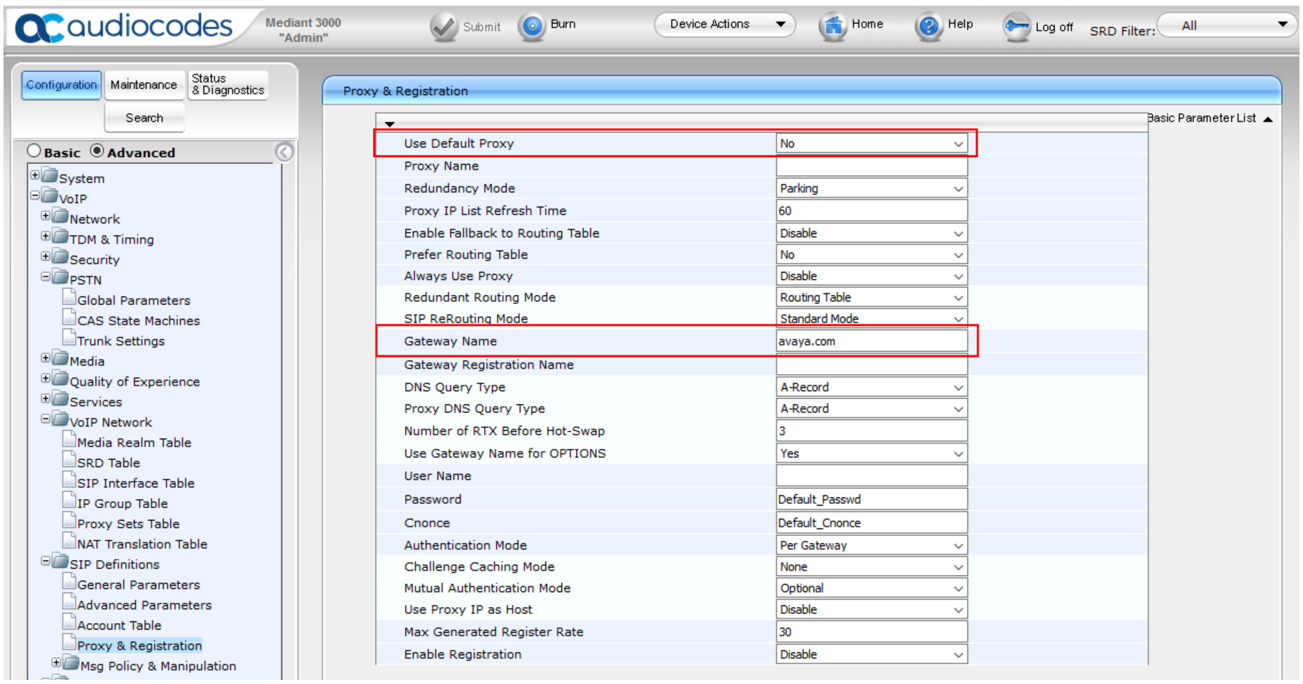
- **Session Expires Method:** **re-INVITE**
- **Fax Signaling Method:** **T-38 Relay**
- **SIP Transport Type:** **TLS**
- **Enable SIPS:** **Enable**
- **SIP Destination Port:** **5061**



7.5.5. Configure Proxy & Registration

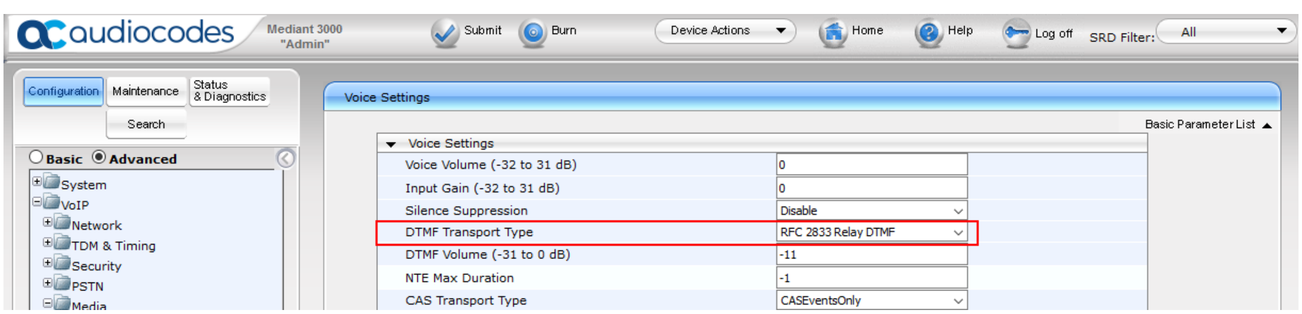
To configure Proxy and Registrations Parameters, navigate to **VoIP → SIP Definitions →Proxy & Registration.** Configure as follows:

- **User Default Proxy:** **No**
- **Gateway Name:** Domain configured in **Section 6.1.**



7.5.6. Configure the Voice parameters

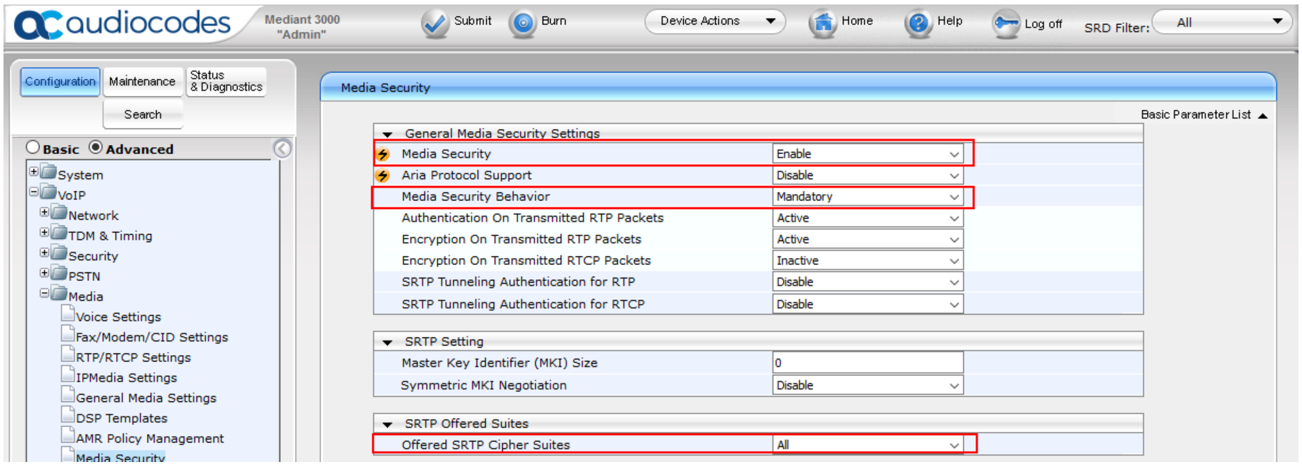
To configure the Voice Settings, navigate to **VoIP → Media → Voice Settings**. Set **DTMF Transport Type** to **RFC2833 Relay DTMF** as shown below, and click the **Submit** button to save changes.



7.5.7. Configure Media Security

To configure SRTP, navigate to **VoIP → Media → Media Security**. Configure as follows:

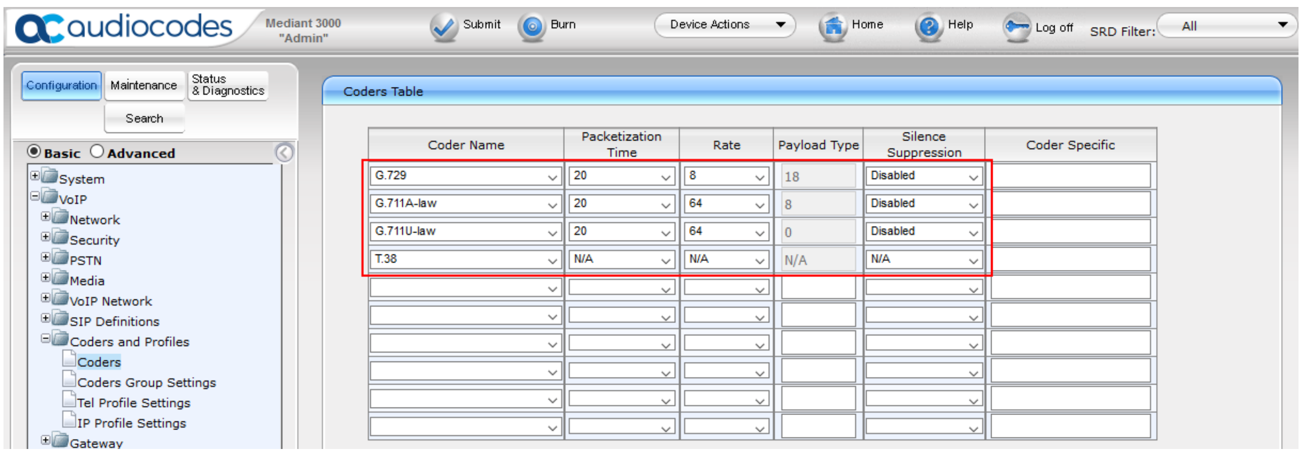
- **Media Security:** **Enable**
- **Media Security Behavior:** **Mandatory**
- **Offered SRTP Cipher Suites:** **All**



7.5.8. Configure Coders

To configure Codecs, navigate to V**oIP → Coders and Profiles → Coders**. Configure as follows:

- From the **Coder Name** drop-down list, select the required coder
- From the **Packetization Time** drop-down list, select the packetization time (in msec) for the selected coder. The packetization time determines how many coder payloads are combined into a single RTP packet
- From the **Rate** drop-down list, select the bit rate (in kbps) for the selected coder
- In the **Payload Type** field, if the payload type (i.e., format of the RTP payload) for the selected coder is dynamic, enter a value from 0 to 120 (payload types of 'well-known' coders cannot be modified)
- From the **Silence Suppression** drop-down list, enable or disable the silence suppression option for the selected coder
- Repeat **Step 2** through **Step 6** for the next optional coders

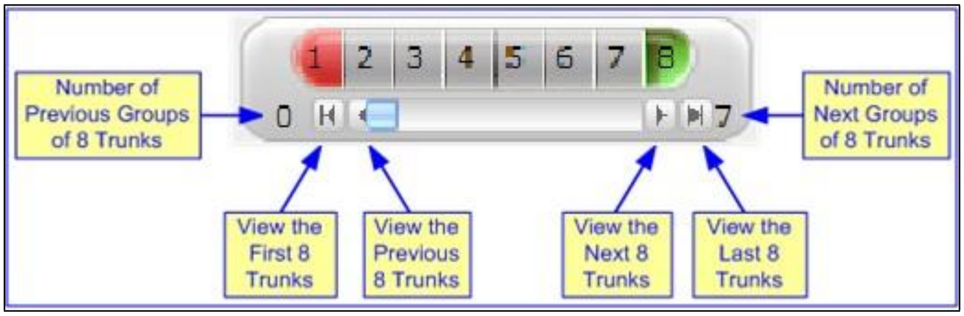


7.6. Configure T1 Interface to Simulated PSTN

This section provides information to configure T1 interface to Simulated PSTN.

7.6.1. Configure Trunk Settings

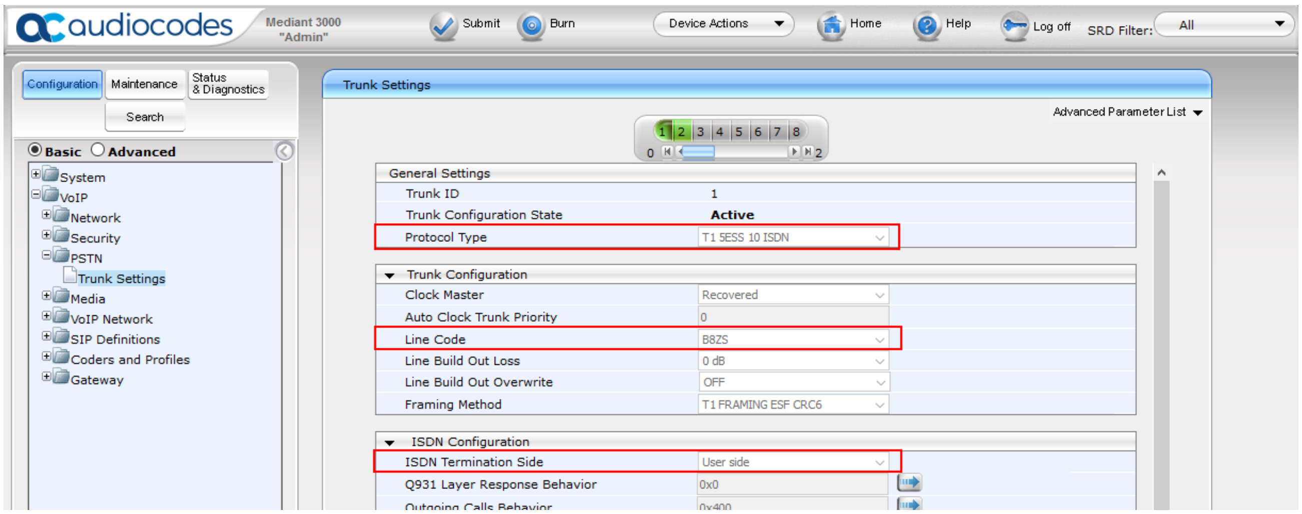
To configure Trunk Settings , navigate to **VoIP → PSTN → Trunk Settings**. Select the trunk to be configured, by clicking the desired Trunk number icon in the right pane. The bar initially displays the first eight trunk number icons (i.e., trunks 1 through 8). To scroll through the trunk number icons (i.e., view the next/last or previous/first group of eight trunks), refer to the figure below:



Click the **Stop Trunk**  button (located at the bottom of the page) to take the trunk out of service to allow configuration of the currently grayed out (unavailable) parameters. (Skip this step to configure parameters that are available when the trunk is active). The stopped trunk is indicated by the **Trunk Configuration State** field displaying **Inactive**. The **Stop Trunk** button is replaced by the **Apply Trunk Settings**  button.

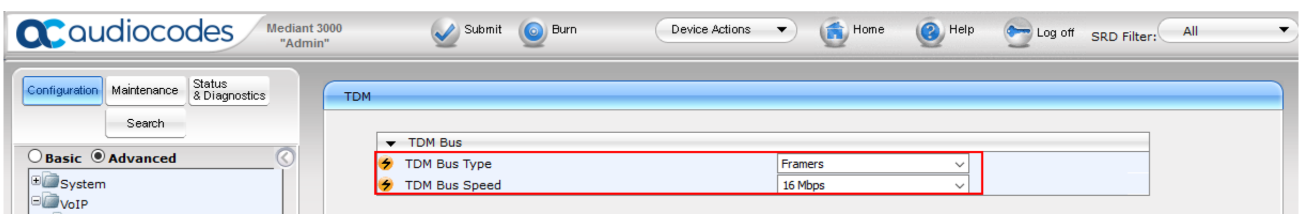
In these Application Notes the PSTN interface was configured as follows:

- **Protocol Type:** **T1 5ESS 10 ISDN**
- **Line Code:** **B8ZS**
- **Framing Method:** **T1 FRAMINIG ESF CRC6**
- **ISDN Termination Side:** **User side**



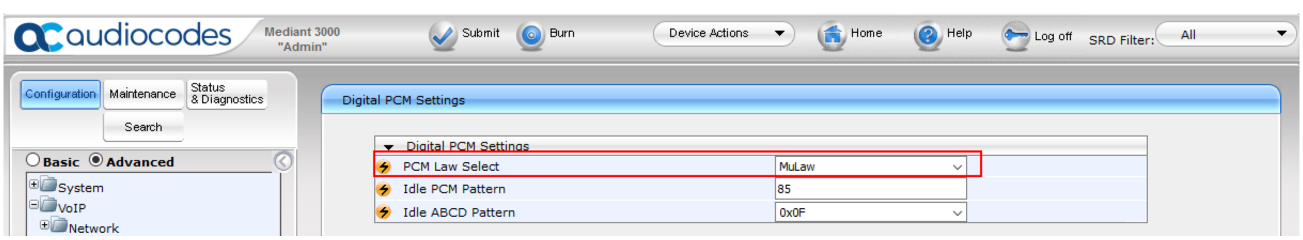
7.6.2. Configure TDM Bus

To configure the TDM Bus settings, navigate to **VoIP → TDM & Timing → TDM**, configure **TDM Bus Type** and **TDM Bus Speed** parameters as required. For T1 set **TDM Bus Type** to **Framers** and **TDM Bus speed** to **16Mbps**.



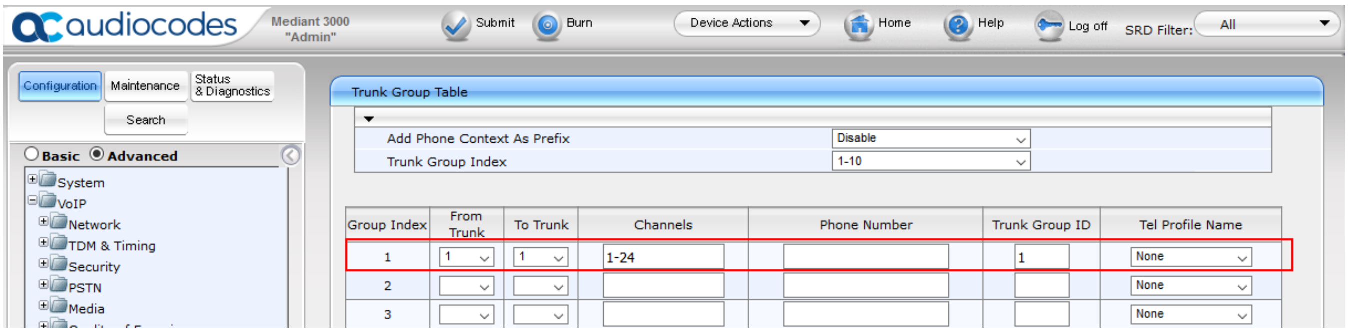
7.6.3. Configure Digital PCM Settings

To configure the digital PCM settings, navigate to **VoIP → TDM & Timing → Digital PCM Settings**. Configure the parameters as required, i.e., **MuLaw** for **PCM Law Select** for T1.



7.6.4. Configure Trunk Group Table

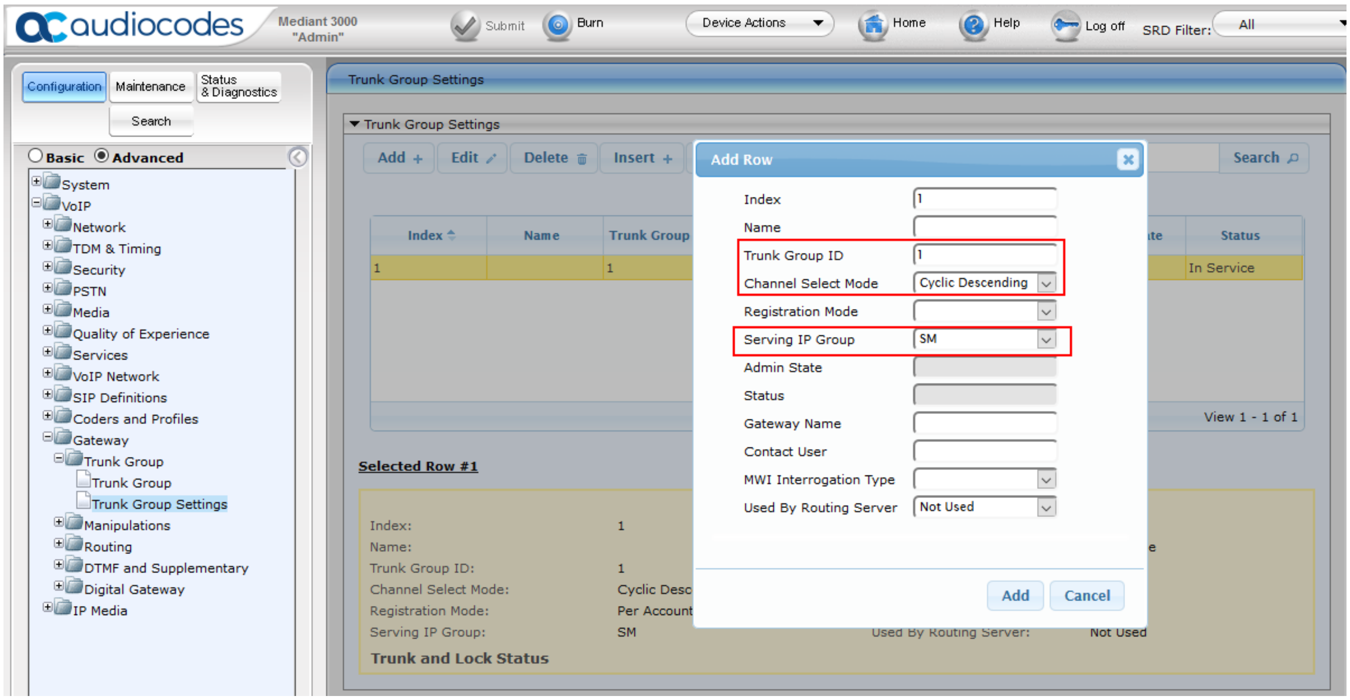
To configure Trunk Group, navigate to **VoIP → Gateway → Trunk Group → Trunk Group**. Select the appropriate **Group Index**, and set the appropriate parameters in the table, i.e., **From /To Trunk, Channels, Phone Number, Trunk Group ID, Tel Profile ID**. For detailed information refer to **[3]**. The screen below illustrates setting used for the compliance test.



7.6.5. Configure Trunk Group Settings

To configure Trunk Group Settings, navigate to **VoIP → Gateway → Trunk Group → Trunk Group Settings.** Select **Add** to add settings for the trunk group created in previous section. Configure as follows:

- **Trunk Group ID:** Trunk Group ID from previous section
- **Channel Select Mode:** **Cyclic Descending**
- **Serving IP Group:** From **Section 7.5.3.**



7.7. Configure Routing

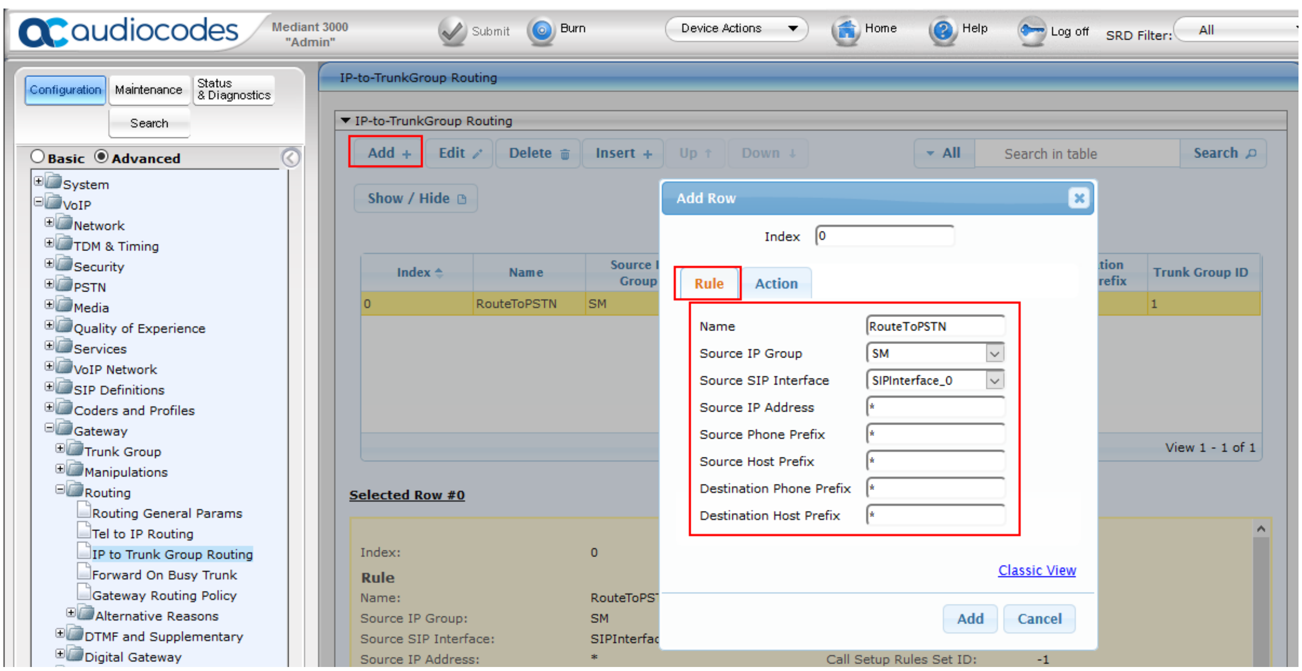
This section provides information to configure routing between Session Manager and Simulated PSTN via Mediant 3000.

7.7.1. Configure IP to Trunk Group Routing Rules

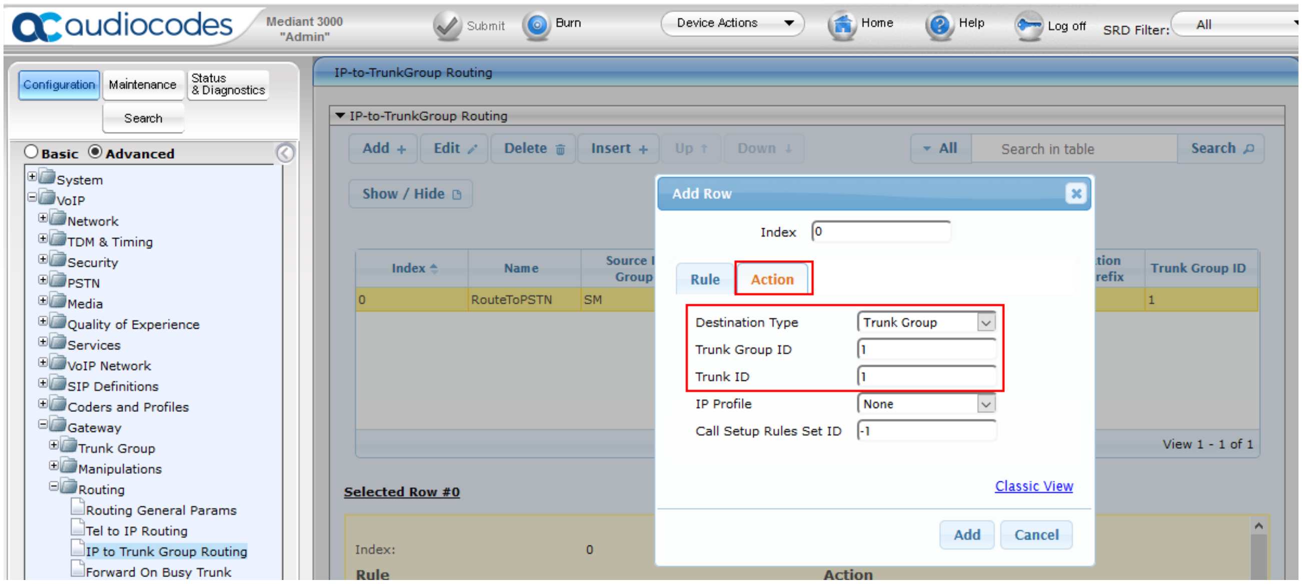
To configure route to Simulated PSTN, navigate to **VoIP → Gateway → Routing → IP to Trunk Group Routing**. To **Add** an entry, select **Add** and configure as follows:

- **Rule** tab:
  - **Name:** Desired name
  - **Source IP Group:** From **Section 7.5.3**
  - **Source SIP Interface:** From **Section 7.5.2**
  - **Source IP Address:** *
  - **Source Phone Prefix** *
  - **Source Host Prefix** *
  - **Destination Phone Prefix** *
  - **Destination Host Prefix** *

Continue to **Action** tab.



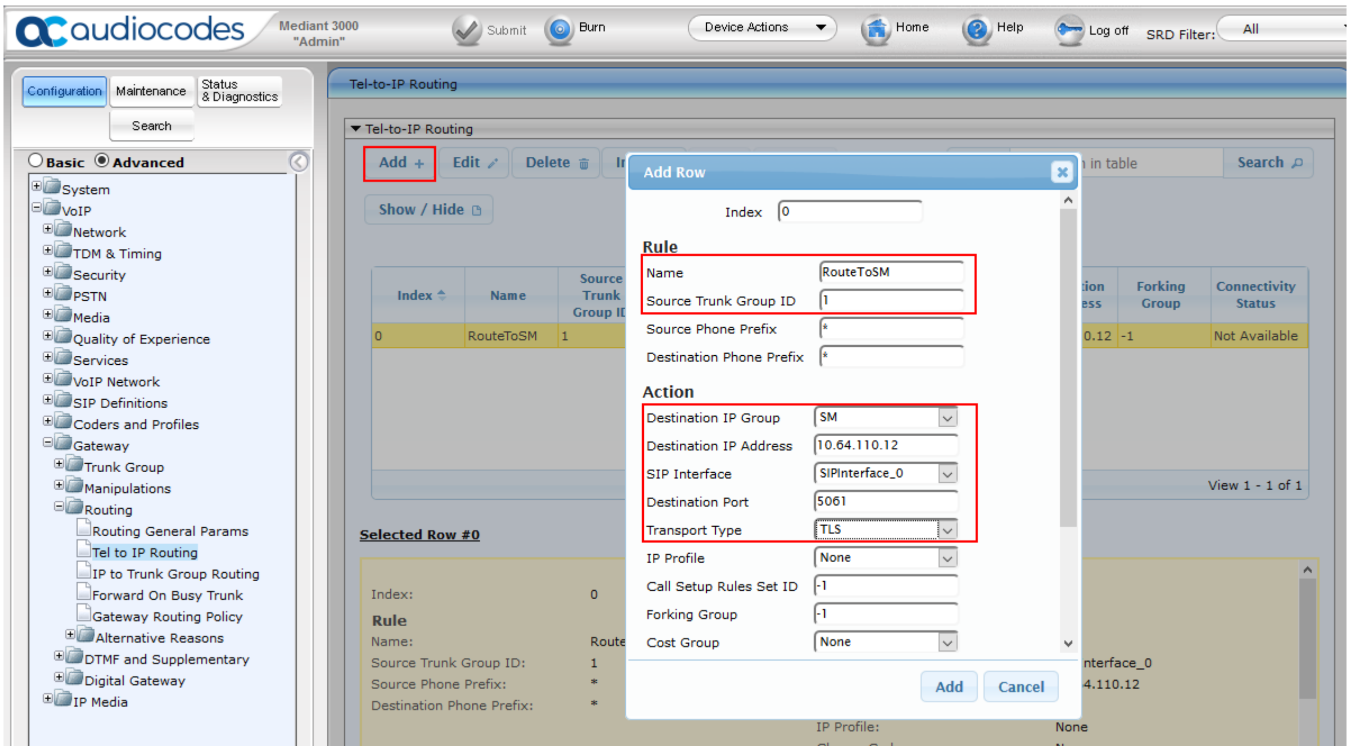
- **Action** Tab:
    - **Destination Type:** **Trunk Group**
    - **Trunk Group ID:** From **Section 7.6.4**
    - **Trunk ID:** From **Section 7.6.1**



7.7.2. Configure Outbound IP Routing Rules

To configure routing to Session Manager, navigate to **VoIP → Gateway → Routing → Tel to IP Routing**. Click the **Add** to add an entry and configure as follows:

- **Name:** Desired Name
- **Source Trunk Group ID:** From **Section 7.6.4**
- **Destination IP Group:** From **Section 7.5.3**
- **Destination IP Address:** SIP Signaling IP Address of Session Manager
- **Destination Port:** **5601**
- **Transport Type:** **TLS**

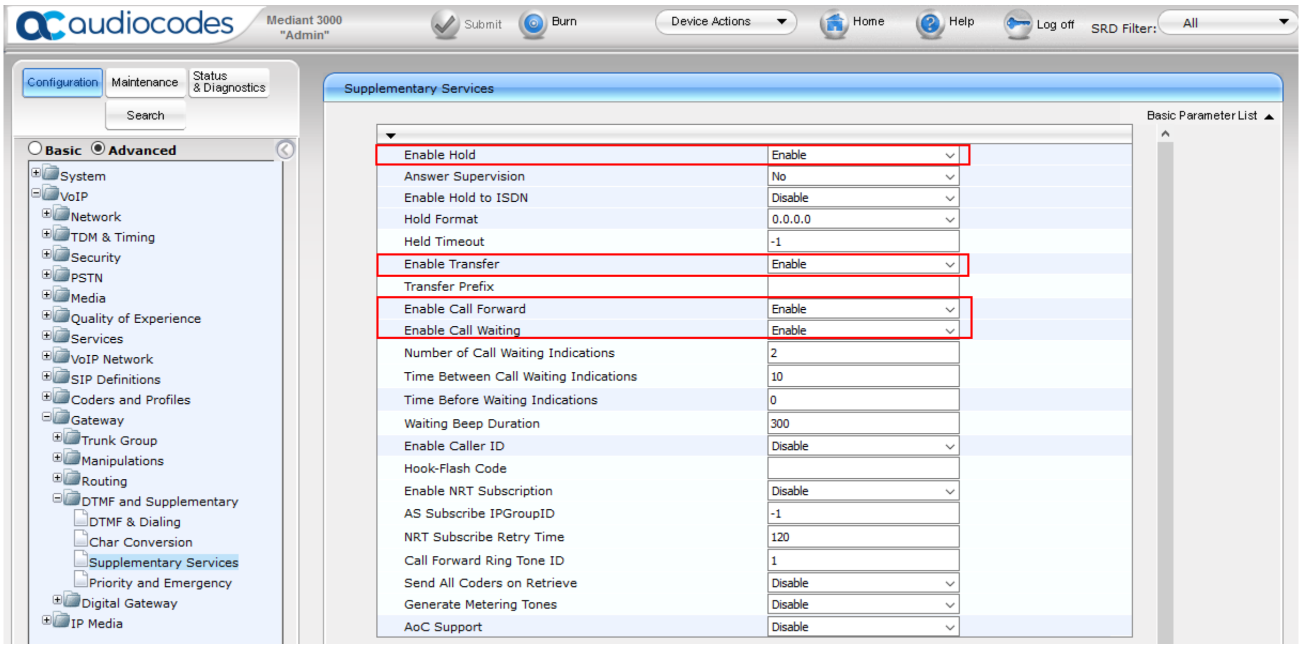


Configure Supplementary Services Parameters

Navigate to **VoIP → Gateway → DTMF and Supplementary → Supplementary Services**. Set the following parameters:

- **Enable Hold:** **Enable**
- **Enable Transfer:** **Enable**
- **Enable Call Forward:** **Enable**
- **Enable Call Waiting:** **Enable**

The screen below illustrates the **Supplementary Services** page.



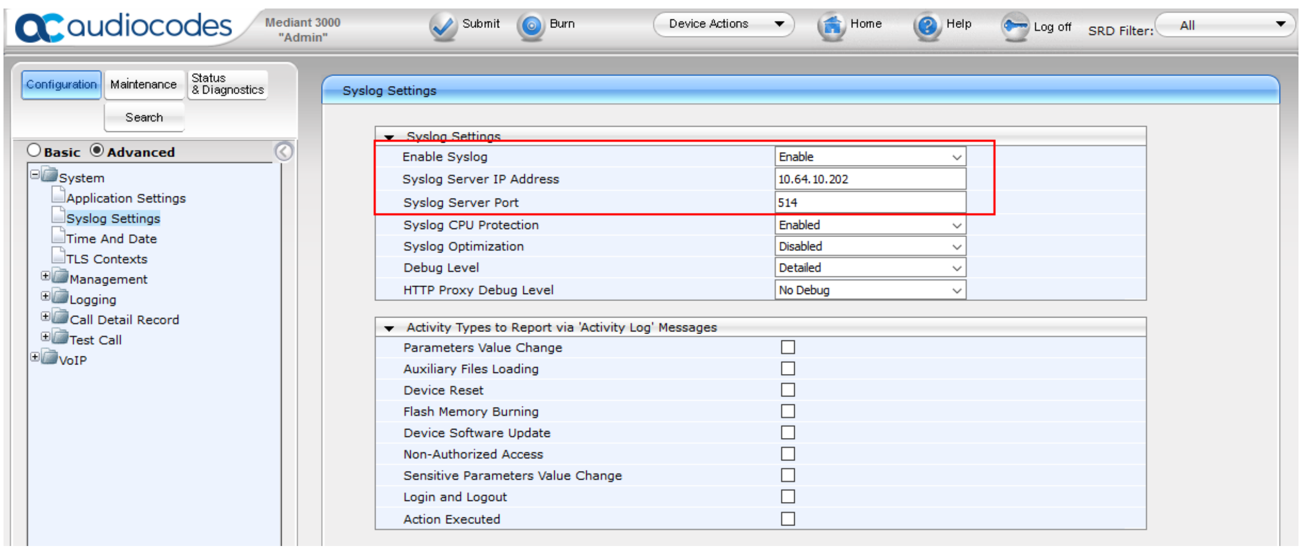
7.8. Configure Syslog Parameters for Debug Assistance

The Mediant 3000 Media Gateway can be configured to output logs to an external Syslog Server for debug assistance. To configure Syslog facility, open the **Syslog Settings** page (**Configuration** tab → **System** → **Sysylog Settings**). Configure the following settings:

- **Enable Syslog:** Set to **Enable**
- **Syslog Server IP Address:** Set to IP address of device running a Syslog Server Application **Syslog Server Port:** Set to port utilized on the Syslog Server listening device (i.e., **514**)
- **Debug Level:** Set to **Detailed** to capture proper level of debug information

Click the **Submit** button to save changes. The screen below illustrates settings used during compliance testing.

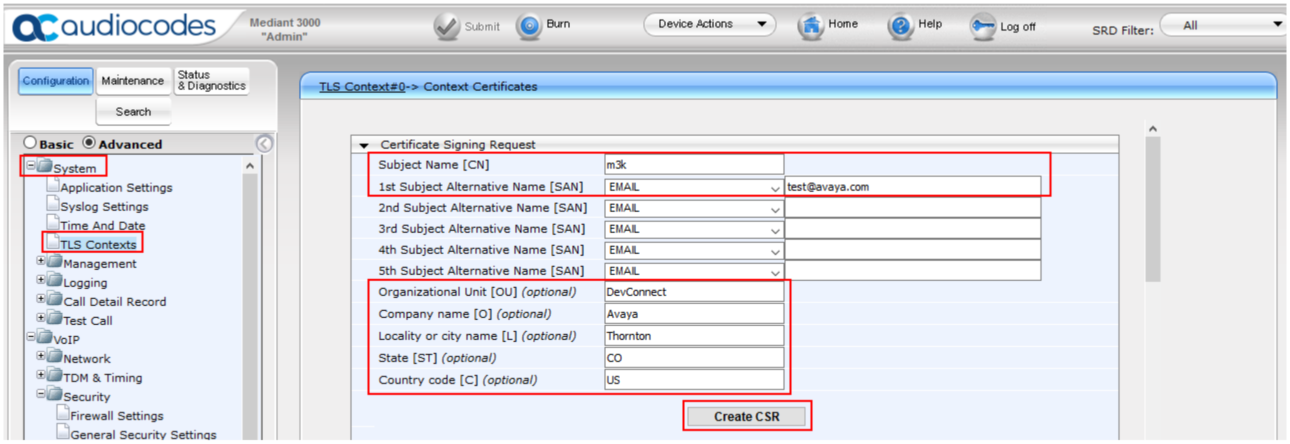
Note: The Syslog facility should be used only for Debugging purposes. **Enable** the Syslog service as needed and revert to **Disable** once troubleshooting is completed.



Note: *Once configuration of the Mediant 3000 is complete refer to Section 7.4 to save the configuration.*

7.9. Configure Certificates

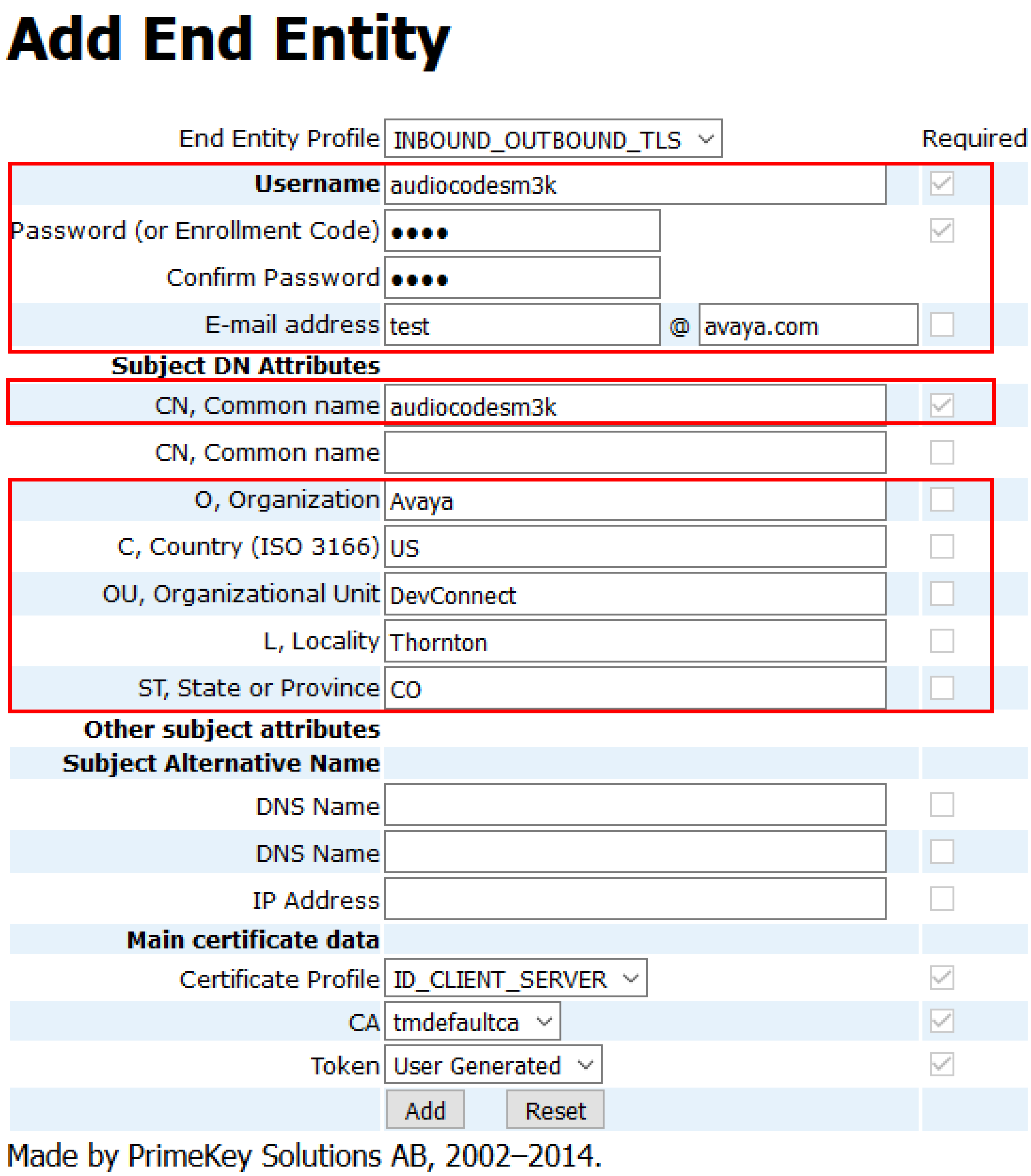
In order for TLS to successfully work, TLS contexts need to be configured. During the compliance testing, System Manager was used as the Certificate Authority for AudioCodes M3K. To configure the certificates, navigate to **System → TLS Contexts → TLS Context Certificates.** Below is an example of the fields configured for generating a CSR during the compliance testing.



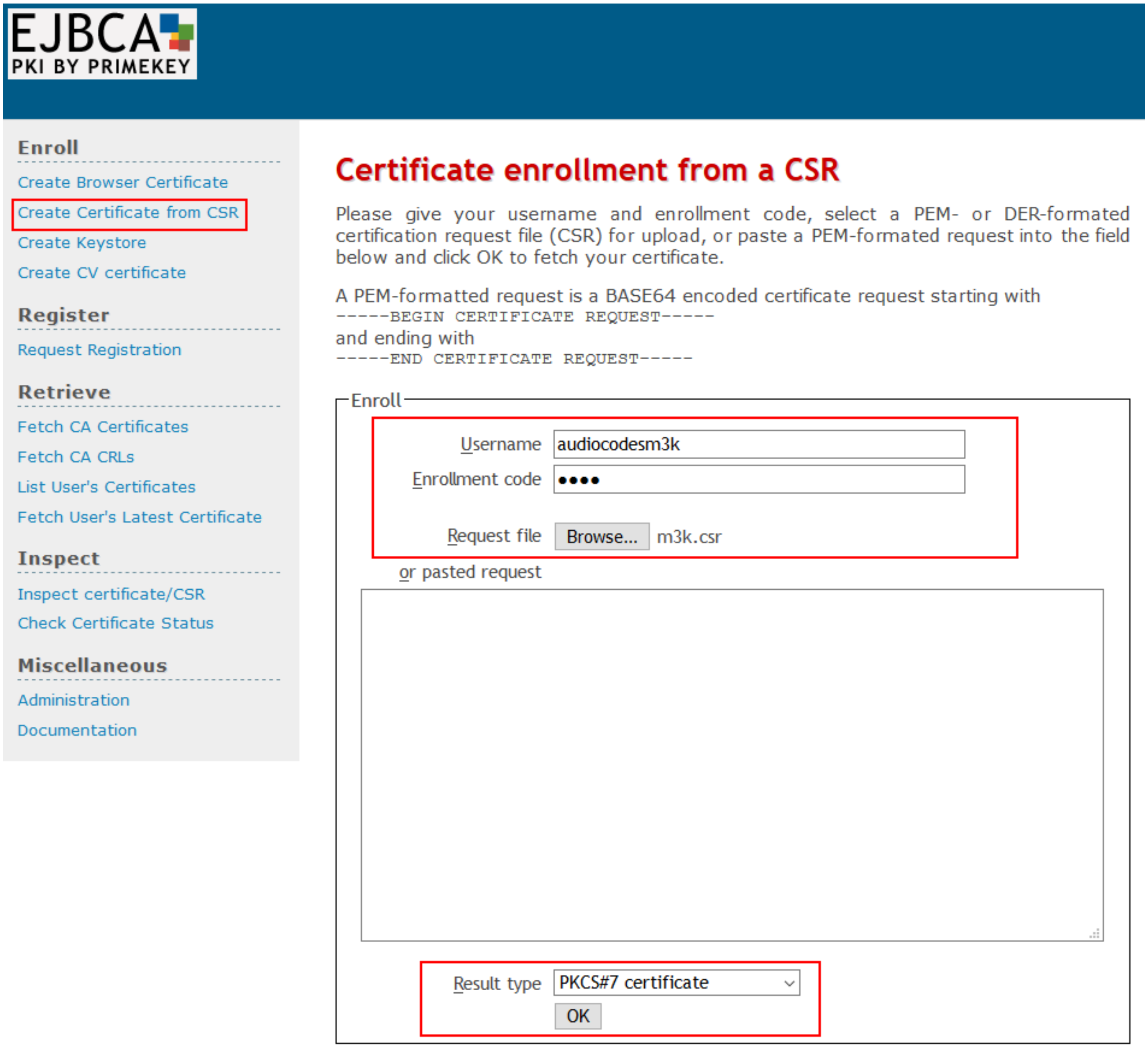
Select **Create CSR** and copy/paste the CSR on a notepad; save it on local PC.


```
-----BEGIN CERTIFICATE REQUEST-----
MIIBzDCCATUCAQMwYDEMMAoGA1UEAwwDbTNrMRMwEQYDVQQLDApEZXZDb25uZWN0
MQ4wDAYDVQQKDAVBdmF5YTERMA8GA1UEBwwIVGhvcm50b24xCzAJBgNVBAgMAkNP
MQswCQYDVQQGEwJVUzCBnzANBgkqhkiG9w0BAQEFAAOBjQAwgYkCgYEAn6onNEsJ
Qd+yxOO8hGUHXEAwKHZEr11mwn24awgtCjX4Ttn7VOQOu1N7AgpnE691nFwNB0tB
96Oat0t/chOKpPybzr1F37sVYNySd6VBsmj9tNyu0upVoGfVumLT3cgHk5XSTnSY
FsSb8tBLfls+fwgoQmEdoq70Exhp3SciIfMCAwEAAaAsMCoGCSqGSIb3DQEJDjEd
MBswGQYDVR0RBBIwEIEOdGVzdEBhdmF5YS5jb20wDQYJKoZIhvcNAQEFBQADgYEA
XoeKsxcnTnCSk2RfVzqcOhfBetqKd6x2aK/oRxcMbcXfvFYMU/kV3aDeoiL/lHkS
ZigKbH83QhE01ahaavUYrA1pboVdTflZrPlkouSZxJQBj6QDV9MpCF7+YrHJx51s
P1yjakKlZ9roX5+MvhVTHTLfR2ldwyaRHAiPnmZKOFQ=
-----END CERTIFICATE REQUEST-----
```

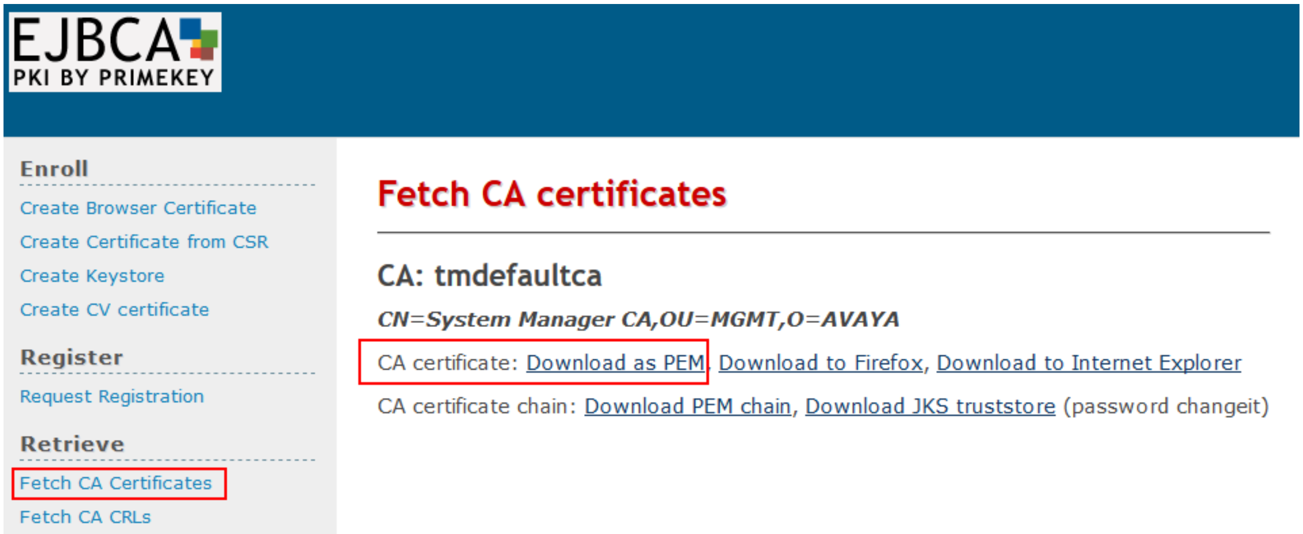
Via a browser, open the System Manager configuration utility and navigate to **Services → Security → Certificates → Authority → Add End Entity.** Below is an example of the fields configured during compliance testing. Select **Add** once done.



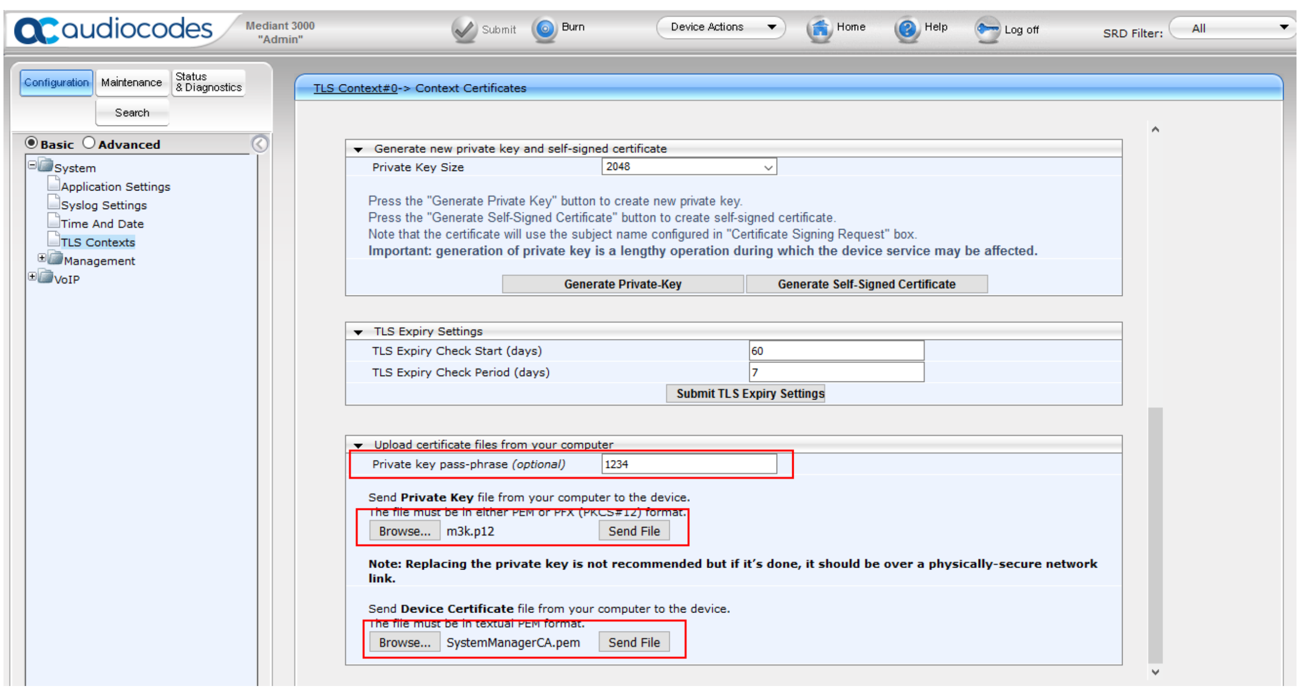
One the left pane select **Public Web (not shown)**. A new browser tab will open. On the left pane select **Generate Certificate from CSR.** Type in the **Username** and **Enrollment Code** from previous page, and browse to the CSR for AudioCodes M3K. Click **OK** and save the certificate to local PC (not shown).



On the left pane select **Fetch CA Certificates** and download the **CA Certificate** by selecting **Download as PEM.**



Return to the AudioCodes M3K webconsole and navigate to **System → TLS Contexts → TLS Context Certificate.** Scroll down to the bottom and select the generate p12 certificate file for AudioCodes M3K and Select the System Manager CA certificate. Type in the **Private key pass-phrase** (Enrollment code used during adding the Enmity on System Manager). Select **Send File** for each certificate.



8. Verification Steps

This section provides the verification steps that may be performed to verify the configuration.

8.1. Verify Avaya Aura® Communication Manager Trunk Status

On Communication Manager, ensure that all the signaling groups are in service by issuing the command status **signaling-group n** where **n** is the signaling group number.

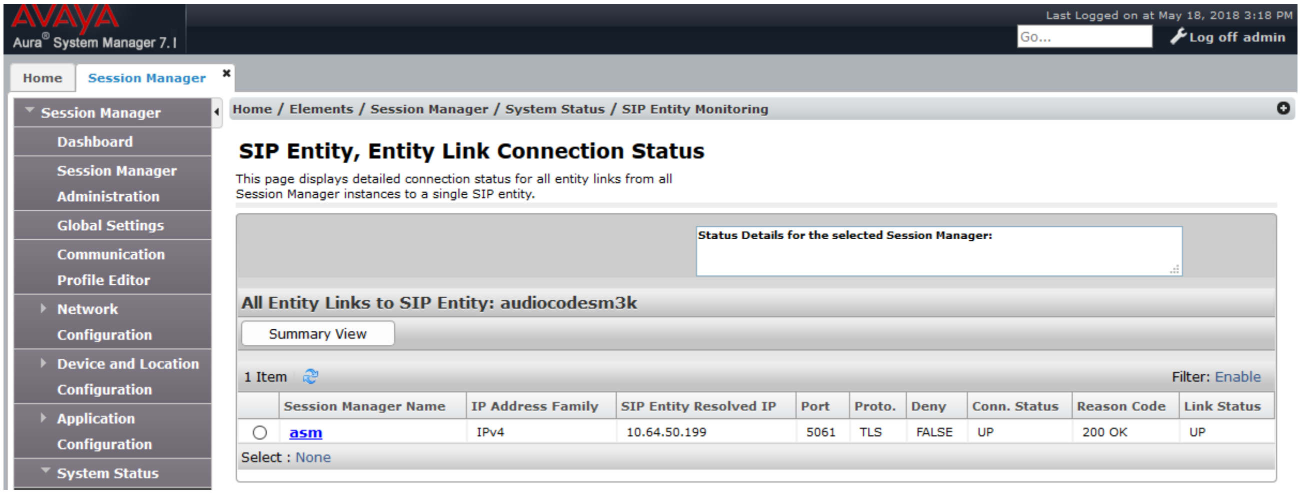
```
status signaling-group 1
                         STATUS
   SIGNALING GROUP

        Group ID: 1
      Group Type: sip

     Group State: in-service
```

8.2. SIP Monitoring on Avaya Aura® Session Manager

From System Manager's Home screen, navigate to **Elements →Session Manager→System Status→SIP Entity Monitoring**. Verify that none of the links to the defined SIP entities are down, indicating that they are all reachable for call routing. The screen below shows the link status between Session Manager and the Mediant 3000.

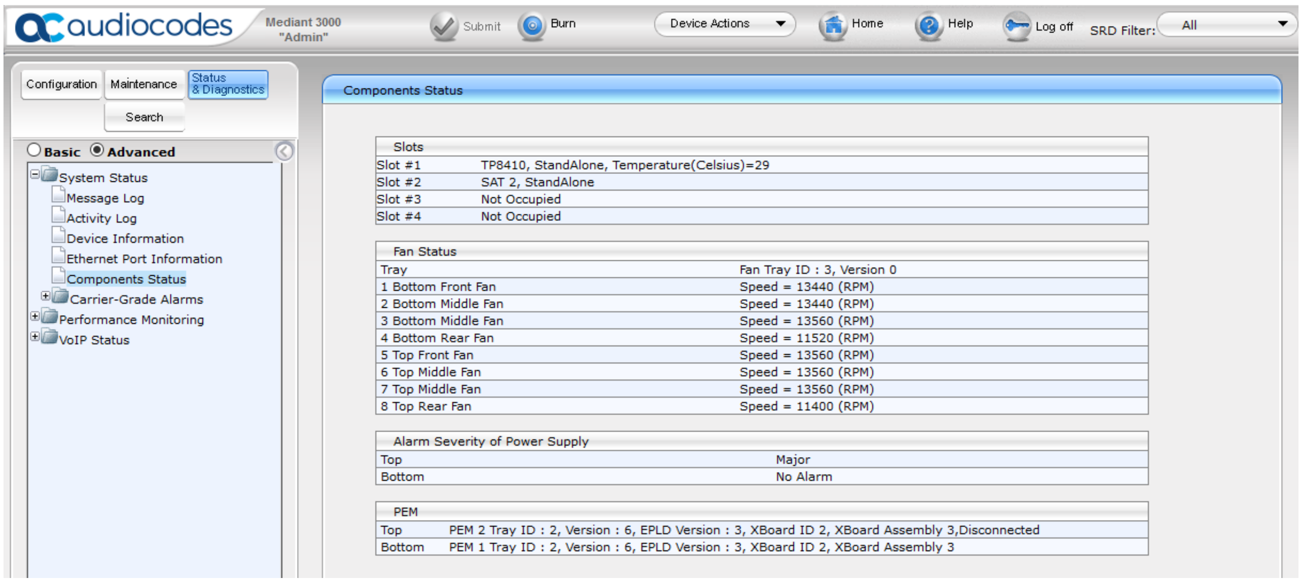


8.3. Utilizing Mediant 3000 Web Interface to Observe Status

The **Status & Diagnostics** menu is used to view and monitor the device's channels, Syslog messages, hardware and software product information, and to assess the device's statistics and IP connectivity information.

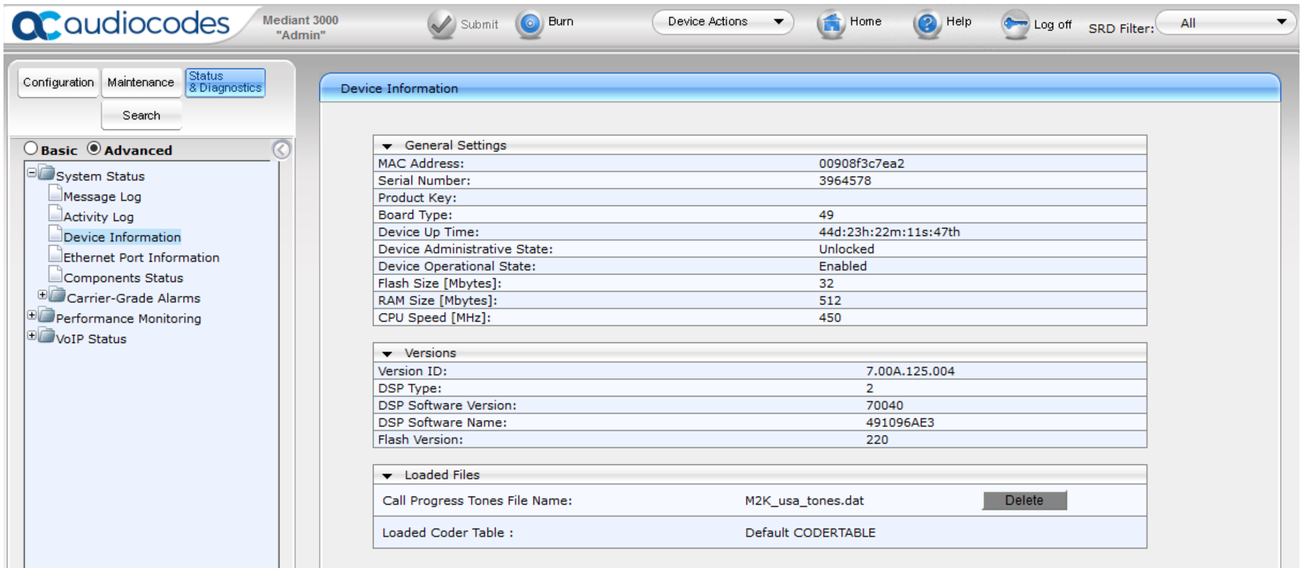
8.3.1. Device Status

To view the status of the device's hardware components, open the **Components Status** page (**Status & Diagnostics** tab → **System Status** → **Components Status**). The screen below illustrates the **Component Status** page for the gateway where the TP8410 board in slot 1 is active.



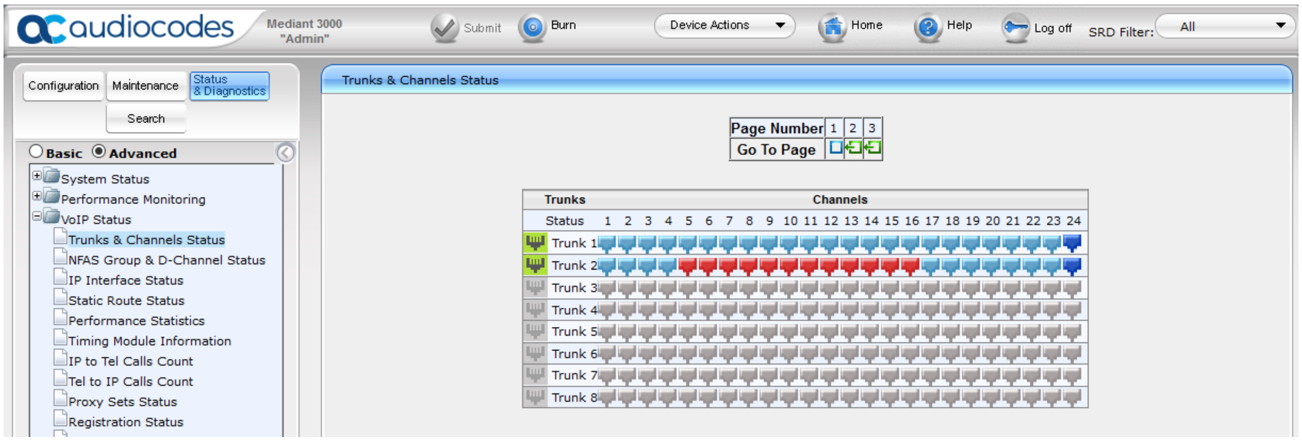
8.3.2. Device Information

Open the **Device Information** page (**Status & Diagnostics** tab → **System Status** → **Device Information**).



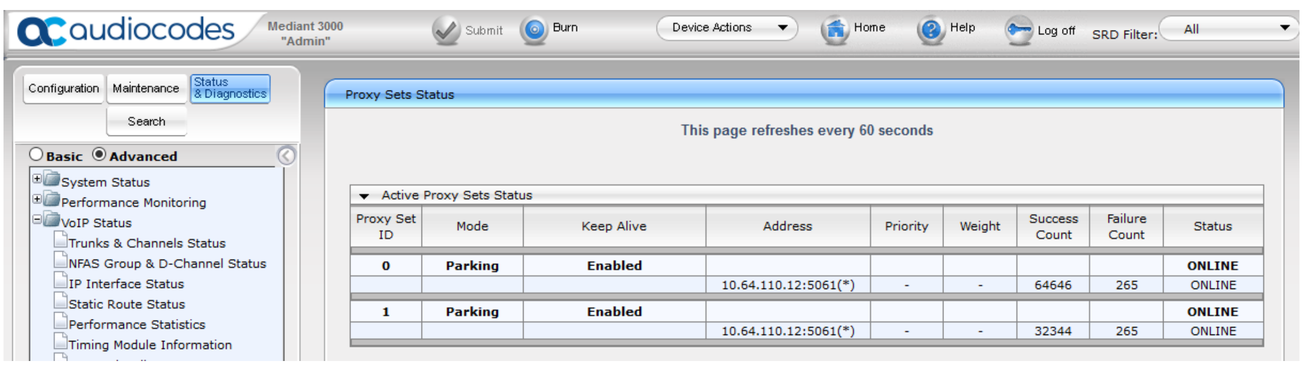
8.3.3. Trunks and Channels Status

To view the status of the device's trunks and the trunks' channels (Simulated PSTN), open the **Trunks & Channels Status** page (**Status & Diagnostics** tab → **VoIP Status** → **Trunks & Channels Status**). The following screen illustrates the **Trunks and Channel Status** page, where the symbols of the port in green represent channels engaged with a call.



8.3.4. Proxy Sets Status

To view the status of the SIP trunk to Session Manager, open the **Proxy Sets Status** page (**Status & Diagnostics** tab → **VoIP Status** → **Proxy Sets Status**). The following screen illustrates the **Proxy Sets Status** page; note the Status of ONLINE for Proxy Set ID 1.



9. Conclusion

These Application Notes describe the procedures required to configure the AudioCodes Mediant 3000 Gateway to interoperate with Avaya Aura® Session Manager and Avaya Aura® Communication Manager. The AudioCodes Mediant 3000 Gateway successfully passed compliance testing.

10. Additional References

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

[1] *Administering Avaya Aura® Communication Manager*, Document 03-300509
[2] *Administering Avaya Aura® Session Manager*, Document 03-603324
[3] *User's Manual Mediant™ 3000 Gateway & E-SBC* Version 7.0

Product documentation for Avaya products may be found at http://support.avaya.com.

Product documentation for AudioCodes products may be found at http://www.audiocodes.com.

11. Appendix

The AudioCodes M3000 .ini file was generated after completing the compliance test. Its contents were copied below. Please use it only for reference purposes.

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

;Board: Mediant 3000
;HW Board Type: 63  FK Board Type: 49
;M3K Board Type: TrunkPack 8410
;Serial Number: 3964578
;Slot Number: 1
;Software Version: 7.00A.125.004
;DSP Software Version: 491096AE3=> 700.40
;Board IP Address: 10.64.50.199
;Board Subnet Mask: 255.255.255.0
;Board Default Gateway: 10.64.50.1
;Ram size: 512M   Flash size: 32M
;Num of DSP Cores: 36  Num DSP Channels: 504
;Profile: NONE
;;;Key features:;Board Type: Mediant 3000 ;PSTN STM1\SONET Interface Not
Supported ;E1Trunks=16 ;T1Trunks=21 ;IP Media: VXML CALEA ;PSTN
Protocols: ISDN IUA=84 CAS V5.2 ;Channel Type: RTP DspCh=504 ;HA ;Coders:
G723 G729 GSM-FR G727 ILBC ;Security: IPSEC MediaEncryption
StrongEncryption EncryptControlProtocol ;DSP Voice features: IpmDetector
AMRPolicyManagement ;Control Protocols: MSFT MGCP MEGACO SIP ;Default
features:;Coders: G711 G726;
;----------------------------------------------


[SYSTEM Params]

;PM_gwSBCRegisteredUsers is hidden but has non-default value
SyslogServerIP = 10.64.10.202
EnableSyslog = 1
;VpFileLastUpdateTime is hidden but has non-default value
TLSPkeySize = 2048
NTPServerIP = '0.0.0.0'
;LastConfigChangeTime is hidden but has non-default value
;RootFileLastUpdateTime is hidden but has non-default value
;PkeyFileLastUpdateTime is hidden but has non-default value

[BSP Params]

PCMLawSelect = 3
TDMBusClockSource = 4
UdpPortSpacing = 10
EnterCpuOverloadPercent = 99
```

```
ExitCpuOverloadPercent = 95

[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

[PSTN Params]

ProtocolType_0 = 13
ProtocolType_1 = 13
ProtocolType_2 = 0
ProtocolType_3 = 0
ProtocolType_4 = 0
ProtocolType_5 = 0
ProtocolType_6 = 0
ProtocolType_7 = 0
ProtocolType_8 = 0
ProtocolType_9 = 0
ProtocolType_10 = 0
ProtocolType_11 = 0
ProtocolType_12 = 0
ProtocolType_13 = 0
ProtocolType_14 = 0
ProtocolType_15 = 0
ProtocolType_16 = 0
ProtocolType_17 = 0
ProtocolType_18 = 0
ProtocolType_19 = 0
ProtocolType_20 = 0
FramingMethod_0 = D
FramingMethod_1 = D
FramingMethod_2 = 0
FramingMethod_3 = 0
FramingMethod_4 = 0
FramingMethod_5 = 0
FramingMethod_6 = 0
FramingMethod_7 = 0
FramingMethod_8 = 0
FramingMethod_9 = 0
FramingMethod_10 = 0
FramingMethod_11 = 0
```

```
FramingMethod_12 = 0
FramingMethod_13 = 0
FramingMethod_14 = 0
FramingMethod_15 = 0
FramingMethod_16 = 0
FramingMethod_17 = 0
FramingMethod_18 = 0
FramingMethod_19 = 0
FramingMethod_20 = 0

[SS7 Params]


[Voice Engine Params]

CallProgressTonesFilename = 'M2K_usa_tones.dat'
IdlePCMPattern = 85
AnswerDetectorSilenceTime = 0
AnswerDetectorSensitivity = 0
EnergyDetectorQualityFactor = 0
EnergyDetectorThreshold = 0
ENABLEMEDIASECURITY = 1
RTCPEncryptionDisableTx = 1

[WEB Params]

;HTTPSPkeyFileName is hidden but has non-default value

[SIP Params]

SIPDESTINATIONPORT = 5061
GWDEBUGLEVEL = 5
;ISPRACKREQUIRED is hidden but has non-default value
SIPGATEWAYNAME = 'avaya.com'
USEGATEWAYNAMEFOROPTIONS = 1
ISFAXUSED = 1
SIPTRANSPORTTYPE = 2
ENABLESIPS = 1
MEDIASECURITYBEHAVIOUR = 1
ENABLETCPCONNECTIONREUSE = 0
MSLDAPPRIMARYKEY = 'telephoneNumber'
FIRSTTXDTMFOPTION = 4
;GWAPPCONFIGURATIONVERSION is hidden but has non-default value

[SCTP Params]


[VXML Params]


[IPsec Params]
```

```
[Audio Staging Params]


[SNMP Params]

;ContextEngineID is hidden but has non-default value

[Video 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, 10.64.50.199, 24, 10.64.50.1, 1, "if 0",
0.0.0.0, 0.0.0.0;

[ \InterfaceTable ]


[ DspTemplates ]

FORMAT DspTemplates_Index = DspTemplates_DspTemplateNumber,
DspTemplates_DspResourcesPercentage;
DspTemplates 0 = 0, 100;

[ \DspTemplates ]


[ WebUsers ]

FORMAT WebUsers_Index = WebUsers_Username, WebUsers_Password,
WebUsers_Status, WebUsers_PwAgeInterval, WebUsers_SessionLimit,
WebUsers_SessionTimeout, WebUsers_BlockTime, WebUsers_UserLevel,
WebUsers_PwNonce;
WebUsers 0 = "Admin",
"$1$YQVaV1dWB1JaD1pcXwkKDEAUQUpHFkVCQE9DQ09JGk21t7rlsbyysr256+66uO3tqaX0q
qT0866hrPur/qmoqcU=", 1, 0, 5, 15, 60, 200,
"39ea427ac3a5abe249eb8e0e3bb18e26";
WebUsers 1 = "User",
"$1$Wj44aG9rbVFTV1IHVVVTXgxZWFUMWAwUF0UREhAXThlKTk4YRBhI5eK2s+Hhsb+6vem8t
bjqvvGjo6eg8fei+fo=", 1, 0, 2, 15, 60, 50,
"bb9a70129d690ca6545321ae6d4e1999";

[ \WebUsers ]
```

```
[ TLSContexts ]

FORMAT TLSContexts_Index = TLSContexts_Name, TLSContexts_TLSVersion,
TLSContexts_DTLSVersion, TLSContexts_ServerCipherString,
TLSContexts_ClientCipherString, TLSContexts_OcspEnable,
TLSContexts_OcspServerPrimary, TLSContexts_OcspServerSecondary,
TLSContexts_OcspServerPort, TLSContexts_OcspDefaultResponse,
TLSContexts_DHKeySize;
TLSContexts 0 = "default", 4, 2, "RC4:AES128", "DEFAULT", 0, 0.0.0.0,
0.0.0.0, 2560, 0, 2048;

[ \TLSContexts ]


[ CpMediaRealm ]

FORMAT CpMediaRealm_Index = CpMediaRealm_MediaRealmName,
CpMediaRealm_IPv4IF, CpMediaRealm_IPv6IF, CpMediaRealm_PortRangeStart,
CpMediaRealm_MediaSessionLeg, CpMediaRealm_PortRangeEnd,
CpMediaRealm_IsDefault, CpMediaRealm_QoeProfile, CpMediaRealm_BWProfile;
CpMediaRealm 0 = "DefaultRealm", "if 0", "", 6000, 2016, 26159, 1, "",
"";

[ \CpMediaRealm ]


[ SBCRoutingPolicy ]

FORMAT SBCRoutingPolicy_Index = SBCRoutingPolicy_Name,
SBCRoutingPolicy_LCREnable, SBCRoutingPolicy_LCRAverageCallLength,
SBCRoutingPolicy_LCRDefaultCost, SBCRoutingPolicy_LdapServerGroupName;
SBCRoutingPolicy 0 = "Default_SBCRoutingPolicy", 0, 1, 0, "";

[ \SBCRoutingPolicy ]


[ SRD ]

FORMAT SRD_Index = SRD_Name, SRD_BlockUnRegUsers, SRD_MaxNumOfRegUsers,
SRD_EnableUnAuthenticatedRegistrations, SRD_SharingPolicy,
SRD_UsedByRoutingServer, SRD_SBCOperationMode, SRD_SBCRoutingPolicyName,
SRD_SBCDialPlanName;
SRD 0 = "DefaultSRD", 0, -1, 1, 0, 0, 0, "Default_SBCRoutingPolicy", "";

[ \SRD ]


[ SIPInterface ]

FORMAT SIPInterface_Index = SIPInterface_InterfaceName,
SIPInterface_NetworkInterface, SIPInterface_ApplicationType,
SIPInterface_UDPPort, SIPInterface_TCPPort, SIPInterface_TLSPort,
```

```
SIPInterface_SRDName, SIPInterface_MessagePolicyName,
SIPInterface_TLSContext, SIPInterface_TLSMutualAuthentication,
SIPInterface_TCPKeepaliveEnable,
SIPInterface_ClassificationFailureResponseType,
SIPInterface_PreClassificationManSet, SIPInterface_EncapsulatingProtocol,
SIPInterface_MediaRealm, SIPInterface_SBCDirectMedia,
SIPInterface_BlockUnRegUsers, SIPInterface_MaxNumOfRegUsers,
SIPInterface_EnableUnAuthenticatedRegistrations,
SIPInterface_UsedByRoutingServer;
SIPInterface 0 = "SIPInterface_0", "if 0", 0, 5060, 5060, 5061,
"DefaultSRD", "", "default", -1, 0, 500, -1, 0, "DefaultRealm", 0, -1, -
1, -1, 0;

[ \SIPInterface ]


[ ProxySet ]

FORMAT ProxySet_Index = ProxySet_ProxyName,
ProxySet_EnableProxyKeepAlive, ProxySet_ProxyKeepAliveTime,
ProxySet_ProxyLoadBalancingMethod, ProxySet_IsProxyHotSwap,
ProxySet_SRDName, ProxySet_ClassificationInput, ProxySet_TLSContextName,
ProxySet_ProxyRedundancyMode, ProxySet_DNSResolveMethod,
ProxySet_KeepAliveFailureResp, ProxySet_GWIPv4SIPInterfaceName,
ProxySet_SBCIPv4SIPInterfaceName, ProxySet_SASIPv4SIPInterfaceName,
ProxySet_GWIPv6SIPInterfaceName, ProxySet_SBCIPv6SIPInterfaceName,
ProxySet_SASIPv6SIPInterfaceName, ProxySet_MinActiveServersLB,
ProxySet_SuccessDetectionRetries, ProxySet_SuccessDetectionInterval,
ProxySet_FailureDetectionRetransmissions;
ProxySet 0 = "ProxySet_0", 1, 60, 0, 0, "DefaultSRD", 0, "default", -1, -
1, "", "SIPInterface_0", "", "", "", "", "", 1, 1, 10, -1;
ProxySet 1 = "AvayaSM", 1, 120, 0, 0, "DefaultSRD", 0, "", -1, -1, "",
"SIPInterface_0", "", "", "", "", "", 1, 1, 10, -1;

[ \ProxySet ]


[ IPGroup ]

FORMAT IPGroup_Index = IPGroup_Type, IPGroup_Name, IPGroup_ProxySetName,
IPGroup_SIPGroupName, IPGroup_ContactUser, IPGroup_SipReRoutingMode,
IPGroup_AlwaysUseRouteTable, IPGroup_SRDName, IPGroup_MediaRealm,
IPGroup_ClassifyByProxySet, IPGroup_ProfileName,
IPGroup_MaxNumOfRegUsers, IPGroup_InboundManSet, IPGroup_OutboundManSet,
IPGroup_RegistrationMode, IPGroup_AuthenticationMode, IPGroup_MethodList,
IPGroup_EnableSBCClientForking, IPGroup_SourceUriInput,
IPGroup_DestUriInput, IPGroup_ContactName, IPGroup_Username,
IPGroup_Password, IPGroup_UUIFormat, IPGroup_QOEProfile,
IPGroup_BWProfile, IPGroup_MediaEnhancementProfile,
IPGroup_AlwaysUseSourceAddr, IPGroup_MsgManUserDef1,
IPGroup_MsgManUserDef2, IPGroup_SIPConnect, IPGroup_SBCPSAPMode,
IPGroup_DTLSContext, IPGroup_CreatedByRoutingServer,
```

```
IPGroup_UsedByRoutingServer, IPGroup_SBCOperationMode,
IPGroup_SBCRouteUsingRequestURIPort, IPGroup_SBCKeepOriginalCallID,
IPGroup_SBCDialPlanName;
IPGroup 0 = 0, "Default_IPG", "ProxySet_0", "", "", -1, 0, "DefaultSRD",
"", 0, "", -1, -1, -1, 0, 0, "", 0, -1, -1, "", "", "$1$gQ==", 0, "", "",
"", 0, "", "", 0, 0, "default", 0, 0, -1, 0, 0, "";
IPGroup 1 = 0, "SM", "AvayaSM", "10.64.110.12", "", -1, 0, "DefaultSRD",
"DefaultRealm", 1, "", -1, -1, -1, 0, 0, "", 0, -1, -1, "", "",
"$1$gQ==", 0, "", "", "", 0, "", "", 0, 0, "default", 0, 0, -1, 0, 0, "";

[ \IPGroup ]


[ PREFIX ]

FORMAT PREFIX_Index = PREFIX_RouteName, PREFIX_DestinationPrefix,
PREFIX_DestAddress, PREFIX_SourcePrefix, PREFIX_ProfileName,
PREFIX_MeteringCodeName, PREFIX_DestPort, PREFIX_DestIPGroupName,
PREFIX_TransportType, PREFIX_SrcTrunkGroupID,
PREFIX_DestSIPInterfaceName, PREFIX_CostGroup, PREFIX_ForkingGroup,
PREFIX_CallSetupRulesSetId, PREFIX_ConnectivityStatus;
PREFIX 0 = "RouteToSM", "*", "10.64.110.12", "*", "", "", 5061, "SM", 2,
1, "SIPInterface_0", "", -1, -1, "Not Available";

[ \PREFIX ]


[ TrunkGroup ]

FORMAT TrunkGroup_Index = TrunkGroup_TrunkGroupNum,
TrunkGroup_FirstTrunkId, TrunkGroup_FirstBChannel,
TrunkGroup_LastBChannel, TrunkGroup_FirstPhoneNumber,
TrunkGroup_ProfileName, TrunkGroup_LastTrunkId, TrunkGroup_Module;
TrunkGroup 0 = 1, 0, 1, 24, "", "", 0, 255;

[ \TrunkGroup ]


[ PstnPrefix ]

FORMAT PstnPrefix_Index = PstnPrefix_RouteName, PstnPrefix_DestPrefix,
PstnPrefix_TrunkGroupId, PstnPrefix_SourcePrefix,
PstnPrefix_SourceAddress, PstnPrefix_ProfileName,
PstnPrefix_SrcIPGroupName, PstnPrefix_DestHostPrefix,
PstnPrefix_SrcHostPrefix, PstnPrefix_SrcSIPInterfaceName,
PstnPrefix_TrunkId, PstnPrefix_CallSetupRulesSetId, PstnPrefix_DestType;
PstnPrefix 0 = "RouteToPSTN", "*", 1, "*", "*", "", "SM", "*", "*",
"SIPInterface_0", 1, -1, 0;

[ \PstnPrefix ]
```

```
[ ProxyIp ]

FORMAT ProxyIp_Index = ProxyIp_ProxySetId, ProxyIp_ProxyIpIndex,
ProxyIp_IpAddress, ProxyIp_TransportType;
ProxyIp 0 = "0", 0, "10.64.110.12", 2;
ProxyIp 1 = "1", 0, "10.64.110.12", -1;

[ \ProxyIp ]


[ TrunkGroupSettings ]

FORMAT TrunkGroupSettings_Index = TrunkGroupSettings_TrunkGroupId,
TrunkGroupSettings_ChannelSelectMode,
TrunkGroupSettings_RegistrationMode, TrunkGroupSettings_GatewayName,
TrunkGroupSettings_ContactUser, TrunkGroupSettings_ServingIPGroupName,
TrunkGroupSettings_MWIInterrogationType,
TrunkGroupSettings_TrunkGroupName,
TrunkGroupSettings_UsedByRoutingServer, TrunkGroupSettings_AdminState;
TrunkGroupSettings 1 = 1, 3, 5, "", "", "SM", 255, "", 0, 0;

[ \TrunkGroupSettings ]


[ CodersGroup0 ]

FORMAT CodersGroup0_Index = CodersGroup0_Name, CodersGroup0_pTime,
CodersGroup0_rate, CodersGroup0_PayloadType, CodersGroup0_Sce,
CodersGroup0_CoderSpecific;
CodersGroup0 0 = "g729", 20, 0, -1, 0, "";
CodersGroup0 1 = "g711Alaw64k", 20, 0, -1, 0, "";
CodersGroup0 2 = "g711Ulaw64k", 20, 0, -1, 0, "";
CodersGroup0 3 = "t38fax", 255, 255, -1, 255, "";

[ \CodersGroup0 ]


[ GwRoutingPolicy ]

FORMAT GwRoutingPolicy_Index = GwRoutingPolicy_Name,
GwRoutingPolicy_LCREnable, GwRoutingPolicy_LCRAverageCallLength,
GwRoutingPolicy_LCRDefaultCost, GwRoutingPolicy_LdapServerGroupName;
GwRoutingPolicy 0 = "GwRoutingPolicy", 0, 1, 0, "";

[ \GwRoutingPolicy ]


[ ResourcePriorityNetworkDomains ]

FORMAT ResourcePriorityNetworkDomains_Index =
ResourcePriorityNetworkDomains_Name,
ResourcePriorityNetworkDomains_Ip2TelInterworking;
```

```
ResourcePriorityNetworkDomains 1 = "dsn", 1;
ResourcePriorityNetworkDomains 2 = "dod", 1;
ResourcePriorityNetworkDomains 3 = "drsn", 1;
ResourcePriorityNetworkDomains 5 = "uc", 1;
ResourcePriorityNetworkDomains 7 = "cuc", 1;

[ \ResourcePriorityNetworkDomains ]
```

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.