Avaya Solution & Interoperability Test Lab

Application Notes for Configuring Avaya Aura® Session Manager and Avaya Aura® Communication Manager with AudioCodes Mediant 3000 Gateway for E1 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 E1 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 and Avaya Aura® Communication Manager using SIP trunking along with E1 access to a simulated PSTN.

These Application Notes present a sample configuration for an enterprise network consisting of Avaya Aura® Session Manager and Avaya Aura® Communication Manager, integrated with an AudioCodes Mediant 3000 Gateway using SIP and providing E1 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 E1 access were verified in this compliance test.

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.

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.

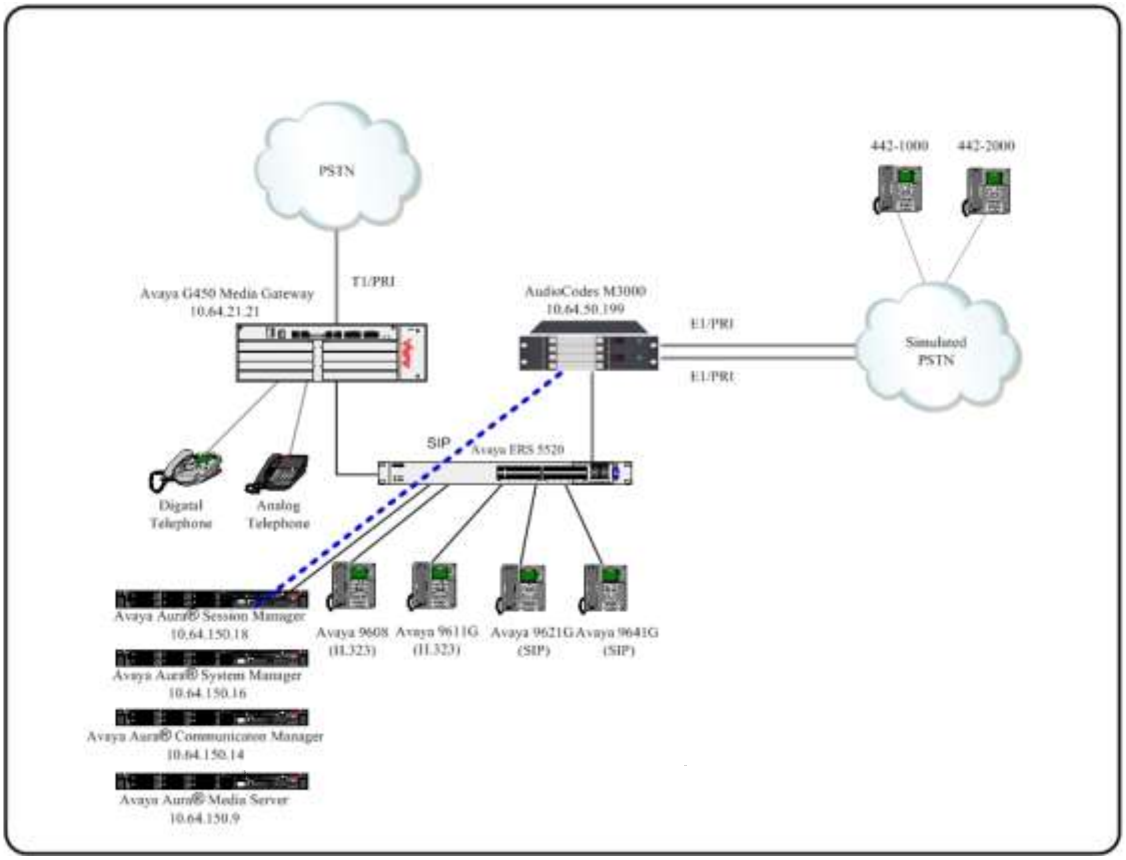


Figure 1: Compliance Test Reference Configuration

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	7.0 SP3
Avaya Aura® Session Manager in a Virtual Environment	7.0 SP2
Avaya Aura® System Manager in a Virtual Environment	7.0 SP2
Avaya Aura® Media Server in a Virtual Environment	7.7.0.226
Avaya 96x1 Deskphone	SIP 7.0, H.323 6.6
Avaya 6211 and 6221 Analog Phone	-
AudioCodes Mediant 3000	7.00A.049.003

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 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  11
                                OPTIONAL FEATURES

IP PORT CAPACITIES                                              USED
                     Maximum Administered H.323 Trunks: 4000  10
           Maximum Concurrently Registered IP Stations: 2400  1
             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  2
                   Maximum Video Capable IP Softphones: 2400  4
                       Maximum Administered SIP Trunks: 4000  22
  Maximum Administered Ad-hoc Video Conferencing Ports: 4000  0
   Maximum Number of DS1 Boards with Echo Cancellation: 80    0
                             Maximum TN2501 VAL Boards: 10    0
                     Maximum Media Gateway VAL Sources: 50    1
           Maximum TN2602 Boards with 80 VoIP Channels: 128   0
          Maximum TN2602 Boards with 320 VoIP Channels: 128   0
   Maximum Number of Expanded Meet-me Conference Ports: 300   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  20
                            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 **10.64.150.14** are entered as **Name** and **IP Address** for the signaling in Communication Manager running in a virtual environment. In addition, **sm15018** and **10.64.150.18** are entered for Session Manager.

```
change node-names ip                                              Page   1 of   2
                                  IP NODE NAMES
    Name                IP Address
procr                 10.64.150.14
sm15018               10.64.150.18
```

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
Location: 1       Authoritative Domain: avaya.com
    Name: CM Local              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: 40000                     IP Audio Hairpinning? y
   UDP Port Max: 65535
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.

Retain the default values for the remaining fields.

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


     Media Encryption
 1: none
 2:
 3:
```

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., **sm15018**
- **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 2                                              Page   1 of   2
                                SIGNALING GROUP

 Group Number: 2              Group Type: sip
  IMS Enabled? n        Transport Method: tls
        Q-SIP? n
     IP Video? y                                  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: sm15018
 Near-end Listen Port: 5061                  Far-end Listen Port: 5061
                                          Far-end Network Region: 1

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

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.**, sm15018**)
- **TAC:** An available trunk access code (i.e., **102**)
- **Service Type:** **tie**
- **Signaling Group:** The number of the signaling group associated (i.e., **2**)
- **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 2                                                Page   1 of  21
                                TRUNK GROUP

Group Number: 2                   Group Type: sip           CDR Reports: y
  Group Name: sm15018                    COR: 1        TN: 1        TAC: 102
   Direction: two-way        Outgoing Display? n
 Dial Access? n                                   Night Service:
Queue Length: 0
Service Type: tie                  Auth Code? n
                                          Member Assignment Method: auto
                                                   Signaling Group: 2
                                                 Number of Members: 20
```

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

```
add trunk-group 2                                                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., **sm15018**)
- **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
- **No. Del Dgts:** **0** was entered to delete zero digits
- **Numbering Format:** This was set to **lev0-pvt** in the tested configuration

```
change route-pattern 2                                          Page   1 of   3
                    Pattern Number: 2      Pattern Name: sm15018
    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: 2    0                    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: 1
                                                           Maximum Entries: 540
```

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 2** (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      2        lev0           n
```

5.9. Administer ARS Analysis

This section provides sample Auto Route Selection (ARS) used for routing calls with dialed digits beginning with **1303442** which correspond to numbers accessible via the Mediant 3000. Use the **change ars analysis 1303442** 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 **11**
- **Total Max:** Maximum number of digits, in this case **11**
- **Route Pattern:** The route pattern number from **Section 5.6**, i.e., **2**
- **Call Type:** hnpa

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

```
change ars analysis 1303442                                Page   1 of   2
                             ARS DIGIT ANALYSIS TABLE
                                   Location: all          Percent Full: 2

          Dialed           Total     Route     Call   Node   ANI
          String          Min  Max  Pattern    Type   Num    Reqd
    1303442                11   11     2       hnpa
```

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

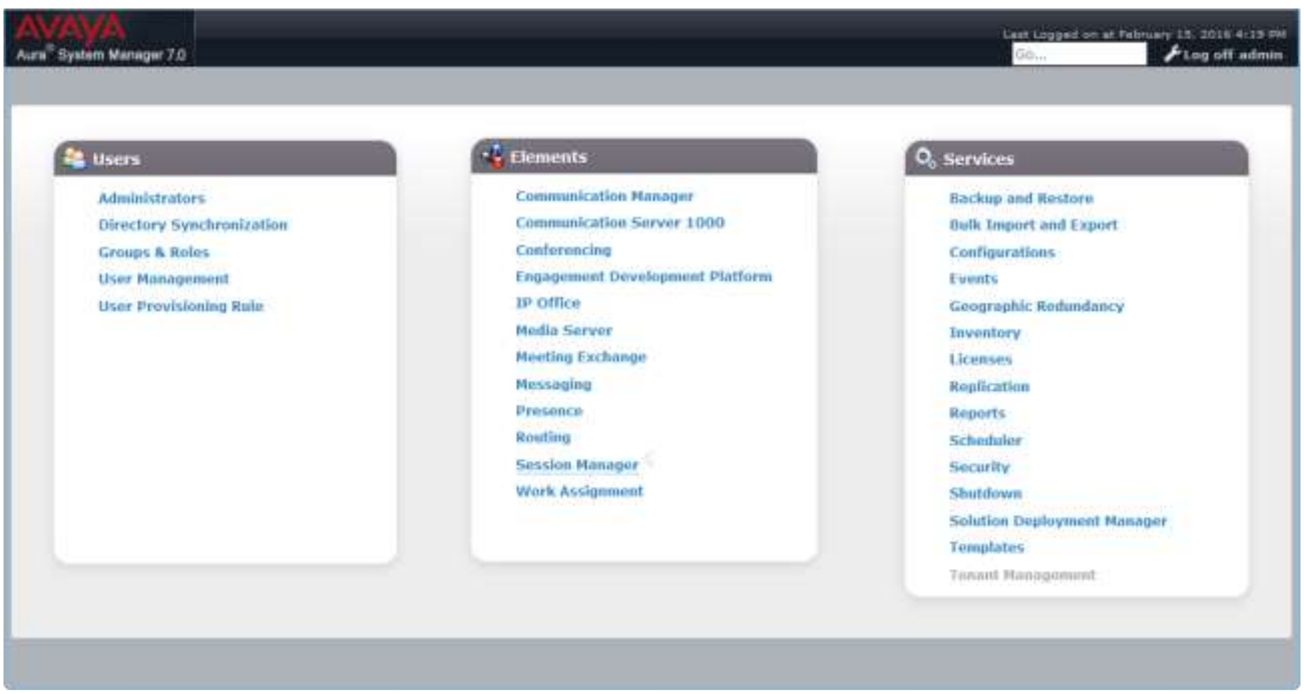
Use the **save translation** command to save all changes.

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

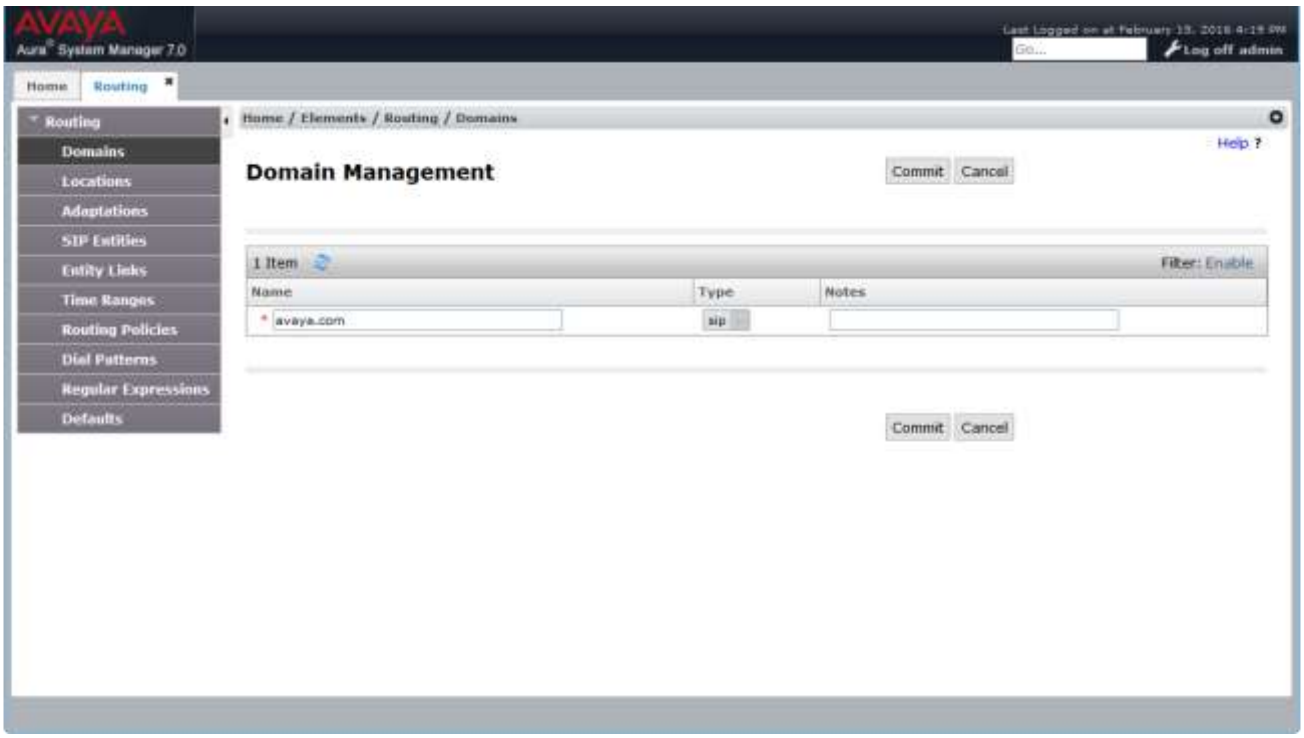
Configuration is accomplished by accessing the browser-based GUI of System Manager, using the URL **http://<ip-address>/SMGR**, where **<ip-address>** is the IP address of System Manager. Log in with the appropriate credentials. The home screen as shown below is displayed. Expand the **Routing** Link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

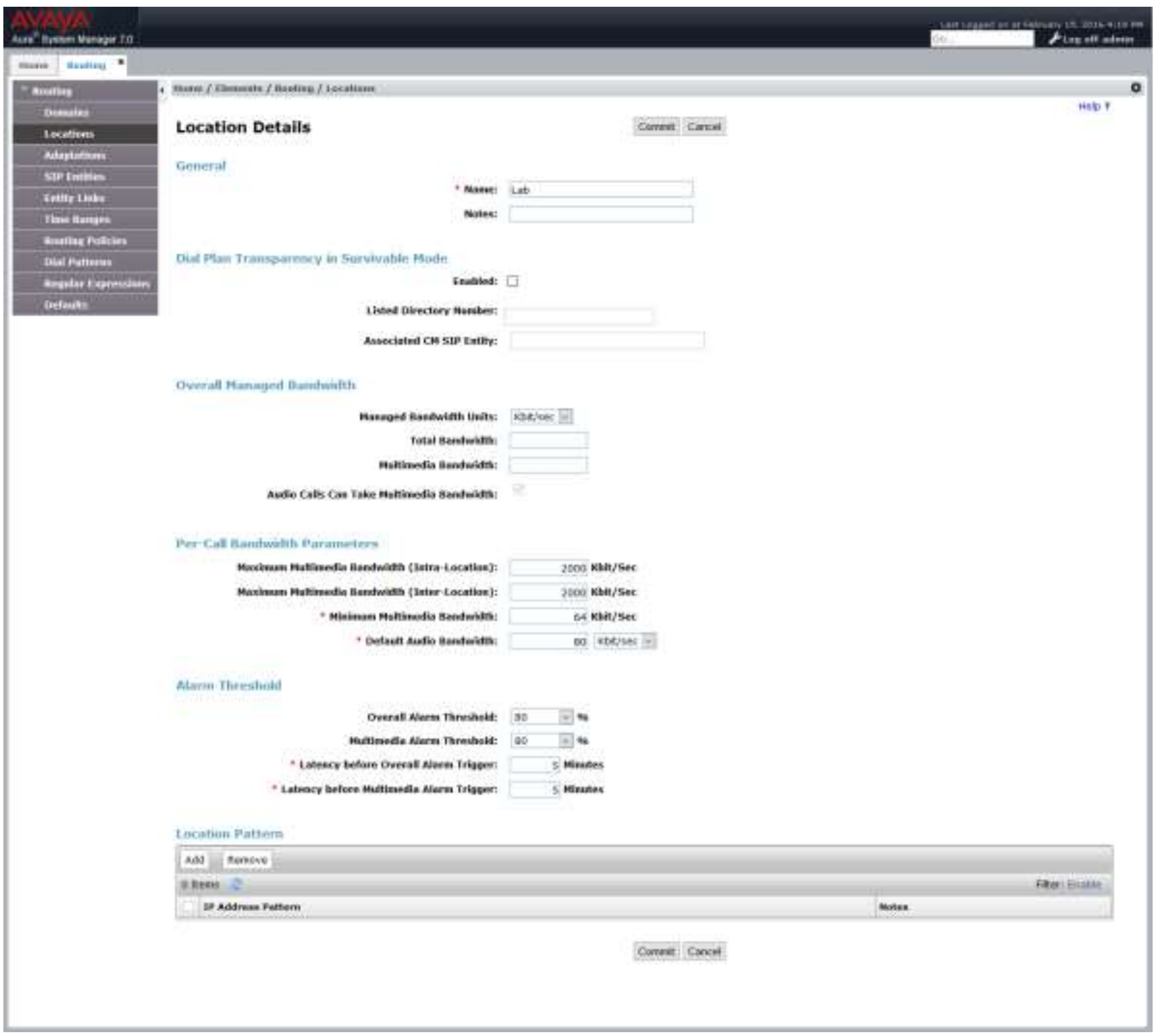
Under **General**:

- **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 **Lab** location, which includes all the components of the compliance test environment. Click **Commit** to save.



6.3. Add Adaptations

In order to maintain digit manipulation centrally on Session Manager, an adaptation module can be configured with a numbering plan offered from the PSTN Service Provider. To add an adaptation, select **Adaptations** on the left and click on the **New** button on the right. The following screen will then be shown. Fill in the following:

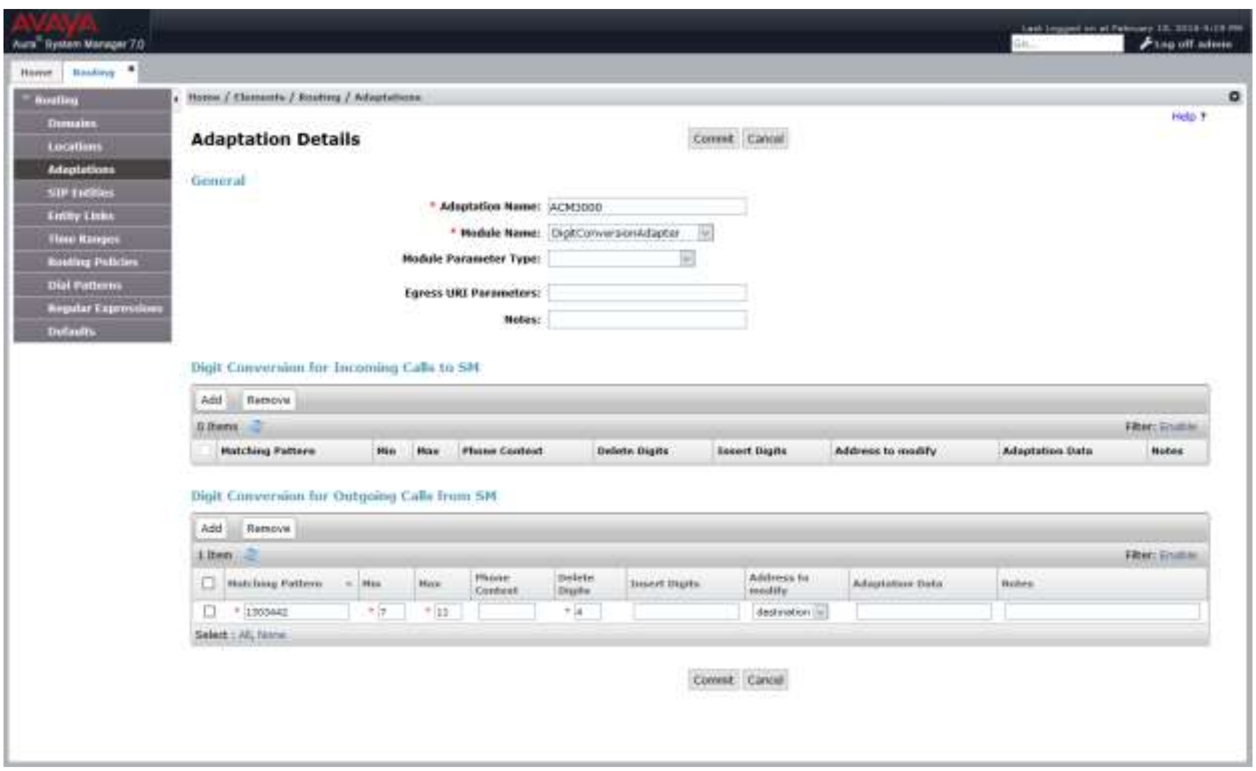
Under **General**:

- **Adaptation Name:** A descriptive name i.e., **ACM3000**
- **Module Name:** From the dropdown list select **DigitConversionAdapter**
- **Module Parameter Type:** Leave it blank

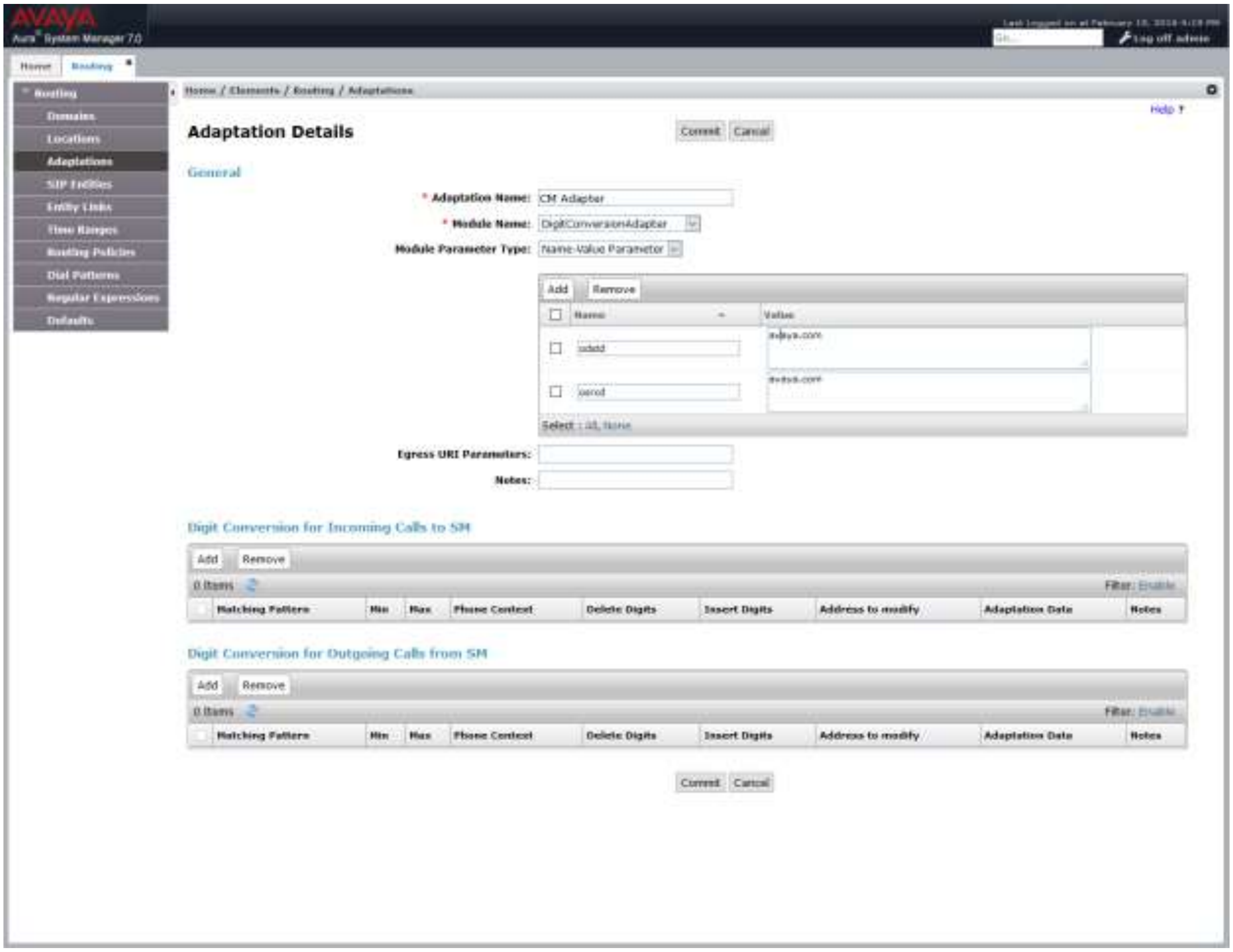
Under **Digit Conversion for Outgoing Calls From SM:**

- **Matching Pattern**: The dialed number to the simulated PSTN
- **Min/Max**: Minimum/Maximum number of digits
- **Delete Digits**: Digits to be deleted In this test the leading 4 digits for dialed numbers beginning with 1303442 were removed by Session Manager and sent to the Mediant 3000 to reach phone numbers that match 442xxxx.
- **Insert Digits**: Digit to be added
- **Address to modify**: Select **destination**

The screen below is the Adaptation detail page. Click **Commit** to save the changes.



Additionally an Adaptation was used between Session Manager and Communication Manager to replace the domain in various SIP Headers with avaya.com when communicating with Communication Manager.



6.4. 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.

6.4.1. 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).

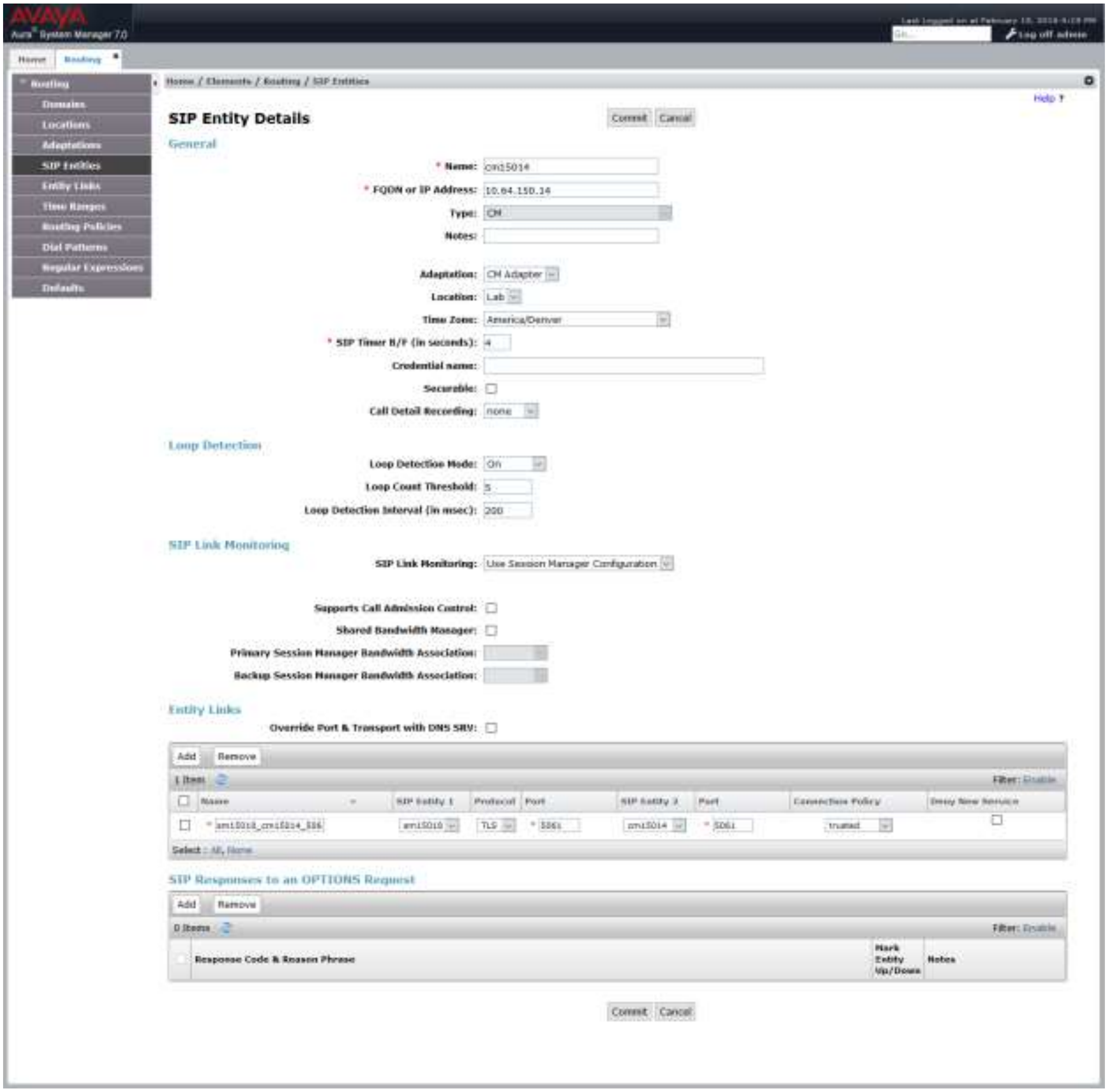
Under **General**:

- **Name:** A descriptive name, i.e., **cm15014**
- **FQDN or IP Address:** IP address of the Communication Manager i.e., **10.64.150.14**
- **Type:** Select **CM**
- **Adaptation:** Select CM Adapter
- **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.


Aura System Manager 7.0
Home
Routing
Home / Elements / Routing / SIP Entities
Routing
Domains
Locations
Adaptations
SIP Entities
Entity Links
Time Ranges
Routing Policies
Dial Patterns
Regular Expressions
Defaults
SIP Entity Details
Commit
Cancel
Help ?
General
* Name: cm15014
* FQDN or IP Address: 10.64.150.14
Type: CM
Notes:
Adaptation: CM Adapter
Location: Lab
Time Zone: America/Denver
* SIP Timer B/F (in seconds): 4
Credential name:
Securable:
Call Detail Recording: none
Loop Detection
Loop Detection Mode: On
Loop Count Threshold: 5
Loop Detection Interval (in msec): 200
SIP Link Monitoring
SIP Link Monitoring: Use Session Manager Configuration
Supports Call Admission Control:
Shared Bandwidth Manager:
Primary Session Manager Bandwidth Association:
Backup Session Manager Bandwidth Association:
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
TLS
5061
cm15014
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

6.4.2. 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).

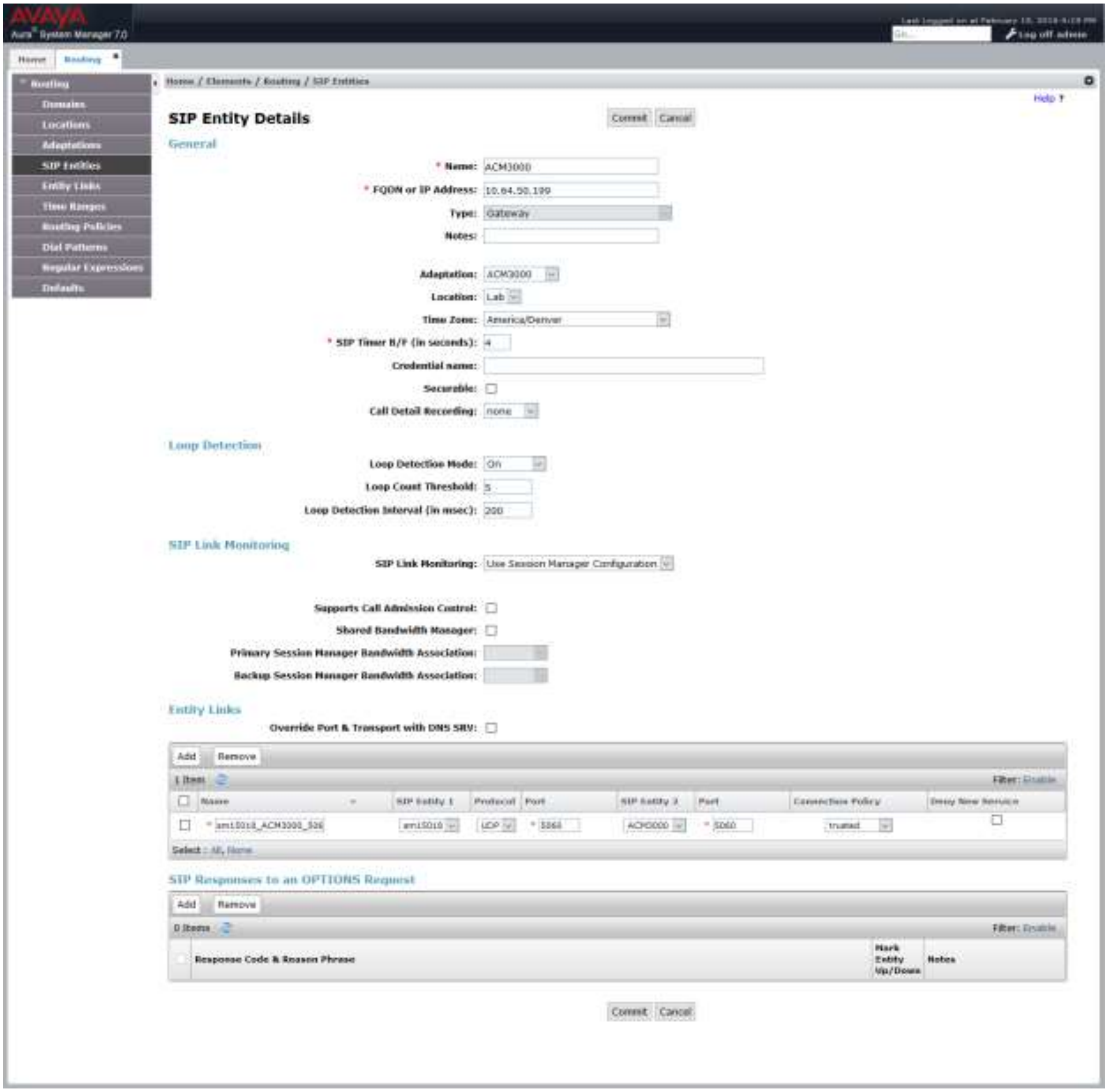
Under **General**:

- **Name:** A descriptive name, i.e., **ACM3000**
- **FQDN or IP Address:** IP address of the Mediant 3000 i.e., **10.64.50.199**
- **Type:** Select **Gateway**
- **Adaptation:** Select **ACM3000**. This was configured in **Section 6.3**
- **Location:** Select one of the locations defined previously
- **Time Zone:** Time zone for this entity
- **SIP Link Monitoring:** **Use Session Manager Configuration**

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 7.0
Home
Routing
Home / Elements / Routing / SIP Entities
Help ?
SIP Entity Details
Commit
Cancel
General
* Name: ACM3000
* FQDN or IP Address: 10.64.50.199
Type: Gateway
Notes:
Adaptation: ACM3000
Location: Lab
Time Zone: America/Denver
* SIP Timer B/F (in seconds): 4
Credential name:
Securable:
Call Detail Recording: none
Loop Detection
Loop Detection Mode: On
Loop Count Threshold: 5
Loop Detection Interval (in msec): 200
SIP Link Monitoring
SIP Link Monitoring: Use Session Manager Configuration
Supports Call Admission Control:
Shared Bandwidth Manager:
Primary Session Manager Bandwidth Association:
Backup Session Manager Bandwidth Association:
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
UDP
5060
ACM3000
5060
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
Routing
Domains
Locations
Adaptations
SIP Entities
Entity Links
Time Ranges
Routing Policies
Dial Patterns
Regular Expressions
Defaults

6.5. Add Routing Policies

Routing policies describe the condition under which calls will be routed to the SIP Entities specified in **Section 6.4**. 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**

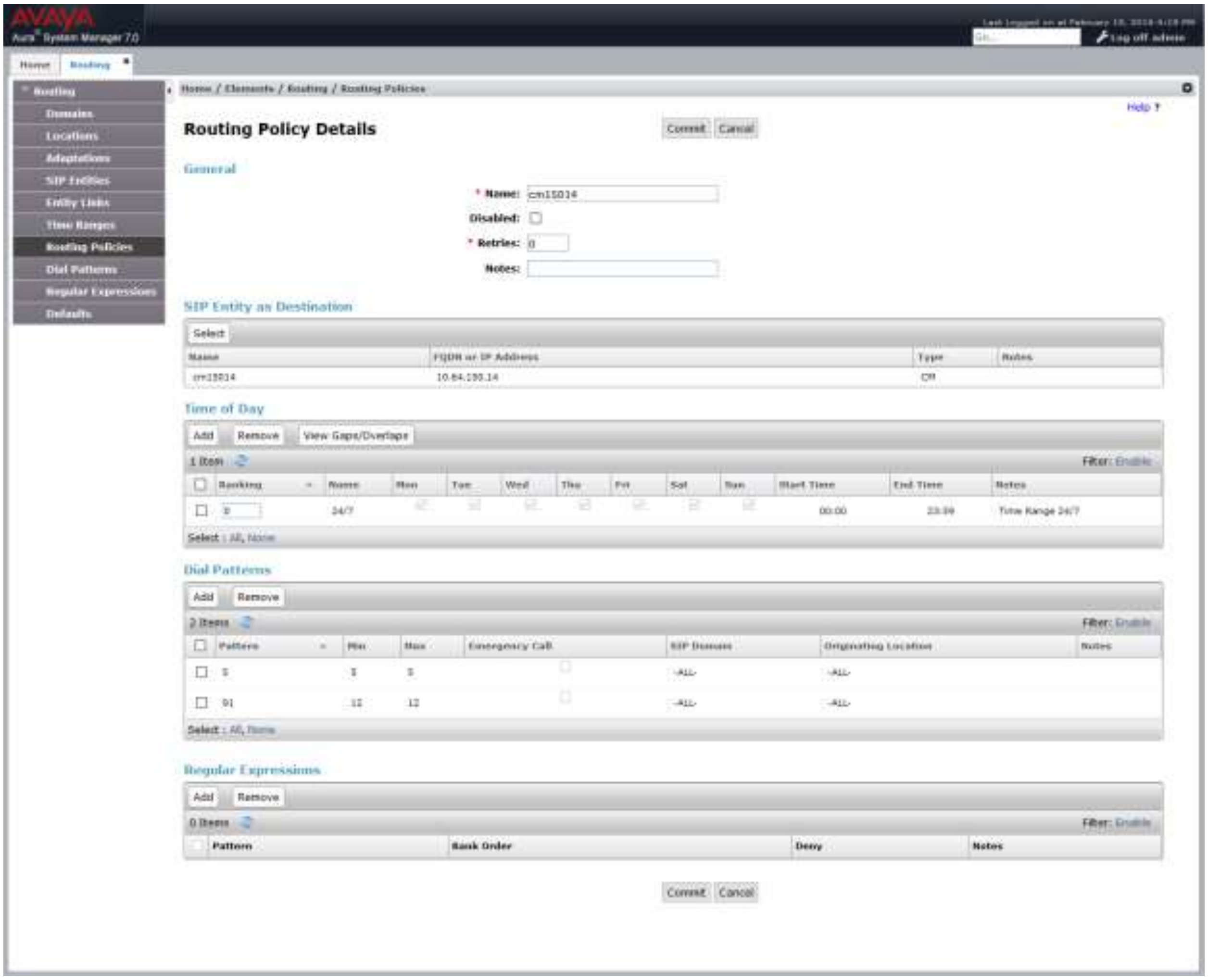
Under **SIP Entity as Destination**

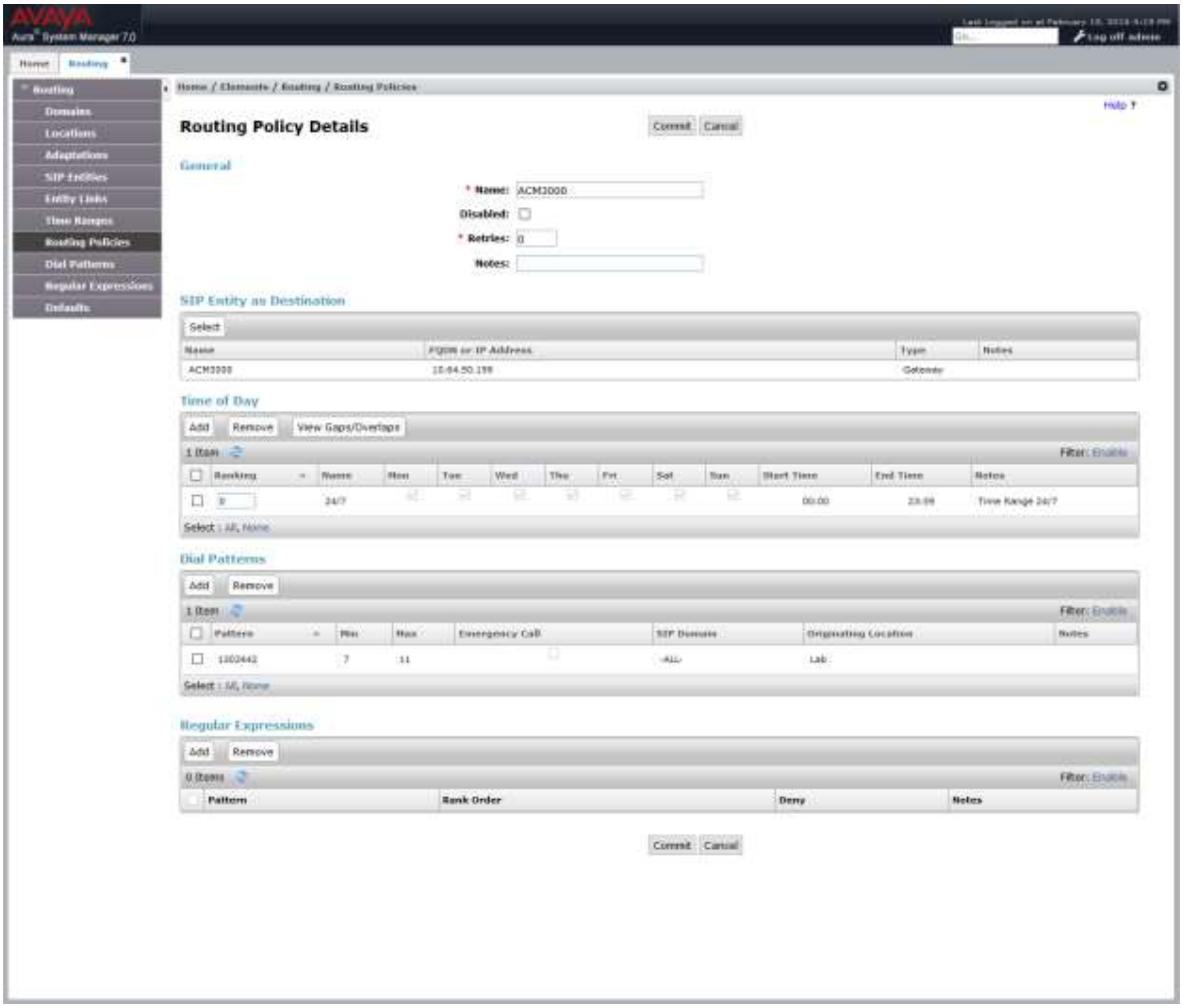
- Click **Select**, and then select the appropriate SIP entity to which this routing policy applies

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. Note that **Dial Patterns** (to be configured in **Section 6.6**), when configured, will be automatically displayed in the **Routing Policy Details** page.




Routing Policy Details
Commit
Cancel
General
* Name: ACM3000
Disabled:
* Retries: 0
Notes:
SIP Entity as Destination
Select
Name
FQDN or IP Address
Type
Notes
ACM3000
Gateway
Time of Day
Add
Remove
View Gaps/Overlaps
1 Item
Filter: Enable
Ranking
Name
Mon
Tue
Wed
Thu
Fri
Sat
Sun
Start Time
End Time
Notes
0
24/7
00:00
23:59
Time Range 24/7
Select : All, None
Dial Patterns
Add
Remove
1 Item
Filter: Enable
Pattern
Min
Max
Emergency Call
SIP Domain
Originating Location
Notes
7
11
-ALL-
Lab
Select : All, None
Regular Expressions
Add
Remove
0 Items
Filter: Enable
Pattern
Rank Order
Deny
Notes
Commit
Cancel

6.6. Add Dial Patterns

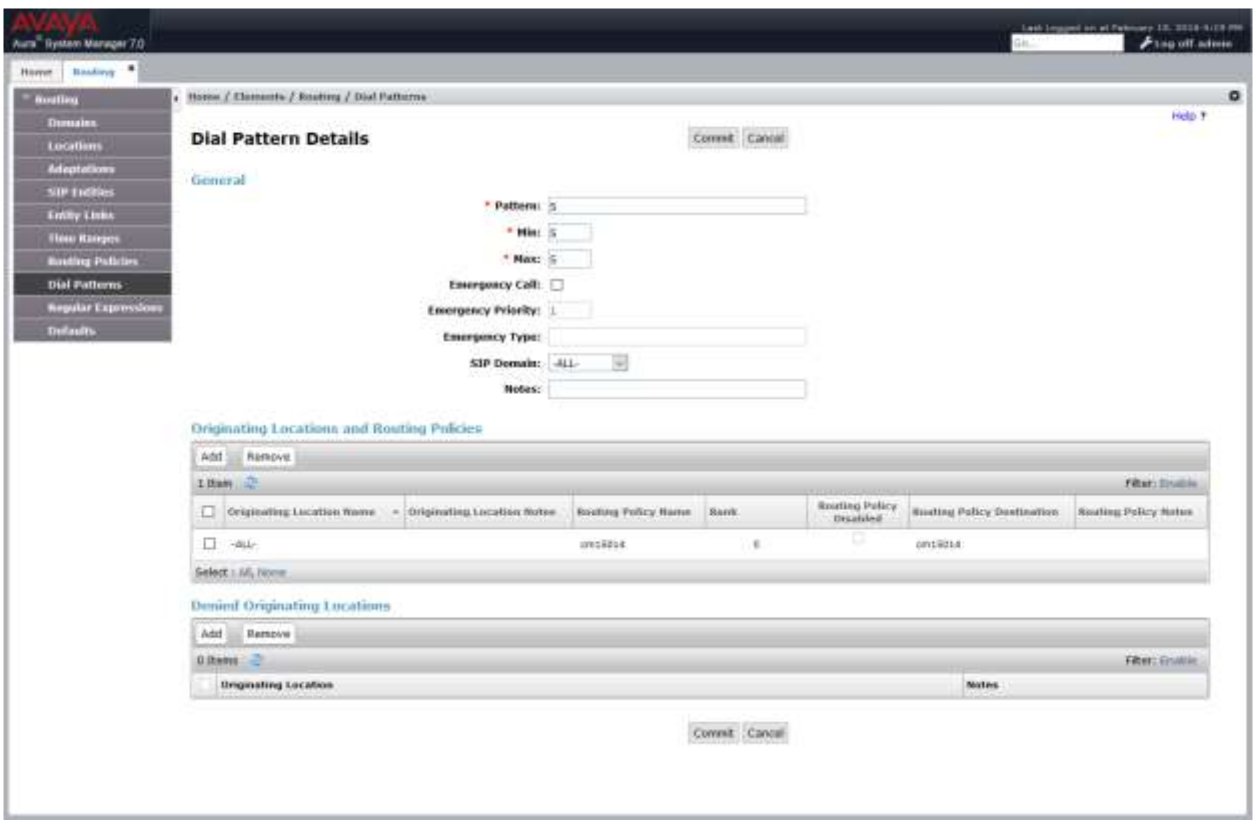
Dial patterns must be defined that will direct calls to the appropriate SIP Entity. In the sample configuration numbers beginning with **5** with 5-digit length reside in the Enterprise network. To add a dial pattern, select **Dial Patterns** on the left and click on the **New** button on the right (not 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**:

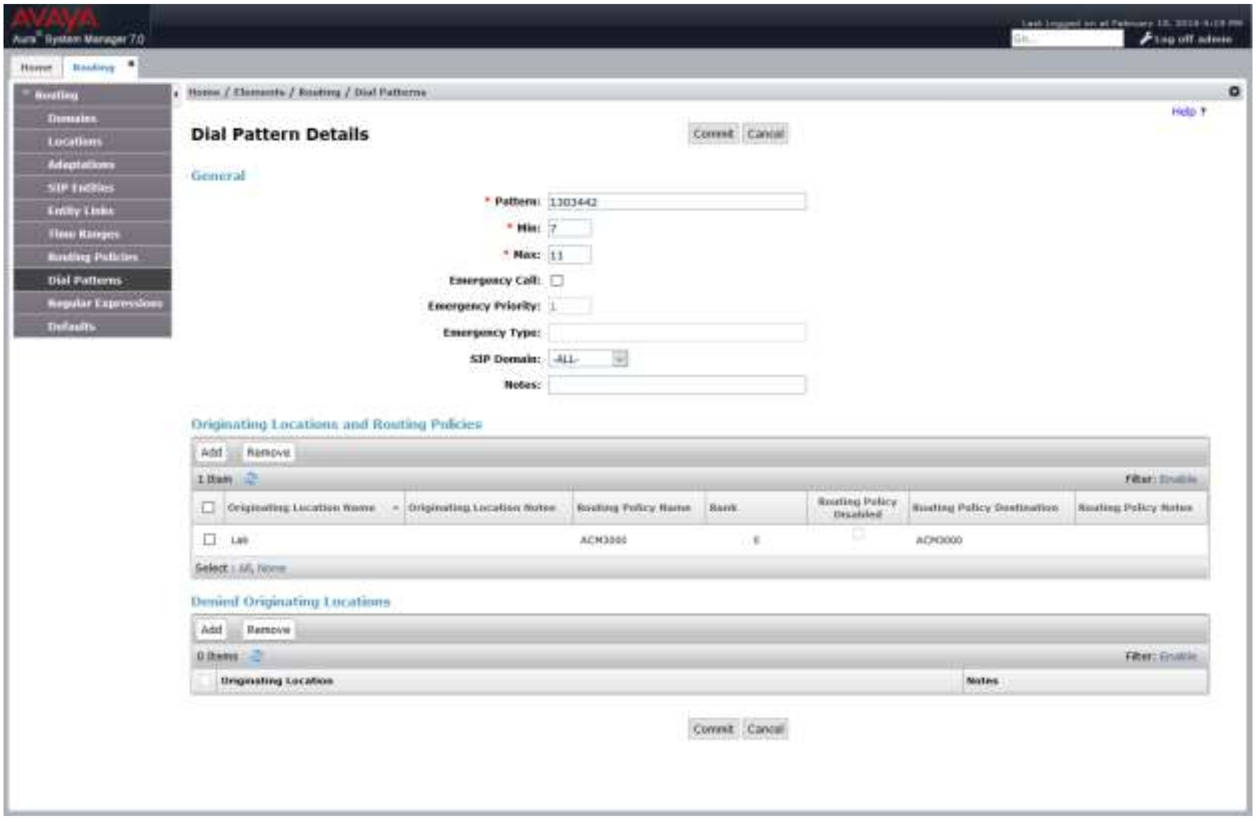
- **Pattern:** Dialed number or prefix i.e., **5**
- **Min:** Minimum length of dialed number i.e., **5**
- **Max:** Maximum length of dialed number i.e., **5**
- **SIP Domain:** Select **ALL**

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.

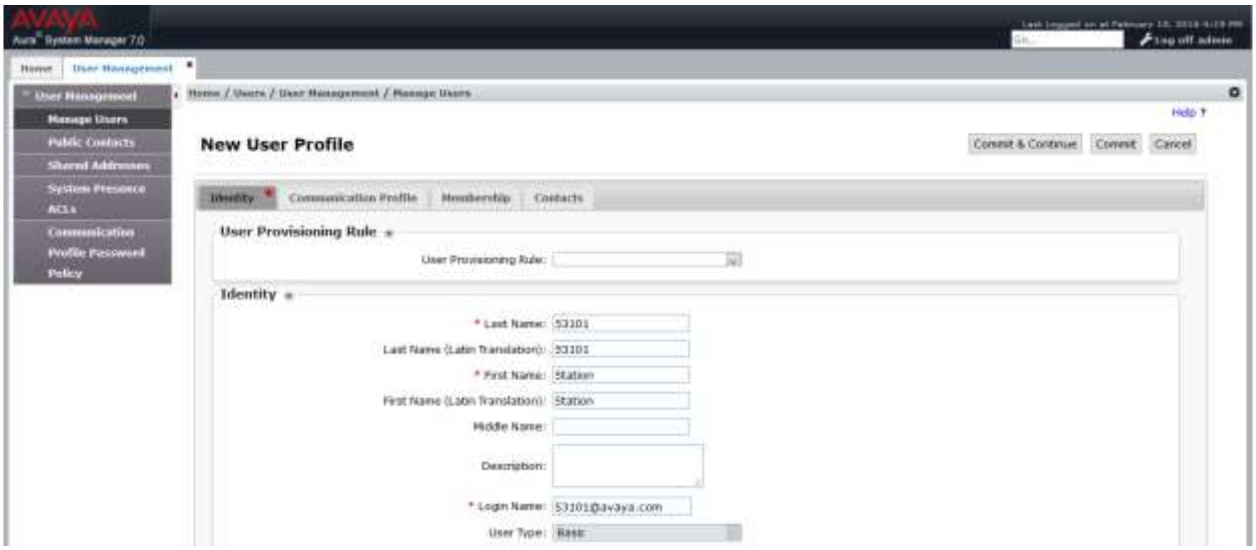


6.7. Add Users for SIP Phones

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

6.7.1. Identity

The **New User Profile** screen is displayed. Enter desired **Last Name** and **First Name**. For **Login Name**, enter “n@z”, where “n” is the user extension and “z” is the domain name, in this case “**avaya.com**” used for compliance testing. Retain the default values in the remaining fields.



6.7.2. Communication Profile

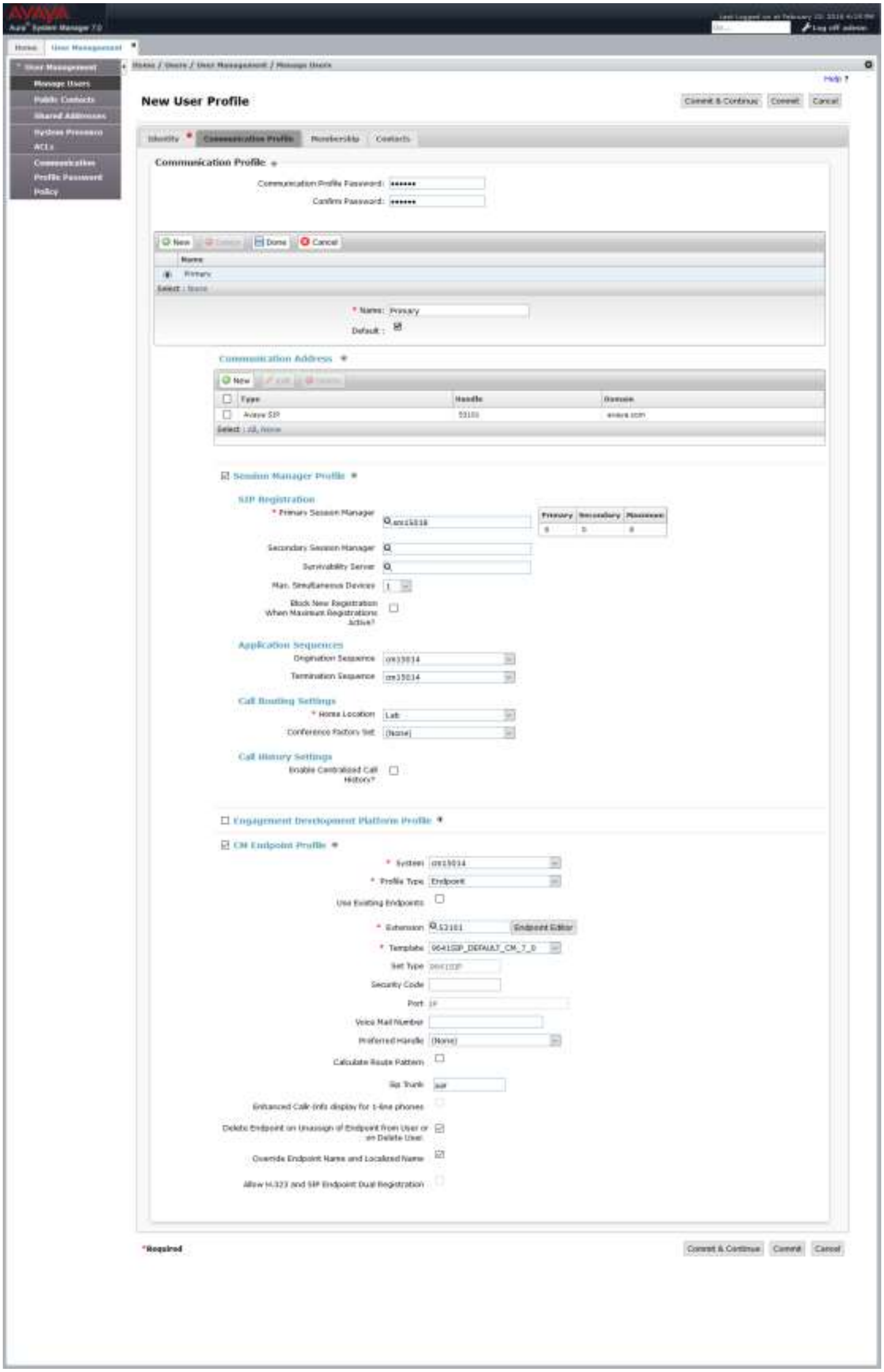
Select the **Communication Profile** tab. For **Communication Profile Password** and **Confirm Password**, enter the desired password for the SIP user to use for registration. Scroll down to the **Communication Address** sub-section, and click **New** to add a new address.

For **Type**, retain "Avaya SIP". For **Fully Qualified Address**, enter and select the SIP user extension and domain configured in **Section 6.7.1**. Click **Add**.

Scroll down to check and expand **Session Manager Profile**. For **Primary Session Manager**, **Origination Application Sequence**, **Termination Application Sequence**, and **Home Location**, select the values corresponding to the applicable Session Manager. Retain the default values in the remaining fields. These settings are configured during the initial setup of Session Manager.

Scroll down to check and expand **CM Endpoint Profile**. For **System**, select the value corresponding to the applicable Communication Manager. For **Extension**, enter or select the SIP user extension configured in **Section 6.7.1**. For **Template**, select corresponding Telephone type. Retain the default values in the remaining fields.

Click **Commit** to complete the creation of the new user.


New User Profile
Commit & Continue
Commit
Cancel
Communication Profile
Communication Profile Password:
Confirm Password:
New
Done
Cancel
Primary
* Name: Primary
Default :
Communication Address
Avaya SIP
Session Manager Profile
SIP Registration
* Primary Session Manager
Secondary Session Manager
Survivability Server
Max. Simultaneous Devices
Application Sequences
Origination Sequence
Termination Sequence
Call Routing Settings
* Home Location
Lab
Conference Factory Set
(None)
Call History Settings
Enable Centralized Call History?
Engagement Development Platform Profile
CM Endpoint Profile
* System
* Profile Type
Endpoint
Use Existing Endpoints
* Extension
53101
Endpoint Editor
* Template
9641SIP_DEFAULT_CM_7_0
Set Type
Security Code
Port
Voice Mail Number
Preferred Handle
(None)
Calculate Route Pattern
Sip Trunk
aar
Enhanced Callr-Info display for 1-line phones
Delete Endpoint on Unassign of Endpoint from User or on Delete User.
Override Endpoint Name and Localized Name
Allow H.323 and SIP Endpoint Dual Registration
*Required
Commit & Continue
Commit
Cancel

7. AudioCodes Mediant 3000 Configuration

This section describes the configuration for enabling the Mediant 3000 to interoperate with Session Manager. The procedures require five distinct operations:

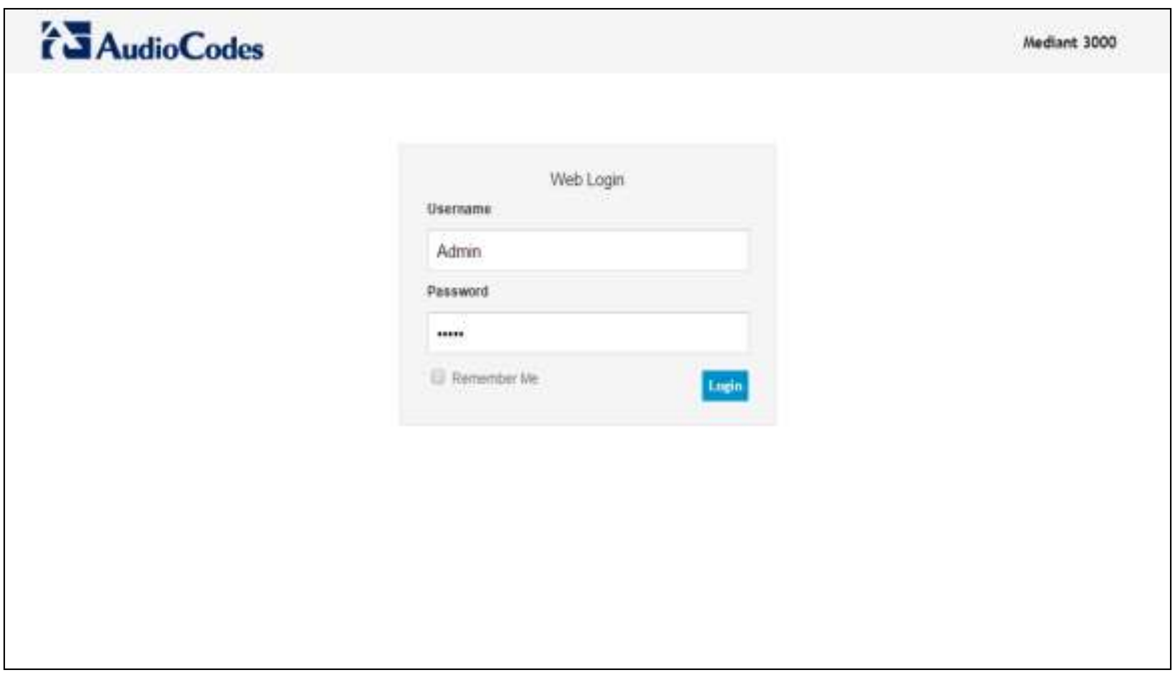
- Configuring the Media Gateway Host IP Network Parameters
- Configuring the Media Gateway TDM and Timing Parameters
- Configuring the Media Gateway Media Settings
- Configuring the Media Gateway Telephony/PSTN Interfaces Parameters
- Configuring the Media Gateway SIP Protocol Parameters

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

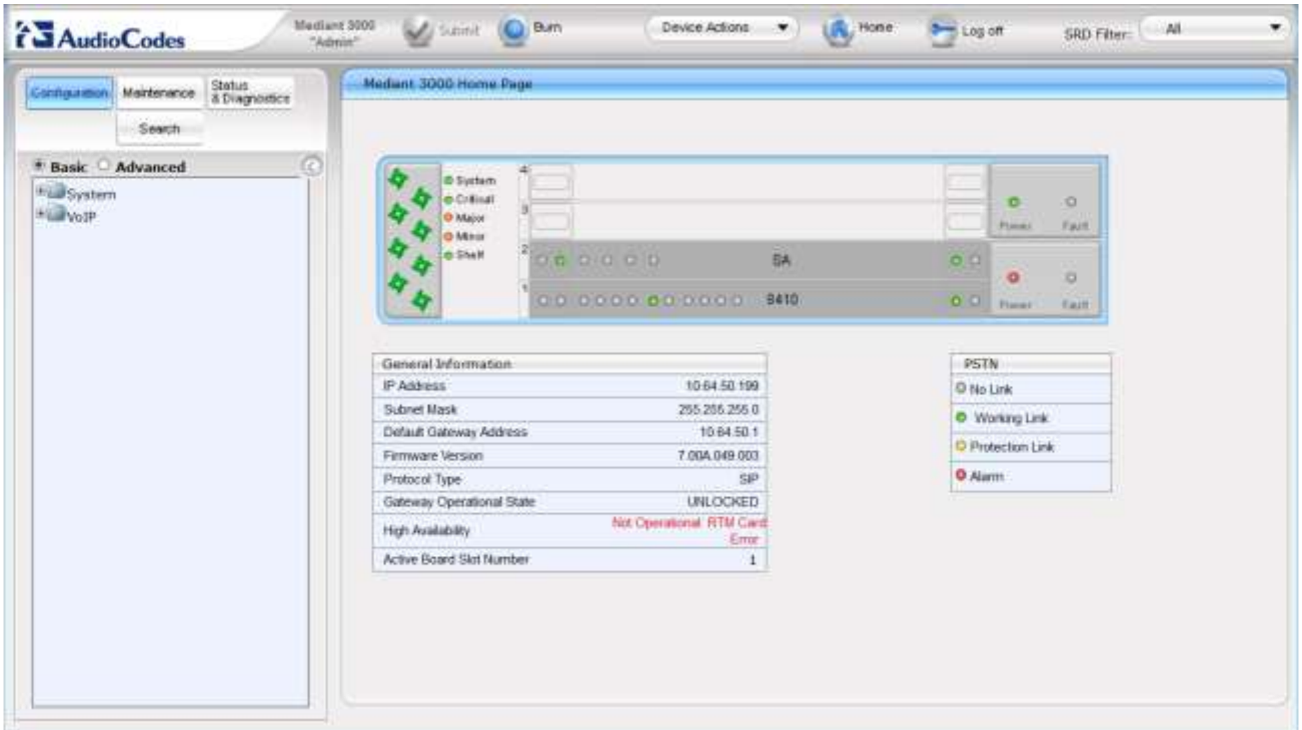
The configuration of the Mediant 3000 Gateway is done via a Web browser. To access the device, enter the **IP address** of the Mediant 3000 in the **Address** field of the web browser. The IP address was provisioned during initial installation.

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



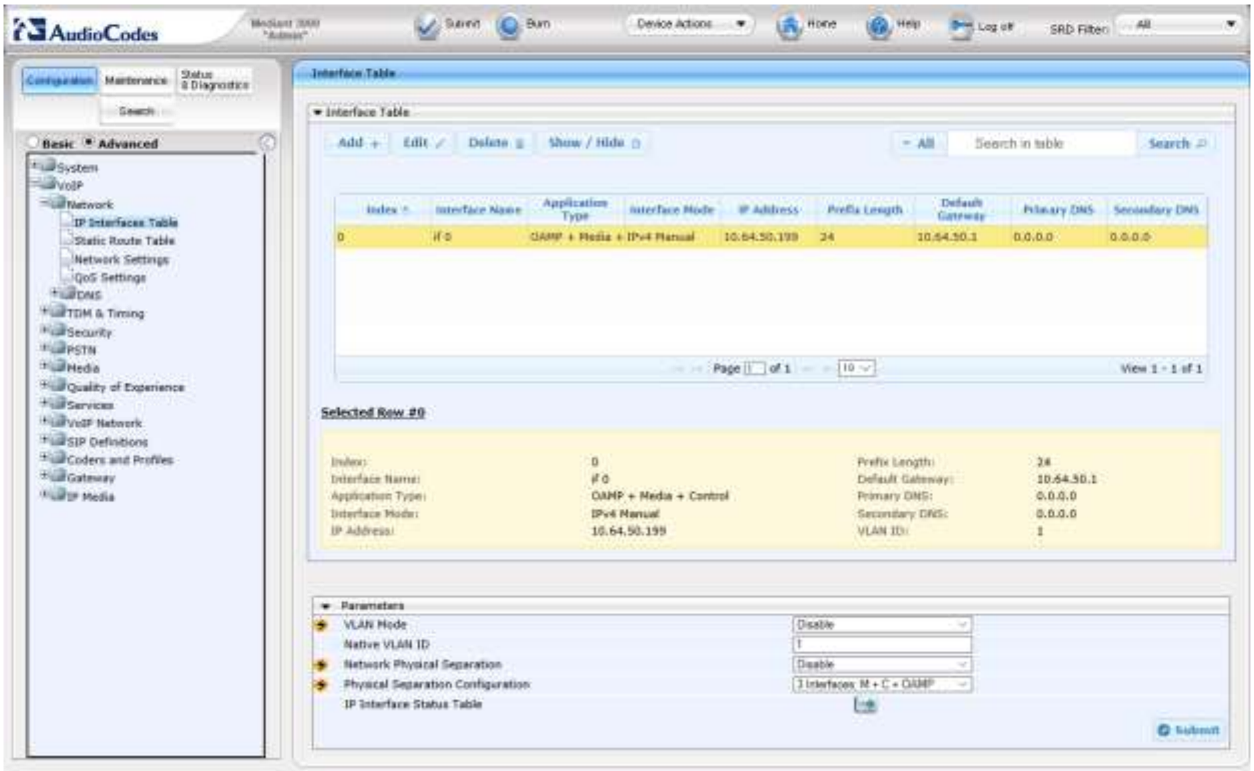
Mediant 3000 Home Page

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



7.2. Configure Media Gateway IP Network Parameters

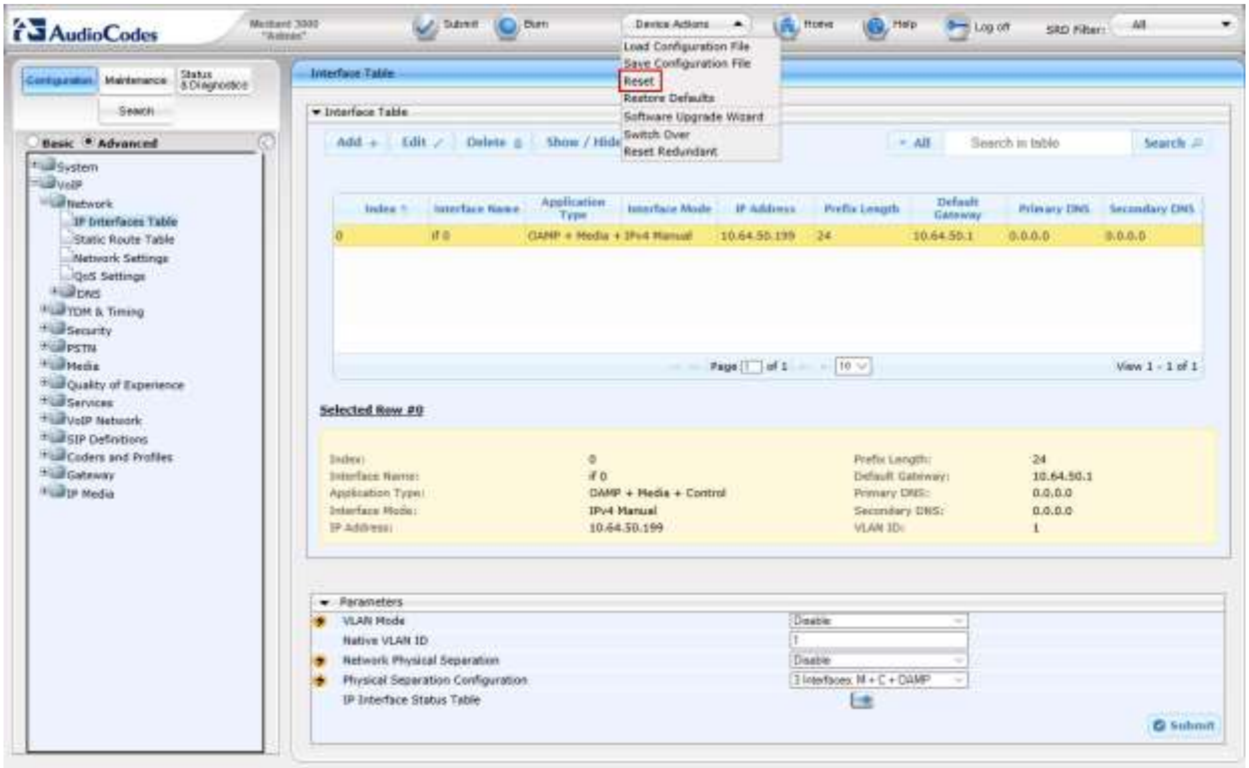
To configure the network parameters, navigate to **VoIP → Network → IP Interfaces Table** and click on the **Add** button to add an index with **Application Type** of **OAMP + Media + Control** and ensure the **Interface Mode** is set to **IPv4 Manual** and that **IP Address** (i.e., **10.64.50.199**),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



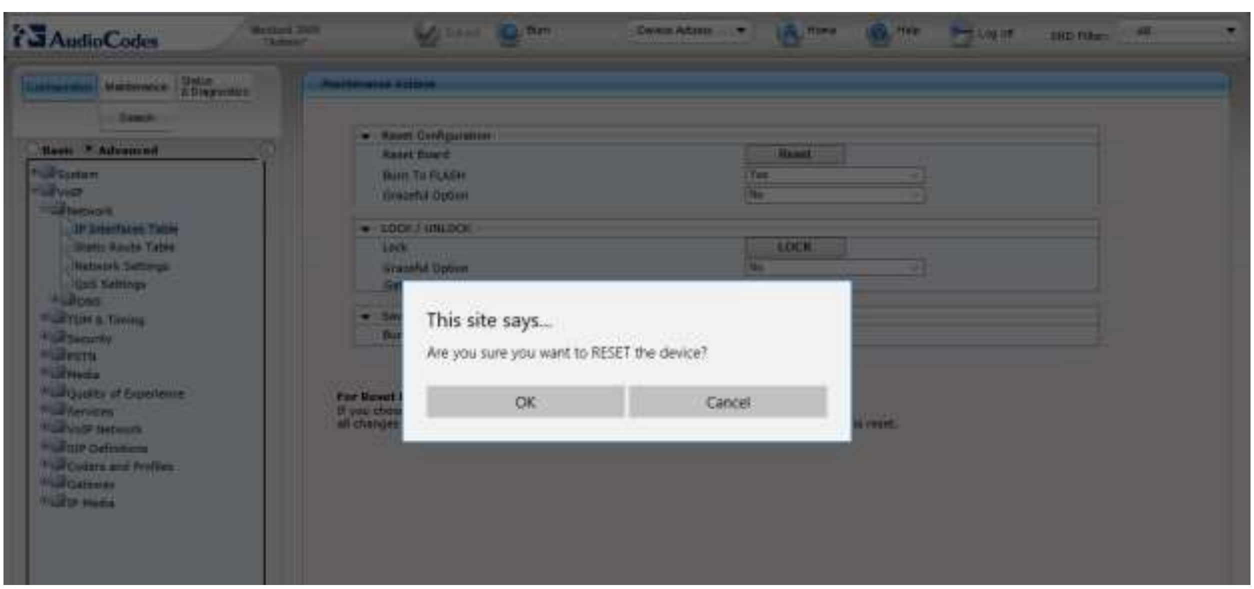
7.2.1. Saving Configuration and Resetting Mediant 3000

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.



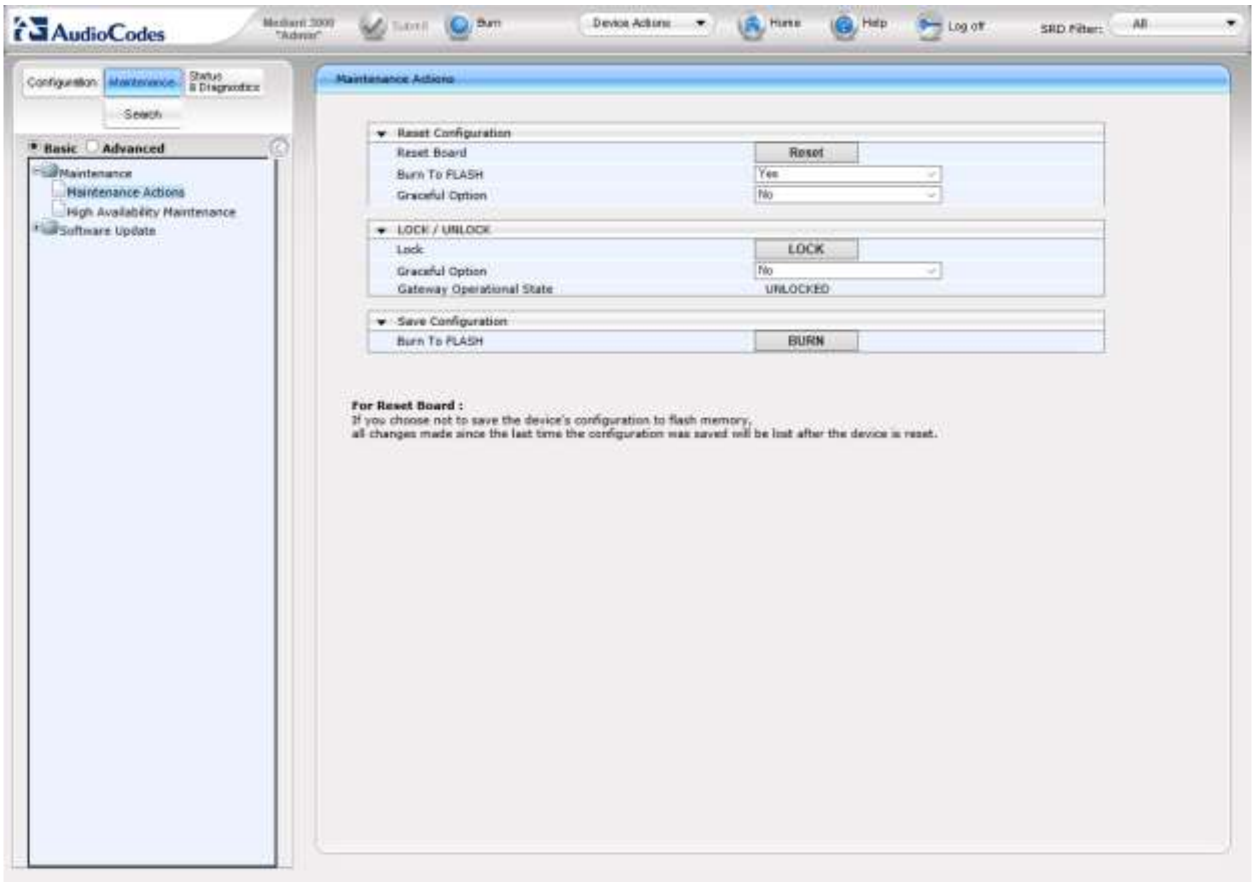
Make sure **Burn To FLASH** is set to **Yes**, and then click the **Reset** button then click **OK** for confirmation. The device's new configuration is saved (burned) to the flash memory and the device resets.



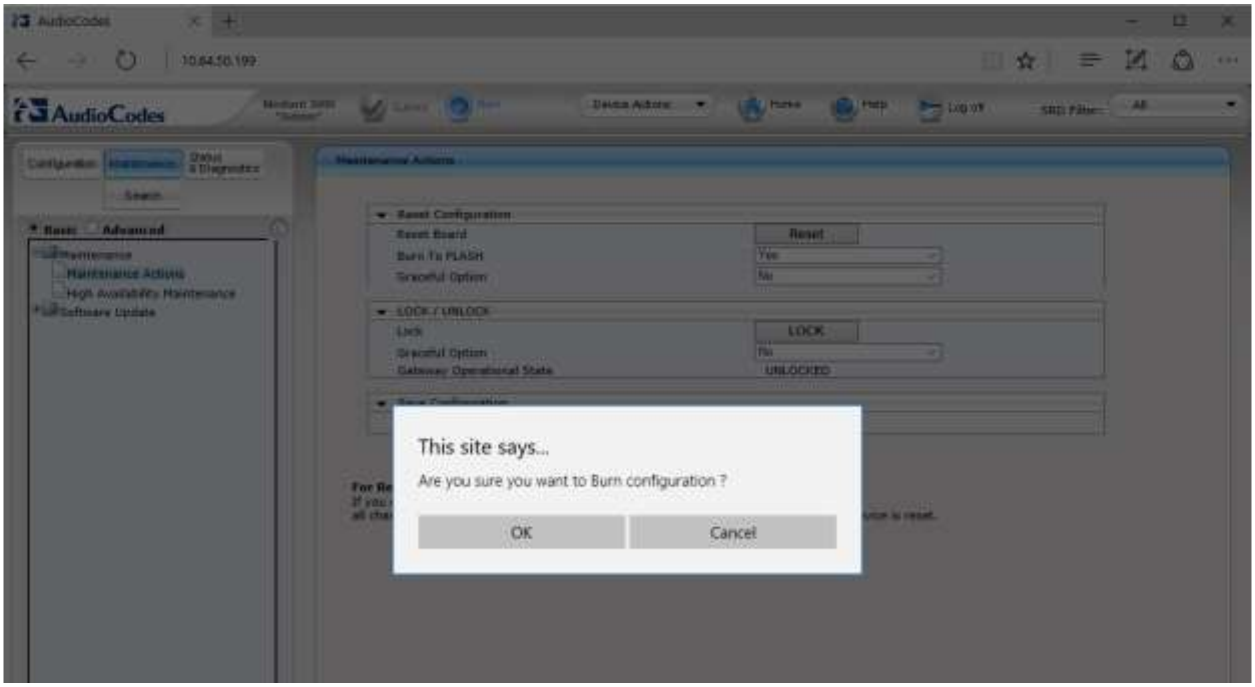
Note: if any parameter with the lightening symbol beside it (see the screenshot is **Section 7.4.1** for example) is changed, a Reset with Burn To Flash is required. The reset does not have to be done until all configuration is completed, and there will be a red reset notification at the top of the page (not shown).

7.3. 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.



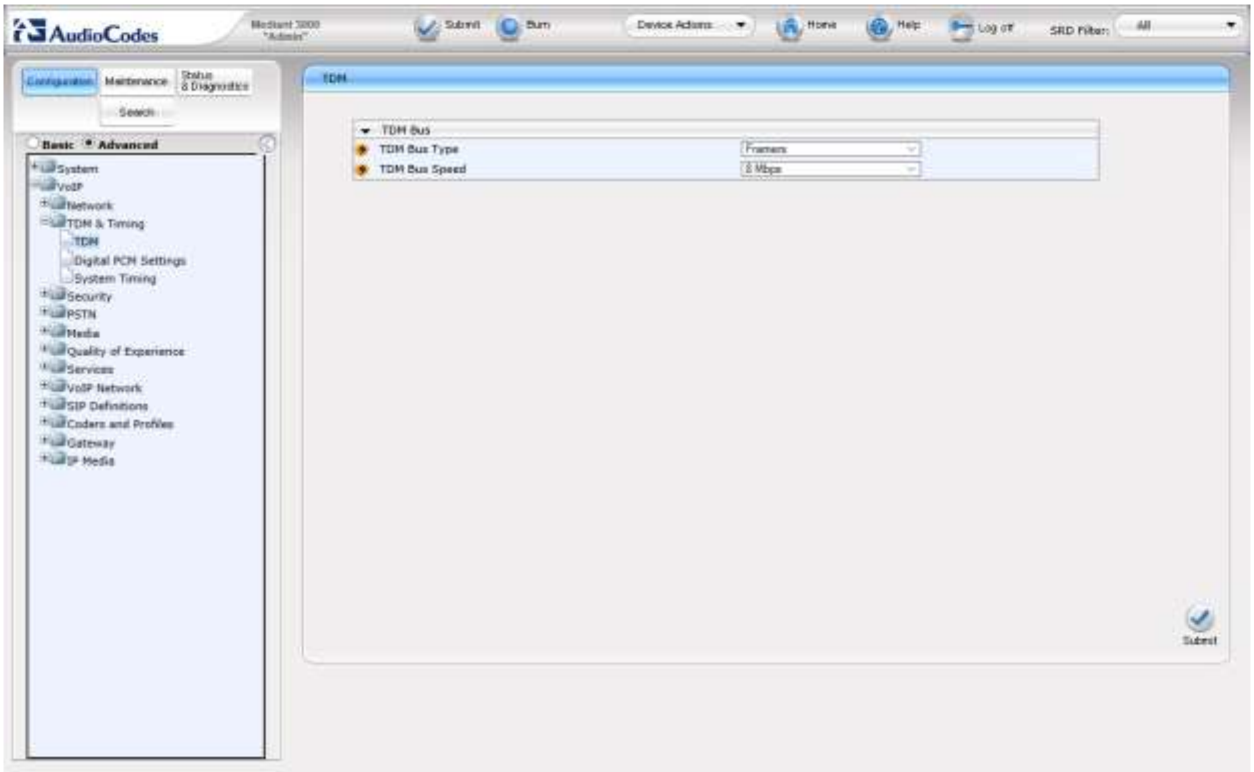
Also note the **Burn** button at the top of the screen. This is the shortest path to do a burn and can be used at any time. When clicked, it will present a pop-up similar to the one shown below.


This site says...
Are you sure you want to Burn configuration ?
OK
Cancel

7.4. Configure Media Gateway TDM and Timing Parameters

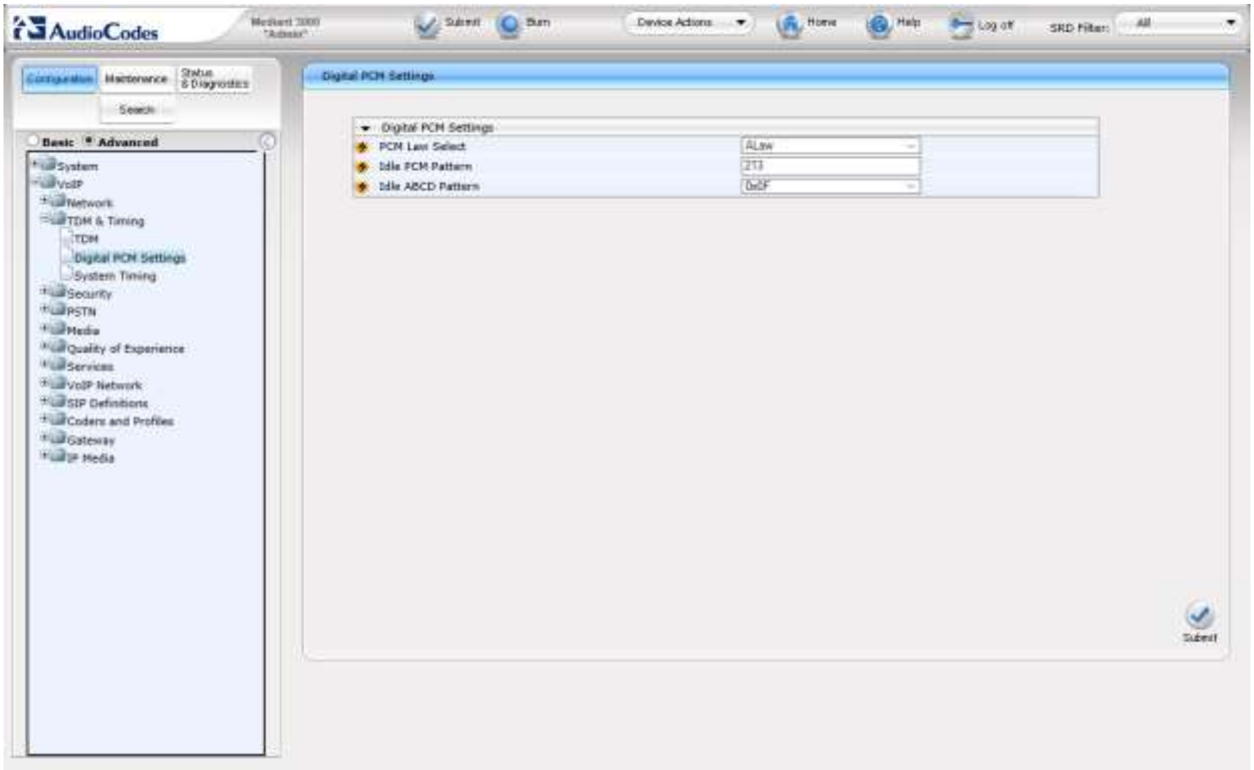
7.4.1. Configure TDM Bus

To configure the TDM Bus settings, open the **TDM** page (**Configuration** tab → **VoIP** → **TDM & Timing** → **TDM**), configure **TDM Bus Type** and **TDM Bus Speed** parameters as required. For E1 set **TDM Bus Type** to **Framers** and **TDM Bus speed** to **8Mbps**. Click the **Submit** button to save changes.



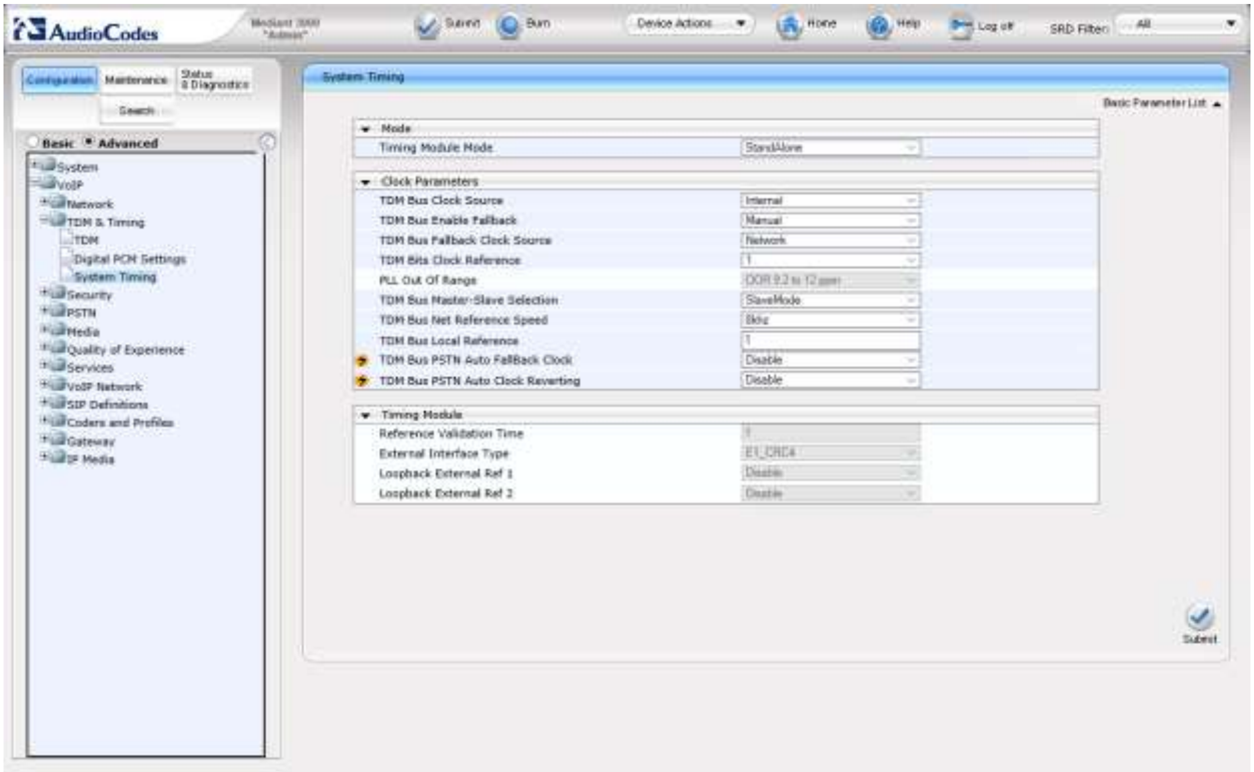
7.4.2. Configure Digital PCM Settings

To configure the digital PCM settings, open the **Digital PCM Settings** page (**Configuration** tab → **VoIP** → **TDM & Timing** → **Digital PCM Settings**). Configure the parameters as required, i.e., **ALaw** for **PCM Law Select** for E1 and click the **Submit** button to save changes.



7.4.3. Configure System Timing

To configure the device's system timing, open the **System Timing** page (**Configuration** tab → **VoIP** → **TDM & Timing** → **System Timing**). Configure the parameters as required. Click the **Submit** button to save changes. The screen below illustrates the configuration of system timing where the Mediant 3000 is configured as Master Clock Source as used in these Application Notes.

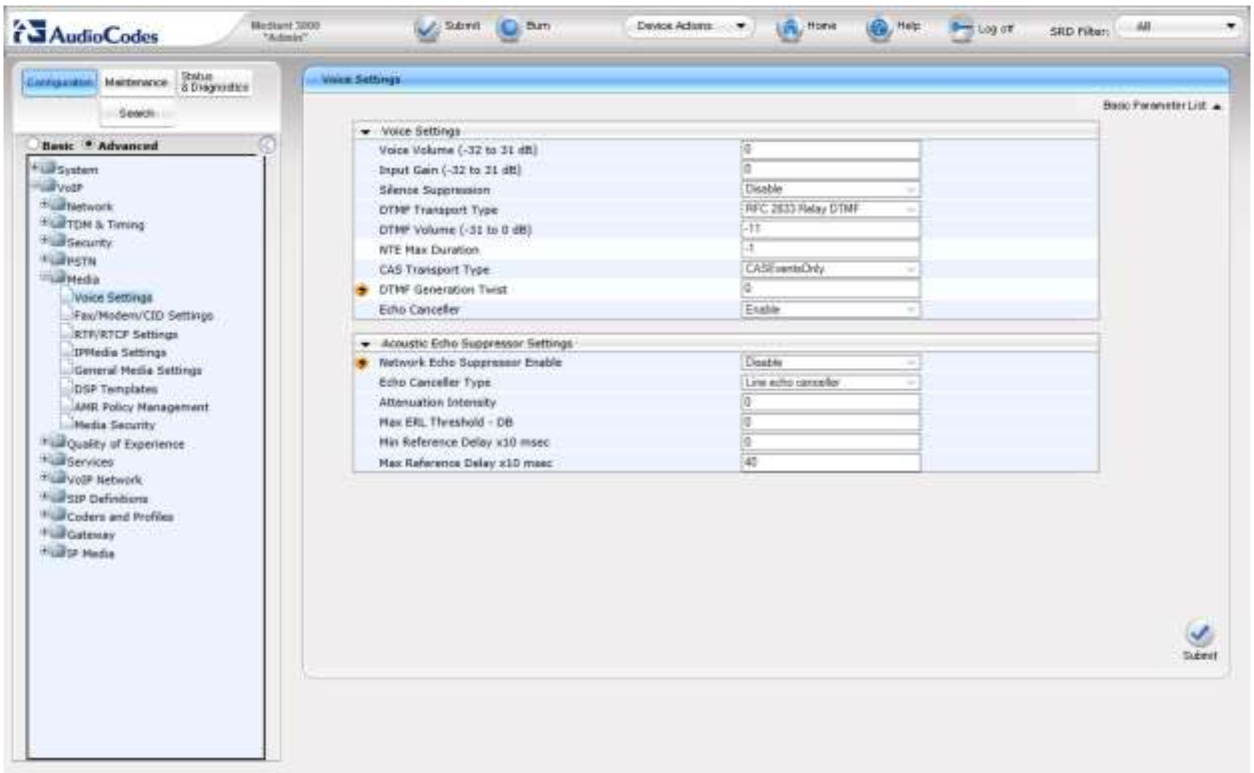


7.5. Configure Media Gateway Media Settings

The Media Settings of the Mediant 3000 Media Gateway can be configured using the web interface.

7.5.1. Configure the Voice parameters

Open the **Voice Settings** page (**Configuration** tab → **VoIP** → **Media** → **Voice Settings**). Set **DTMF Transport Type** to **RFC2833 Relay DTMF** as shown below, and click the **Submit** button to save changes.

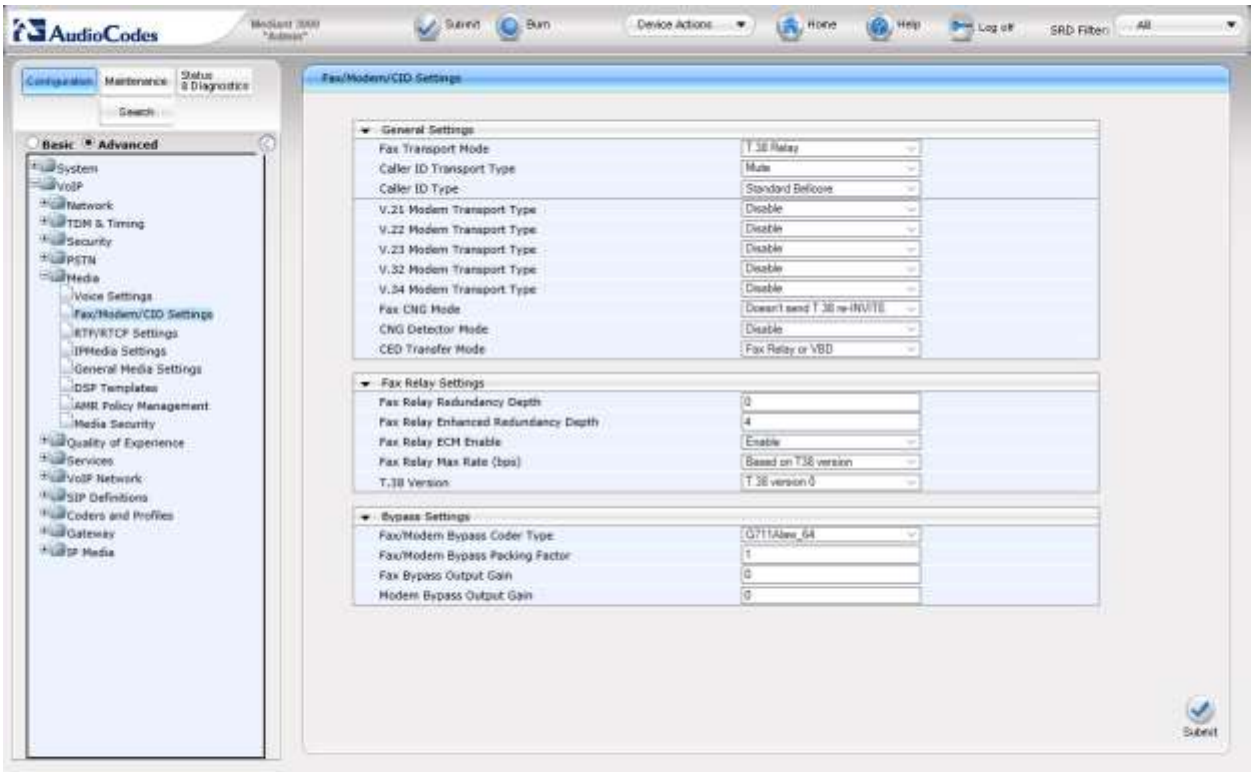


7.5.2. Configure the Fax Parameters

To configure FAX support, open the **Fax/Modem/CID Settings** page (**Configuration** tab→ **VoIP** → **Media** → **Fax/Modem/CID Settings**). Set the following values:

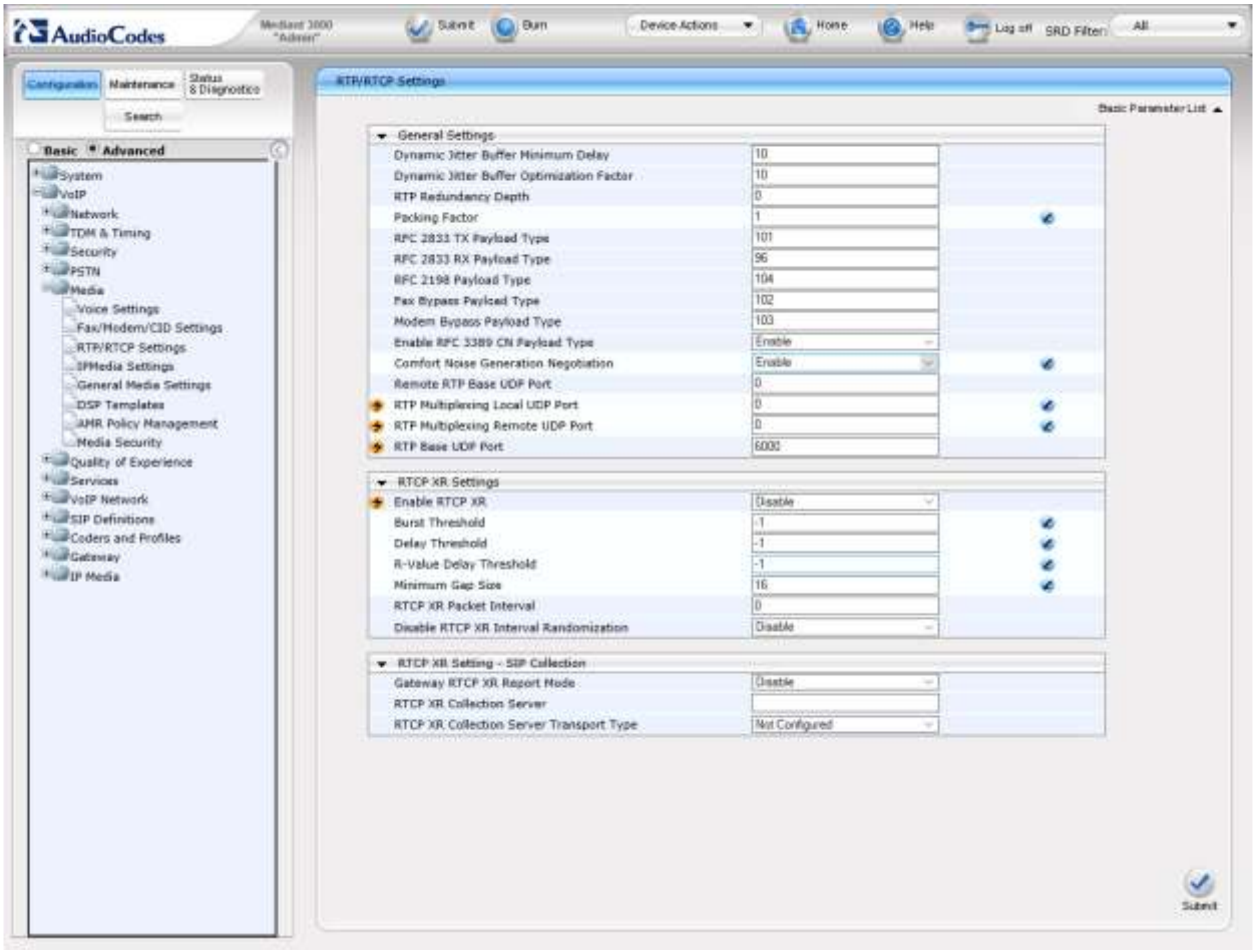
- **Fax Transport Mode:** **T.38 Relay**
- **Fax Relay Max Rate:** **Based on T38 version**

Click the **Submit** button to save changes. The screen below illustrates the Fax settings on the Mediant 3000.



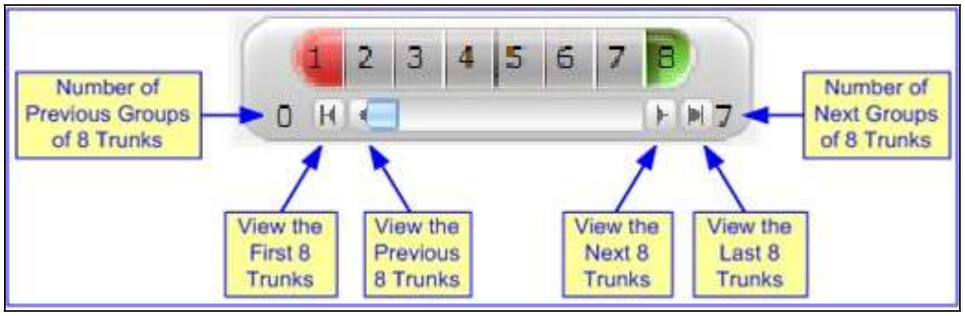
7.5.3. Configure the RTP/RTCP Parameters

Verify and configure RTP parameters by opening the **RTP/RTCP Settings** page (**Configuration** tab → **VoIP** → **Media** → **RTP / RTCP Settings**). Configure the settings as shown below, then click the **Submit** button to save changes. The figure below illustrates settings used for the compliance test.



7.6. Configure Media Gateway Telephony/PSTN Interface Parameters

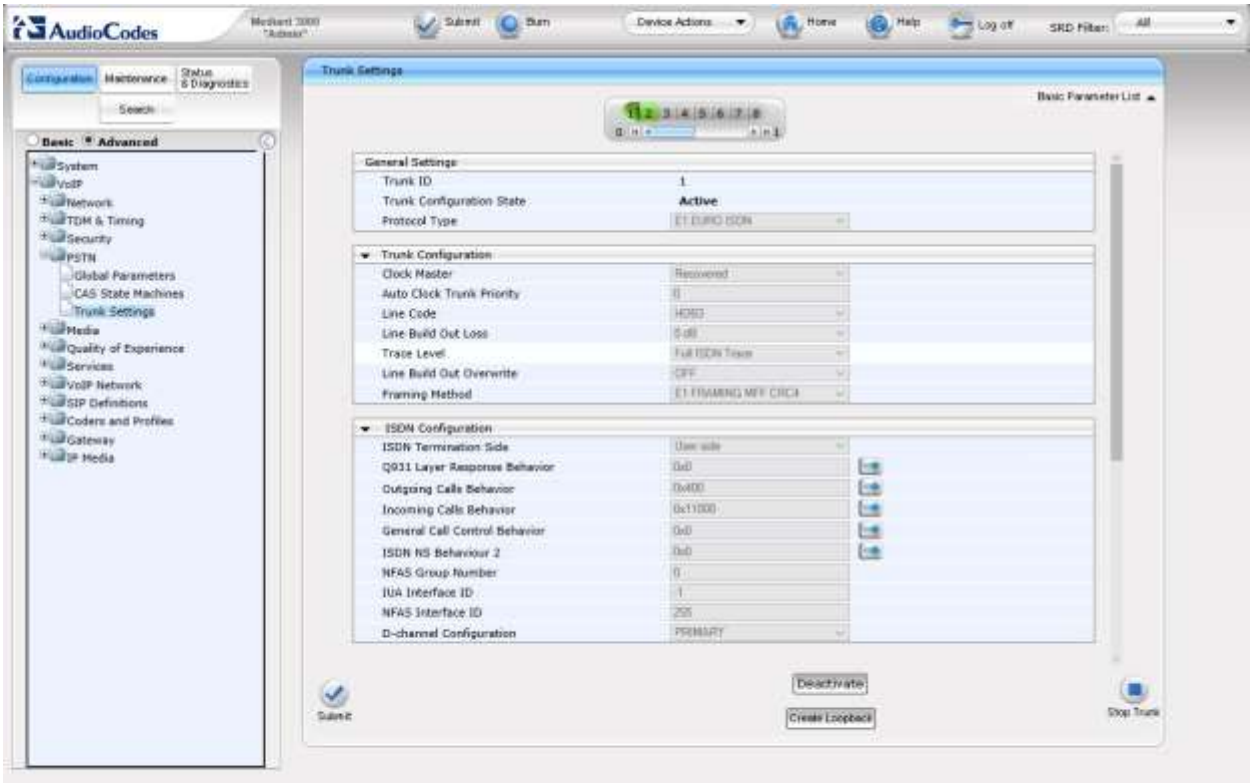
Open the **Trunk Settings** page (**Configuration** tab → **VoIP** → **PSTN** → **Trunk Settings**). Select the trunk to be configured, by clicking the desired Trunk number icon in the right pane. The bar initially displays the first eight trunk number icons (i.e., trunks 1 through 8). To scroll through 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.

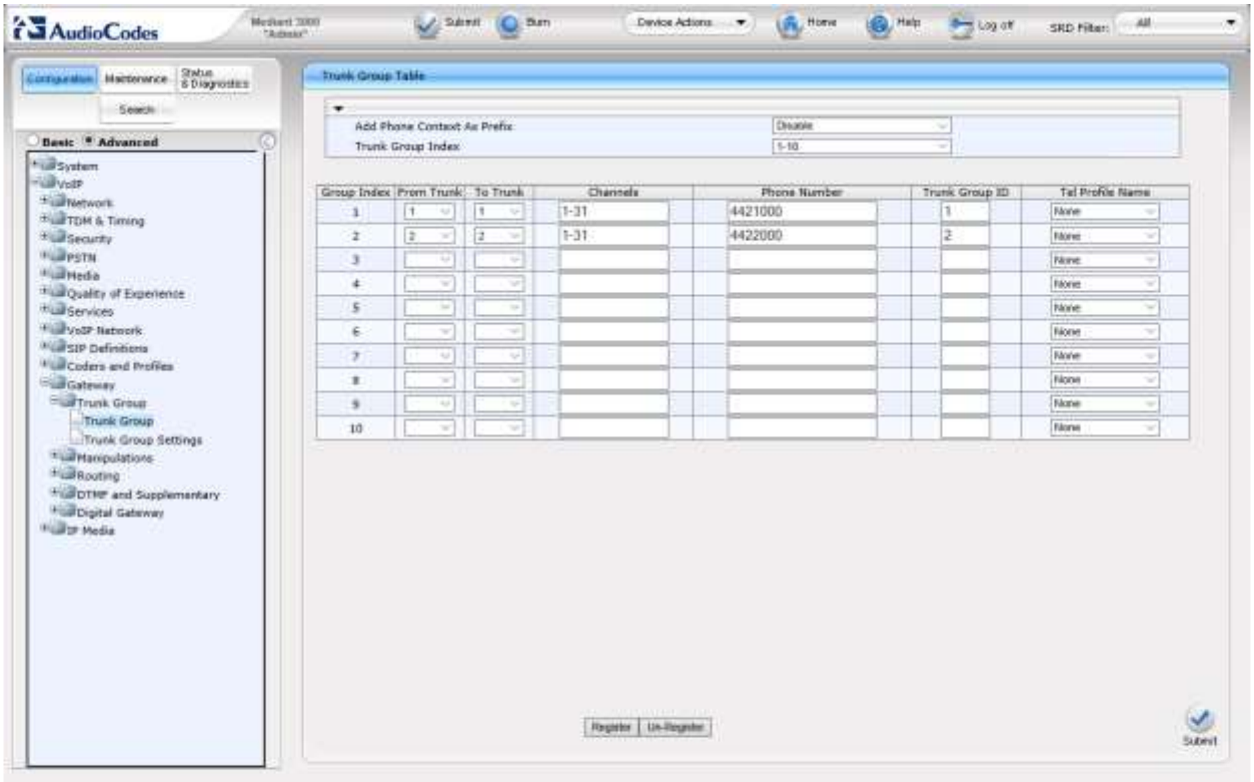
In these Application Notes the PSTN interface was configured as follows:

- **Protocol Type:** **E1 EURO ISDN**
- **Line Code:** **HDB3**
- **Framing Method:** **E1 FRAMINIG MFF CRC4**
- **ISDN Termination Side:** **User side**


AudioCodes
Submit
Burn
Device Actions
Home
Help
Log off
SRD Filter:
All
Configuration
Maintenance
Status & Diagnostics
Search
Basic
Advanced
System
VoIP
Network
TDM & Timing
Security
PSTN
Global Parameters
CAS State Machines
Trunk Settings
Media
Quality of Experience
Services
VoIP Network
SIP Definitions
Coders and Profiles
Gateway
IP Media
Trunk Settings
Basic Parameter List
General Settings
Trunk ID
1
Trunk Configuration State
Active
Protocol Type
Trunk Configuration
Clock Master
Auto Clock Trunk Priority
Line Code
Line Build Out Loss
Trace Level
Line Build Out Overwrite
Framing Method
ISDN Configuration
ISDN Termination Side
Q931 Layer Response Behavior
Outgoing Calls Behavior
Incoming Calls Behavior
General Call Control Behavior
ISDN NS Behaviour 2
NFAS Group Number
IUA Interface ID
NFAS Interface ID
D-channel Configuration
Deactivate
Create Loopback
Submit
Stop Trunk

7.6.1. Configure Trunk Group Table

Open the **Trunk Group Table** page (**Configuration** tab → **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]**. Click the **Submit** button to save changes. The screen below illustrates setting used for the compliance test.



7.7. Configure SIP Protocol Parameters

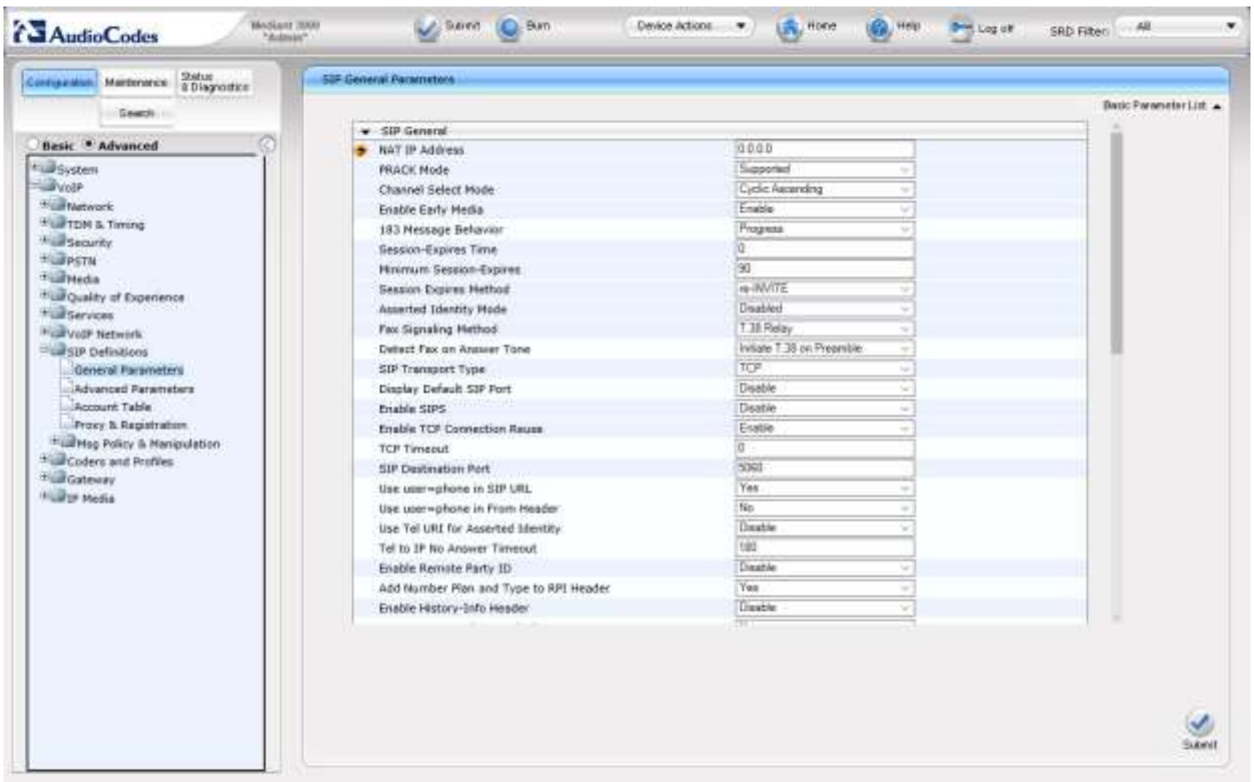
The SIP protocol interface is configured through a series of configuration steps.

7.7.1. Configure General SIP Protocol Parameters

Open the **SIP General Parameters** page (**Configuration** tab → **VoIP** → **SIP Definitions** → **General Parameters**). Set the following values:

- **Enable Early Media:** **Enable**
- **Fax Signaling Method:** **T.38 Relay**
- **SIP Transport Type:** Align with setting in the entity link definition on Session Manager for the Mediant 3000, i.e. **TCP**
- **Use Tel URI for Asserted Identity:** Set to **Disable**

Click the **Submit** button to save changes. The figure below illustrates the **SIP General Parameters** page.

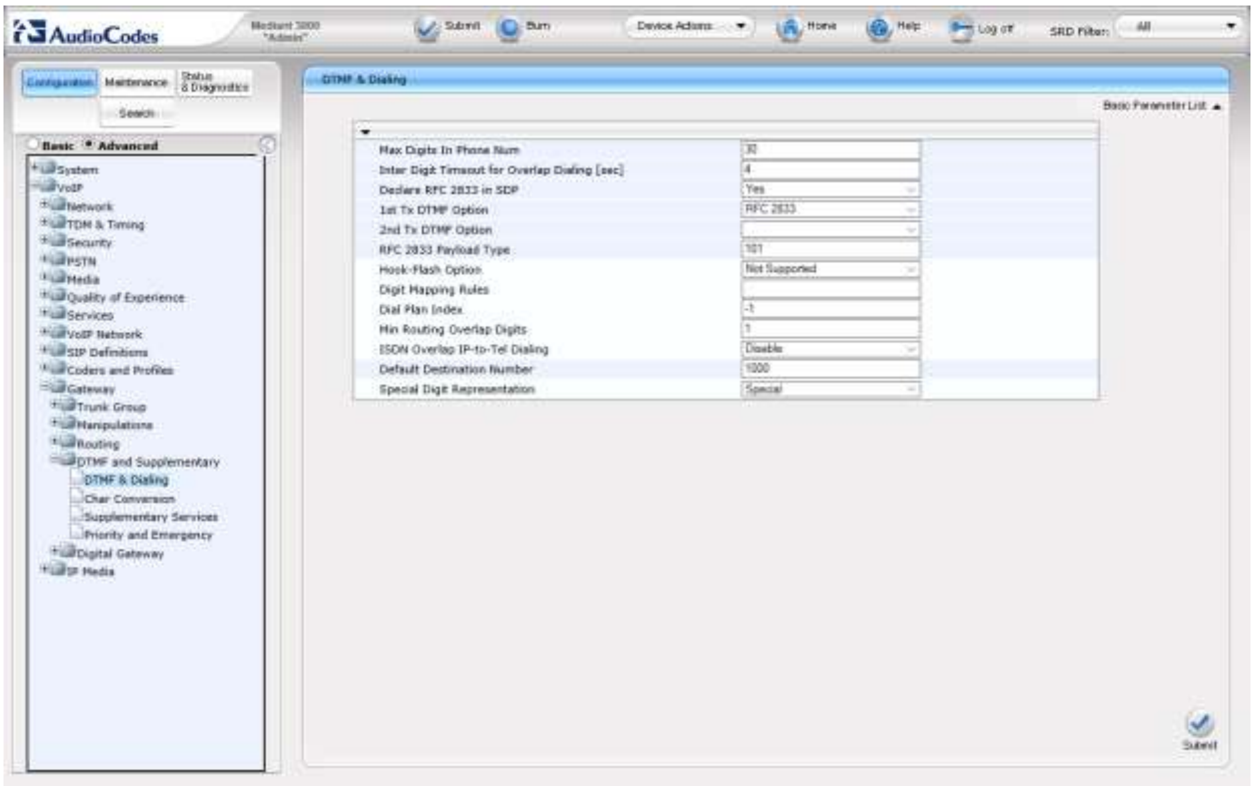


7.7.2. Configure DTMF and Dialing Parameters

Open the **DTMF & Dialing** page (**Configuration** tab → **VoIP** → **Gateway** → **DTMF and Supplementary** → **DTMF & Dialing**). Set the following values:

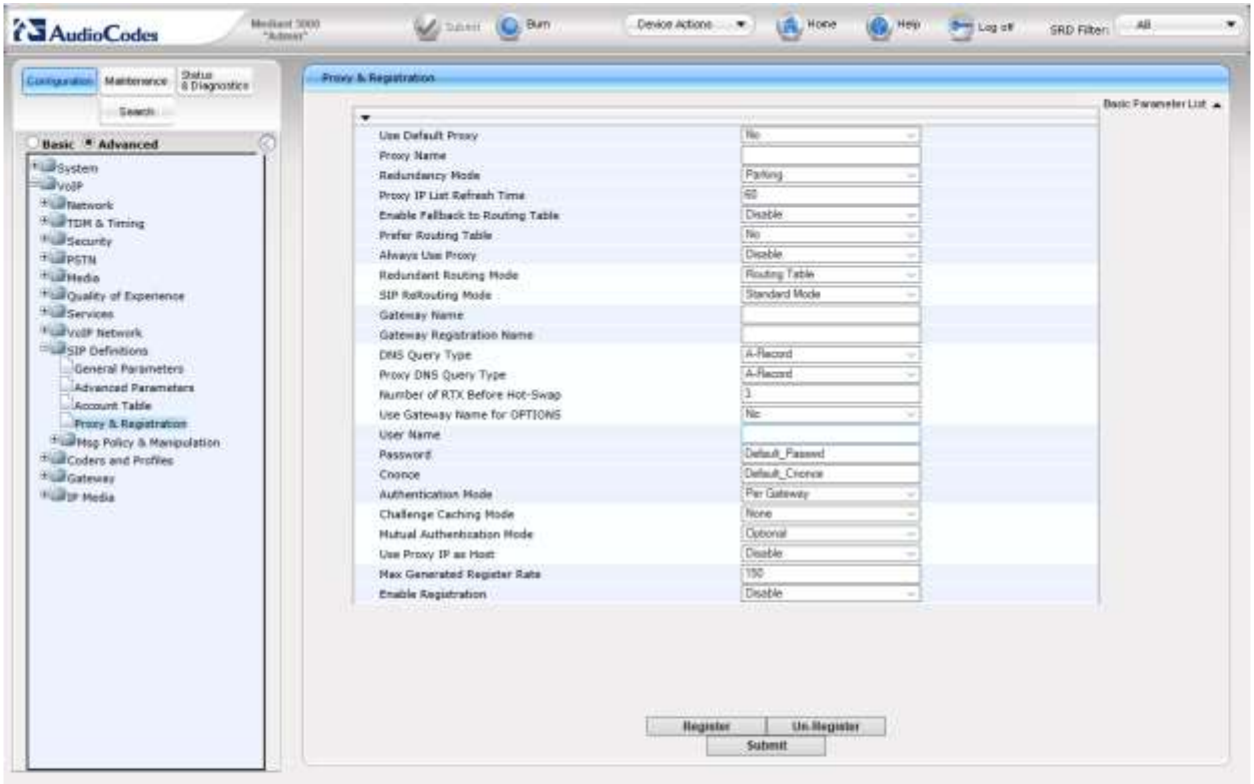
- **Declare RFC 2833 in SDP:** **Yes**
- **1st Tx DTMF Option:** **RFC 2833**

Click the **Submit** button to save changes. The figure below illustrates the **DTMF & Dialing** page.



7.7.3. Configure Proxy & Registration Parameters

Open the **Proxy & Registration** page (**Configuration** tab → **VoIP** → **SIP Definitions** → **Proxy & Registration**). Ensure that **Use Default Proxy** is set to **No** and **Enable Registration** is set to **Disable**. Click the **Submit** button to save changes. The screen below displays the **Proxy & Registration** page for the system used for the compliance test.

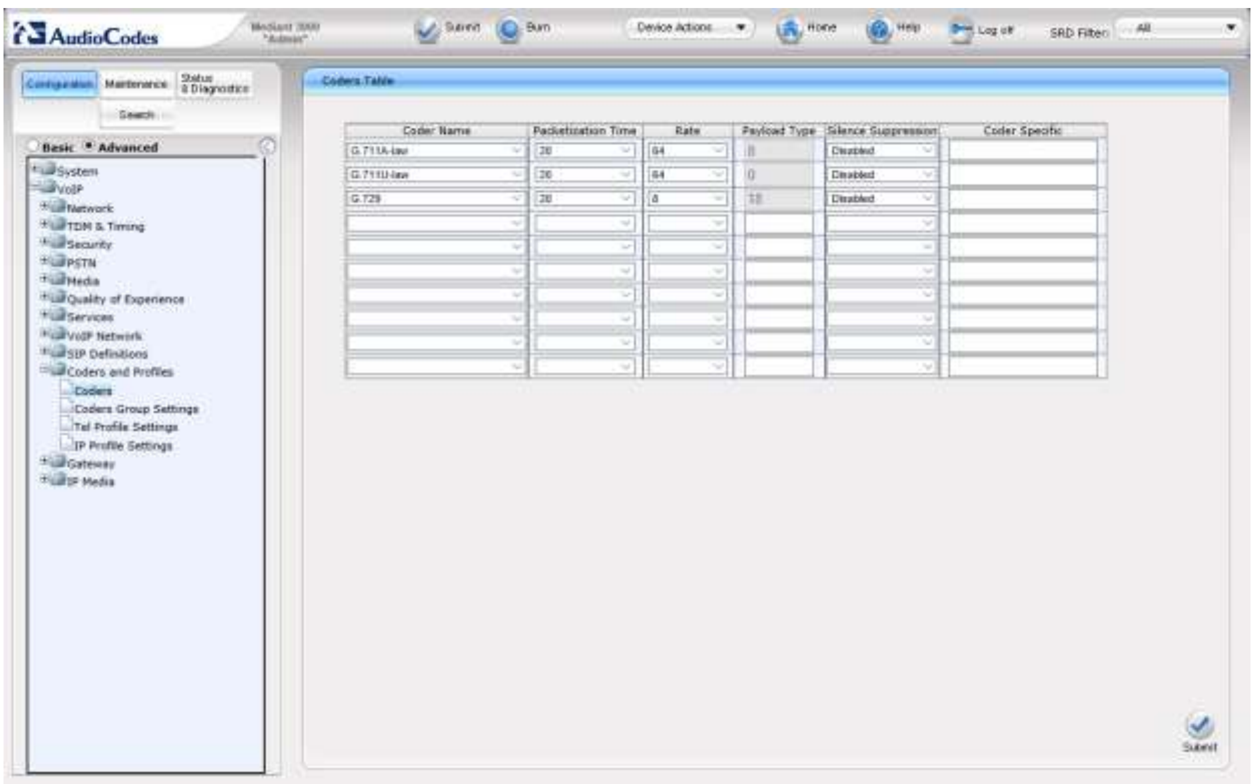


7.7.4. Configure Device's Coders

Open the **Coders** page (**Configuration** tab → **VoIP** → **Coders and Profiles** → **Coders**).

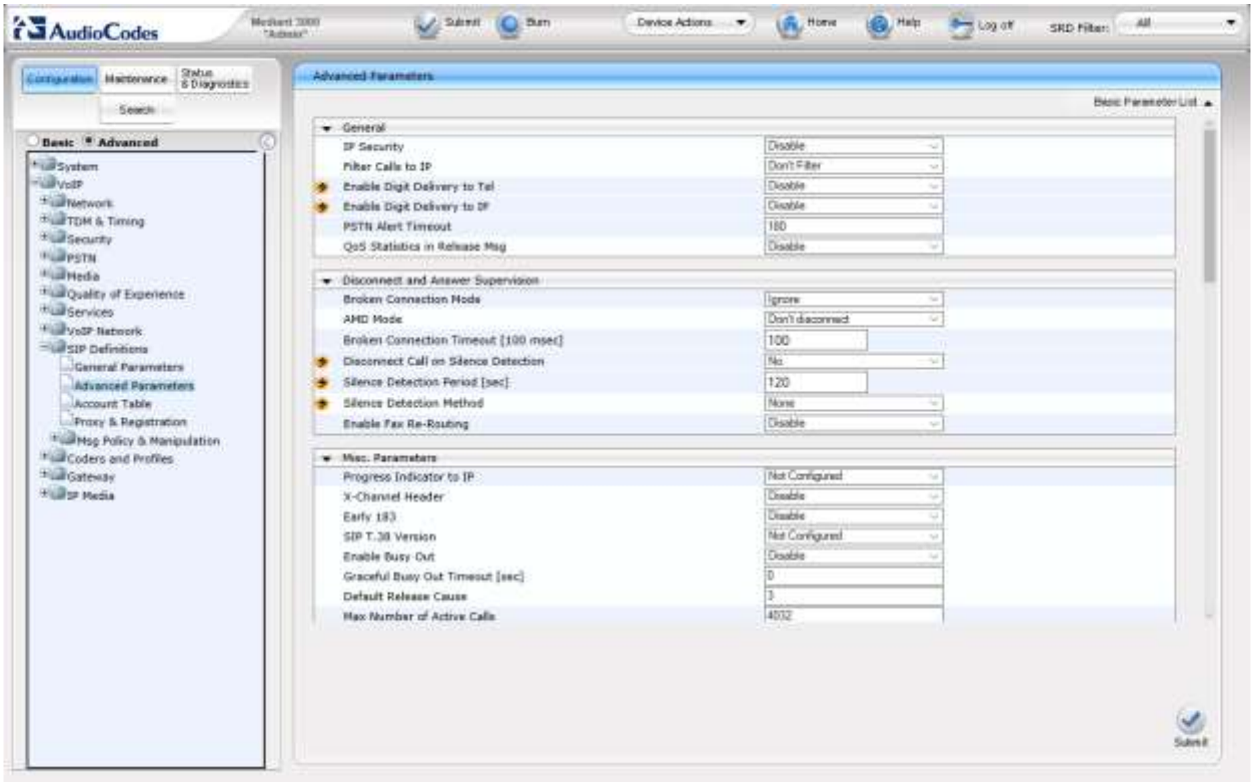
- From the **Coder Name** drop-down list, select the required coder
- From the **Packetization Time** drop-down list, select the packetization time (in msec) for the selected coder. The packetization time determines how many coder payloads are combined into a single RTP packet
- From the **Rate** drop-down 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

Click the **Submit** button to save changes. The following screen presented the codecs configured for the compliance test.



7.7.5. Configure Advanced General Protocol Parameters

Open the **Advanced Parameters** page (**Configuration** tab → **VoIP** → **SIP Definitions** → **Advanced Parameters**). This page allows the configuration of the defaults protocol parameters in case there is no mach on the previously configured protocol parameters. For the compliance test, only the **Broken Connection Mode** parameter was set to **Ignore**, other configurations may require special care. Click the **Submit** button to save changes.

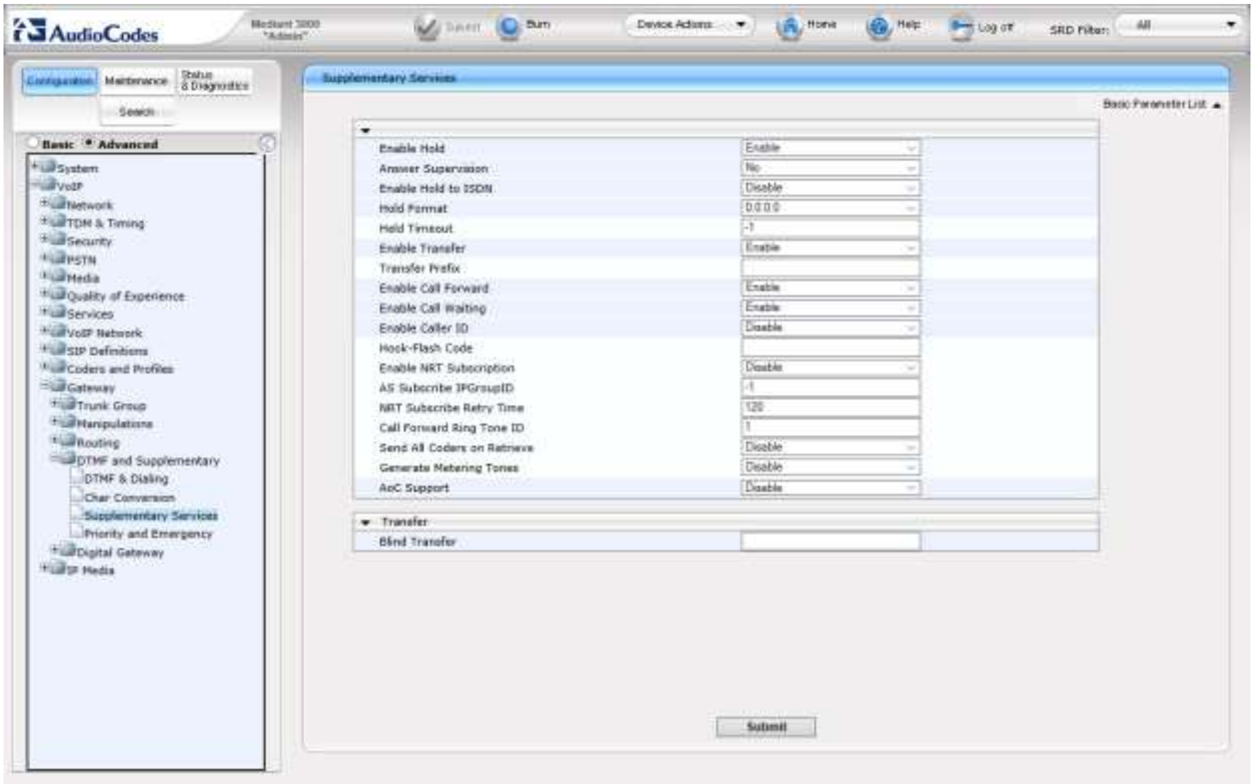


7.7.6. Configure Supplementary Services Parameters

Open the **Supplementary Services** page (**Configuration** tab → **VoIP** → **Gateway** → **DTMF and Supplementary** → **Supplementary Services**). Set the following parameters:

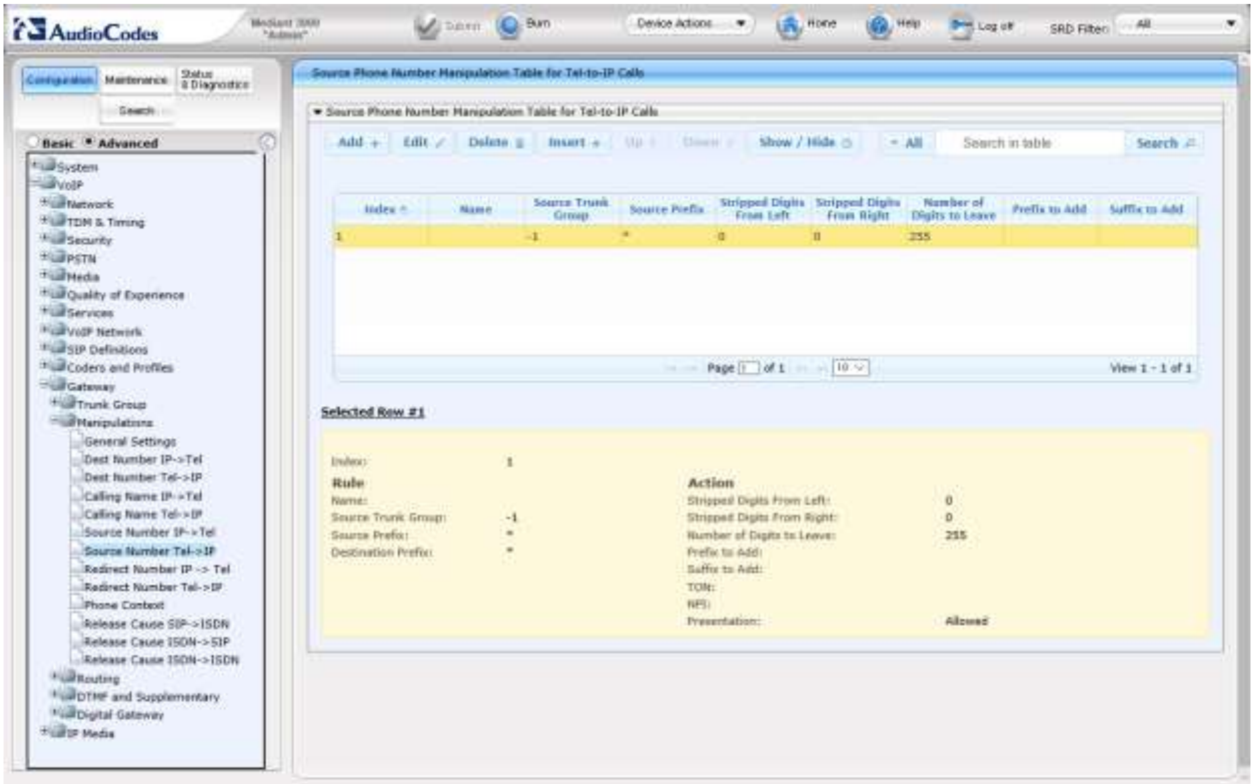
- **Enable Hold:** **Enable**
- **Enable Transfer:** **Enable**
- **Enable Call Forward:** **Enable**
- **Enable Call Waiting:** **Enable**

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



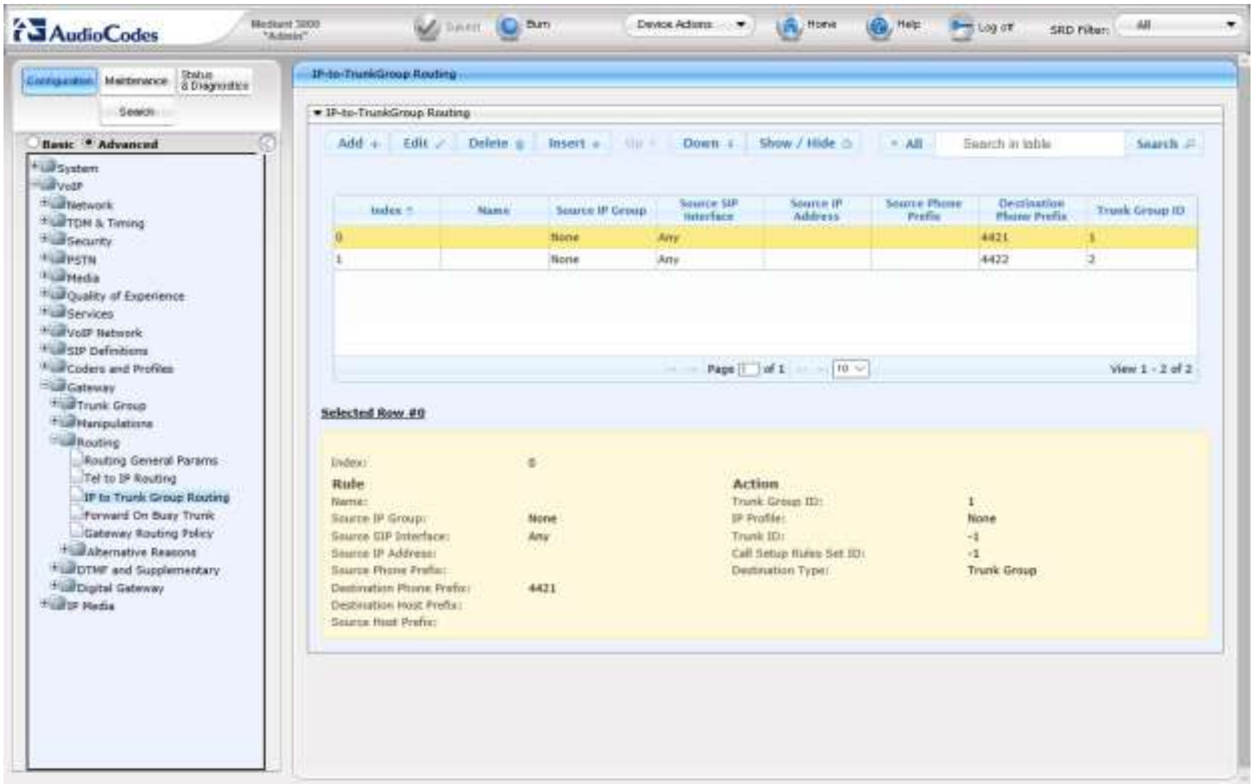
7.7.7. Configure Number Manipulation Tables

Open the required **Source Phone Number Manipulation Table for Tel-to-IP Calls** page (**Configuration** tab → **VoIP** → **Gateway** → **Manipulations** → **Source Number Tel→IP**). The relevant Manipulation table page is displayed. The screen below shows the manipulation rule for Tel-to-IP source phone number manipulation, used in the compliance test.



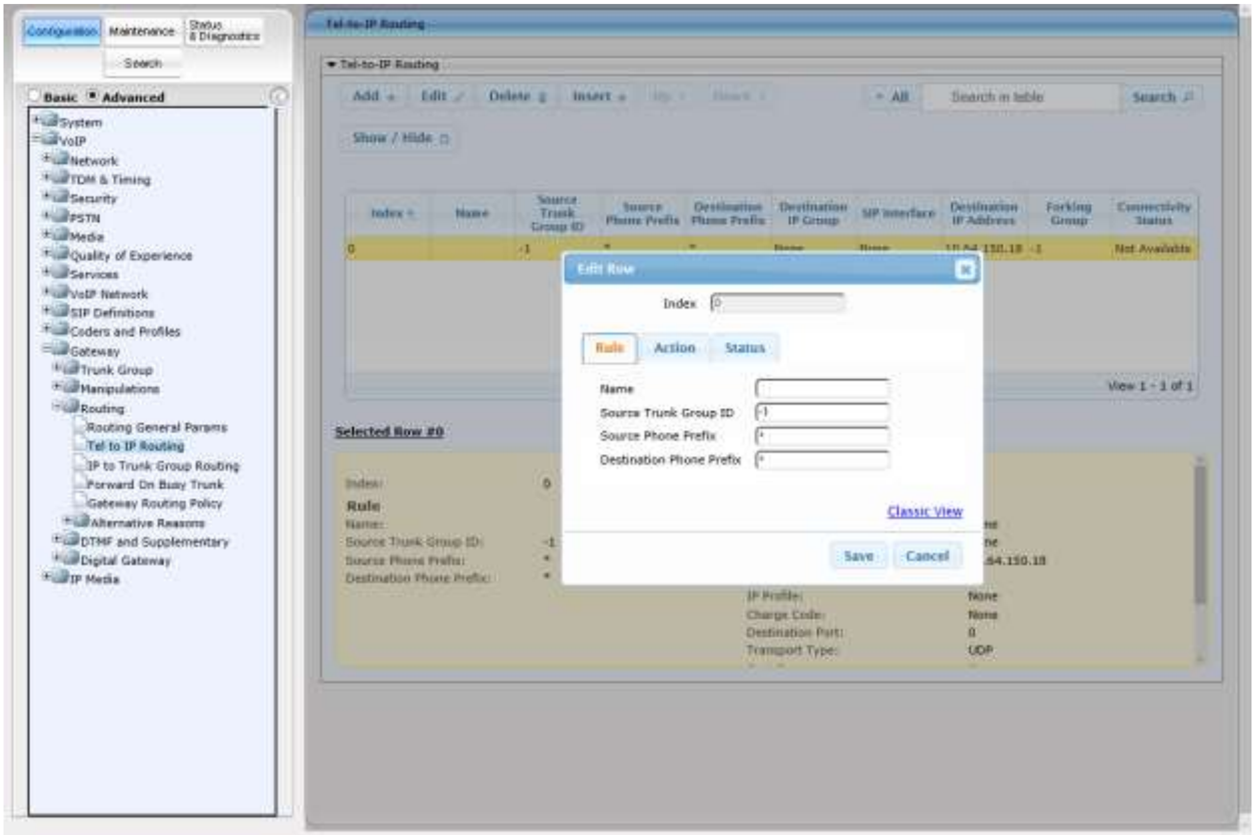
7.7.8. Configure IP to Trunk Group Routing Rules

Open the **IP to Trunk Group Routing Table** page (**Configuration** tab → **VoIP** → **Gateway** → **Routing** → **IP to Trunk Group Routing**). Configure the inbound IP routing rules. The screen below illustrates the Inbound IP Routing Table used in the compliance test.

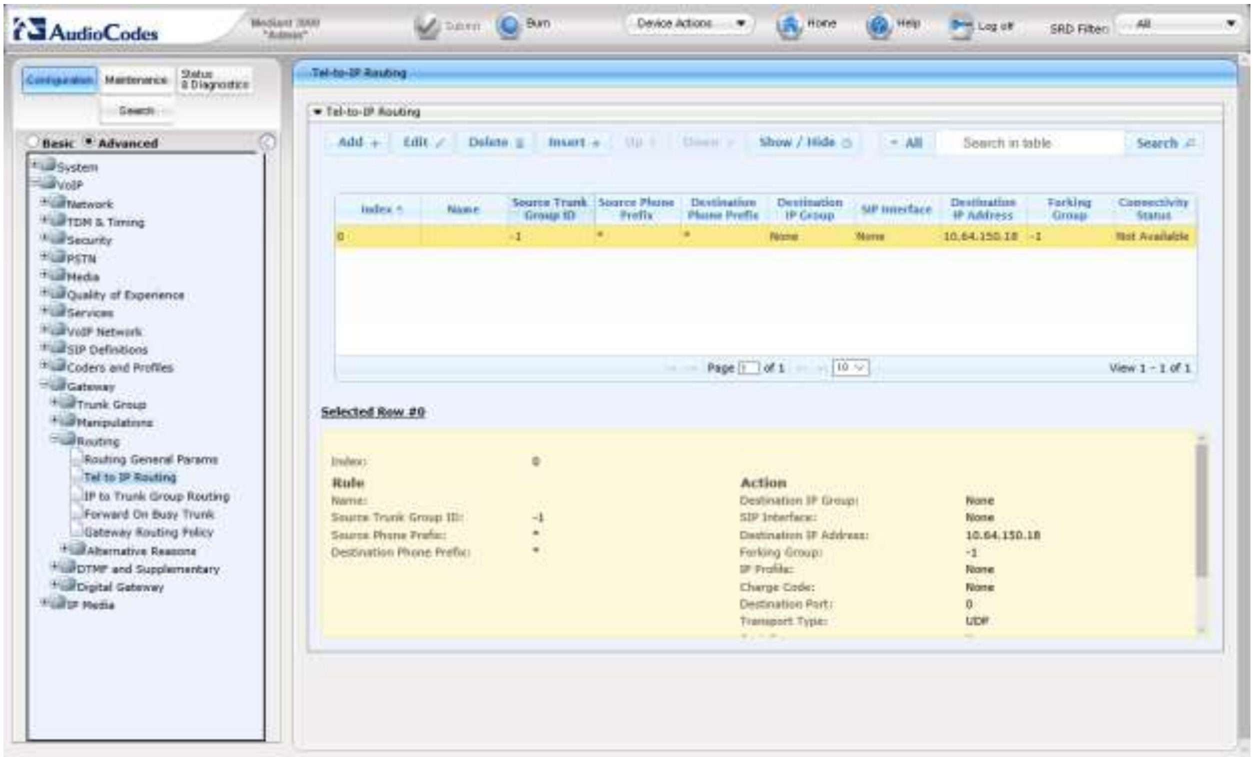


7.7.9. Configure Outbound IP Routing Rules

Open the **Tel to IP Routing** page (**Configuration** tab → **VoIP** → **Gateway** → **Routing** → **Tel to IP Routing**). Click the **Add** or **Edit** button to add or edit an index row.



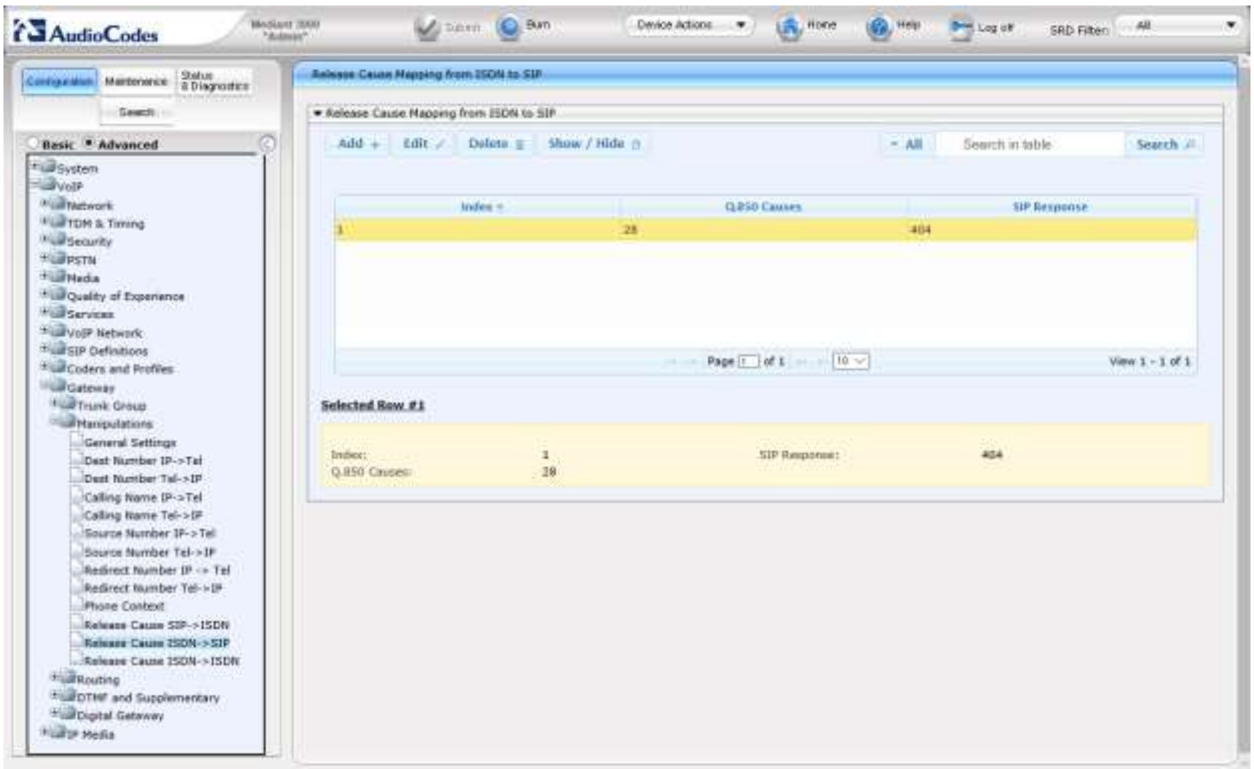
Configure the **Source Trunk Group ID** with the appropriate trunk number, **Destination Phone Prefix**, **Source Phone Prefix** with the appropriate patterns (i.e., *) and **Destination IP Address** with the IP Address of signalling interface of Session Manager (i.e., **10.64.150.18**). Set the **Transport Type** to the appropriate value (i.e, **UDP**). Click on the **Submit** button to save changes. The following screen illustrates the configuration done for the compliance test.



7.7.10. Configure Release Cause Mapping

Open the **Release Cause Mapping from ISDN to SIP** page (**Configuration** tab → **VoIP** → **Gateway** → **Manipulations** → **Release Cause ISDN -> SIP**).

In these Application Notes, mapping from **Q.850 Cause** value **28** is mapped into **SIP Response** message **404**, this was used to ensure the mapping of Invalid Number in the Q.850 was mapped to a SIP 404 for the appropriate interworking. Click the **Submit** button to save changes. The screen below illustrates the **Release Cause Mapping from ISDN to SIP** page.



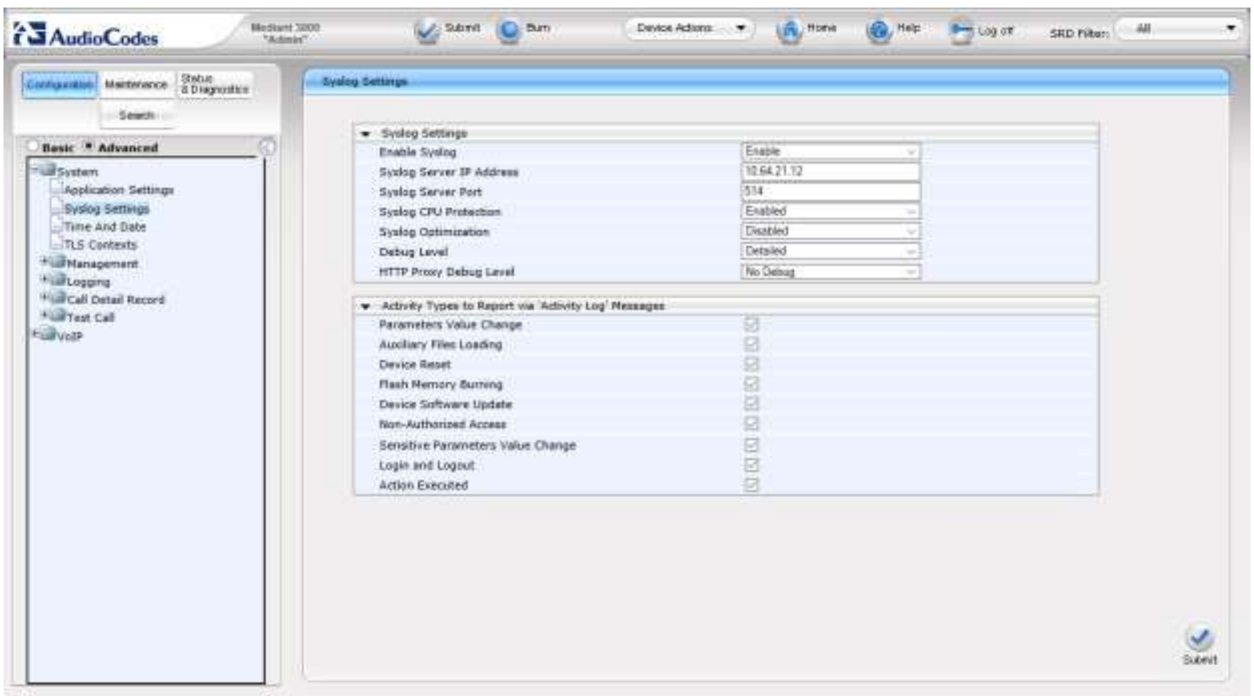
7.7.11. Configure Syslog Parameters for Debug Assistance

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 (i.e., **10.64.21.12**)
- **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 7.3 to save the configuration.*

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 signalling groups are in service by issuing the command status **signalling-group n** where **n** is the signalling group number.

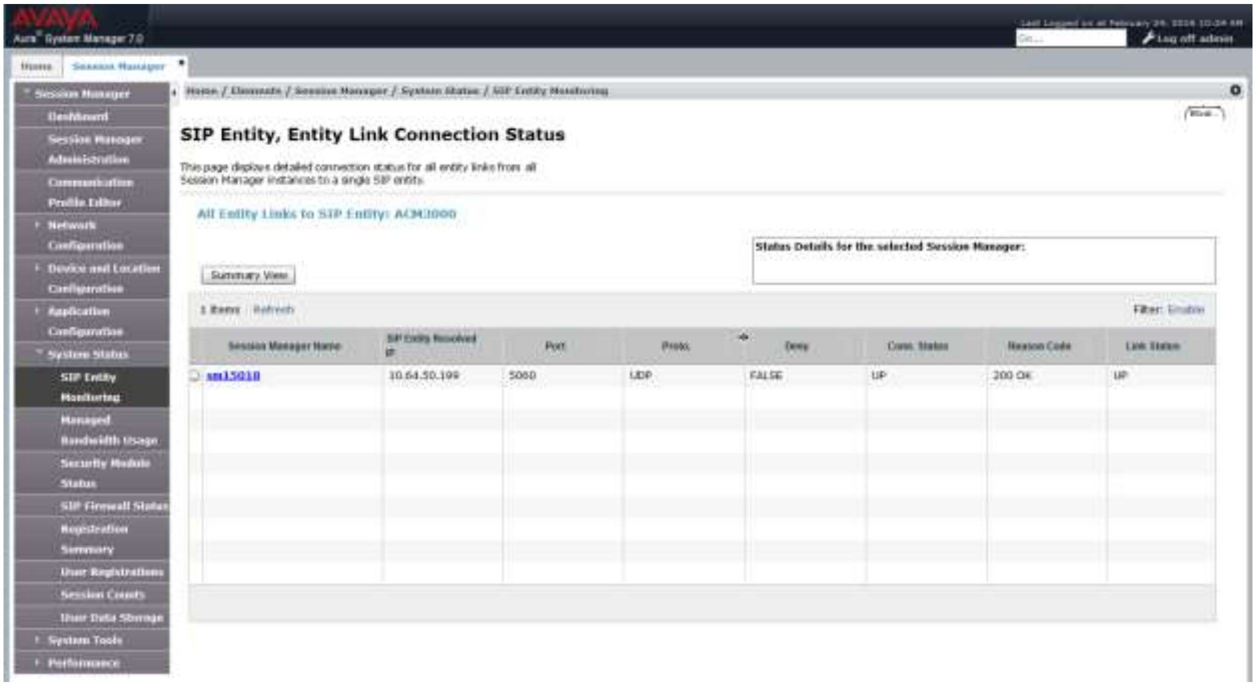
```
status signaling-group 2
                         STATUS
   SIGNALING GROUP

        Group ID: 2
      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.

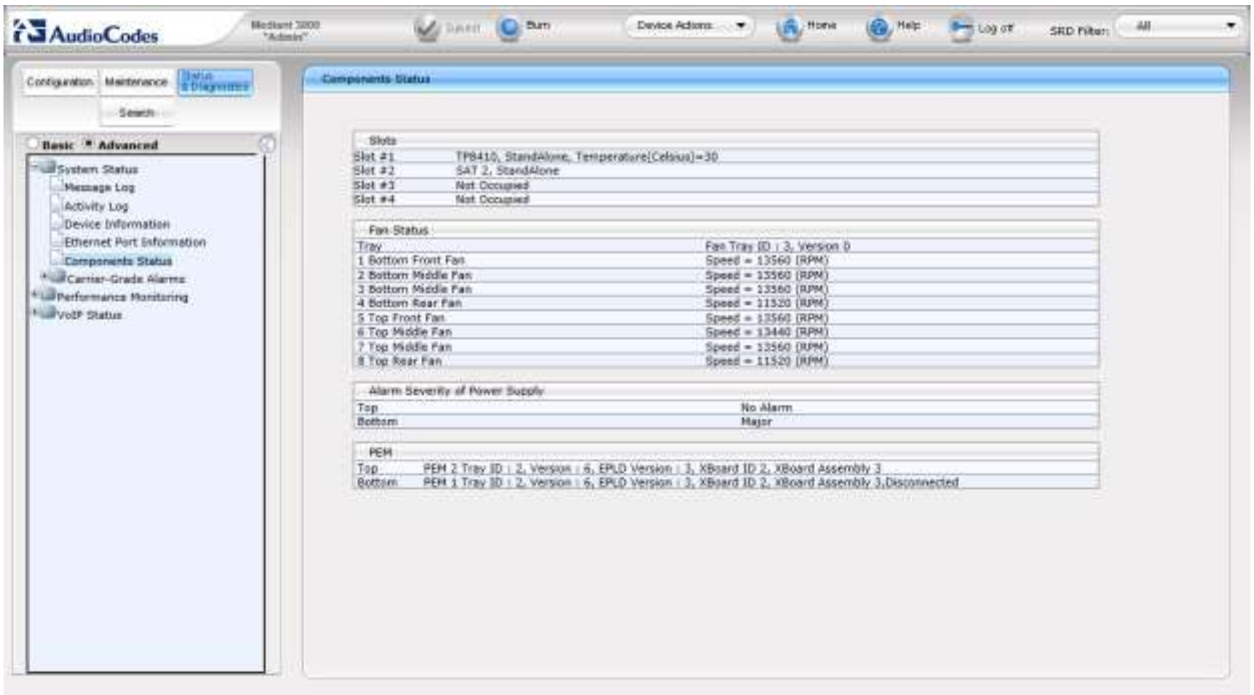


8.3. Utilizing Mediant 3000 Web Interface to Observe Status

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.

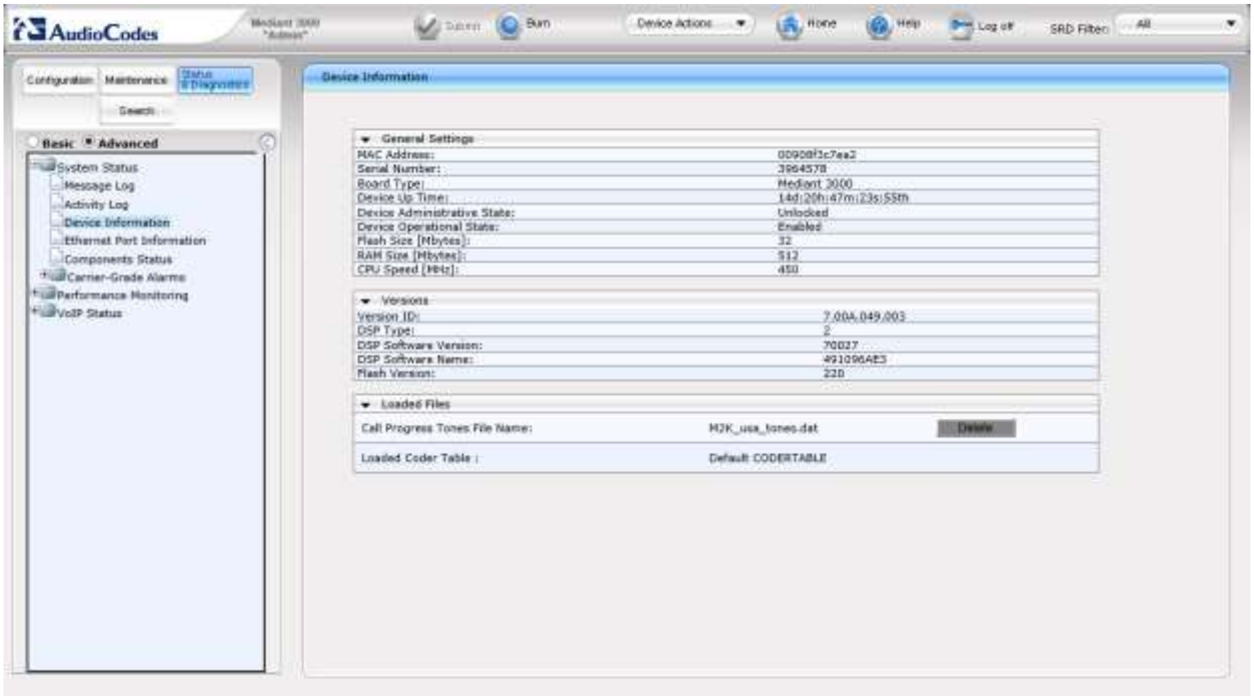
8.3.1. Device Status

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, 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.

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 6.8 November 2014 Document # LTRT-89725

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.049.003
;DSP Software Version: 491096AE3=> 700.27
;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
;PM_gwSBCTranscodingSessions is hidden but has non-default value
SyslogServerIP = 10.64.21.12
EnableSyslog = 1
ENABLEPARAMETERSMONITORING = 1
ActivityListToLog = 'pvc', 'afl', 'dr', 'fb', 'swu', 'naa', 'spc', 'll',
'ae'
;VpFileLastUpdateTime is hidden but has non-default value
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 = 1
```

```
TDMBusSpeed = 3
UdpPortSpacing = 10
EnterCpuOverloadPercent = 99
ExitCpuOverloadPercent = 95

[ControlProtocols Params]

AdminStateLockControl = 0
cpRecordCoder = 'PCMA'

[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]

TraceLevel_0 = 1
TraceLevel_1 = 0
TraceLevel_2 = 0
TraceLevel_3 = 0
TraceLevel_4 = 0
TraceLevel_5 = 0
TraceLevel_6 = 0
TraceLevel_7 = 0
TraceLevel_8 = 0
TraceLevel_9 = 0
TraceLevel_10 = 0
TraceLevel_11 = 0
TraceLevel_12 = 0
TraceLevel_13 = 0
TraceLevel_14 = 0
TraceLevel_15 = 0
ProtocolType_0 = 1
ProtocolType_1 = 1
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
FramingMethod_0 = b
FramingMethod_1 = b
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
LineCode_0 = 2
LineCode_1 = 2
LineCode_2 = 0
LineCode_3 = 0
LineCode_4 = 0
LineCode_5 = 0
LineCode_6 = 0
LineCode_7 = 0
LineCode_8 = 0
LineCode_9 = 0
LineCode_10 = 0
LineCode_11 = 0
LineCode_12 = 0
LineCode_13 = 0
LineCode_14 = 0
LineCode_15 = 0
CASProtocolEnable = 0

[SS7 Params]


[Voice Engine Params]

CallProgressTonesFilename = 'M2K_usa_tones.dat'
V22ModemTransportType = 0
V23ModemTransportType = 0
V32ModemTransportType = 0
V34ModemTransportType = 0
RFC2833TxPayloadType = 101
AnswerDetectorSilenceTime = 0
AnswerDetectorSensitivity = 0
EnergyDetectorQualityFactor = 0
EnergyDetectorThreshold = 0
```

```
[WEB Params]

LogoWidth = '145'
HTTPSCipherString = 'RC4:EXP'
;HTTPSPkeyFileName is hidden but has non-default value
;HTTPSCertFileName is hidden but has non-default value

[SIP Params]

GWDEBUGLEVEL = 5
;ISPRACKREQUIRED is hidden but has non-default value
ENABLEEARLYMEDIA = 1
DISCONNECTONBROKENCONNECTION = 0
ISFAXUSED = 1
SIPTRANSPORTTYPE = 1
BCHANNELNEGOTIATIONFORTRUNK_0 = 0
BCHANNELNEGOTIATIONFORTRUNK_1 = -1
BCHANNELNEGOTIATIONFORTRUNK_2 = -1
BCHANNELNEGOTIATIONFORTRUNK_3 = -1
BCHANNELNEGOTIATIONFORTRUNK_4 = -1
BCHANNELNEGOTIATIONFORTRUNK_5 = -1
BCHANNELNEGOTIATIONFORTRUNK_6 = -1
BCHANNELNEGOTIATIONFORTRUNK_7 = -1
BCHANNELNEGOTIATIONFORTRUNK_8 = -1
BCHANNELNEGOTIATIONFORTRUNK_9 = -1
BCHANNELNEGOTIATIONFORTRUNK_10 = -1
BCHANNELNEGOTIATIONFORTRUNK_11 = -1
BCHANNELNEGOTIATIONFORTRUNK_12 = -1
BCHANNELNEGOTIATIONFORTRUNK_13 = -1
BCHANNELNEGOTIATIONFORTRUNK_14 = -1
BCHANNELNEGOTIATIONFORTRUNK_15 = -1
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, 2, 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_ServerCipherString, TLSContexts_ClientCipherString,
TLSContexts_OcspEnable, TLSContexts_OcspServerPrimary,
TLSContexts_OcspServerSecondary, TLSContexts_OcspServerPort,
TLSContexts_OcspDefaultResponse;
```

```
TLSContexts 0 = "default", 0, "RC4:EXP", "ALL:!ADH", 0, 0.0.0.0, 0.0.0.0,
2560, 0;

[ \TLSContexts ]


[ IpProfile ]

FORMAT IpProfile_Index = IpProfile_ProfileName, IpProfile_IpPreference,
IpProfile_CodersGroupID, IpProfile_IsFaxUsed,
IpProfile_JitterBufMinDelay, IpProfile_JitterBufOptFactor,
IpProfile_IPDiffServ, IpProfile_SigIPDiffServ, IpProfile_SCE,
IpProfile_RTPRedundancyDepth, IpProfile_RemoteBaseUDPPort,
IpProfile_CNGmode, IpProfile_VxxTransportType, IpProfile_NSEMode,
IpProfile_IsDTMFUsed, IpProfile_PlayRBTone2IP,
IpProfile_EnableEarlyMedia, IpProfile_ProgressIndicator2IP,
IpProfile_EnableEchoCanceller, IpProfile_CopyDest2RedirectNumber,
IpProfile_MediaSecurityBehaviour, IpProfile_CallLimit,
IpProfile_DisconnectOnBrokenConnection, IpProfile_FirstTxDtmfOption,
IpProfile_SecondTxDtmfOption, IpProfile_RxDTMFOption,
IpProfile_EnableHold, IpProfile_InputGain, IpProfile_VoiceVolume,
IpProfile_AddIEInSetup, IpProfile_SBCExtensionCodersGroupID,
IpProfile_MediaIPVersionPreference, IpProfile_TranscodingMode,
IpProfile_SBCAllowedMediaTypes, IpProfile_SBCAllowedCodersGroupID,
IpProfile_SBCAllowedVideoCodersGroupID, IpProfile_SBCAllowedCodersMode,
IpProfile_SBCMediaSecurityBehaviour, IpProfile_SBCRFC2833Behavior,
IpProfile_SBCAlternativeDTMFMethod, IpProfile_SBCAssertIdentity,
IpProfile_AMDSensitivityParameterSuit, IpProfile_AMDSensitivityLevel,
IpProfile_AMDMaxGreetingTime, IpProfile_AMDMaxPostSilenceGreetingTime,
IpProfile_SBCDiversionMode, IpProfile_SBCHistoryInfoMode,
IpProfile_EnableQSIGTunneling, IpProfile_SBCFaxCodersGroupID,
IpProfile_SBCFaxBehavior, IpProfile_SBCFaxOfferMode,
IpProfile_SBCFaxAnswerMode, IpProfile_SbcPrackMode,
IpProfile_SBCSessionExpiresMode, IpProfile_SBCRemoteUpdateSupport,
IpProfile_SBCRemoteReinviteSupport,
IpProfile_SBCRemoteDelayedOfferSupport, IpProfile_SBCRemoteReferBehavior,
IpProfile_SBCRemote3xxBehavior, IpProfile_SBCRemoteMultiple18xSupport,
IpProfile_SBCRemoteEarlyMediaResponseType,
IpProfile_SBCRemoteEarlyMediaSupport, IpProfile_EnableSymmetricMKI,
IpProfile_MKISize, IpProfile_SBCEnforceMKISize,
IpProfile_SBCRemoteEarlyMediaRTP, IpProfile_SBCRemoteSupportsRFC3960,
IpProfile_SBCRemoteCanPlayRingback, IpProfile_EnableEarly183,
IpProfile_EarlyAnswerTimeout, IpProfile_SBC2833DTMFPayloadType,
IpProfile_SBCUserRegistrationTime, IpProfile_ResetSRTPStateUponRekey,
IpProfile_AmdMode, IpProfile_SBCReliableHeldToneSource,
IpProfile_GenerateSRTPKeys, IpProfile_SBCPlayHeldTone,
IpProfile_SBCRemoteHoldFormat, IpProfile_SBCRemoteReplacesBehavior,
IpProfile_SBCSDPPtimeAnswer, IpProfile_SBCPreferredPTime,
IpProfile_SBCUseSilenceSupp, IpProfile_SBCRTPRedundancyBehavior,
IpProfile_SBCPlayRBTToTransferee, IpProfile_SBCRTCPMode,
IpProfile_SBCJitterCompensation,
IpProfile_SBCRemoteRenegotiateOnFaxDetection,
```

```
IpProfile_JitterBufMaxDelay,
IpProfile_SBCUserBehindUdpNATRegistrationTime,
IpProfile_SBCUserBehindTcpNATRegistrationTime,
IpProfile_SBCSDPHandleRTCPAttribute,
IpProfile_SBCRemoveCryptoLifetimeInSDP, IpProfile_SBCIceMode,
IpProfile_SBCRTCPMux, IpProfile_SBCMediaSecurityMethod,
IpProfile_SBCHandleXDetect, IpProfile_SBCRTCPFeedback,
IpProfile_SBCRemoteRepresentationMode, IpProfile_SBCKeepVIAHeaders,
IpProfile_SBCKeepRoutingHeaders, IpProfile_SBCKeepUserAgentHeader,
IpProfile_SBCRemoteMultipleEarlyDialogs,
IpProfile_SBCRemoteMultipleAnswersMode, IpProfile_SBCDirectMediaTag,
IpProfile_SBCAdaptRFC2833BWToVoiceCoderBW;
IpProfile 1 = "IpProfile_1", 1, 0, 1, 10, 10, 46, 40, 0, 0, 0, 0, 0, 0,
0, 1, 1, -1, 1, 0, 0, -1, 0, 4, -1, 1, 1, 0, 0, "", -1, 0, 0, "", -1, -1,
0, 0, 0, 0, 0, 0, 8, 300, 400, 0, 0, 0, -1, 0, 0, 1, 3, 0, 2, 2, 1, 0, 0,
1, 0, 1, 0, 0, 0, 0, 0, 1, 0, 0, 0, 0, 0, 0, 1, 0, 0, 0, 0, 0, 0, 0, 0,
0, 0, 0, 0, 300, -1, -1, 0, 0, 0, 0, 0, 0, 0, -1, -1, -1, -1, -1, 0, "",
0;

[ \IpProfile ]


[ 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, 4032, 46319, 1, "",
"";

[ \CpMediaRealm ]


[ SBCRoutingPolicy ]

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

[ \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, "", 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 0 = "ProxySet_0", 0, 60, 0, 0, "DefaultSRD", 0, "", -1, -1, "",
"SIPInterface_0", "", "", "", "", "";

[ \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",
"", 1, "", -1, -1, -1, 0, 0, "", 0, -1, -1, "", "", "$1$gQ==", 0, "", "",
"", 0, "", "", 0, 0, "", 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 = "", "*", "10.64.150.18", "*", "", "", 0, "", 0, -1, "", "", -
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, 31, "4421000", "", 0, 255;
TrunkGroup 1 = 2, 1, 1, 31, "4422000", "", 1, 255;

[ \TrunkGroup ]


[ SourceNumberMapTel2Ip ]

FORMAT SourceNumberMapTel2Ip_Index =
SourceNumberMapTel2Ip_ManipulationName,
SourceNumberMapTel2Ip_DestinationPrefix,
SourceNumberMapTel2Ip_SourcePrefix, SourceNumberMapTel2Ip_NumberType,
SourceNumberMapTel2Ip_NumberPlan, SourceNumberMapTel2Ip_RemoveFromLeft,
SourceNumberMapTel2Ip_RemoveFromRight,
SourceNumberMapTel2Ip_LeaveFromRight, SourceNumberMapTel2Ip_Prefix2Add,
SourceNumberMapTel2Ip_Suffix2Add,
SourceNumberMapTel2Ip_IsPresentationRestricted,
SourceNumberMapTel2Ip_SrcTrunkGroupID;
```

```
SourceNumberMapTel2Ip 1 = "", "*", "*", 255, 255, 0, 0, 255, "", "", 0, -
1;

[ \SourceNumberMapTel2Ip ]


[ 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 = "", "4421", 1, "", "", "", "", "", "", "Any", -1, -1, 0;
PstnPrefix 1 = "", "4422", 2, "", "", "", "", "", "", "Any", -1, -1, 0;

[ \PstnPrefix ]


[ CauseMapIsdn2Sip ]

FORMAT CauseMapIsdn2Sip_Index = CauseMapIsdn2Sip_IsdnReleaseCause,
CauseMapIsdn2Sip_SipResponse;
CauseMapIsdn2Sip 1 = 28, 404;

[ \CauseMapIsdn2Sip ]


[ 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 0 = 1, 1, 255, "", "", "", 255, "", 0, 0;
TrunkGroupSettings 1 = 2, 1, 255, "", "", "", 255, "", 0, 0;

[ \TrunkGroupSettings ]


[ CodersGroup0 ]

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

```
[ \CodersGroup0 ]


[ RoutingRuleGroups ]

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

[ \RoutingRuleGroups ]


[ GwRoutingPolicy ]

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

[ \GwRoutingPolicy ]


[ 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 ]
```

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.