Avaya Solution & Interoperability Test Lab

Application Notes for Configuring Aura™ Session Manager and Avaya Aura™ Communication Manager Feature Server with AudioCodes Mediant 3000 Gateway to access E1 PSTN - Issue 1.0

Abstract

These Application Notes describe the procedure to configure an Enterprise network built on Avaya Aura™ Session Manager, and Avaya Aura™ Communication Manager Feature Server to interoperate with AudioCodes Mediant 3000 Gateway to access E1 PSTN using SIP trunking.

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 present a sample configuration for an Enterprise network consisting on Avaya Aura™ Session Manager and Avaya Aura™ Communication Manager Feature Server as SIP infrastructure to access the PSTN with AudioCodes Mediant 3000 Gateway using SIP. The AudioCodes Mediant 3000 is a carrier-grade VoIP gateway that supports both media and signaling in a single chassis. It provides any-to-any voice network connectivity and can deliver SIP services into legacy PRI, CAS, and SS7 networks, as well as IP-to-IP transcoding and multimedia border element functions, such as SIP mediation for network edge applications. Its compact 2U high-density design features integrated SS7 termination across multiple gateways, GUI-based management, and software licensing for in-service capacity expansion.

1.1. AudioCodes Mediant 3000

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.

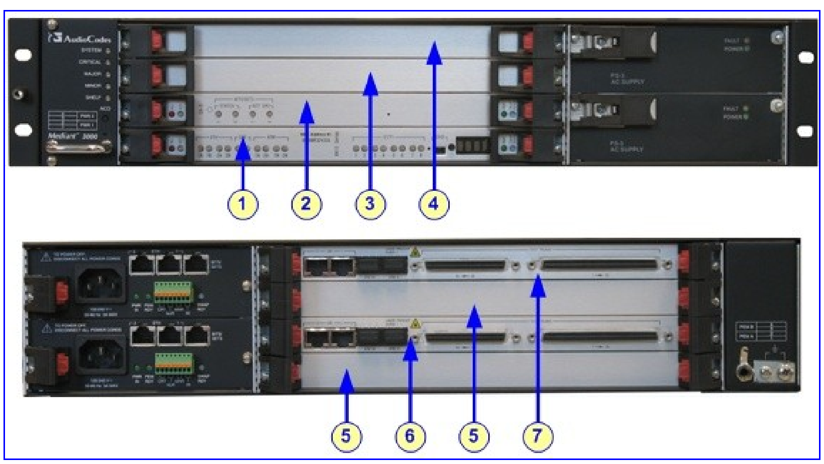


Figure 1: Front and Rear Panel Slot Assignment for AudioCodes Mediant 3000 Simplex with 8410 Blades

Legend:

1. Slot 1 front panel: 8410 blade (active blade for AudioCodes Mediant 3000 HA only).
2. Slot 2 front panel: SA/M3K blade (active blade for AudioCodes Mediant 3000 HA only).
3. Slot 3 front panel: Standby (redundant) 8410 blade (applicable only to AudioCodes Mediant 3000 HA). In Simplex mode, this slot is covered with a blank panel.

4. Slot 4 front panel: Standby (redundant) Alarm and Status blade (applicable only to AudioCodes Mediant 3000 HA). In Simplex mode, this slot is covered with a blank panel.
5. Blank panels covering unoccupied slots.
6. Slot 2 rear panel: RTM-8410 providing PSTN E1/T1 (Trunks 1 to 42, or 1 to 16) and dual Gigabit Ethernet interfaces.
7. Slot 4 rear panel: RTM-8410 providing PSTN E1/T1 (Trunks 43 to 84) interfaces and Gigabit Ethernet interfaces.

1.2. Interoperability Compliance Testing

The primary focus of testing is to verify SIP trunking interoperability between an Avaya Aura™ SIP-based network and AudioCodes Mediant 3000 Gateway using SIP. Test cases are selected to exercise a sufficiently broad segment of functionality to have a reasonable expectation of interoperability in production configurations.

Basic Interoperability:

- PSTN calls delivered via the AudioCodes Mediant 3000 to an Enterprise endpoint
- PSTN calls sent via the AudioCodes Mediant 3000 from an Enterprise endpoint
- Calling with various Avaya SIP telephone models
- Verify ITU-T codecs: G.711A G.711MU G.729A G.729B support
- Various PTSN dialing plans including national and international calling, toll-free, operator, directory assistance and direct inward dialed calling
- SIP transport using UDP and TCP

Advanced Interoperability:

- 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
- EC500 for Avaya Aura™ Communication Manager

1.3. Support

Technical Support on AudioCodes Mediant 3000 Gateway can be obtained through email notification to support@audiocodes.com

2. 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 PSTN. The Mediant 3000 is managed by using the web interface, other administration capabilities are available, refer to **[15-18]** for additional information. Session Manager, with its SM-100 (Security Module) network interface, routes the 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 System Manager via the management network interface.

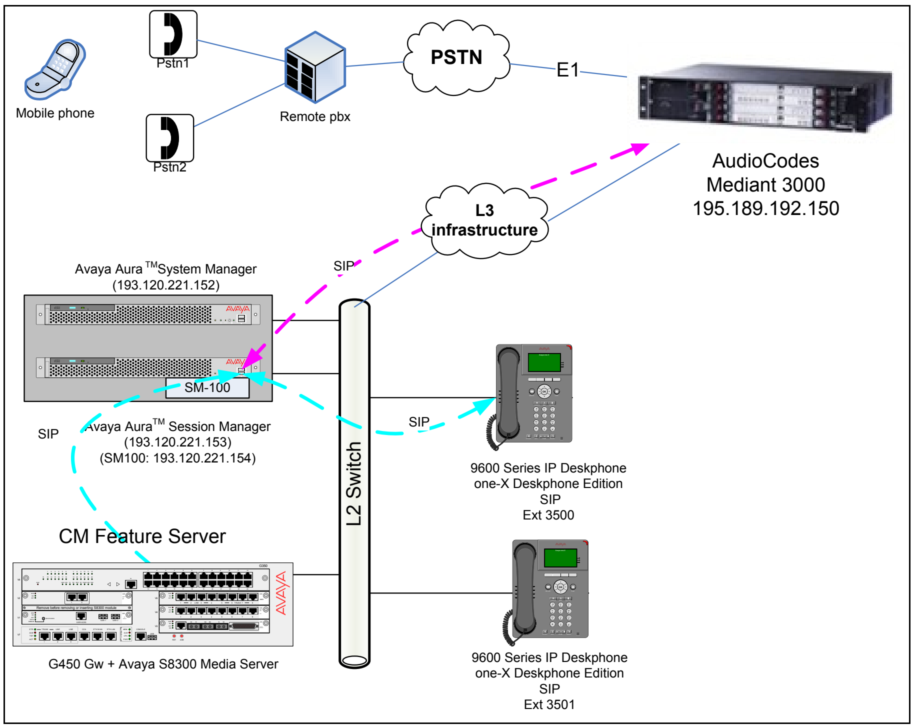


Figure 2: Sample configuration for Avaya Aura™ Communication Manager Feature Server and Avaya Aura™ Session Manager with AudioCodes Mediant 3000 using SIP Trunking

For the sample configuration shown in **Figure 1**, Session Manager runs on an Avaya S8510 Server and Communication Manager Feature Server runs on an Avaya S8300D inside an Avaya G450 Media Gateway. For the Communication Manager Feature Server, the results in these Application Notes are applicable to other supported Communication Manager Server and Media Gateway combinations. These Application Notes will focus on the configuration of the SIP trunks and call routing. Detailed administration of the endpoint telephones will not be described. Refer to the appropriate documentation in **Section 10**.

3. Equipment and Software Validated

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

Avaya Product / Hardware Platform	**Software Version**
Avaya Aura™ Session Manager on Avaya S8510 Server	Avaya Aura™ Session Manager 5.2 5.2.1.1.521012 – 5.2.1 SP1
Avaya Aura™ System Manager Template running on Avaya System Platform S8510 Server	Avaya Aura™ System Manager 5.2 5.2.1.0.521001 - 05_02_GA_01_Dec10
Avaya Aura™ System Platform on Avaya S8510 Server	Avaya Aura™ System Platform Version 1.1.1.0.2
Avaya Aura™ Communication Manager – Feature Server – Avaya Media Server S8300C	Avaya Aura™ Communication Manager R015x.02.1.016.4 – patch 18250 (SP3)
Avaya Media Gateway G450	Firmware 30.13.2
Avaya IP Telephones: 9630 (SIP) 9620 (SIP)	Avaya one-X™ Deskphone SIP 2.5.0
AudioCodes	
Product /Hardware Platform	**Software Version**
AudioCodes Mediant 3000 chassis equipped with: SA/M3K - Alarm, Status and Synchronization blade TP8410 blades – Trunk Pack RTM-8410, Rear module, proving the I/O connections to the supported interfaces (Gigabit Ethernet and DS1 PSTN).	Mediant 3000 TP 8410 based software 6.00A.014.005 Firmware load: TP8410_SIP_F6.00A.014.005.cmp

4. Configure Avaya Aura™ Communication Manager Feature Server

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 **References [10]** and **[13].** The procedures include the following areas:

- Verify Avaya Aura™ 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

4.1. 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  10
                                OPTIONAL FEATURES

IP PORT CAPACITIES                                            USED
                     Maximum Administered H.323 Trunks: 100   0
           Maximum Concurrently Registered IP Stations: 450   0
             Maximum Administered Remote Office Trunks: 0     0
Maximum Concurrently Registered Remote Office Stations: 0     0
              Maximum Concurrently Registered IP eCons: 0     0
  Max Concur Registered Unauthenticated H.323 Stations: 100   0
                        Maximum Video Capable Stations: 100   0
                    Maximum Video Capable IP Softphones: 100   0
                       Maximum Administered SIP Trunks: 100   50
```

4.2. 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.

Note: This feature poses significant security risk and must be used with caution. As an alternative, the trunk-to-trunk feature can be implemented using Class Of Restriction or Class Of Service levels.

```
change system-parameters features                              Page   1 of  18
                            FEATURE-RELATED SYSTEM PARAMETERS
                               Self Station Display Enabled? y
                                    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/Tone on Hold: none
              Music (or Silence) on Transferred Trunk Calls? no
                       DID/Tie/ISDN/SIP Intercept Treatment: attd
    Internal Auto-Answer of Attd-Extended/Transferred Calls: transferred
                  Automatic Circuit Assurance (ACA) Enabled? n
```

4.3. 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 **193.120.221.180** are entered as **Name** and **IP Address** for the signaling in Communication Manager running on the Avaya S8300 Server. In addition, **SM100** and **193.120.221.154** are entered for Session Manager.

```
change node-names ip                                          Page   1 of   2
                                  IP NODE NAMES
    Name                  IP Address

default                 0.0.0.0
procr                   193.120.221.180
sm100                   193.120.221.154
```

4.4. 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  19
                               IP NETWORK REGION
  Region: 1
Location: 1       Authoritative Domain: avaya.com
    Name: Enterprise
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                  RTCP Reporting Enabled? y
 Call Control PHB Value: 46      RTCP MONITOR SERVER PARAMETERS
        Audio PHB Value: 46       Use Default Server Parameters? y
        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. A list of supported by the interoperability compliance testing is presented in **Section 1.2.** The ITU G.711A-law is described here. Configure the IP Codec Set as follows:

- **Audio Codec** Set **G.711A**

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:
 3:
```

4.5. Administer SIP Trunks with Avaya Aura™ Session Manager

In the test configuration, since Communication Manager acts as a Feature Server in this case, trunks with Session Manager must be IMS enabled. Two SIP trunks are needed for the configuration presented in these notes: one for calls with Mediant 3000 and another one for calls within the Enterprise. To administer a SIP Trunk on Communication Manger, two intermediate steps are required: the creation of a signaling group and a trunk group

4.5.1. Add SIP Signaling Group for Calls within the Enterprise

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**
- **IMS Enabled:** **y**
- **Near-end Node Name:** **procr**
- **Far-end Node Name:** Session Manager node name from **Section 5.3** i.e.**sm100**
- **Near-end Listen Port:** **5061**
- **Far-end Listen Port:** **5061**
- **Far-end Domain:** **avaya.com**
- **DTMF over IP:** **rtp-payload**
- **Direct IP-IP Audio Connections:** **y**

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

 Group Number: 1              Group Type: sip
                        Transport Method: tls
  IMS Enabled? y
     IP Video? n


   Near-end Node Name: procr                  Far-end Node Name: sm100
 Near-end Listen Port: 5061                 Far-end Listen Port: 5061
                                         Far-end Network Region: 1
Far-end Domain: avaya.com

                                            Bypass If IP Threshold Exceeded? n
Incoming Dialog Loopbacks: eliminate                 RFC 3389 Comfort Noise? n
         DTMF over IP: rtp-payload             Direct IP-IP Audio Connections? y
Session Establishment Timer(min): 3                    IP Audio Hairpinning? n
         Enable Layer 3 Test? n                     Direct IP-IP Early Media? n
H.323 Station Outgoing Direct Media? n        Alternate Route Timer(sec): 30
```

4.5.2. Configure a SIP Trunk Group for Calls within the Enterprise

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**. with-SessionManager**)
- **TAC:** An available trunk access code (i.e. **101**)
- **Service Type:** **tie**
- **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 licensed verified in **Section 4.1**)

```
add trunk-group 1                                           Page   1 of  21
                               TRUNK GROUP

Group Number: 1                   Group Type: sip          CDR Reports: y
  Group Name: with-SessionManager       COR: 1        TN: 1        TAC: 101
   Direction: two-way       Outgoing Display? n
 Dial Access? n                                Night Service:
Queue Length: 0
Service Type: tie                    Auth Code? n

                                                       Signaling Group: 1
                                                     Number of Members: 20
```

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
```

4.5.3. Add SIP Signaling Group for Mediant 3000

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**
- **IMS Enabled:** **y**
- **Near-end Node Name:** **procr**
- **Far-end Node Name:** Session Manager node name from **Section 4.3** i.e. **sm100**
- **Near-end Listen Port:** **5061**
- **Far-end Listen Port:** **5061**
- **Far-end Domain:** Leave it blank
- **DTMF over IP:** **rtp-payload**
- **Direct IP-IP Audio Connections:** **y**

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

 Group Number: 3              Group Type: sip
                        Transport Method: tls
  IMS Enabled? y
     IP Video? n


   Near-end Node Name: procr                   Far-end Node Name: sm100
 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? n                    Direct IP-IP Early Media? n
H.323 Station Outgoing Direct Media? n       Alternate Route Timer(sec): 30
```

4.5.4. Configure a SIP Trunk Group for Mediant 3000

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**. OUTSIDE CALL**)
- **TAC:** An available trunk access code (i.e. **103**)
- **Service Type:** **tie**
- **Signaling Group:** The number of the signaling group associated (i.e. **3**)
- **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 licensed verified in **Section 4.1**)

Note: The number of members determines how many simulataneous calls can be processed by the trunk through Session Manager.

```
add trunk-group 3                                                  Page   1 of  21
                                TRUNK GROUP

Group Number: 3                    Group Type: sip           CDR Reports: y
  Group Name: OUTSIDE CALL                  COR: 1       TN: 1      TAC: 103
   Direction: two-way       Outgoing Display? n
 Dial Access? n                                  Night Service:
Queue Length: 0
Service Type: tie                 Auth Code? n

                                              Signaling Group: 3
                                             Number of Members: 30
```

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

```
add trunk-group 3                                                  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
```

4.6. Configure Route Patterns

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

4.6.1. Route Pattern for Enterprise Calls

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

- **Pattern Name:** A descriptive name (i.e. **toSessionManager**)
- **Grp No:** The trunk group number from **Section 4.5.2**
- **FRL:** Enter a level that allows access to this trunk, with **0** being least restrictive

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

     BCC VALUE  TSC CA-TSC    ITC BCIE Service/Feature PARM  No. Numbering
LAR
    0 1 2 M 4 W     Request                                 Dgts Format
                                                         Subaddress
 1: y y y y y n  n            unre                                    none
 2: y y y y y n  n            rest                                    none
none
```

4.7. 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 **4**-digit extension (**Ext Len**) beginning with **35** (**Ext Code**) will result in a **4**-digit calling number (**Total Len**). The calling party number will be in the SIP "From" header.

```
change private-numbering 0
                           NUMBERING - PRIVATE FORMAT

Ext Ext              Trk        Private          Total
Len Code             Grp(s)     Prefix           Len
 4  35                                           4     Total Administered: 1
                                                          Maximum Entries: 540
```

4.8. Administer Dial Plan and AAR analysis

Configure the dial plan for dialing 4-digit extensions beginning with **30** to stations registered with Communication Manager Feature Server (not shown in these Application Notes). Use the **change dialplan analysis** command to define **Dialed String 350** as an **aar Call Type**.

```
change dialplan analysis                                   Page   1 of  12
                             DIAL PLAN ANALYSIS TABLE
                                  Location:  all          Percent Full:    2

       Dialed   Total  Call     Dialed   Total  Call      Dialed   Total  Call
       String   Length Type     String   Length Type      String   Length Type
     1            3    dac
     30           4    aar
     35           4    ext
     9            1    fac
     *            1    fac
```

Use the **change aar analysis n** command where **n** is the dial string pattern to configure an **aar** entry for **Dialed String 30** (Extensions on Communication Manager Feature Server**)** to use **Route Pattern 1** (defined in **Section 4.6.1**).

```
change aar analysis 0                                       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
    30                    4    4      1         aar          n
    35                    4    4      1         aar          n
```

4.9. Administer ARS Analysis

This section provides sample Auto Route Selection (ARS) used for routing calls with dialed digits beginning with **0** corresponding to national numbers accessible via the Mediant 3000. Use the **change ars analysis 0** 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. Submit these changes.

- **Dialed String:** Dialed prefix digits to match on, in this case **0**
- **Total Min:** Minimum number of digits, in this case **3**
- **Total Max:** Maximum number of digits, in this case **25**
- **Route Pattern:** The route pattern number from **Section 4.6.1** i.e. **1**
- **Call Type:** **pubu**

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

```
change ars analysis 0                                          Page   1 of   2
                             ARS DIGIT ANALYSIS TABLE
                                   Location:  all         Percent Full:    1

          Dialed           Total     Route     Call   Node   ANI
          String          Min  Max  Pattern    Type   Num    Reqd
    0                     3    25    1         pubu          n
```

4.10. Administer Feature Access Code

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

```
change feature-access-codes                                    Page   1 of   8
                               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: 9
    Auto Route Selection (ARS) - Access Code 1: *     Access Code 2:
                 Automatic Callback Activation:        Deactivation:
```

4.11. Save Changes

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

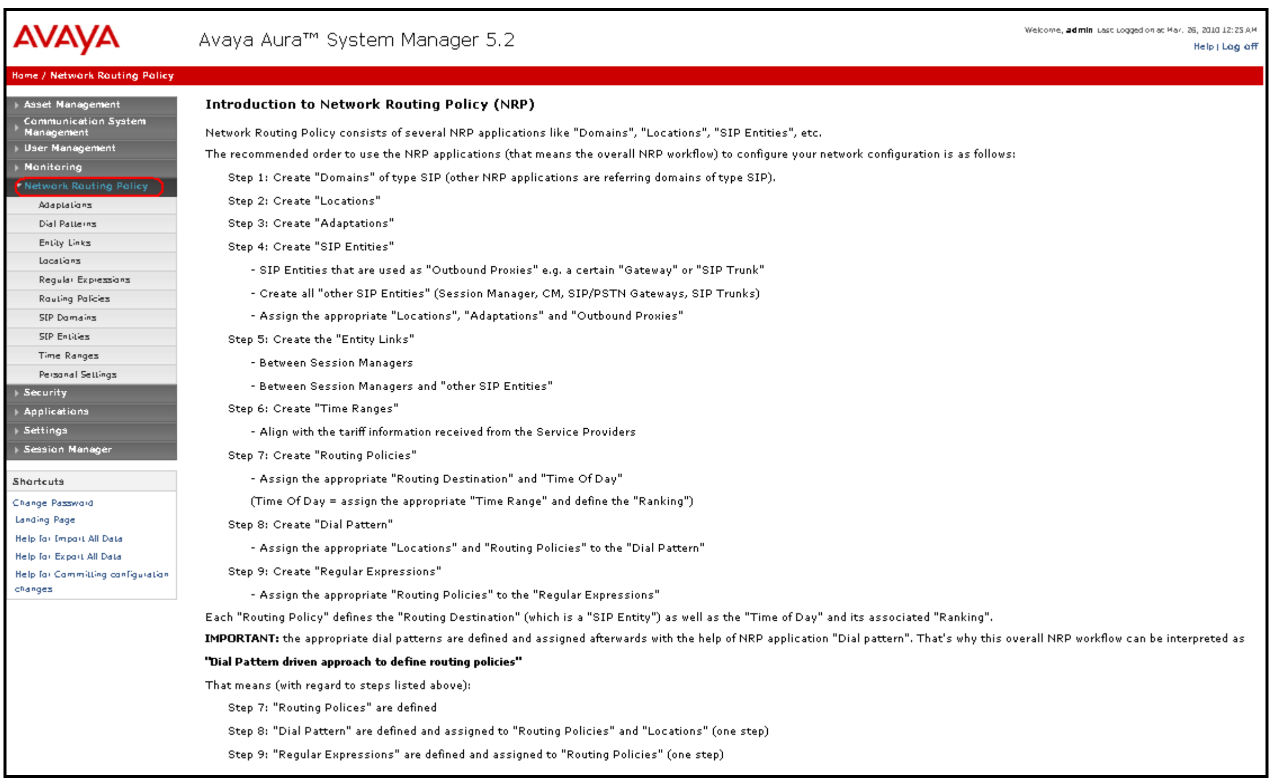
```
save translation
                               SAVE TRANSLATION
           Command Completion Status                              Error Code
           Success                                                  0
```

5. Configure Avaya Aura^{TM} Session Manager

This section provides the procedures for configuring Session Manager, assuming it has been installed and licensed as described in **Reference [3]**. The procedures include adding the following items:

- Specify SIP Domain
- Add Locations
- Add Adaptations
- Add SIP Entities
- Add Entity Links
- Add Routing Policies
- Add Dial Patterns
- Add Session Manager
- Add Avaya Aura^{TM} Communication Manager as Feature Server
- Add Users for SIP Phones

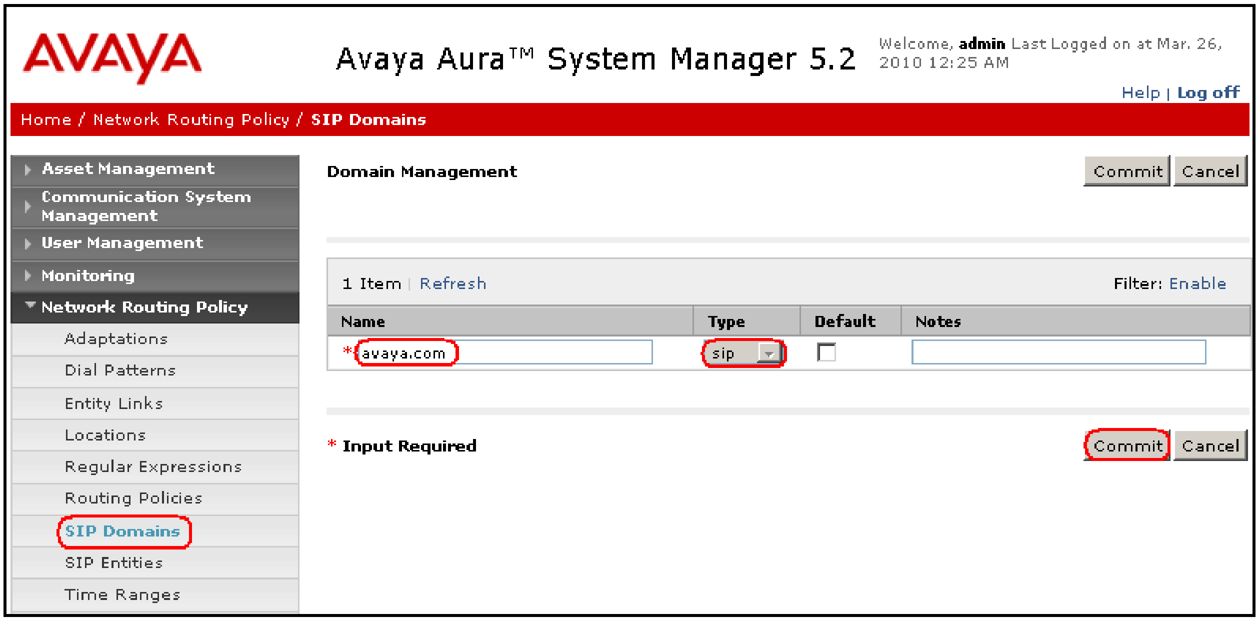
Configuration is accomplished by accessing the browser-based GUI of System Manager, using the URL **http://<ip-address>/SMGR**, where **<ip-address>** is the IP address of System Manager. Log in with the appropriate credentials and accept the Copyright Notice. The menu shown below is displayed. Expand the **Network Routing Policy** Link on the left side as shown.



5.1. Specify SIP Domain

Add the SIP domain for which the communications infrastructure will be authoritative. Do this by selecting **SIP Domains** on the left and clicking the **New** button on the right. 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)



5.2. 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 Feature Server and Mediant 3000 Gateway. To add a location, select **Locations** on the left and click on the **New** button on the right. The following screen will then be shown. Fill in the following:

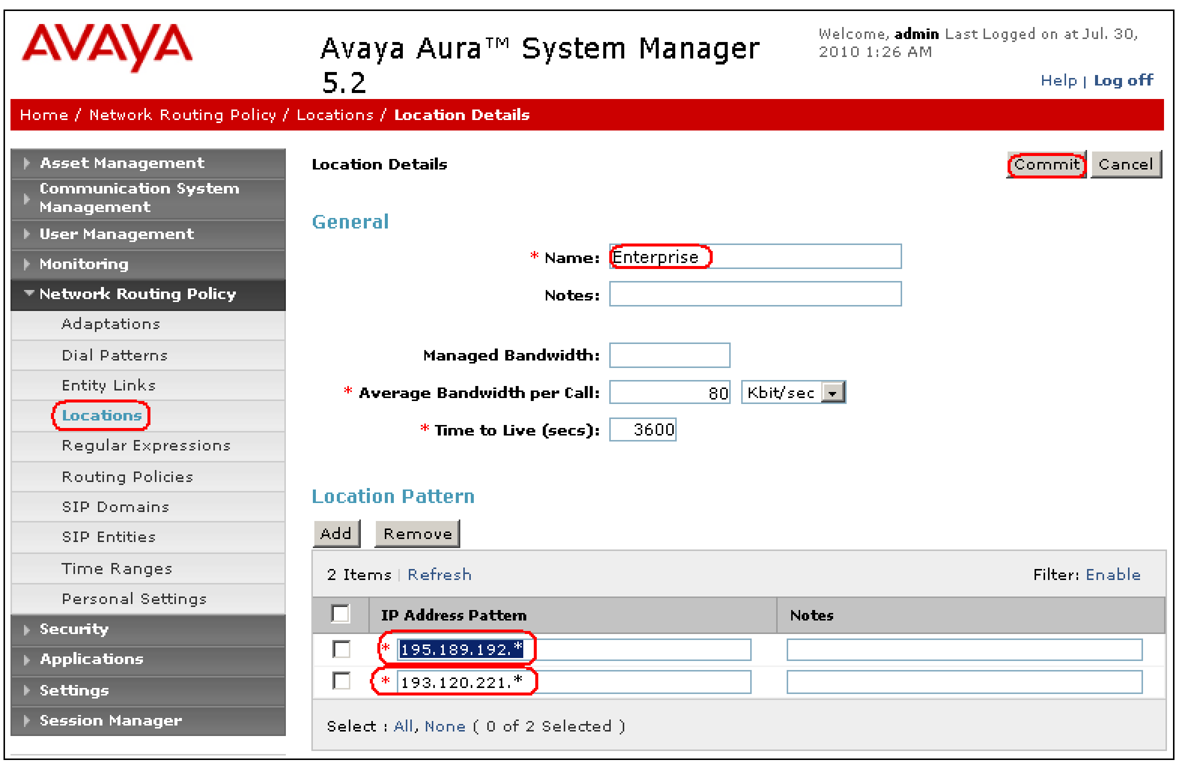
Under **General**:

- **Name:** A descriptive name
- **Notes:** Descriptive text (optional)
- **Managed Bandwidth:** Leave the default or customize as described in **[5]**

Under **Location Pattern**:

- **IP Address Pattern:** A pattern used to logically identify the location. In these Application Notes, the pattern selected defined the networks involved e.g. **193.120.221.*** for referring the Enterprise network and **195.189.192.*** for IP network where the Mediant 3000 Gateway resides.
- **Notes:** Descriptive text (optional)

The screen below shows addition of the **Enterprise** location, which includes all the components of the compliance environment. Click **Commit** to save.



5.3. 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. Alternatively the numbering plan translation can be implemented in the Mediant 3000 Gateway. Note that the **Digit Conversion for Outgoing Calls from SM** will modify the P-AI field in the SIP invite, requiring the Mediant 3000 privacy setting to be configured as described in **Section 6.5.2**. To add an adaptation, under the **Network Routing Policy** 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**:

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

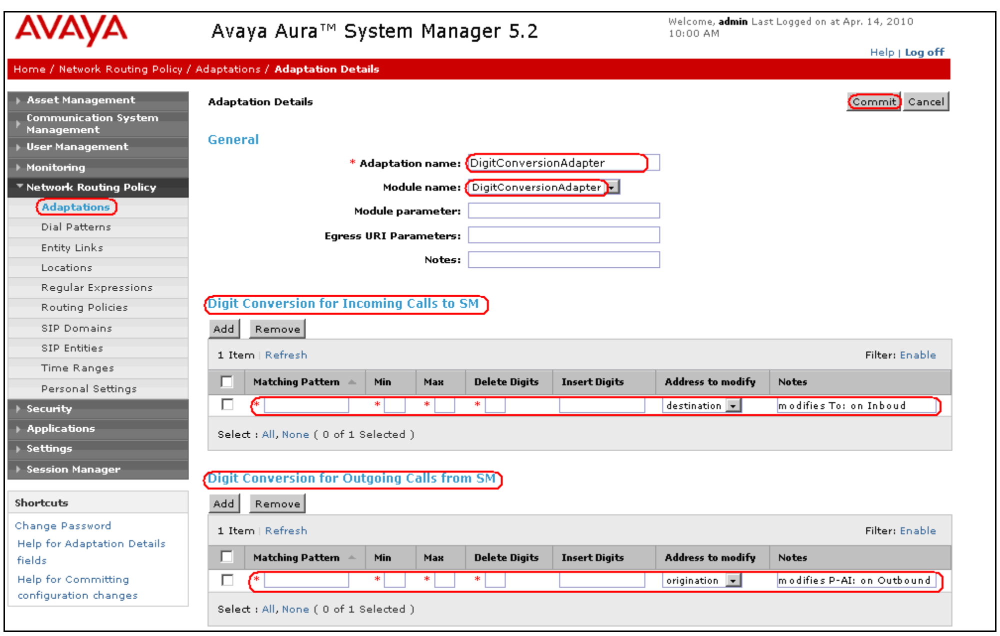
Under **Digit Conversion for Incoming Calls to SM**:

- **Matching Pattern**: The dialed number from the PSTN
- **Min/Max**: Minimum/Maximum number of digits
- **Delete Digits**: Digits to be deleted
- **Insert Digits**: Digit to be added
- **Address to modify**: Select **destination**

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

- **Matching Pattern**: The dialed number from enterprise network
- **Min/Max**: Minimum/ Maximum number of digits
- **Delete Digits**: Digits to be deleted
- **Insert Digits**: Digit to be added
- **Address to modify**: Select **origination**

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



5.4. Add SIP Entities

A SIP Entity must be added for Session Manager and for each SIP-based telephony system supported by it using SIP trunks. In the sample configuration, a SIP Entity is added for the Session Manager the Proc interface for the Communication Manager Feature Server and the Mediant 3000 Gateway IP interface.

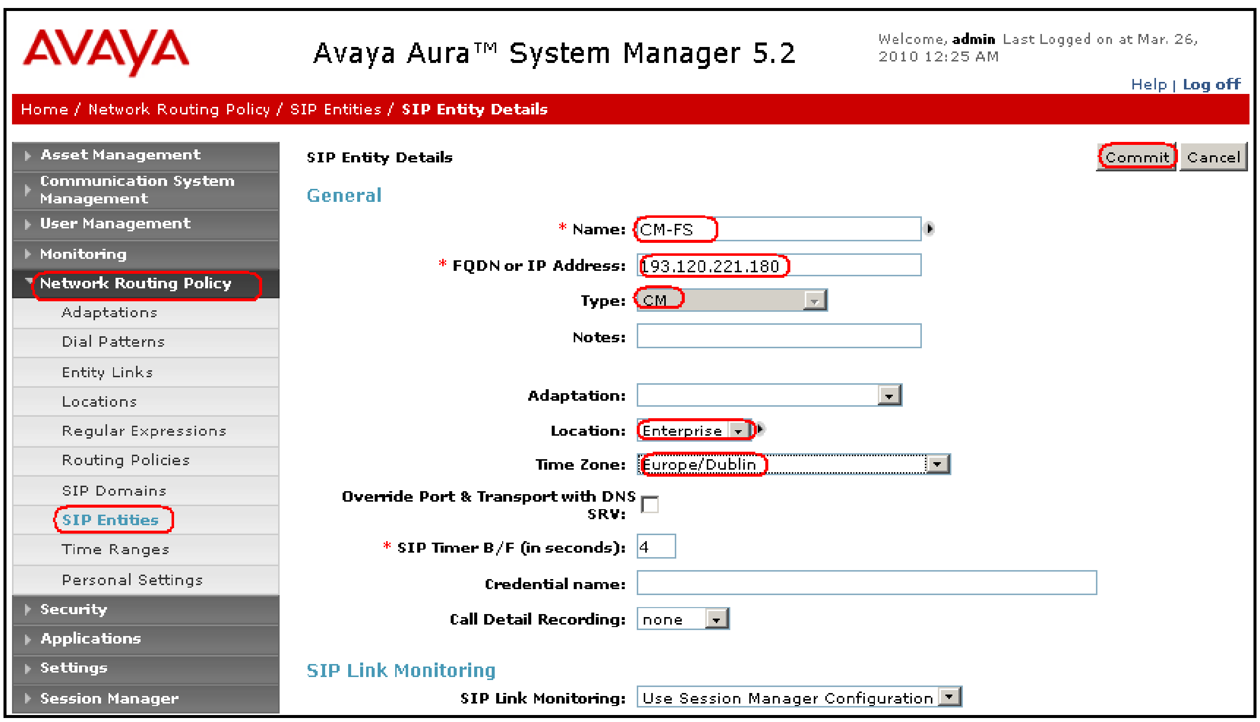
5.4.1. Adding Avaya Aura™ Communication Manager Feature Server SIP Entity

To add a SIP Entity, navigate **Network Routing Policy → SIP Entities** on the left and click on the **New** button on the right.

Under **General**:

- **Name:** A descriptive name (i.e. **CM-FS**)
- **FQDN or IP Address:** IP address of the Proc interface of S8300 Server, i.e. **193.120.221.180**
- **Type:** Select **CM**
- **Location:** Select one of the locations defined previously i.e. **Enterprise**
- **Time Zone:** Time zone for this entity

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



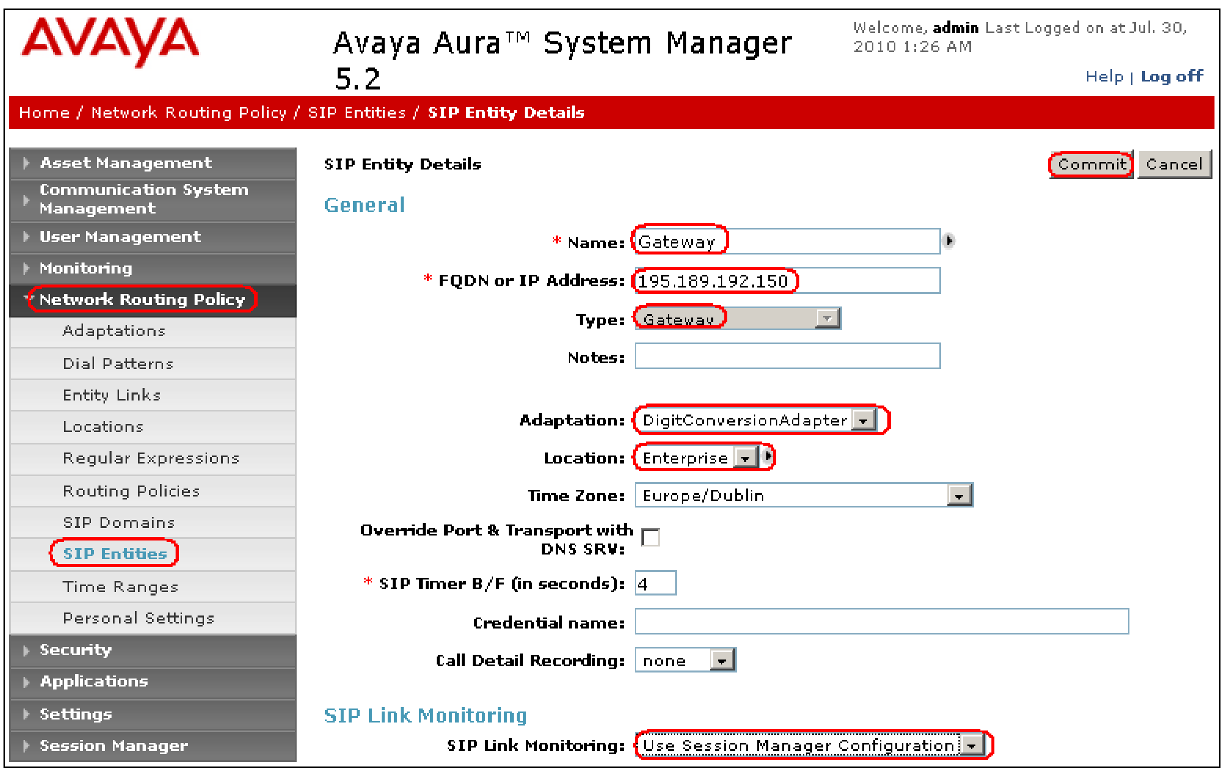
5.4.2. Adding AudioCodes Mediant 3000 Gateway SIP Entity

Navigate **Network Routing Policy → SIP Entities** on the left and click on the **New** button on the right.

Under **General**

- **Name:** A descriptive name (i.e. **Gateway**)
- **FQDN or IP Address:** IP address of the signaling interface of Mediant 3000 Gateway, i.e. **208.209.43.59**
- **Type:** Select **Gateway**
- **Adaptation:** Select the adaptation created in **Section 5.3** i.e. **DigitConversionAdapter**
- **Location:** Select one of the locations defined previously i.e. **Enterprise**
- **Time Zone:** Time zone for this entity

Under **SIP Link Monitoring**, configure **SIP Link Monitoring** as **Use Session Manager Configuration** if Mediant 3000 is in simplex configuration or **Link Monitoring Disabled** for Mediant 3000 Gateway in HA configuration. Defaults can be used for the remaining fields. Click **Commit** to save SIP Entity definition. The screen below shows the configuration of the SIP Entity related to Mediant 3000.



5.4.3. Adding Avaya Aura™ Session Manager SIP Entity

Navigate **Network Routing Policy → SIP Entities** on the left and click on the **New** button on the right.

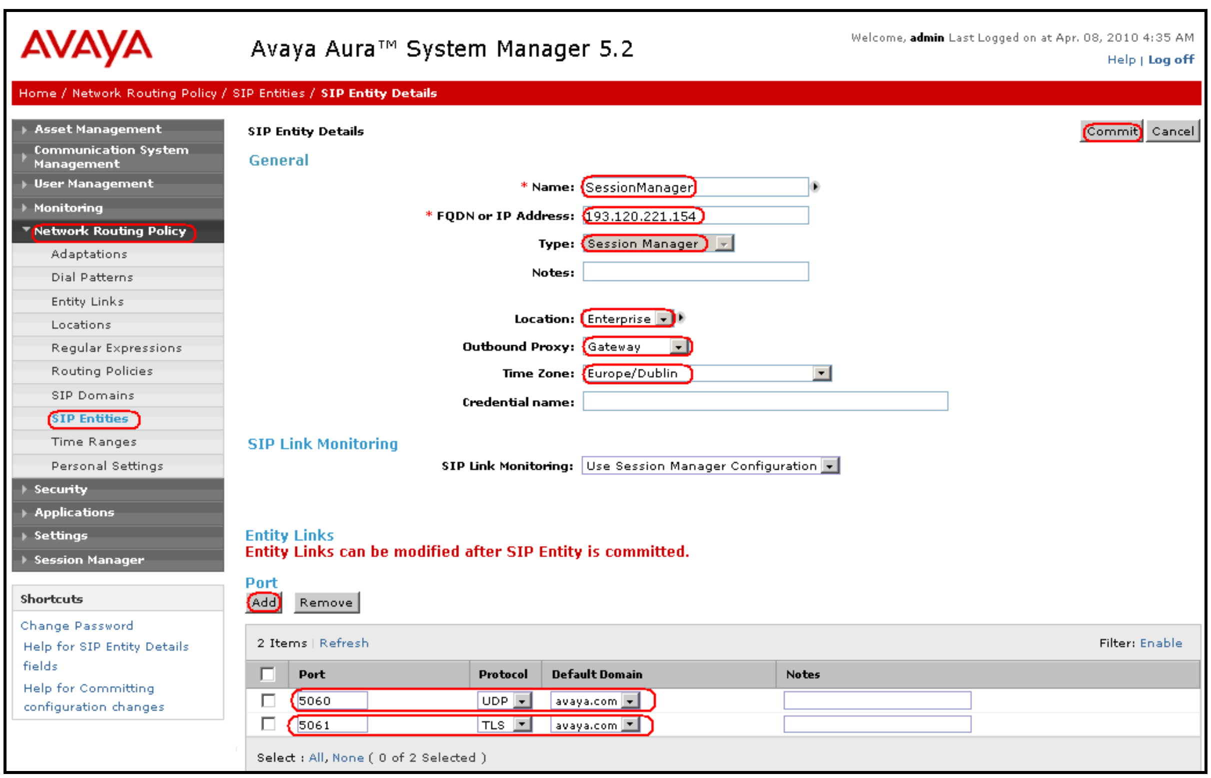
Under **General**:

- **Name:** A descriptive name, i.e. **SessionManager**
- **FQDN or IP Address:** IP address of the Session Manager i.e. **193.120.221.154**, the SM-100 Security Module
- **Type:** Select **Session Manager**
- **Location:** Select one of the locations defined previously
- **Outbound Proxy:** Select the SIP Entity defined previously for Mediant 3000, i.e. **Gateway**
- **Time Zone:** Time zone for this entity

Create two Port definitions, one for **TLS** and one for **UDP**. Under **Port**, click **Add**, and then edit the fields in the resulting new row as shown below:

- **Port:** Port number on which the system listens for SIP requests
- **Protocol:** Transport protocol to be used to send SIP requests
- **Default Domain** The domain used (e.g., **avaya.com**)

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

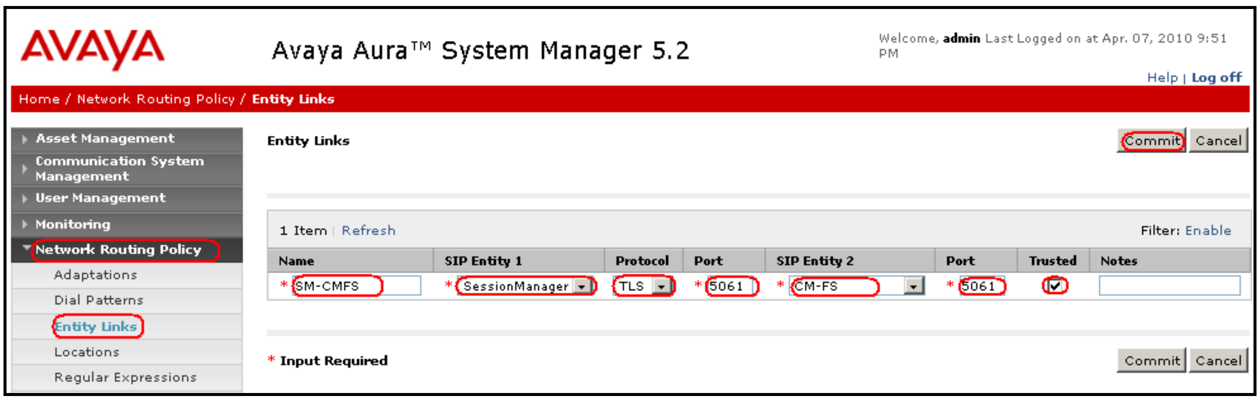


5.5. Add Entity Links

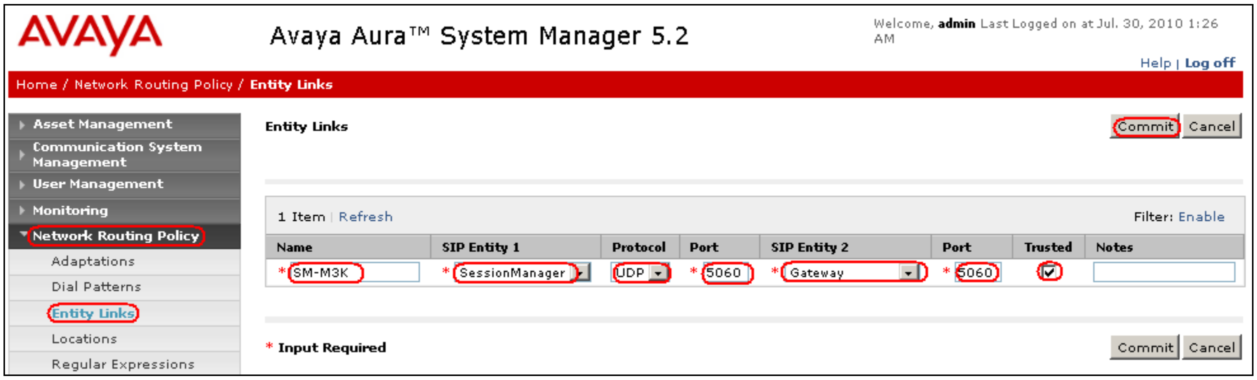
A SIP trunk between Session Manager and a telephony system is described by an Entity link. To add an Entity Link, select **Entity Links** on the left and click on the **New** button on the right. Fill in the following fields in the new row that is displayed:

- **Name:** A descriptive name
- **SIP Entity 1:** Select the **SessionManager** entity
- **Protocol:** Select the transport protocol between **UDP/TCP/TLS** to align with the definition on the **other end o**f the link. In these Application Notes **TLS** was used for **Feature Server** while **UDP** or **TCP** can be used for **Mediant 3000.**
- **Port:** Port number to which the other system sends SIP requests
- **SIP Entity 2:** Select the name of the other system
- **Port:** Port number on which the other system receives SIP requests
- **Trusted:** Check this box, otherwise calls from the associated SIP Entity specified will be denied

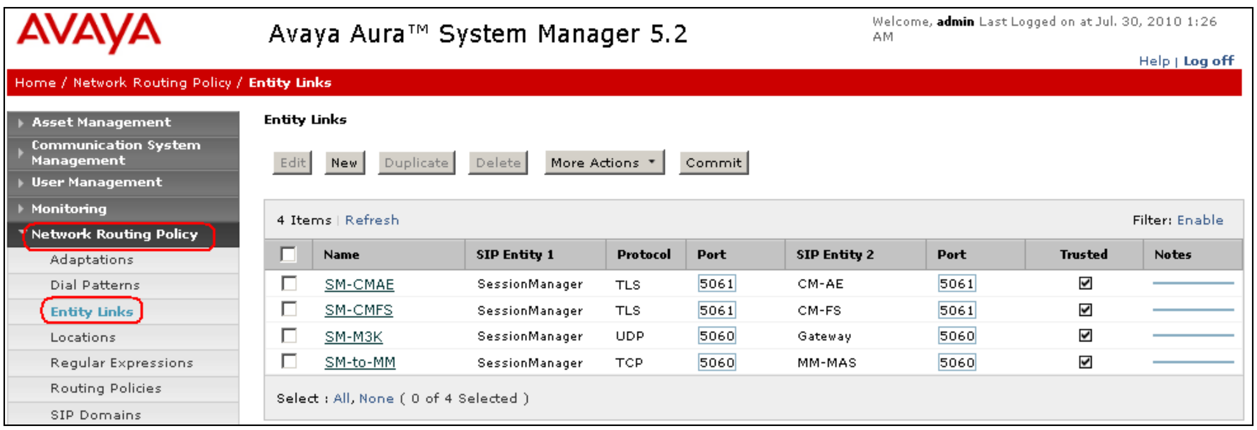
Click **Commit** to save each Entity Link definition**.** The screen below illustrates adding the Entity Link for Communication Manager Feature Server.



The screen below illustrates adding the Entity Link for Mediant 3000 SIP Entity.



The screen below summarizes the Entity Links view after the insertion of the two Entity Links.



5.6. Add Routing Policies

Routing policies describe the condition under which calls will be routed to the SIP Entities specified in **Section 5.4**. A routing policy must be added for the Mediant 3000 Gateway. To add a routing policy, select **Routing Policies** on the left and click on the **New** button on the right. The following screen is displayed. Fill in the following:

Under **General:**

- Enter a descriptive name in **Name**

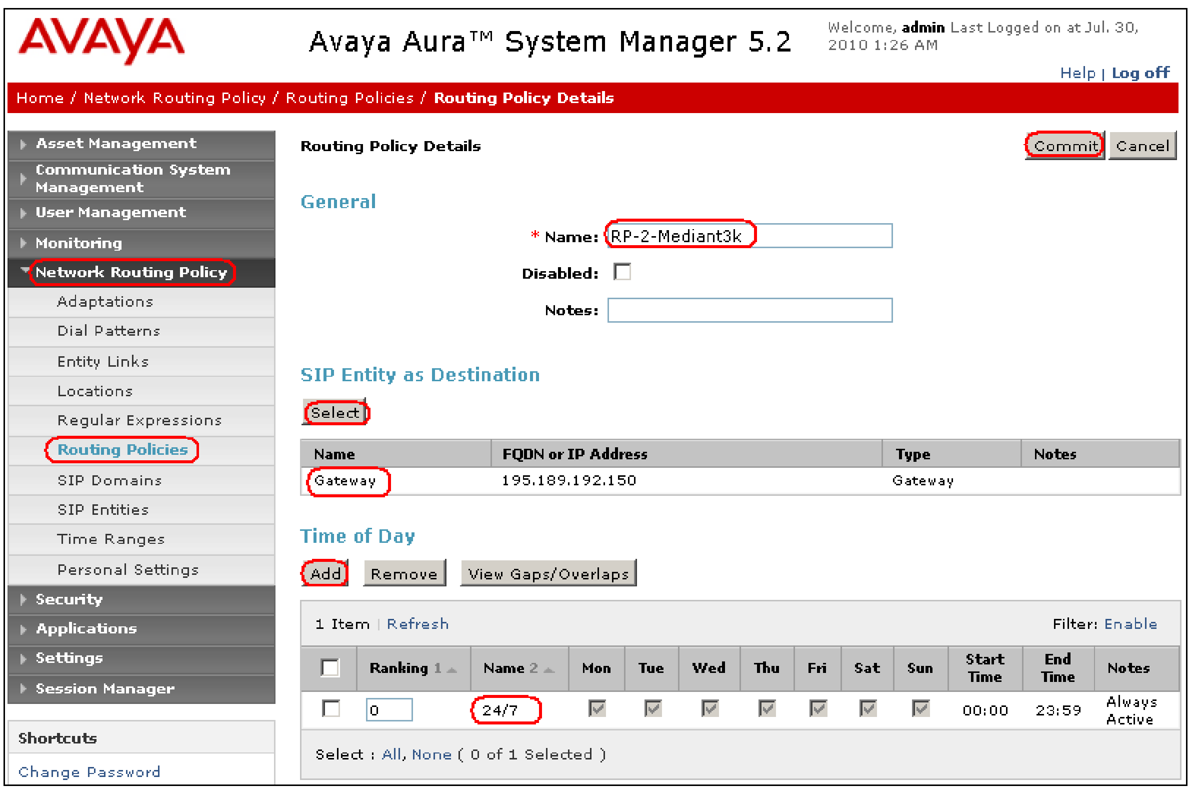
Under **SIP Entity as Destination:**

- 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 screen shows the Routing Policy for Mediant 3000.



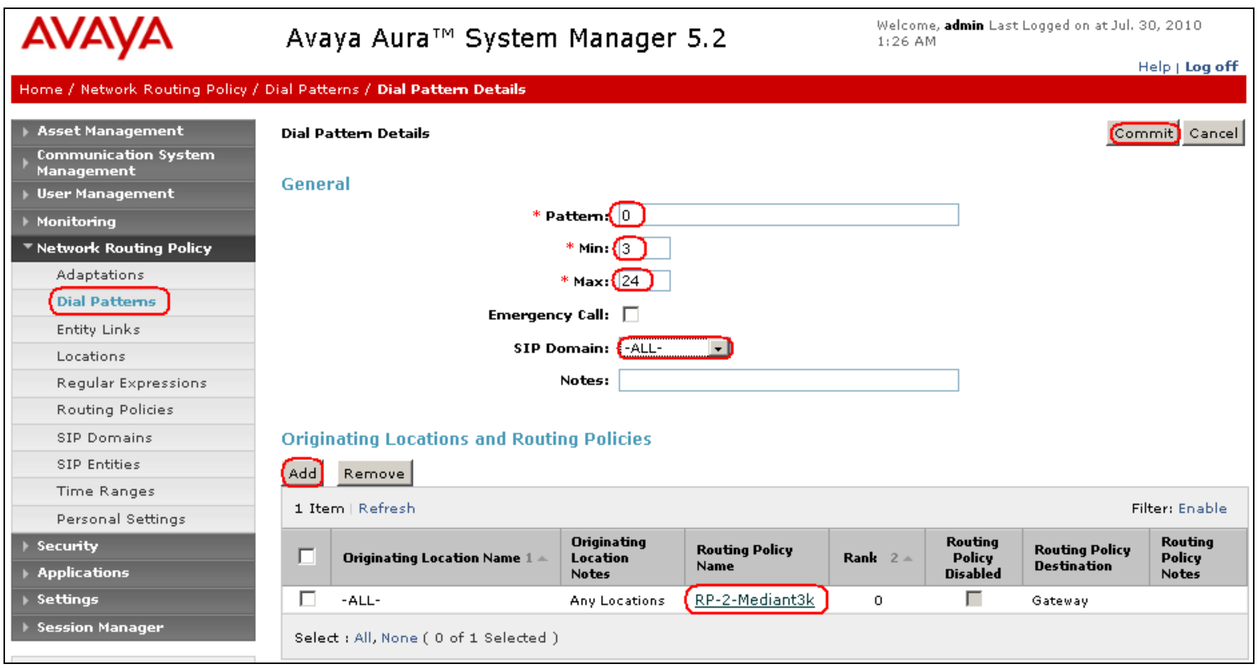
5.7. Add Dial Patterns

Dial patterns must be defined that will direct calls to the appropriate SIP Entity. In the sample configuration numbers beginning with **0** with 3 to 25 digits reside on the Mediant 3000. To add a dial pattern, select **Dial Patterns** on the left and click on the **New** button on the right. Fill in the following, as shown in the screen below, which corresponds to the dial pattern for routing calls to Mediant 3000:

Under **General**:

- **Pattern:** Dialed number or prefix i.e. **0**
- **Min:** Minimum length of dialed number i.e. **3**
- **Max:** Maximum length of dialed number i.e. **24**
- **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 a sample the dial pattern definition for PSTN reachable with Mediant 3000.



5.8. Add Avaya Aura™ Session Manager

To complete the configuration, adding the Session Manager will provide the linkage between System Manager and Session Manager. Expand the **Session Manager** menu on the left and select **Session Manager Administration**. Then click **Add**, and fill in the fields as described below and shown in the following screen:

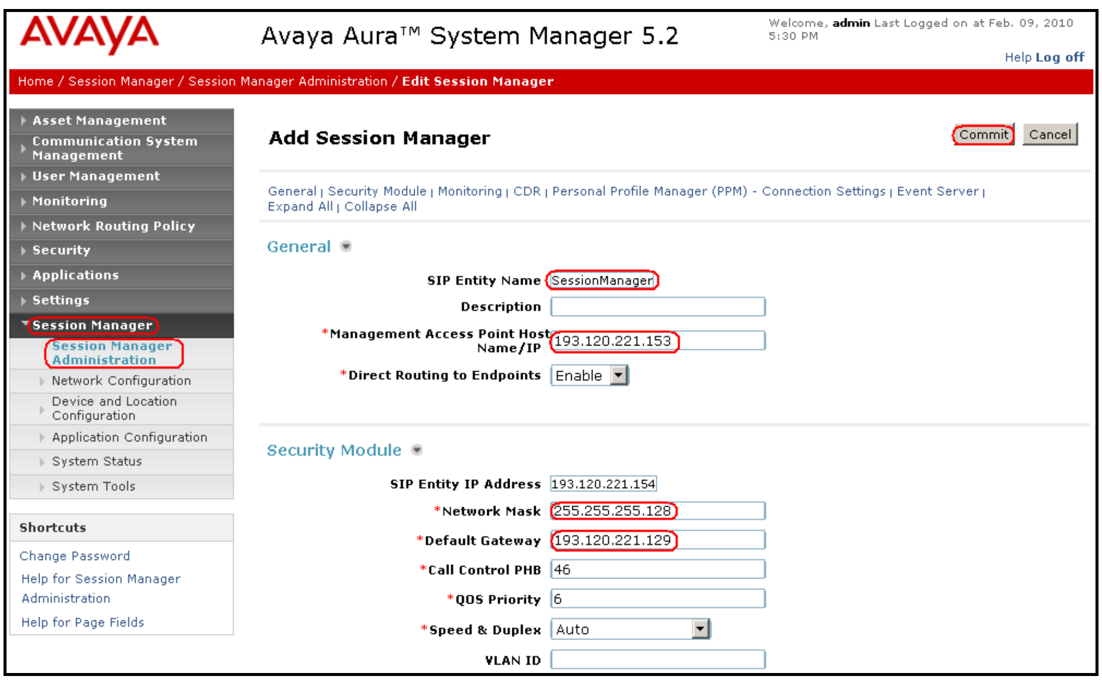
Under **General**:

- **SIP Entity Name**: Select the name of the SIP Entity added for Session Manager
- **Description**: Descriptive comment (optional)
- **Management Access Point Host Name/IP**:
  Enter the IP address of the Session Manager management interface

Under **Security Module**:

- **Network Mask**: Enter the network mask corresponding to the IP address of the SM100 interface (i.e., **255.255.255.128**)
- **Default Gateway**: Enter the IP address of the default gateway for SM100 interface (i.e., **193.120.221.129**)

Use default values for the remaining fields. Click **Commit** to add this configuration to Session Manager.



5.9. Add Avaya Aura™ Communication Manager as a Feature Server

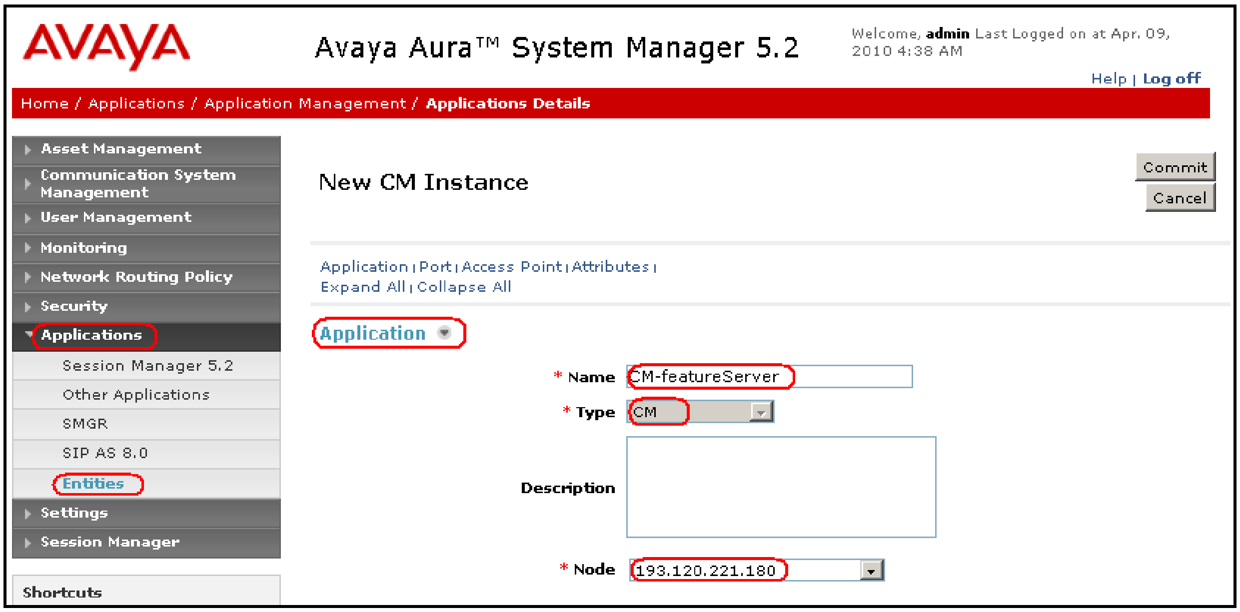
In order for Communication Manager to provide configuration and Feature Server support to SIP phones when they register to Session Manager, Communication Manager must be added as an application.

5.9.1. Create an Application Entry

Expand **Application** menu, select **Entities** on left, click on **New** (not shown). Enter the following fields and retain defaults for the remaining fields.

Under **Application**:

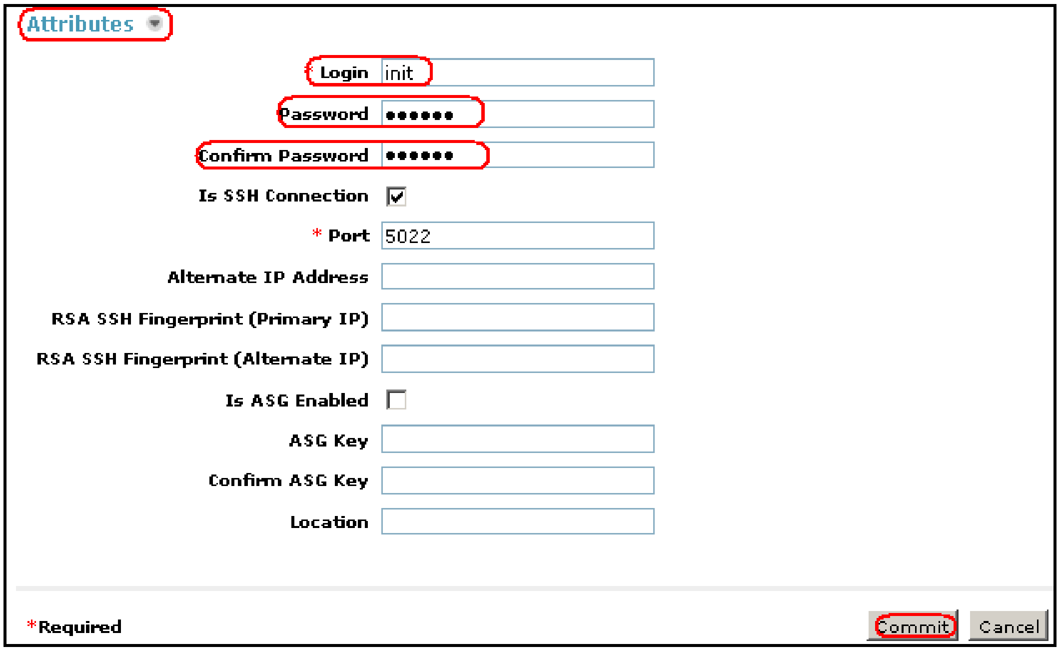
- **Name**: Enter a descriptive name i.e. **CM-featureServer**
- **Type**: Select **CM**
- **Node**: Select **Other..** and enter the IP address for CM SAT access i.e. **193.120.221.180**



Navigate to the **Attributes** section and enter the following:

- **Login**: Login used for SAT access
- **Password**: Password used for SAT access
- **Confirm Password**: Password used for SAT access

Retain default values for the remaining fields. Click **Commit** to save.

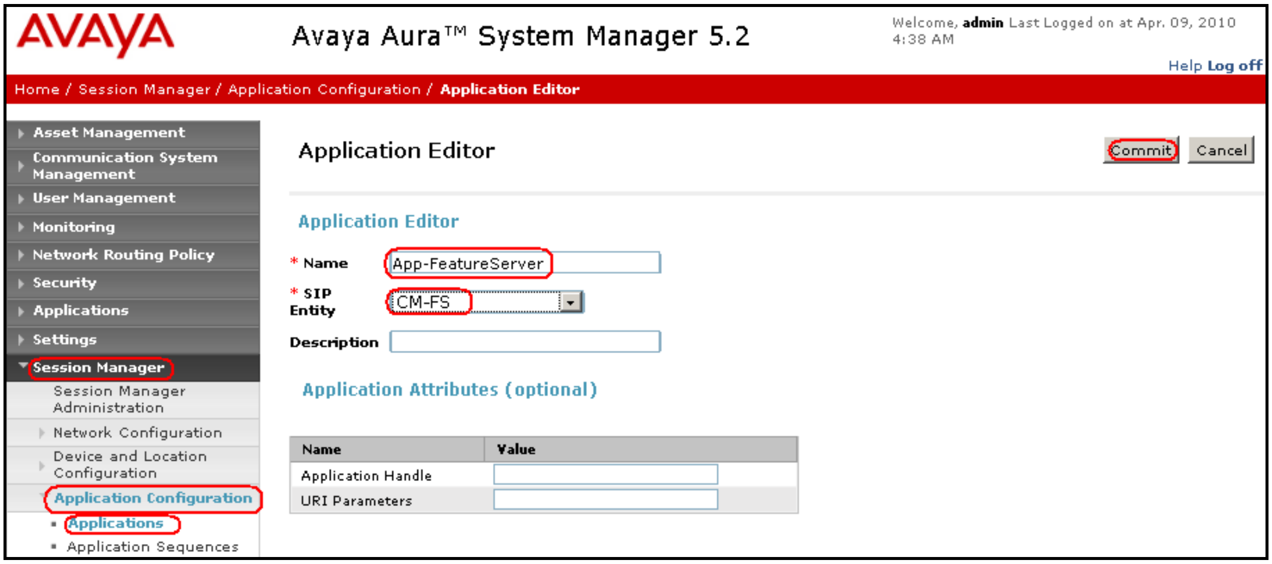


5.9.2. Create a Feature Server Application

Navigate to **Session Manger → Application Configuration → Applications** on the left menu. Click on **New** (not shown). Enter following fields and use defaults for the remaining fields:

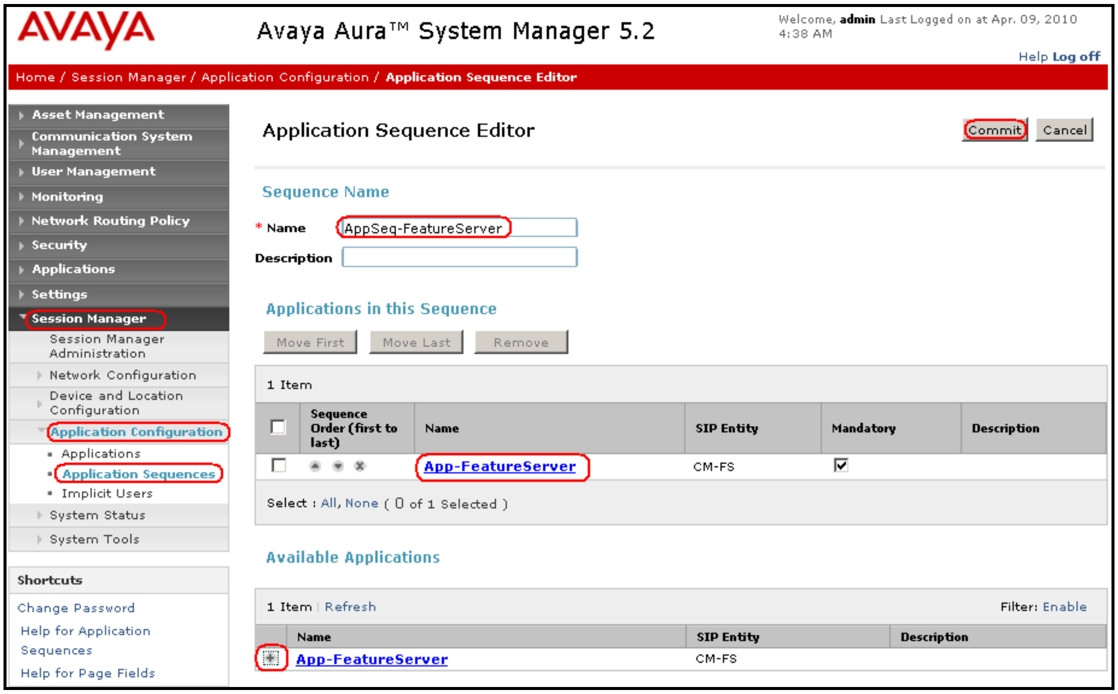
- **Name** A descriptive name
- **SIP Entity** Select the CM SIP Entity defined in **Section 5.4.2**

Click on **Commit** to save.



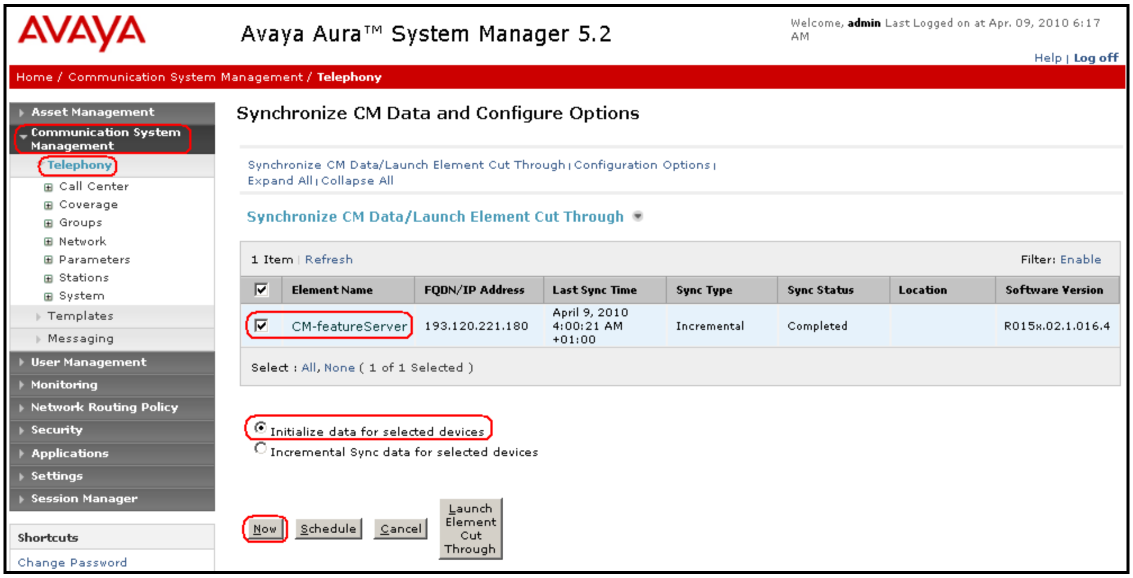
5.9.3. Create a Feature Server Application Sequence

From the left menu, navigate to **Application Sequences** under **Session Manager → Application Configuration**. Click on **New** (not shown). Enter a descriptive **Name**. Click on the + sign next to the appropriate **Available Applications** and they will move up to the **Applications in this Sequence** section. Click on **Commit** to save.



5.9.4. Synchronize Avaya Aura™ Communication Manager Data

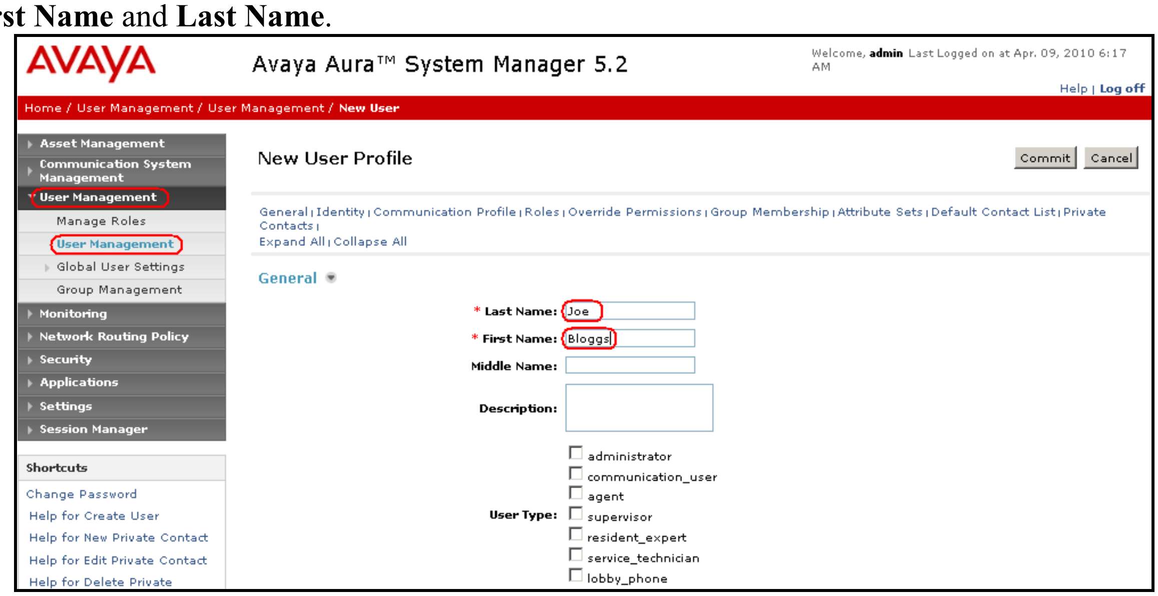
Select **Communications System Management → Telephony** on the left. Select the appropriate **Element Name**. Select **Initialize data for selected devices**, then click on **Now**. This may take some time.



Use the menus on the left under **Monitoring → Scheduler** to determine when the task is complete.

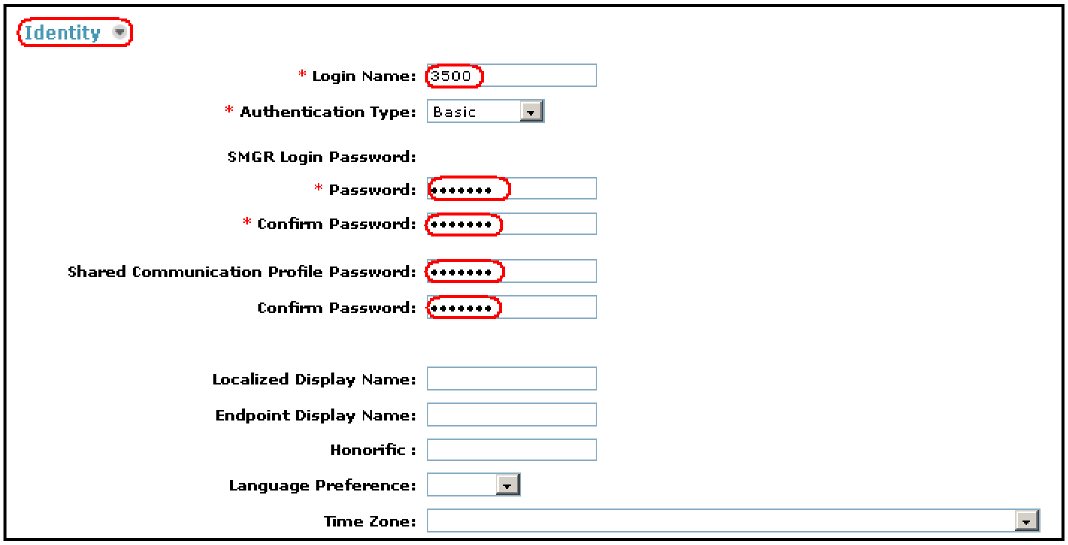
5.10. Add Users for SIP Phones

Users must be added via Session Manager and the details will be updated on the CM. Select **User Management → User Management** on the left. Then click on **New** (not shown). Enter a **First Name** and **Last Name**.



Navigate to the **Identity** section and enter the following and use defaults for other fields:

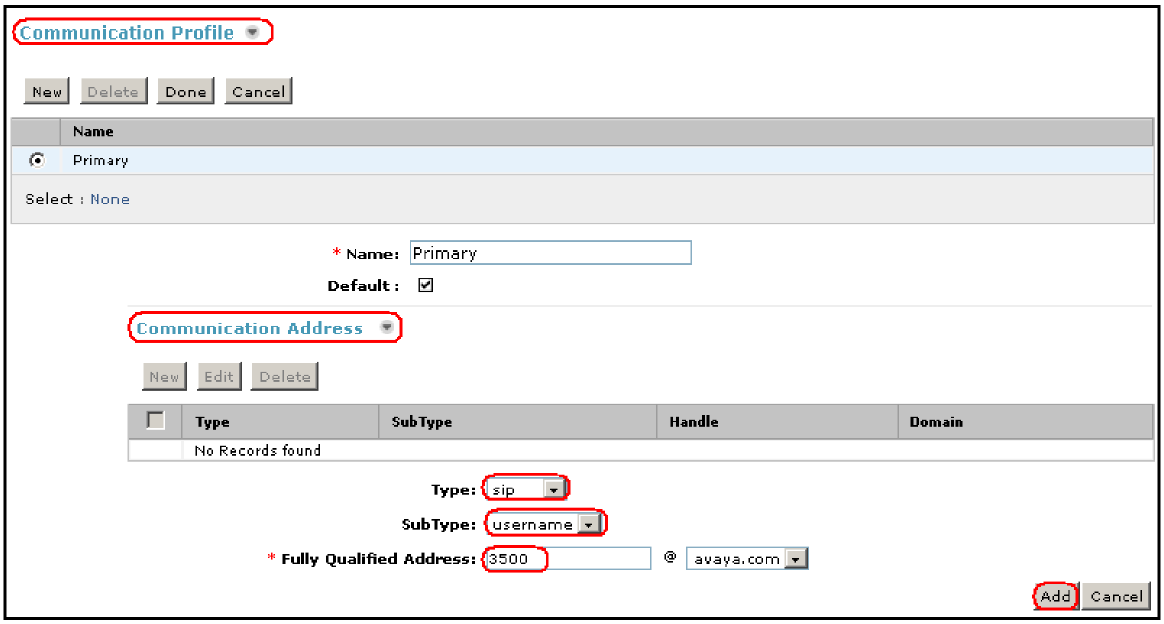
- **Login Name** The desired phone extension number belonging to the domain defined in **Section 5.1**
- **Password** Password for user to log into SMGR
- **Shared Communication Profile Password**
  Password to be entered by the user when logging into the phone



Navigate to and click on the **Communication Profile** section to expand. Then click on **Communication Address** to expand that section. Enter the following and defaults for the remaining fields:

- **Type** Select **SIP**
- **SubType** Select **username**
- **Fully Qualified Address** Enter the extension number i.e. **3500**

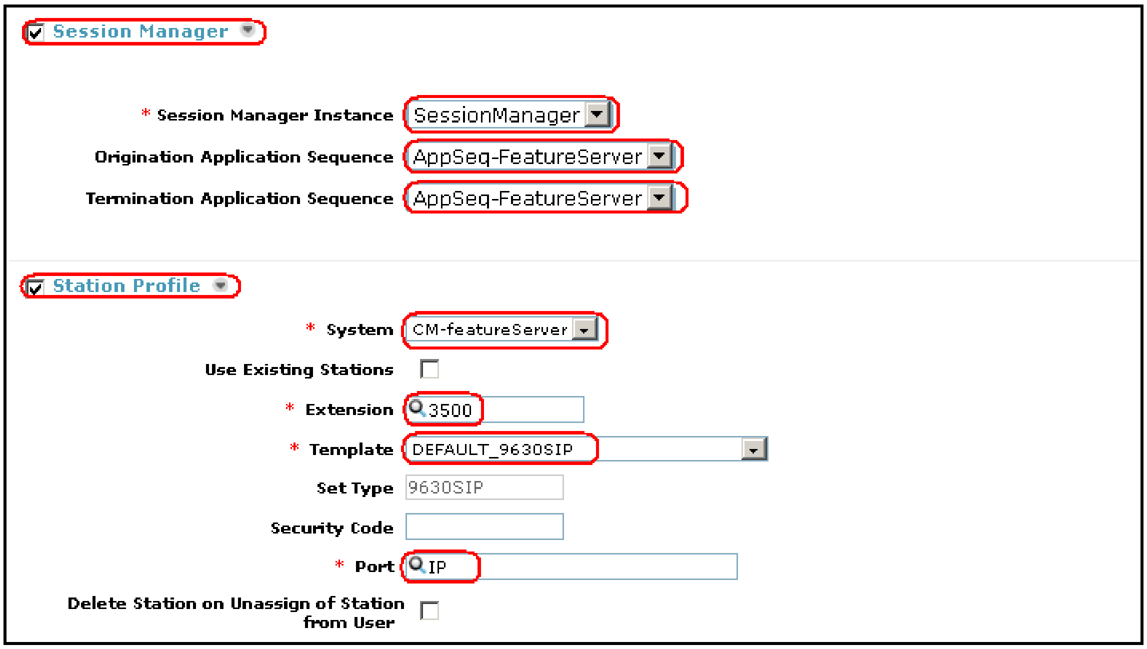
Click on **Add**.



Navigate to and click on **System**. Select the CM Entity.

- **Extension**: Enter a desired extension number i.e. **3500**
- **Template**: Select a telephone type template
- **Port**: Select **IP**

The **Session Manager** section to expand. Select the appropriate Session Manager server for **Session Manager Instance**. For **Origination Application Sequence** and **Termination Application Sequence** select the application sequence created in **Section 5.9.3**. Click on **Station Profile** to expand that section. Enter the following fields and use defaults for the remaining fields:



Click on **Commit** to save (not shown).

6. AudioCodes Mediant 3000 Configuration

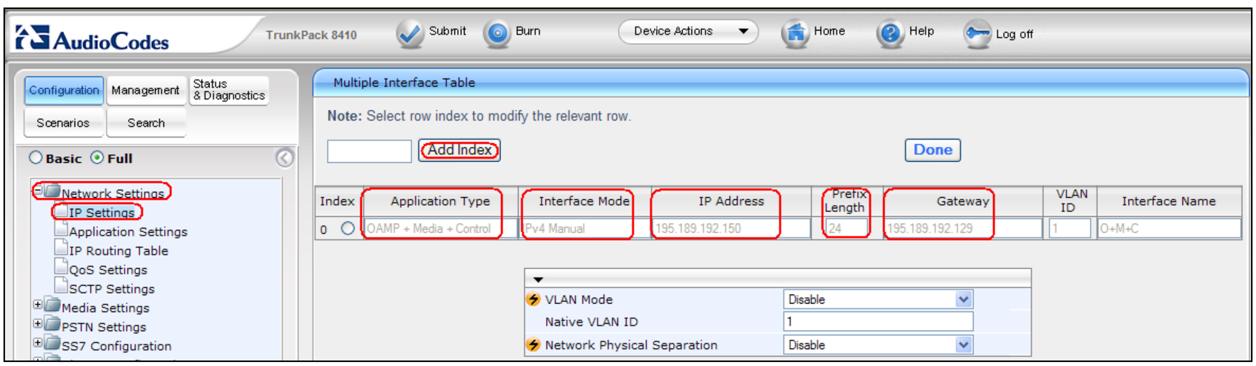
This section displays the configuration for enabling the Mediant 3000 to interoperate with Session Manager. The procedures require five distinct operations:

- 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) Refer to **[15], [16]** and **[17].** 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.

6.1. Configure the Media Gateway IP Network Parameters

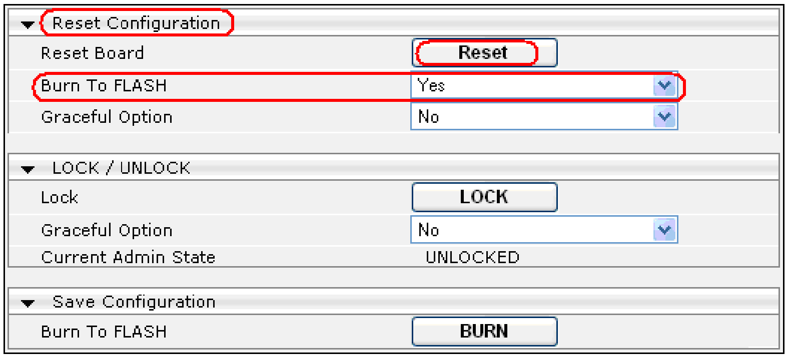
To configure the network parameters click on **Add Index** button to add and index with **Application Type** of **OAMP + Media + Control** and ensure the **Interface Mode** is set to **IPv4** and that **IP Address** (i.e. **195.189.192.150**) **Prefix Length** (i.e**. 24**) and **Gateway** (i.e. **195.180.192.129**) are set according to the expected values.



Save settings to the device's flash memory and reset the device, by performing the following:

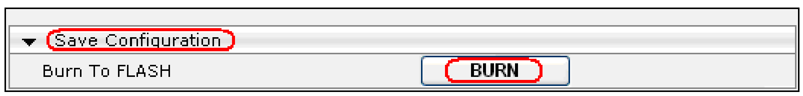
- Navigate (not shown) to the **Maintenance Actions** page (Management tab → **Management Configuration** menu → **Maintenance Actions**).
- Under the **Reset Configuration** group, from the **Burn To FLASH** drop-down list, select **Yes**, and then click the **Reset** button; the device's new configuration (i.e., global IP address) is saved (burned) to the flash memory and the device resets and now enters HA mode (with Active and Redundant blades). The Web interface session terminates (as it's no longer accessible using the blade's private IP address).

The picture below illustrates the saving process for initial IP configuration.



6.1.1. Saving settings

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

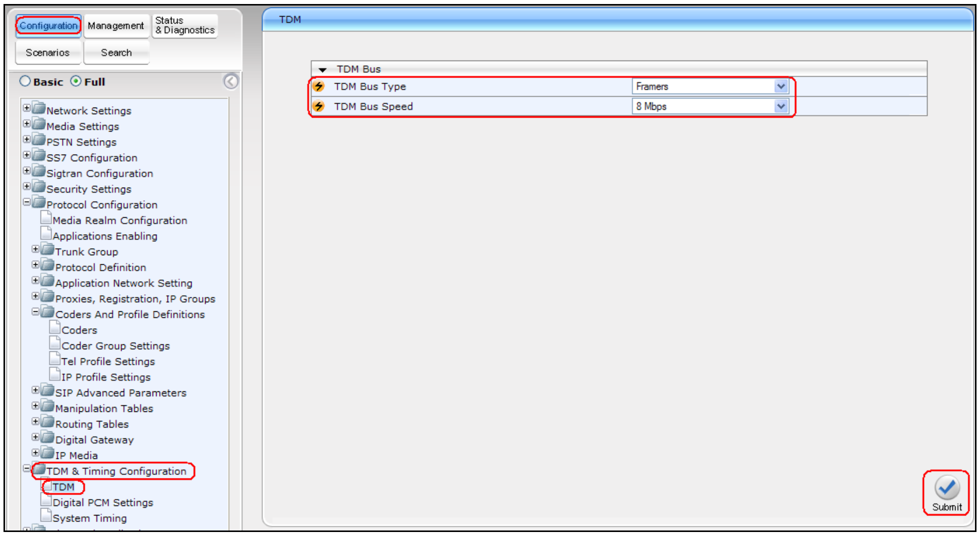


Note: If the value changed is highlighted by a lightning bolt  , the setting will take place after system restart.

6.2. Configure the Media Gateway TDM and Timing Parameters

6.2.1. Configure TDM Bus

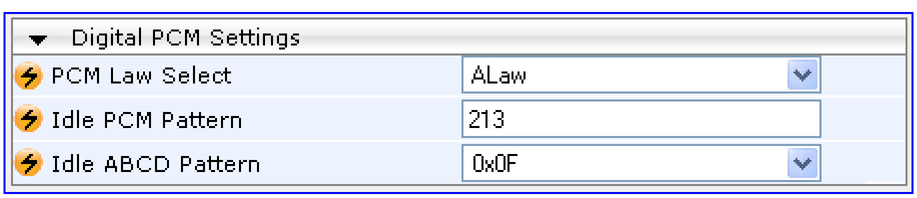
To configure the TDM Bus settings open the **TDM** page (Configuration tab → **TDM & Timing Configuration** menu → **TDM**), configure **TDM Bus Type** and **TDM Bus Speed** parameters as required. (For E1 set **TDM Bus Type** to **Frames** and **TDM Bus speed** to **8Mbps**) Click the **Submit** button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



Note: To save the changes to flash memory, refer to **Section 6.1.1**

6.2.2. Configure digital PCM settings

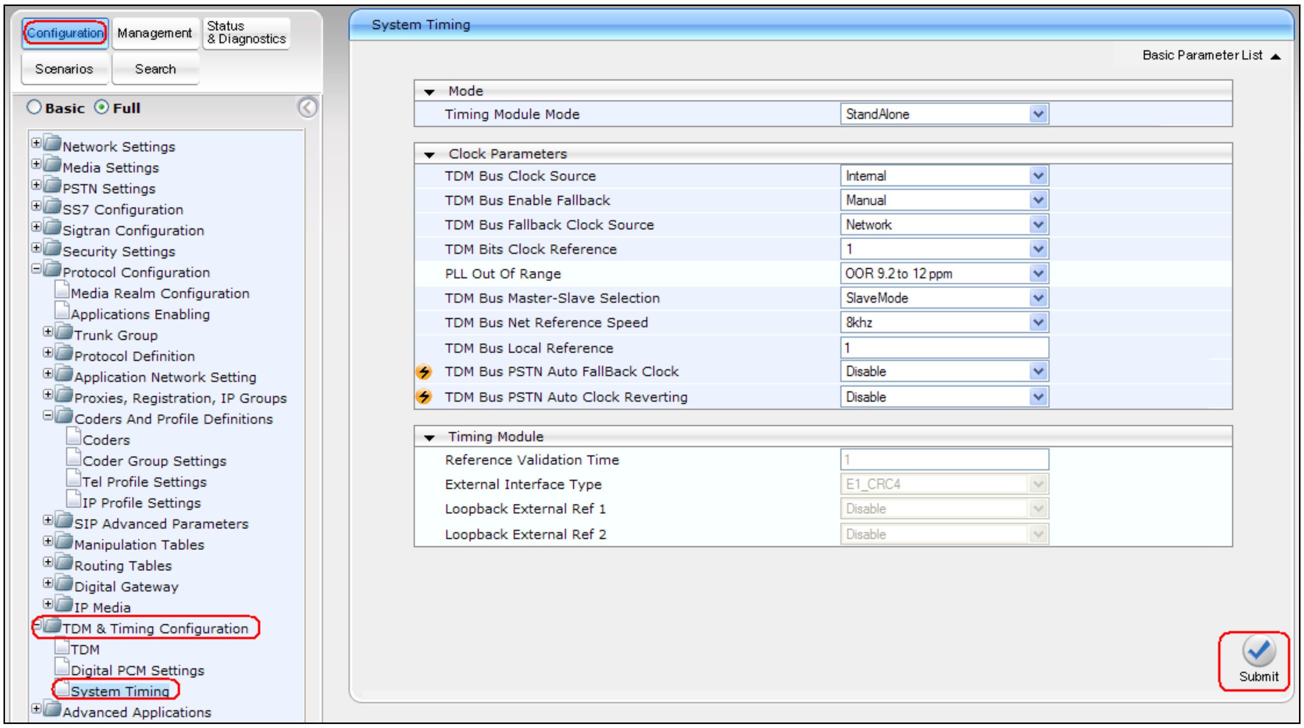
To configure the digital PCM settings, Open the **Digital PCM Settings** page (**Configuration** tab → **TDM & Timing Configuration** menu → **Digital PCM Settings**), configure the parameters as required i.e. **PCM Law Select ALaw** for E1 and click the **Submit** button to save changes.



Note: To save the changes to flash memory, refer to **Section 6.1.1**

6.2.3. Configure system timing

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



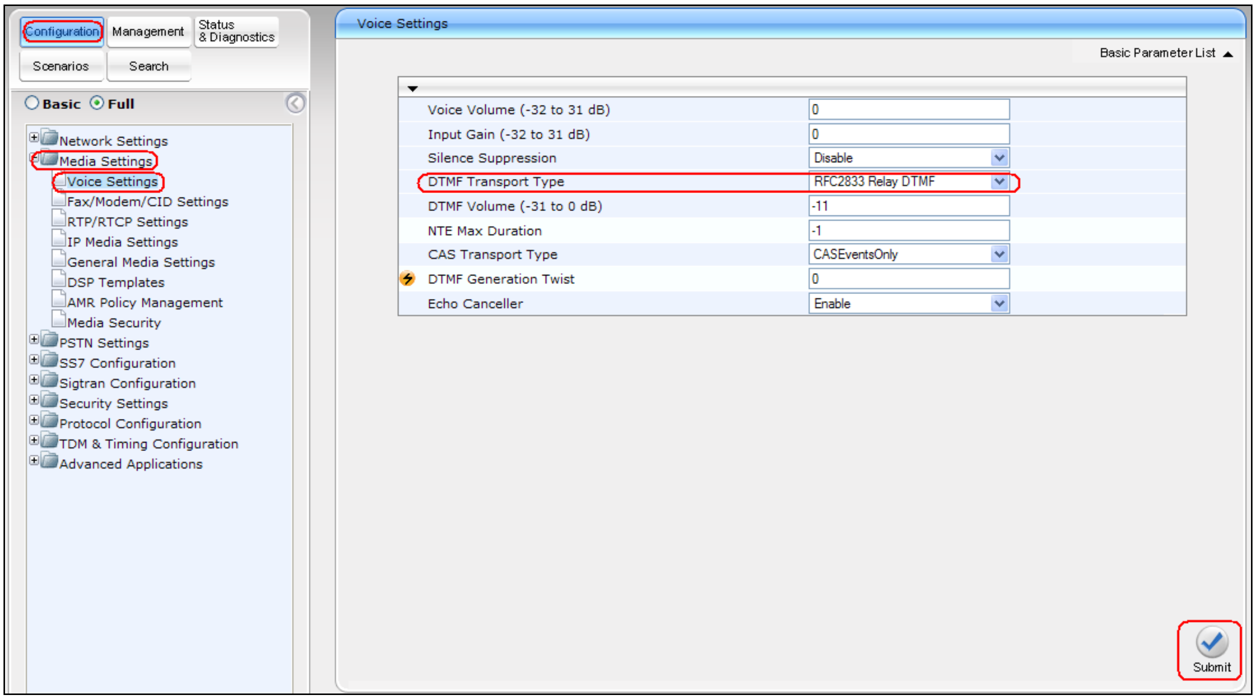
Note: To save the changes to flash memory, refer to **Section 6.1.1**

6.3. Configure the Media Gateway Media Settings

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

6.3.1. Configure the Voice parameters

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



Note: To save the changes to flash memory, refer to **Section 6.1.1**

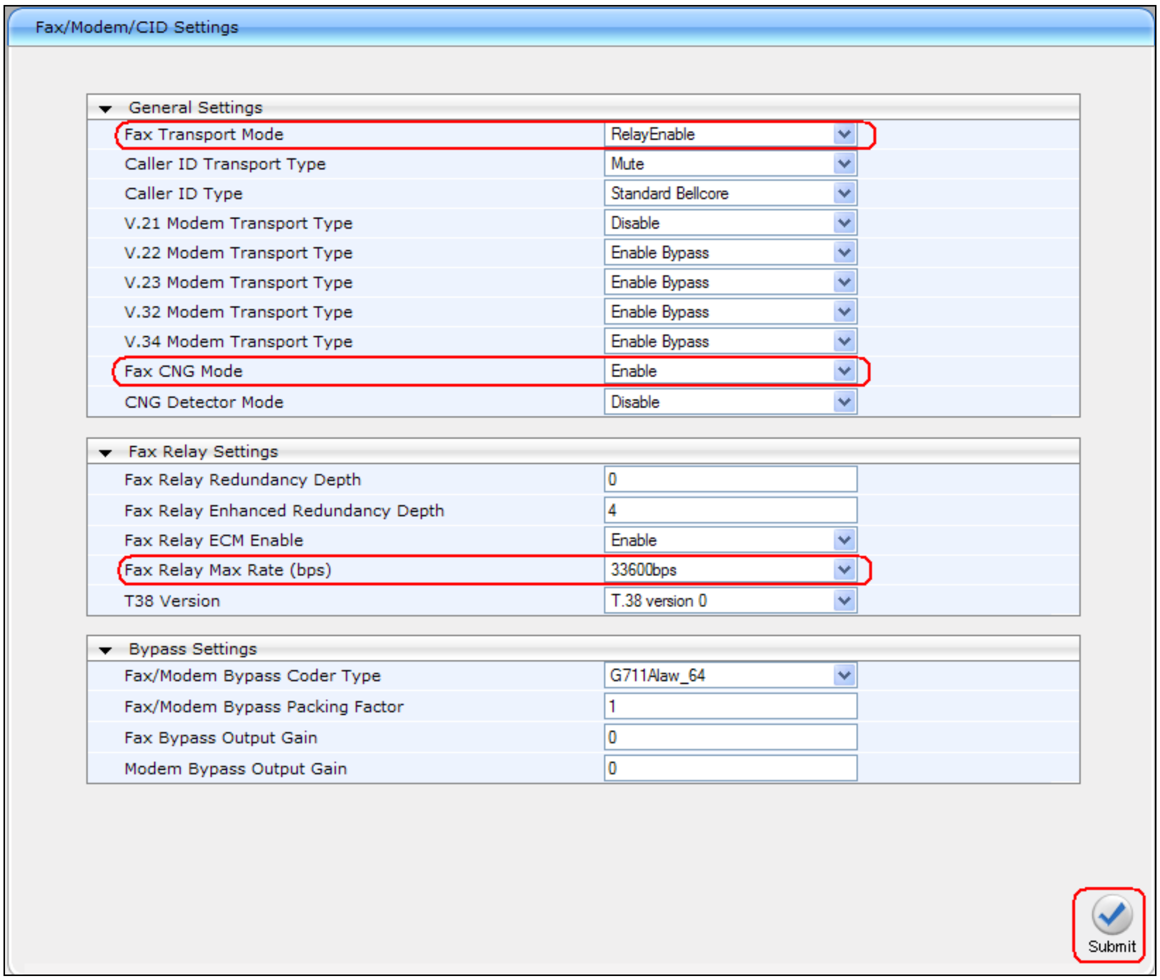
6.3.2. Configure the Fax Parameters

To configure FAX support, open the **Fax/Modem/CID Settings** page (**Configuration** tab→ **Media Settings** menu → **Fax/Modem/CID Settings**).

Set the following values:

- **Fax Transport Mode:** **Relay/Enable**
- **Fax CNG Mode:** **Enable**
- **Fax Relay Max Rate:** **33600bps** (note that supported bit rate by the entire solution is limited by the capabilities of Communication Manger, capped at 9600bps)

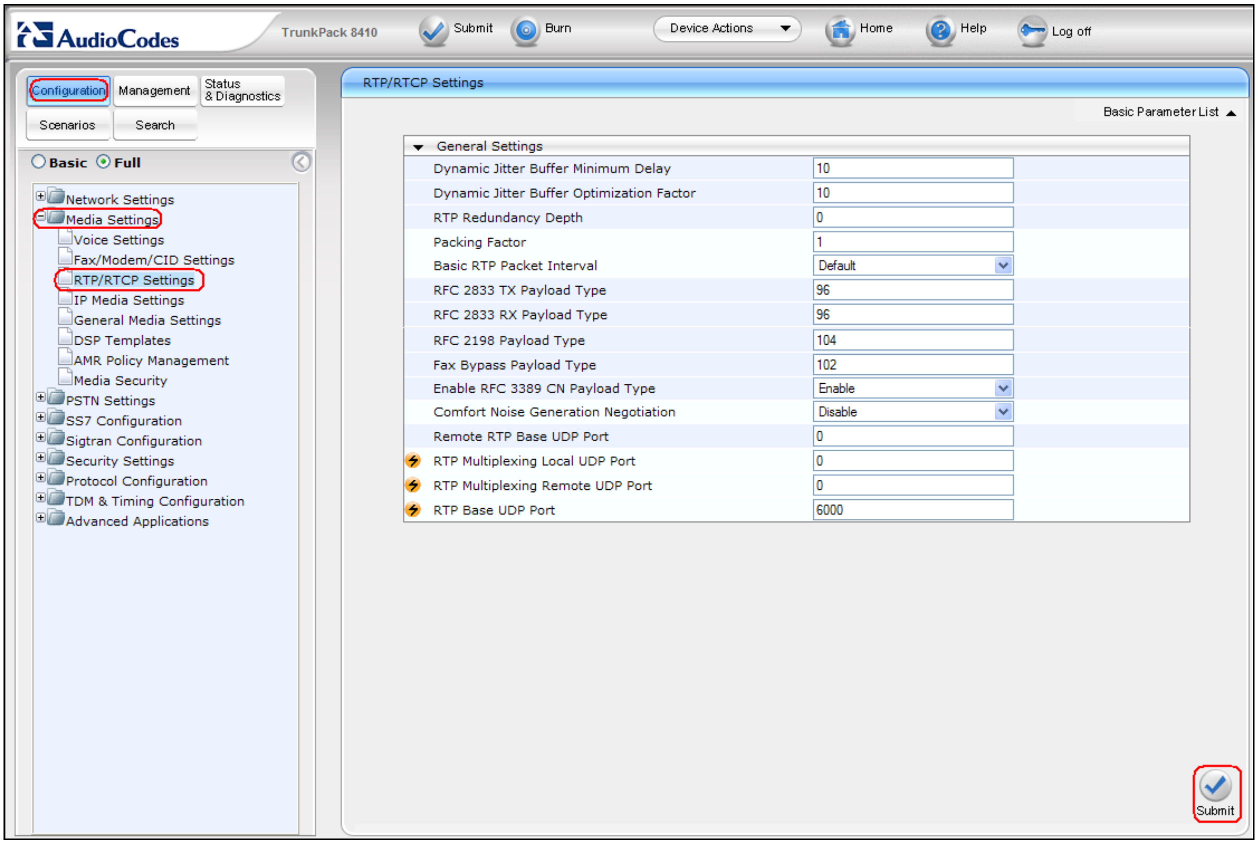
Click the **Submit** button to save changes. The figure below illustrates the Fax settings on the Mediant 3000.



Note: To save the changes to flash memory, refer to **Section 6.1.1**

6.3.3. Configure the RTP/RTCP Parameters

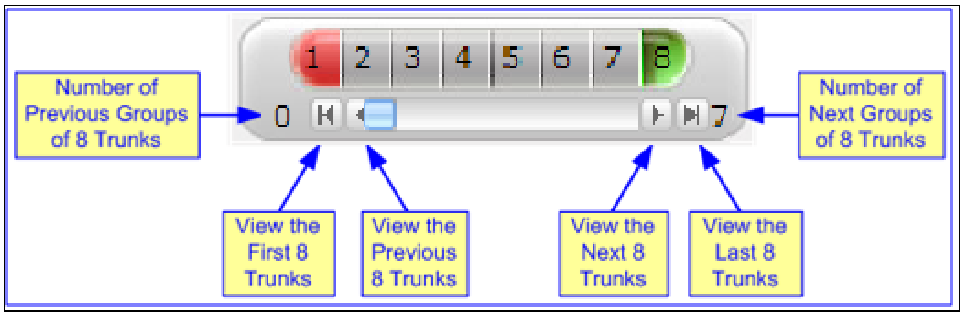
Verify and configure RTP parameters by opening the **RTP/RTCP Settings** page (**Configuration** tab → **Media Settings** menu → **RTP / RTCP Settings**). Click the **Submit** button to save changes. The figure below illustrates setting use in these sample Application Notes.



Note: To save the changes to flash memory, refer to **Section 6.1.1**

6.4. Configure the Media Gateway Telephony/PSTN Interface Parameters

Open the **Trunk Settings** page (**Configuration** tab → **PSTN Settings** menu → **Trunk Settings**). Select the trunk to be configured, by clicking the desired Trunk number icon. 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:



After having selected a trunk, the following is displayed:

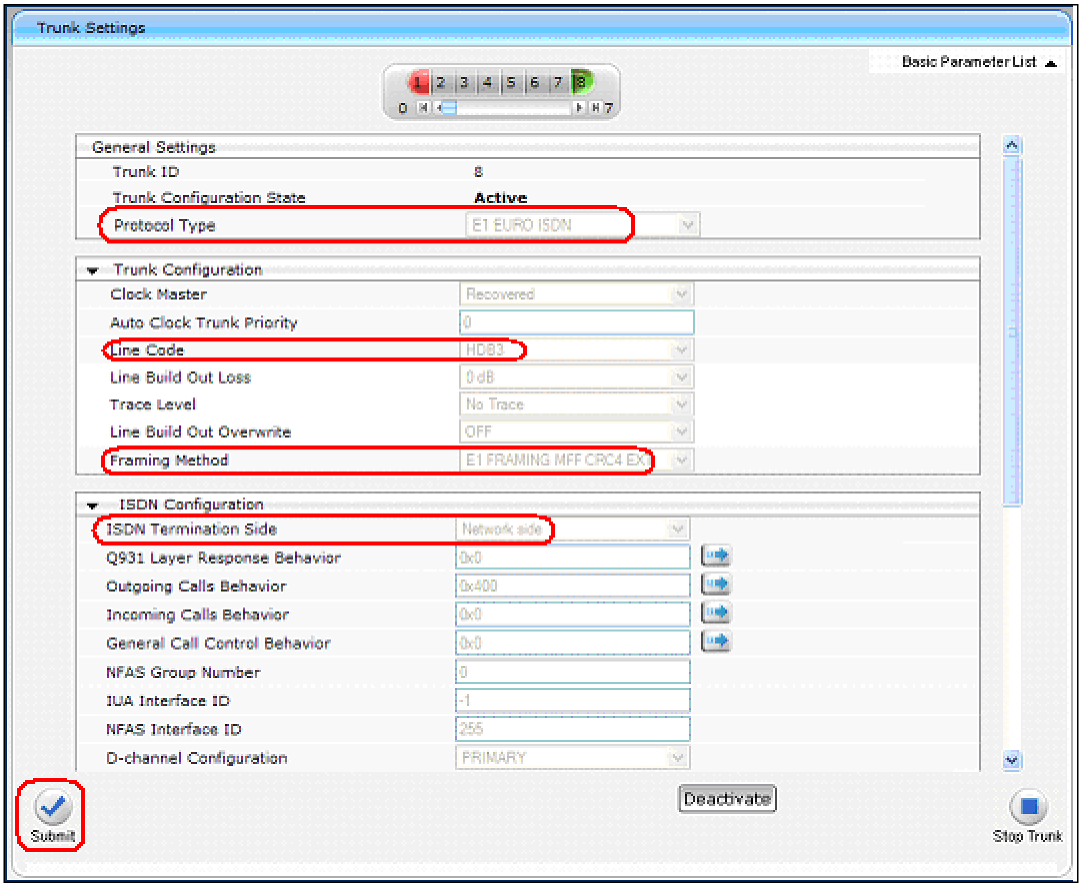
- The read-only **Trunk ID** field displays the selected trunk number.
- The read-only **Trunk Configuration State** displays the state of the trunk (e.g., **Active** or **Inactive**).
- The parameters displayed in the page pertain to the selected trunk only.

Click the **Stop Trunk**  button (located at the bottom of the page) to take the trunk out of service so that you can configure 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. (When all trunks are stopped, the **Apply to All Trunks**  button also appears.) All the parameters are available and can be modified. Configure the desired trunk parameters. Click the **Apply Trunk Settings** button to apply the changes to the selected trunk (or click **Apply to All Trunks** to apply the changes to all trunks); the **Stop Trunk** button replaces **Apply Trunk Settings** and the **Trunk Configuration State** displays **Active**.

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

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

Refer to **[15-18]** to configure the different E1 types.



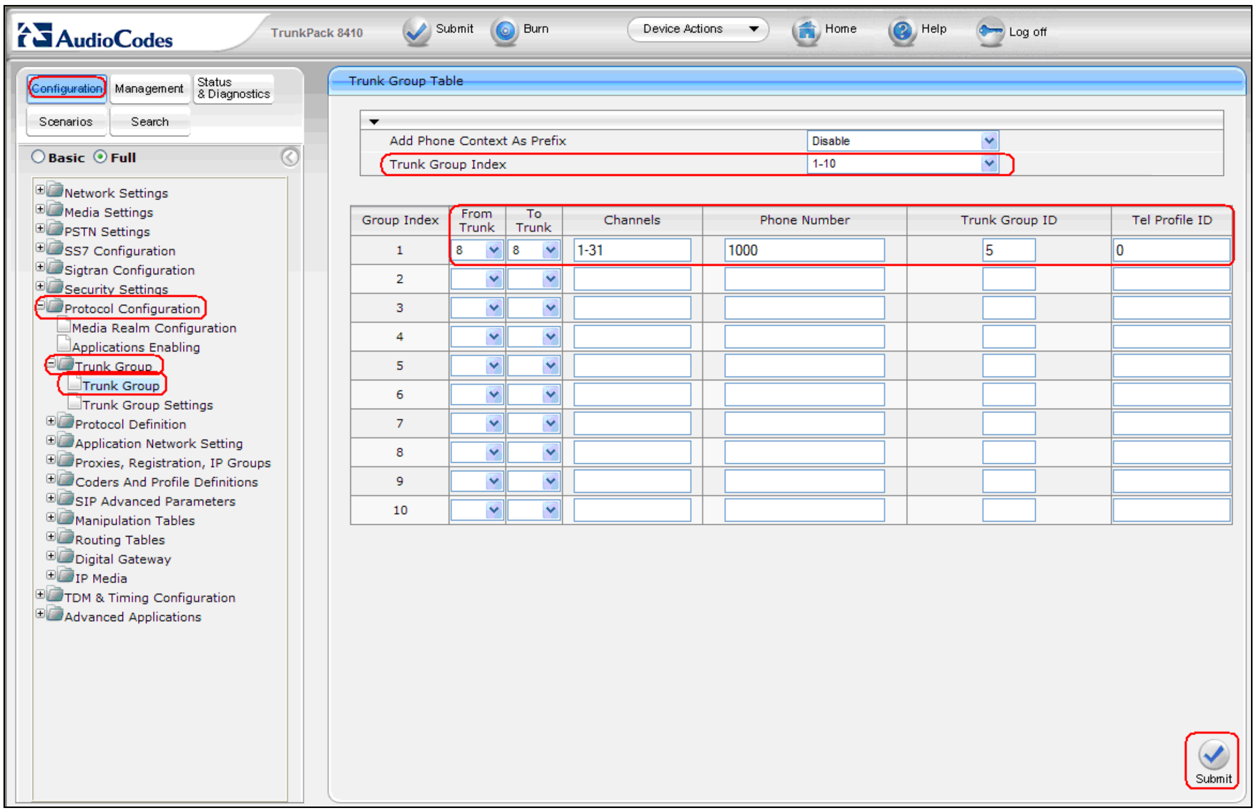
Note: To save the changes to flash memory, refer to **Section 6.1.1**

6.5. Configure the SIP Protocol Parameters

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

6.5.1. Configure the Trunk Group Table

Open the **Trunk Group Table** page (**Configuration** tab → **Protocol Configuration** menu → **Trunk Group** submenu → **Trunk Group).** Select the appropriate **Trunk 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 **[15-18]**.Click the **Submit** button to save changes. The figure below illustrates setting use in these sample Application Notes.



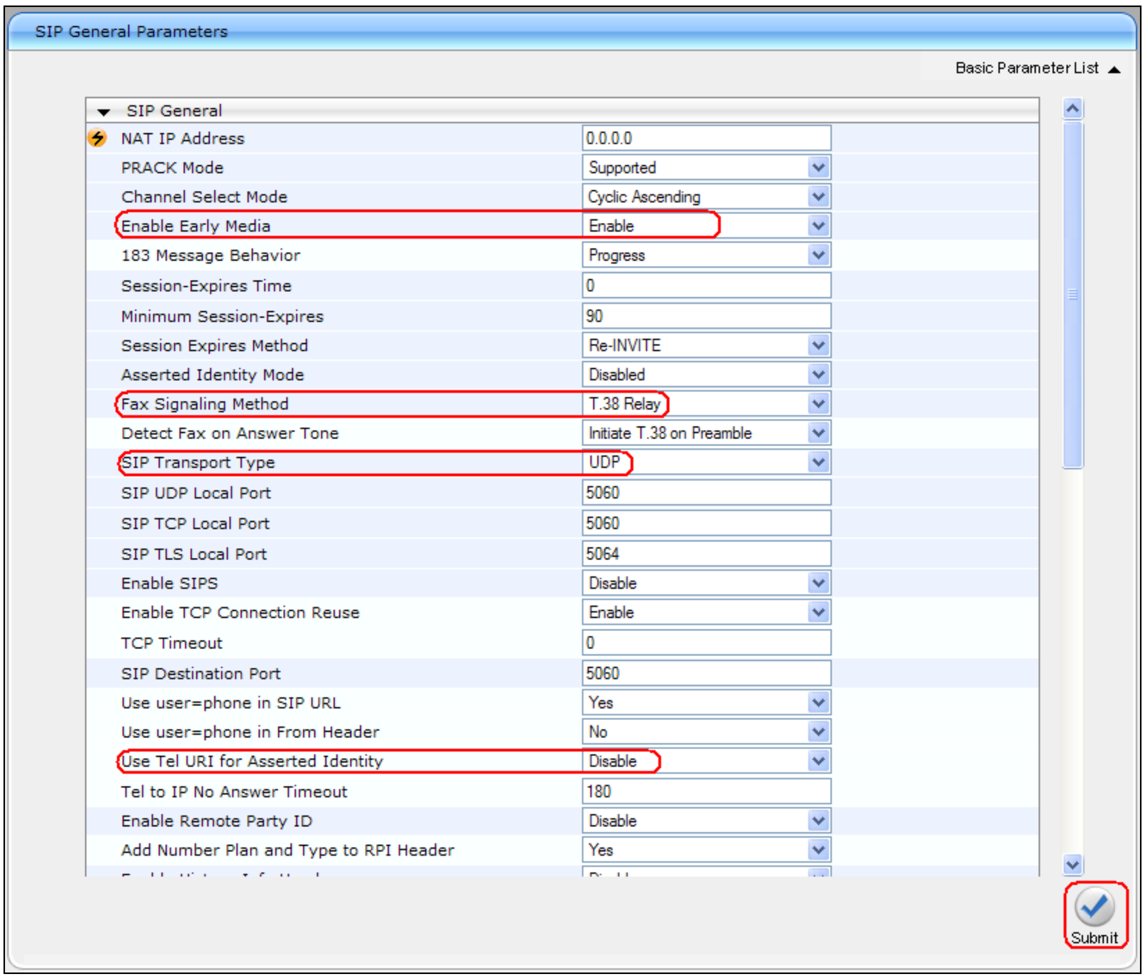
Note: To save the changes to flash memory, refer to **Section 6.1.1**

6.5.2. Configure the General SIP Protocol Parameters

Open the **SIP General Parameters** page (**Configuration** tab → **Protocol Configuration** menu → **Protocol Definition** submenu → **SIP 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. **UDP**.
- **Use Tel URI for Asserted Identity:** Set to **Enable** if Adaptation is used on Session Manager otherwise set to **Disable**

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



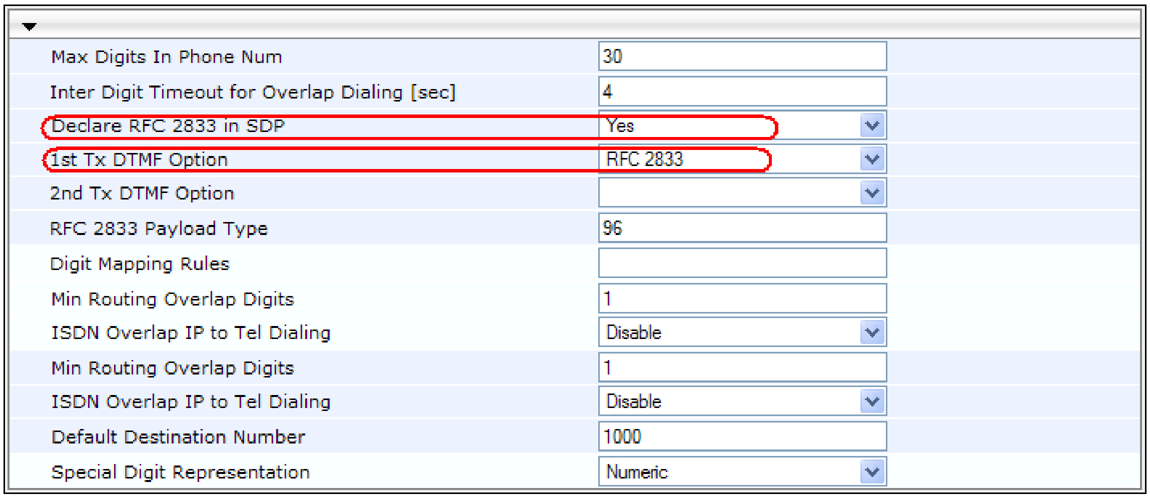
Note: To save the changes to flash memory, refer to **Section 6.1.1**

6.5.3. Configure the DTMF and Dialing Parameters

Open the **DTMF & Dialing** page (**Configuration** tab → **Protocol Configuration** menu → **Protocol Definition** submenu → **DTMF & Dialing**). Set the following values:

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

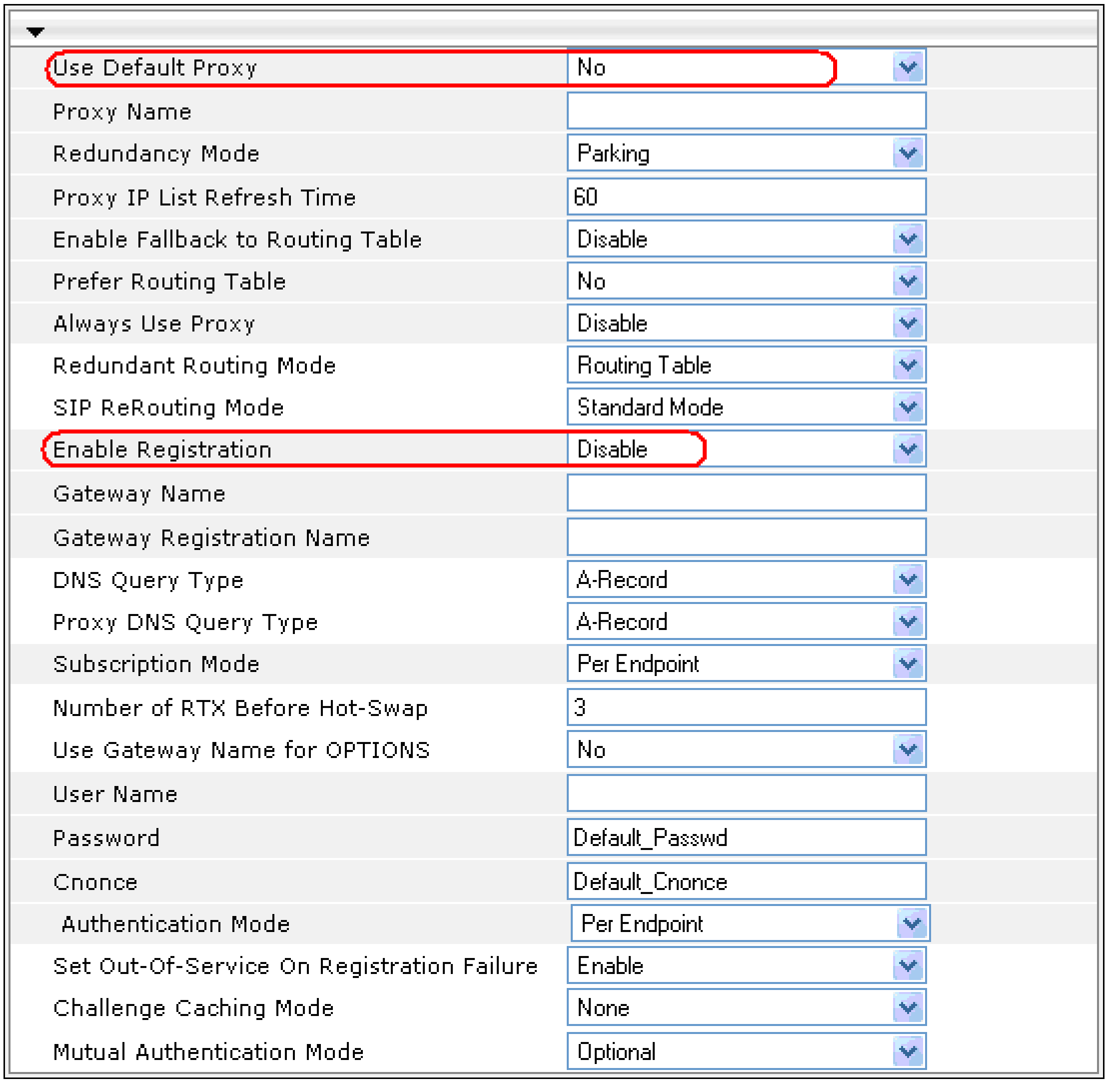
Click the **Submit** button to save changes. The figure below illustrates the SIP General Parameters page.



Note: To save the changes to flash memory, refer to **Section 6.1.1**

6.5.4. Configure the Proxy & Registration Parameters

Open the **Proxy & Registration** page (**Configuration** tab → **Protocol Configuration** menu → **Proxies, Registration, IP Groups** submenu → **Proxy & Registration**). Ensure that **Used Default Proxy** is set to **No** and **Enable Registration** is set to **Disable**. Click the **Submit** button to save your changes. The figure below displays **the Proxy & Registration** page for the system used in these Application Notes.



Note: To save the changes to flash memory, refer to **Section 6.1.1**

6.5.5. Configure the Device's Coders

Open the **Coders** page (**Configuration** tab → **Protocol Configuration** menu → **Coders And Profile Definitions** submenu → **Coders**).

1. From the **Coder Name** drop-down list, select the required coder
2. 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
3. From the **Rate** drop-down list, select the bit rate (in kbps) for the selected coder
4. 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)
5. From the **Silence Suppression** drop-down list, enable or disable the silence suppression option for the selected coder
6. Repeat **Step 2** through **Step 6** for the next optional coders

Click the **Submit** button to save your changes. In the following figure are presented the codecs used in these Application Notes.

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

Note: To save the changes to flash memory, refer to **Section 6.1.1**

The following table describes the Codec Interoperability between Communication Manager and Mediant 3000.

Avaya ip codec set	**AudioCodes codec definition**	
	(G.729 Annex b=no) **Silence suppression=Disabled**	(G.729 Annex b=yes) **Silence Suppression=Enabled**
G.729	**ok**	**ok**
G.729A	**ok**	**ok**
G.729B	**No interop**	**ok**
G.729AB	**No interop**	**ok**

6.5.6. Configure the IP Profile Settings

Open the **IP Profile Settings** page (**Configuration** tab → **Protocol Configuration** menu → **Coders And Profile Definitions** submenu → **IP Profile Settings**). Complete the following steps to define the **IP Profile Settings**:

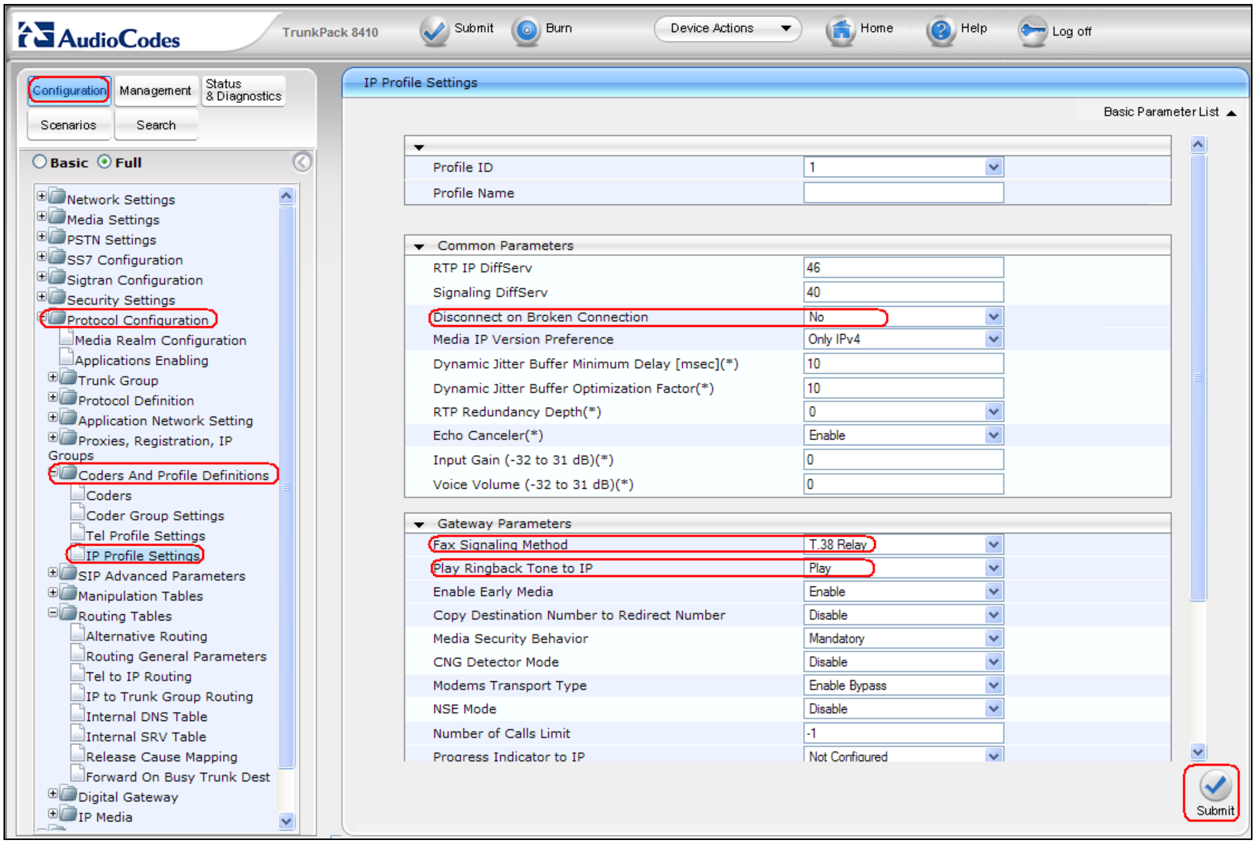
1. From the **Profile ID** drop-down list, select an identification number for the IP Profile.
2. In the **Profile Name** field, enter an arbitrary name that allows you to easily identify the IP Profile.
3. From the **Profile Preference** drop-down list, select the priority of the IP Profile, where **1** is the lowest priority and **20** is the highest. If both IP and Tel profiles apply to the same call, the coders and other common parameters (noted by an asterisk) of the preferred Profile are applied to that call. If the Preference of the Tel and IP Profiles is identical, the Tel Profile parameters are applied.
   Note:
   If the coder lists of both IP and Tel Profiles apply to the same call, only the coders common to both are used. The order of the coders is determined by the preference.
4. Configure the IP Profile's parameters according to your requirements. Parameters that are unique to IP Profile are described in the table below.
5. From the **Coder Group** drop-down list, select the coder group that need to be assigned assign to the IP Profile. The device's default coders can be set, or one of the coder groups defined in the **Coder Group Settings** page.
6. Repeat **Step 2** through **Step 6** for the next IP Profiles (optional).

Click the **Submit** button to save changes.

In these Application Notes, the following values were set:

- **Disconnect on Broken Connection: No**
- **Fax Signaling Method: T.38 Relay**
- **Play Ringback tone to IP: Play**

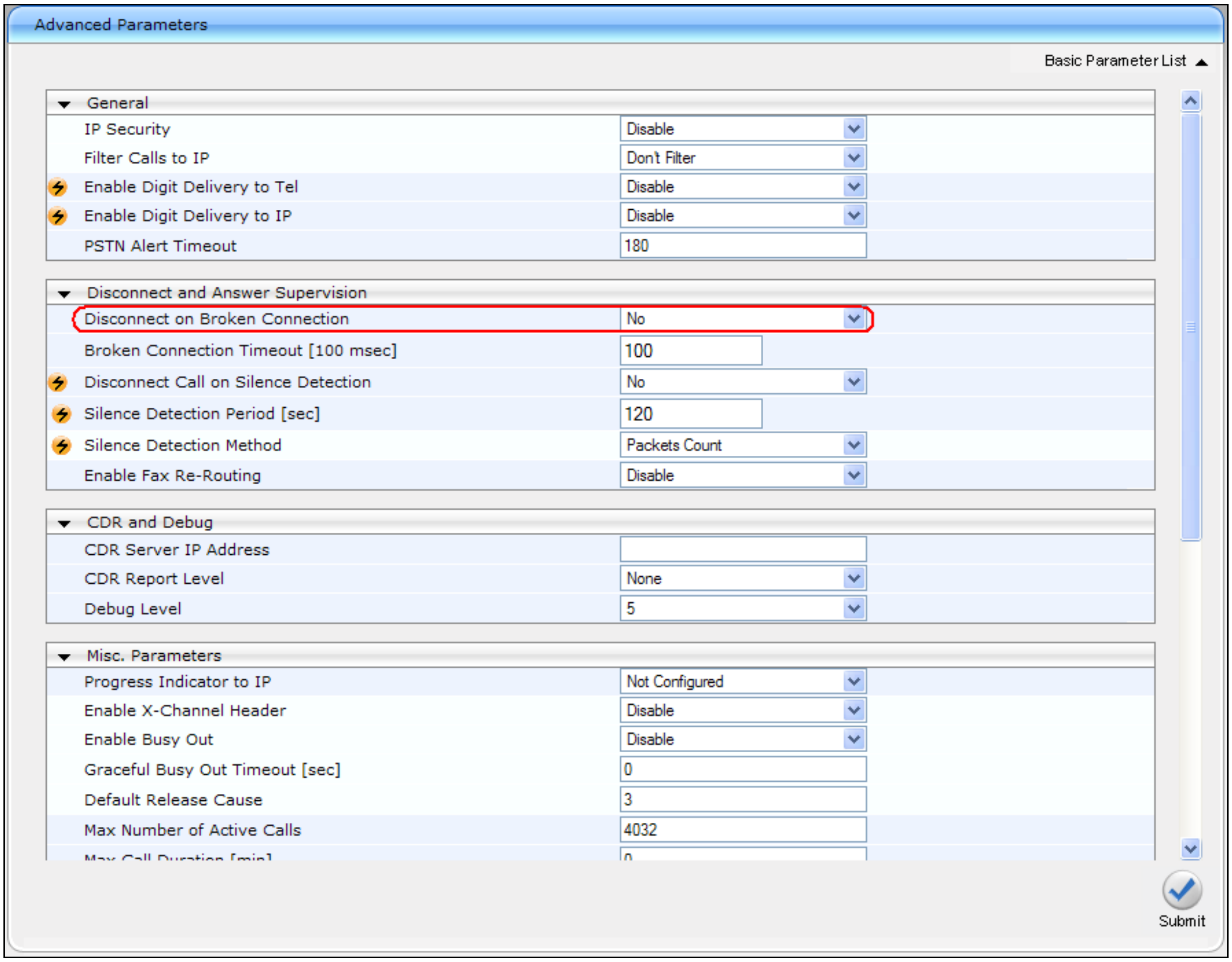
The figure below illustrates the **IP Profile Settings** page.



Note: To save the changes to flash memory, refer to **Section 6.1.1**

6.5.7. Configure the Advanced General Protocol Parameters

Open the **Advanced Parameters** page (**Configuration** tab → **Protocol Configuration** menu → **SIP Advanced Parameters** submenu → **Advanced Parameters**). This page allows the configuration of the defaults protocol parameters in case there is no mach on the previously configured protocol parameters. In these notes only the **Disconnect on Broken Connection** was set to **No**, other configurations may require special care. Refer to **[15-18]** for additional information. Click the **Submit** button to save your changes.



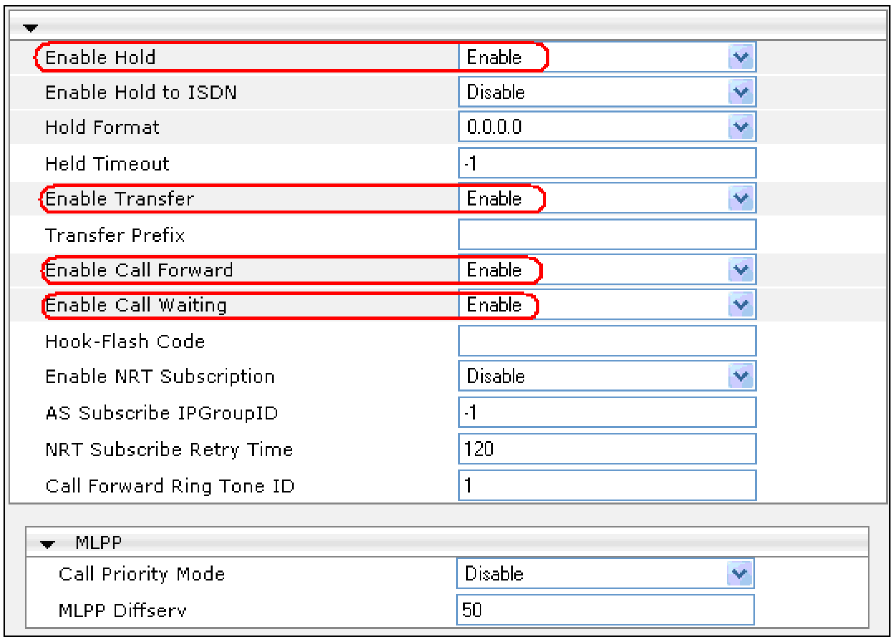
Note: To save the changes to flash memory, refer to **Section 6.1.1**

6.5.8. Configure the Supplementary Services Parameters

Open the **Supplementary Services** page (**Configuration** tab → **Protocol Configuration** menu → **SIP Advanced Parameters** submenu → **Supplementary Services**). Set to **Enable** the following services:

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

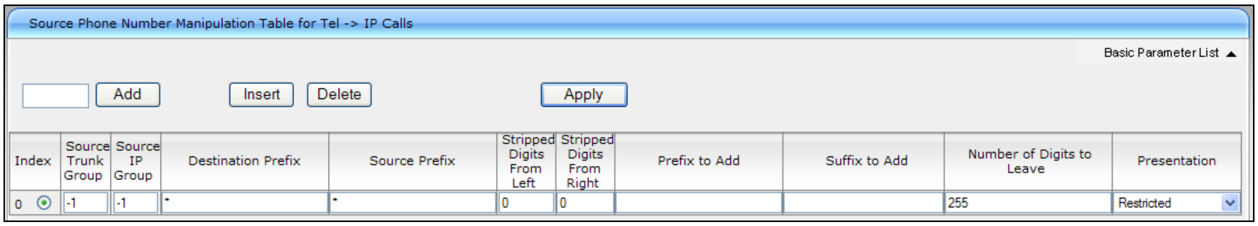
The figure below illustrates the **Supplementary Services** page.



Note: To save the changes to flash memory, refer to **Section 6.1.1**

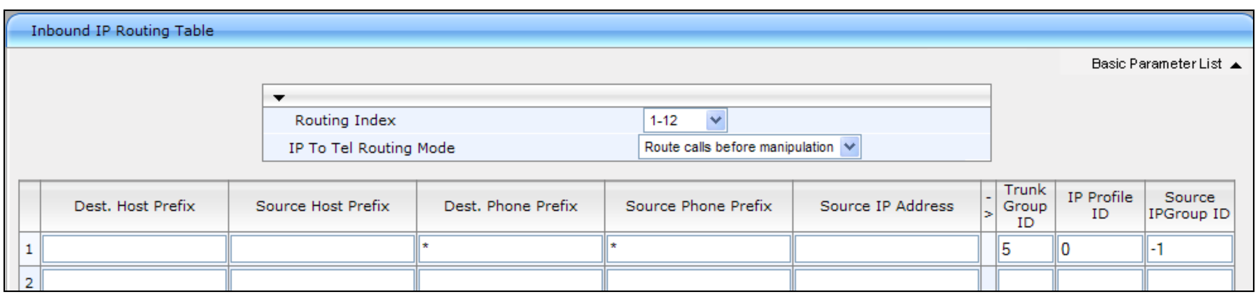
6.5.9. Configure the Number Manipulation Tables

Open the required **Number Manipulation** page (**Configuration** tab→ **Protocol Configuration** menu → **Manipulation Tables** submenu → **Dest Number IP→Tel**, **Dest Number Tel→IP**, **Source Number IP→Tel**, or **Source Number Tel→IP**); the relevant Manipulation table page is displayed (e.g., **Source Phone Number Manipulation Table for Tel→IP Calls** page). The figure shows the manipulation rules for Tel-to-IP source phone number manipulation, used in these Application Notes. For more information on Configuring the Number Manipulation tables refer to **[15-18]**.



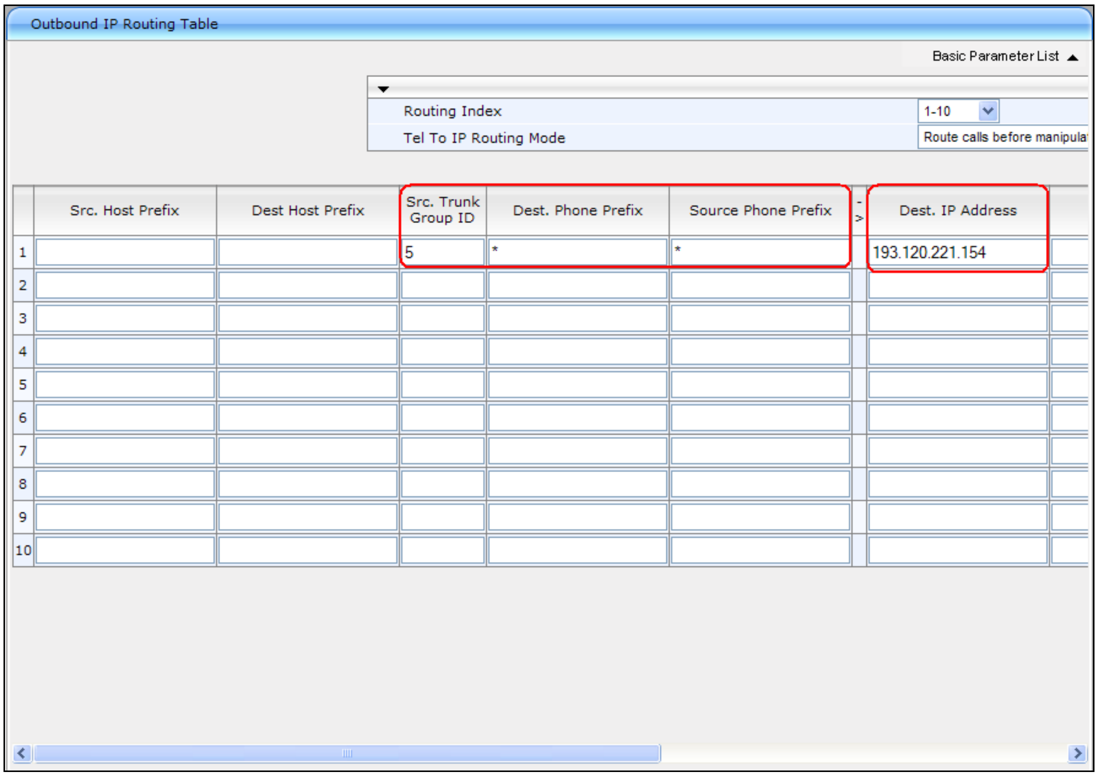
6.5.10. Configure Inbound IP Routing Rules

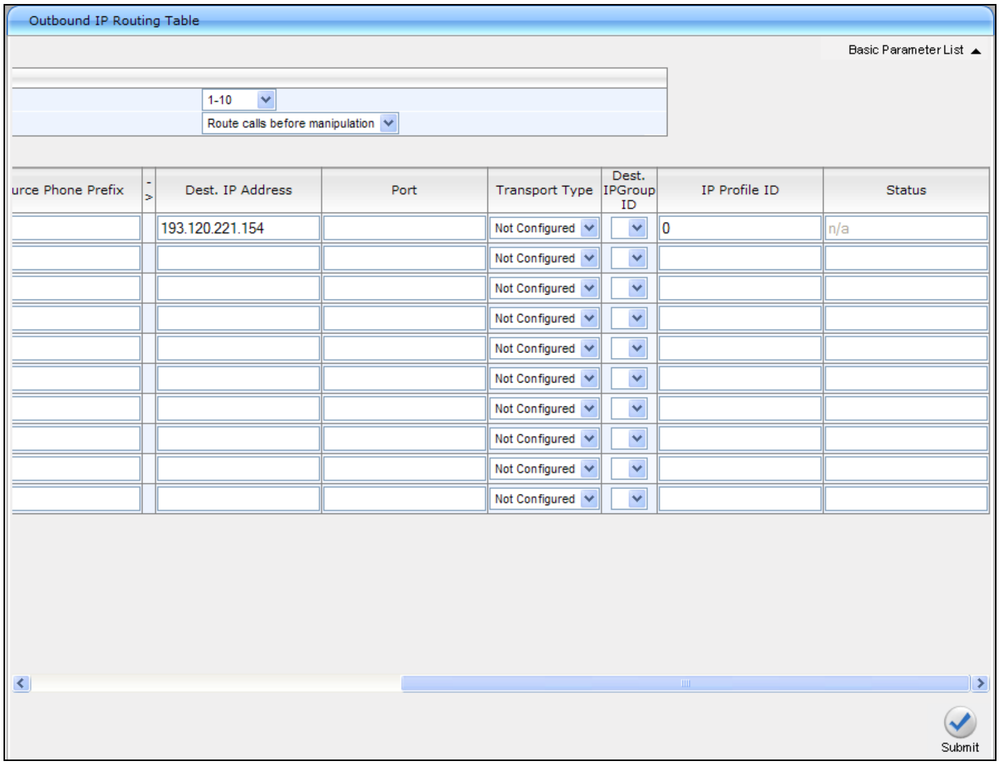
Open the **Inbound IP Routing Table** page (**Configuration** tab → **Protocol Configuration** menu → **Routing Tables** submenu → **IP to Trunk Group Routing**). Configure the inbound IP routing rules, refer to **[15-18]** for additional information on Inbound IP Routing Table. The figure below illustrates the Inbound IP Routing Table used in these Application Notes.



6.5.11. Configure Outbound IP Routing Rules

Open the **Outbound IP Routing Table** page (**Configuration** tab → **Protocol Configuration** menu → **Routing Tables** submenu → **Tel to IP Routing**). Configure the **Src. Trunk Group ID** with the appropriate trunk number (i.e. **5**), **Dest. Phone Prefix**, **Source Phone Prefix** with the appropriate patterns (i.e. *) and **Dest. IP Address** with the IP Address of signalling interface of Session Manager (i.e. **193.120.221.154**). For additional information on configuring Outbound IP Routing Table, refer to **[15-18]**. Click on **Submit** button to save changes. The following pictures illustrate the configuration done in these Application Notes.





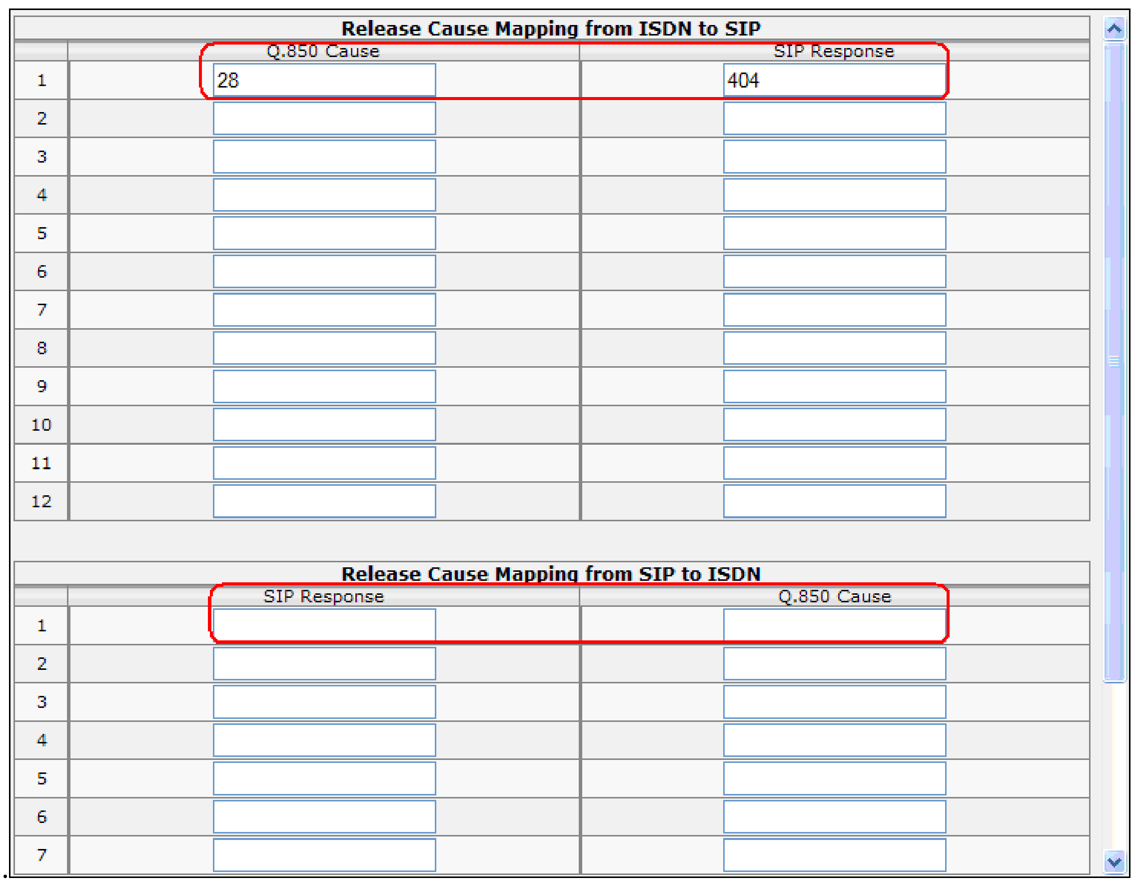
Note: To save the changes to flash memory, refer to **Section 6.1.1**

6.5.12. Configure Release Cause Mapping

Open the **Release Cause Mapping** page (**Configuration** tab→ **Protocol Configuration** menu → **Routing Tables** submenu → **Release Cause Mapping**). The page is separated into two sections:

- In the **Release Cause Mapping from ISDN to SIP** group, map different Q.850 Release Causes to SIP Responses
- In the **Release Cause Mapping from SIP to ISDN** group, map different SIP Responses to Q.850 Release Causes

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 your changes. The figure below illustrates the **Release Cause Mapping** Page.



Note: To save the changes to flash memory, refer to **Section 6.1.1**

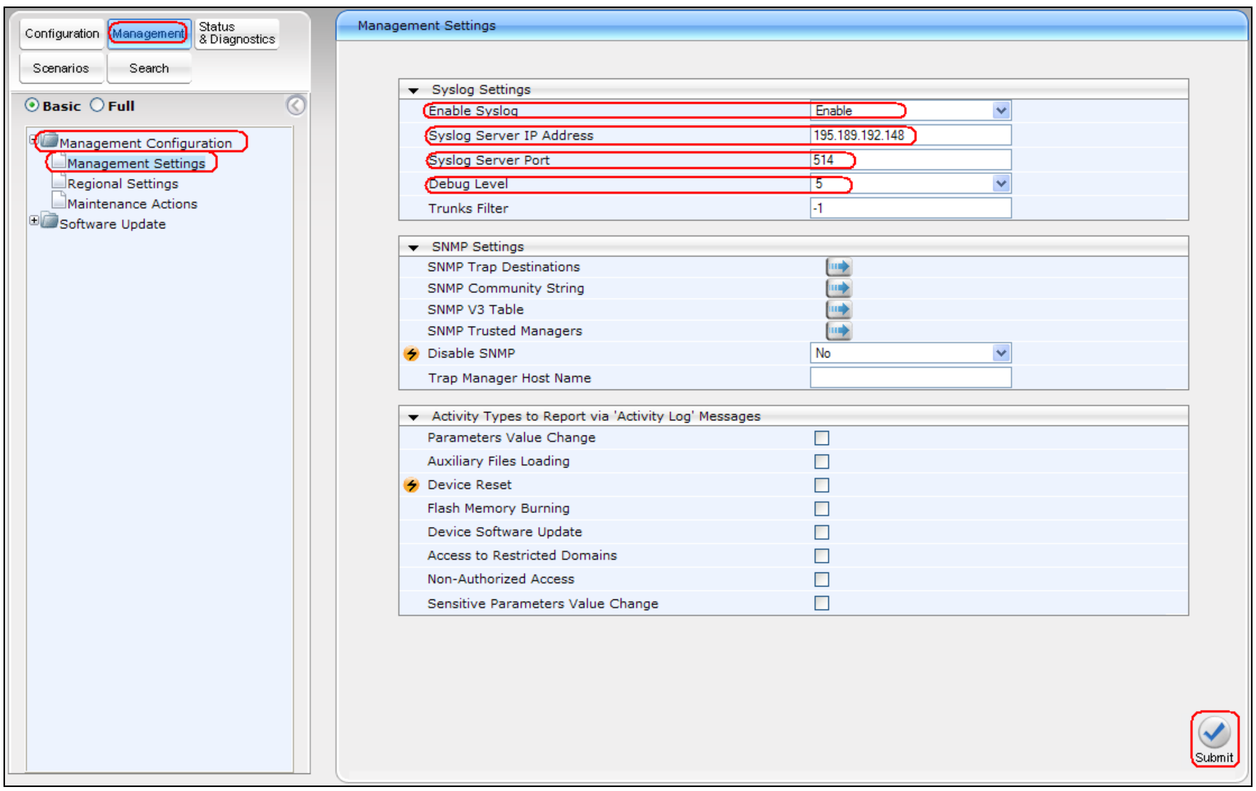
6.6. Configure the Syslog Parameters for Debug Assistance

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 **Management Settings** page (**Management** tab → **Management Configuration** menu → **Management 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. **195.189.192.148**)
- **Syslog Server Port:** Set to port utilized on the Syslog Server listening device (i.e. **514**)
- **Debug Level:** Set to **5** to capture proper level of debug information

Click the **Submit** button to save changes. The figure below illustrates setting use in these sample Application Notes.

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



Note: To save the changes to flash memory, refer to **Section 6.1.1**

7. Verification Steps

This section provides the verification steps that may be performed to verify that Avaya Aura^{TM} enterprise network can establish and receive calls with Mediant 3000.

7.1. Verify Avaya AuraTM Communication Manager Feature Server Trunk Status

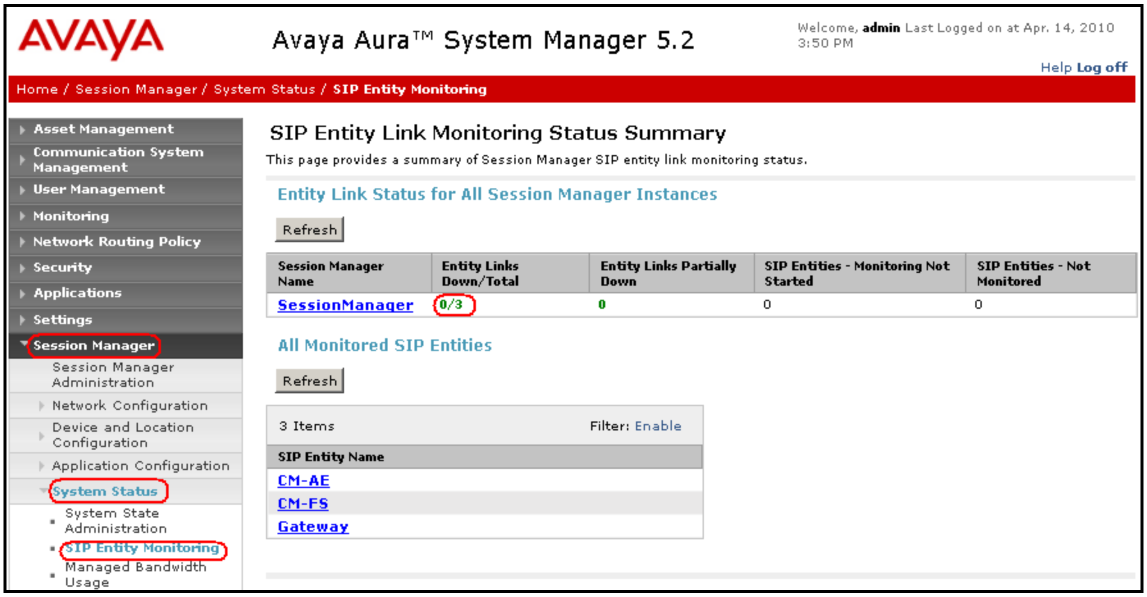
On Communication Manager Feature Server, ensure that all the signalling groups are in-service status by issuing the command status **signalling-group n** where **n** is the signalling group number.

```
status signaling-group 1
                         STATUS SIGNALING GROUP
        Group ID: 2                              Active NCA-TSC Count: 0
      Group Type: sip                             Active CA-TSC Count: 0
  Signaling Type: facility associated signaling
     Group State: in-service
```

```
status signaling-group 3
                         STATUS SIGNALING GROUP
        Group ID: 3                              Active NCA-TSC Count: 0
      Group Type: sip                             Active CA-TSC Count: 0
  Signaling Type: facility associated signaling
     Group State: in-service
```

7.2. SIP Monitoring on Avaya Aura™ Session Manager

Expand the menu on the left and navigate **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.

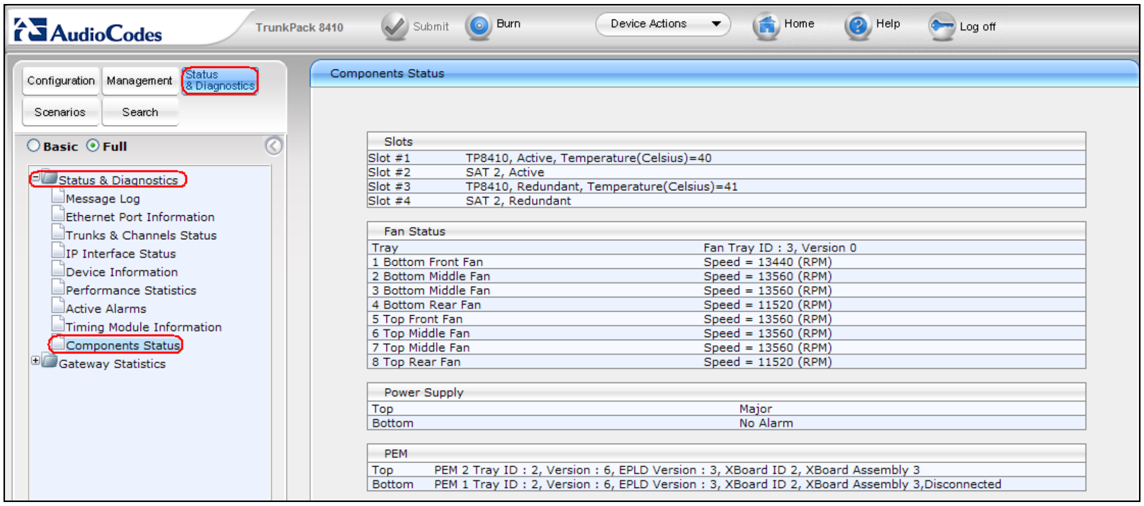


7.3. Utilizing the Web Interface to observe Status

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.

7.3.1. Device Status

To view the status of the device's hardware components, open the **Components Status** page (**Status & Diagnostics** tab → **Status & Diagnostics** menu → **Components Status).** The figure below illustrates **Component Status** page for an HA/Redundant gateway where the TP18410 board in slot 1 is active and the second in slot 3 is Redundant.



7.3.2. Device Information

To access the **Device Information** page Open the **Device Information** page (**Status & Diagnostics** tab → **Status & Diagnostics** menu → **Device Information**).

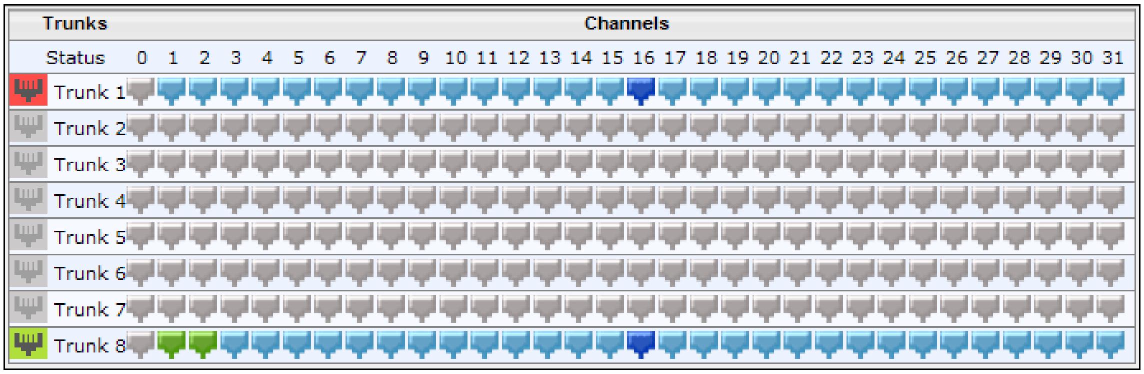
General Settings	
MAC Address:	00908f1e7553
Serial Number:	1996115
Board Type:	TrunkPack 8410
Device Up Time:	0d:4h:11m:30s:88th
Device Administrative State:	Unlocked
Device Operational State:	Enabled
Flash Size [bytes]:	33554432
RAM Size [bytes]:	536870912
CPU Speed [MHz]:	450

Versions	
Version ID:	6.00A.014.005
DSP Type:	2
DSP Software Version:	60017
DSP Software Name:	491096AE3
Flash Version:	217

Loaded Files		
Call Progress Tones File Name:	usa_tones_1221.dat	Delete
Loaded Coder Table :	Default CODERTABLE	

7.3.3. Trunks and Channels Status

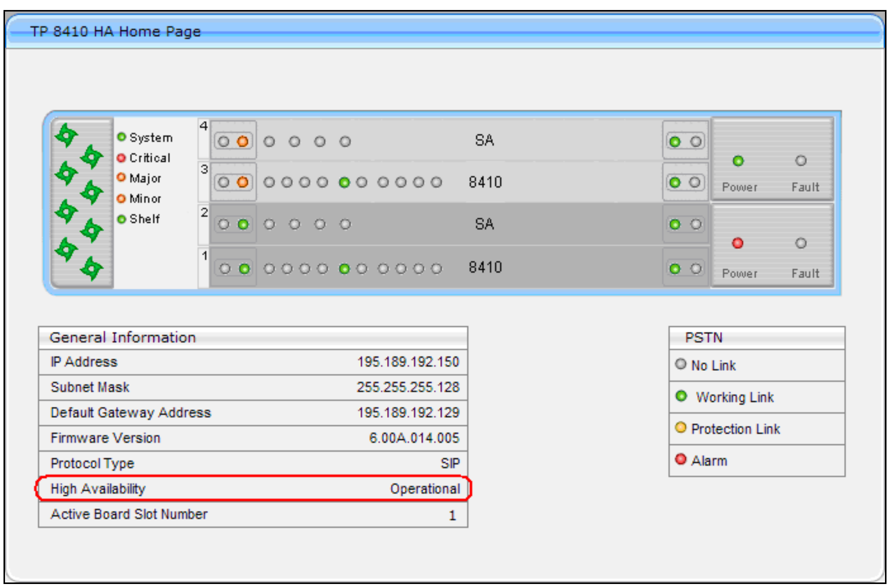
To view the status of the device's trunks and the trunks' channels open the **Trunks & Channels Status** page (**Status & Diagnostics** tab → **Status & Diagnostics** menu → **Trunks & Channels Status**). The following figure illustrates the Trunks and Channel status, where the symbol of the port in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



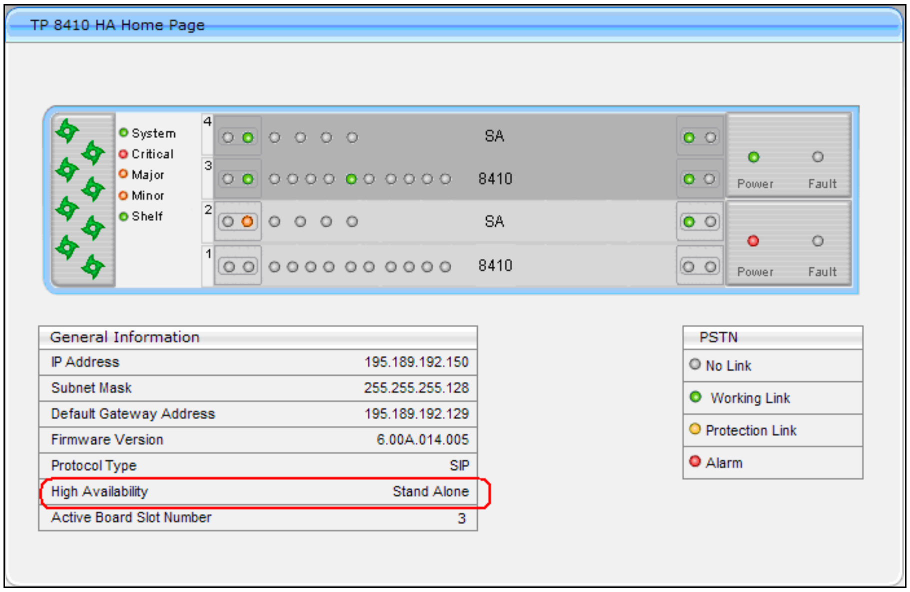
7.3.4. Gateway Home Page

To view the status of the device home page, open the **Home** page be selecting from the top the following ICON: 

. The following figure display and HA system that has both TP8410 modules in service, ready for switchover, as described by the General information table, where **High Availability** is **Operational**.



The following figure display and HA system that has both TP8410 modules in service, but not ready for switchover, as described by the General information table, where **High Availability** is **Stand Alone**.



8. General Test Approach

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

- PSTN calls from and to Avaya IP endpoint
- Calling with various Avaya SIP telephone models
- Support G.711A/MU G.729A/B
- Various PTSN dialing plans including national and international calling, toll-free, operator, directory assistance and direct inward dialed calling
- SIP transport using UDP and TCP

Advanced Interoperability:

- 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
- EC500 for Communication Manager

The serviceability testing focused on verifying the ability of solution to recover from adverse conditions, such as network failures and failover between the Active/ Stanby modules on the gateway.

8.1. Test Results and Remarks

All test cases were executed. During the compliance testing, it has been noted and issue with hold /resume on incoming call to SIP endpoints if shuffling is enabled on the signaling trunk group. A workaround is available by disabling shuffling on the trunk used.

9. Conclusion

As illustrated in these Application Notes, AudioCodes Mediant 3000 Gateway can successfully offer access to E1 PSTN to an enterprise telephony network built on Avaya Aura™ Session Manager and Avaya Aura™ Communication Manager Feature Server.

10. Additional References

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

[1] "Avaya Aura™ Session Manager Overview", Document Number 03-603323, Issue 2, Release 5.2, November 2009

[2] "Installing and Upgrading Avaya Aura™ Session Manager", Document Number 03-603473, Issue 2, Release 5.2, November 2009

[3] "Administering Avaya Aura™ Session Manager", Document Number 03-603324, Issue 2.1, Release 5.2, August 2010

[4] "Avaya Aura™ Session Manager Case Studies", Document Number 03-603478, Issue 3, Release 6.0, June 2010

[5] "Maintaining and Troubleshooting Avaya Aura™ Session Manager, Document Number 03-603325, Issue 1.3, Release 5.2, January 2010

[6] "Installing and Configuring Avaya Aura™ System Platform", Release 1.1, November 2009

[7] "Installing and Upgrading Avaya Aura™ System Manager", Release 5.2, January 2010

[8] "Avaya Aura™ Communication Manager Overview", Document Number 03-300468, Issue 6, Release 5.2, May 2009

[9] "Administering Avaya Aura™ Communication Manager", Document Number 03-300509, Issue 5.0, Release 5.2, May 2009

[10] "Avaya Aura™ Communication Manager Feature Description and Implementation", Document Number 555-245-205, Issue 7.0, Release 5.2, May 2009

[11] "Administering Network Connectivity on Avaya Aura™ Communication Manager", Document Number 555-233-504, Issue 14, May 2009

[12] "SIP Support in Avaya Aura™ Communication Manager Running on Avaya S8xxx Servers", Document Number 555-245-206, Issue 9, May 2009

[13] "Administering Avaya Aura™ Communication Manager as a Feature Server", Document Number 03-603479, Issue 1.2, Release 5.2, January 2010

[14] "Configuring 9600-Series SIP Phones with Avaya Aura™ Session Manager Release 5.2 – Issue 1.0", Application Note, February 2010

AudioCodes Mediant 3000 references, are available at http://www.audiocodes.com/support

[15] LTRT-69017_Mediant_2000_and_Mediant_3000_SIP_Release_Notes_Ver_6.0.pdf

[16] LTRT-89708_Mediant_3000_SIP_User's_Manual_Ver_6.0.pdf

[17] LTRT-94706_Mediant_3000_and_IPmedia_3000_SIP-MGCP-MEGACO_Installation_Manual_Ver 6.0.pdf

[18] LTRT-52305_Product_Reference_Manual_for_SIP_CPE_Devices_Ver_6.0.pdf

APPENDIX

In this section are presented the relevant configuration files for the devices used in the DevConnect compliance testing.

Configure the Number Manipulation tables

Open the required **Number Manipulation** page (**Configuration** tab→ **Protocol Configuration** menu → **Manipulation Tables** submenu → **Dest Number IP→Tel**, **Dest Number Tel→IP**, **Source Number IP→Tel**, or **Source Number Tel→IP**); the relevant Manipulation table page is displayed (e.g., **Source Phone Number Manipulation Table for Tel→IP Calls** page). The figure shows an example of the use of manipulation rules for Tel-to-IP source phone number manipulation:

Index	Source Trunk Group	Source IP Group	Destination Prefix	Source Prefix	Stripped Digits From Left	Stripped Digits From Right	Prefix to Add	Suffix to Add	Number of Digits to Leave	Presentation
1	-1	2	03	201	0	0	971		255	Allowed
2	0	0		1001	4	0	5	23	255	Restricted
3	-1	-1	*	123451001#	0	0		8	4	Not Configured
4	-1	-1	*	[30-40]x	0	1	2		255	Not Configured
5	-1	-1	[6,7,8]	2001	5	0	3		255	Not Configured

Index 1: When the destination number has the prefix 03 (e.g., 035000), source number prefix 201 (e.g., 20155), and from source IP Group ID 2, the source number is changed to, for example, 97120155.

Index 2: When the source number has prefix 1001 (e.g., 1001876), it is changed to 587623.

Index 3: When the source number has prefix 123451001 (e.g., 1234510012001), it is changed to 20018.

Index 4: When the source number has prefix from 30 to 40 and a digit (e.g., 3122), it is changed to 2312.

Index 5: When the destination number has the prefix 6, 7, or 8 (e.g., 85262146), source number prefix 2001, it is changed to 3146.

From the **Table Index** drop-down list, select the range of entries that you want to edit. Configure the Number Manipulation table according to the table below. Click the **Submit** button to save your changes.

Configure outbound IP routing rules

Open the **Outbound IP Routing Table** page (**Configuration** tab → **Protocol Configuration** menu → **Routing Tables** submenu → **Tel to IP Routing**).

Routing Index	1-10
Tel To IP Routing Mode	Route calls before manipulation

	Src. IPGroupID	Src. Host Prefix	Dest Host Prefix	Src. Trunk Group ID	Dest. Phone Prefix	Source Phone Prefix	->	Dest. IP Address	Port	Transport Type	Dest. IPGroup ID	IP Profile ID	Status
1				*	10	100		10.33.45.63		Not Configured		1	n/a
2				0	20	*				Not Configured	1	0	n/a
3				1	[30-40]	*		10.33.45.64		Not Configured		0	n/a
4				*	[5,7-9]	*		domain.com		Not Configured		0	n/a
5				*	00	*		0.0.0.0		Not Configured		0	n/a
6	2	domain.com			*	*		10.33.45.65		Not Configured			

The figure above shows the following configured outbound IP routing rules:

Rule 1: If the called phone prefix is 10 and the caller's phone prefix is 100, the call is assigned settings configured for IP Profile ID 1 and sent to IP address 10.33.45.63.

Rule 2: If the called phone prefix is 20 and the caller is all prefixes (*), the call is sent to the destination according to IP Group 1 (which in turn is associated with a Proxy Set ID providing the IP address).

Rule 3: If the called phone prefix is between 30 and 40, and the caller belongs to Trunk Group ID 1, the call is sent to IP address 10.33.45.64.

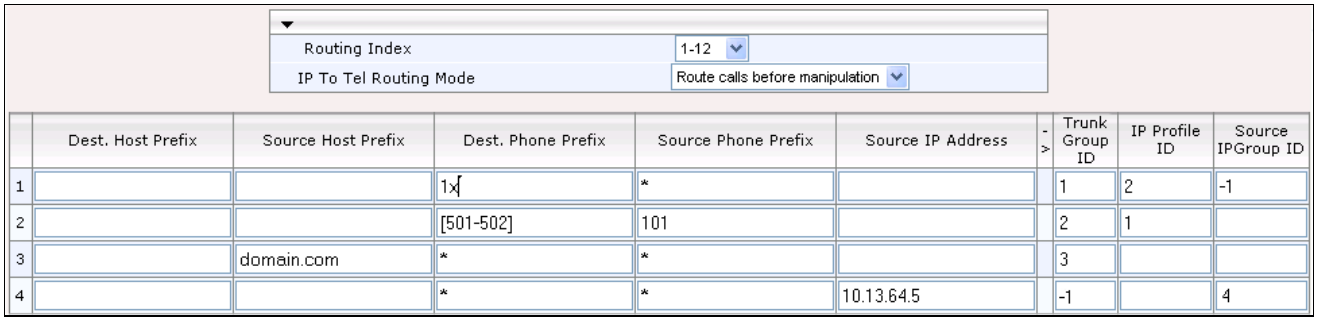
Rule 4: If the called phone prefix is either 5, 7, 8, or 9 and the caller is all (*), the call is sent to domain.com.

Rule 5: If the called phone prefix is 00 and the caller is all (*), the call is discarded.

Rule 6: If an incoming IP call pertaining to Source IP Group 2 with domain.com as source host prefix in its Request URI, the IP call is sent to IP address 10.33.45.65. From the **Routing Index** drop-down list, select the range of entries that you want to add. Configure the outbound IP routing rules according to the table below. Click the **Submit** button to apply your changes.

Configure inbound IP routing rules

Open the **Inbound IP Routing Table** page (**Configuration** tab → **Protocol Configuration** menu → **Routing Tables** submenu → **IP to Trunk Group Routing**).



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

	Dest. Host Prefix	Source Host Prefix	Dest. Phone Prefix	Source Phone Prefix	Source IP Address	->	Trunk Group ID	IP Profile ID	Source IPGroup ID
1			1x	*			1	2	-1
2			[501-502]	101			2	1	
3		domain.com	*	*			3		
4			*	*	10.13.64.5		-1		4

The previous figure shows the following configured inbound IP routing rules:

Rule 1: If the incoming IP call destination phone prefix is between 10 and 19, the call is assigned settings configured for IP Profile ID 2 and routed to Trunk Group ID 1.

Rule 2: If the incoming IP call destination phone prefix is between 501 and 502, and source phone prefix is 101, the call is assigned settings configured for IP Profile ID 1 and routed to Trunk Group ID 2.

Rule 3: If the incoming IP call has a From URI host prefix as domain.com, the call is routed to Trunk Group ID 3.

Rule 4: If the incoming IP call has IP address 10.13.64.5 in the INVITE's Contact header, the call is considered an IP-to-IP call and assigned to Source IP Group 4. This call is later routed according to the outbound IP routing rules for this Source IP Group configured in the **Outbound IP Routing Table**.

From the **Routing Index** drop-down list, select the range of entries that you want to add. Configure the inbound IP routing rule according to the table below. Click the **Submit** button to save your changes.

AudioCodes Mediant 3000 configuration file

Here it is presented the Mediant 3000 **ini** file used in these Application Notes.

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

;Board: TrunkPack 8410
;Serial Number: 1996115
;Slot Number: 1
;Software Version: 6.00A.014.005
;DSP Software Version: 491096AE3 => 600.17
;Board IP Address: 195.189.192.150
;Board Subnet Mask: 255.255.255.0
;Board Default Gateway: 195.189.192.129
;Ram size: 512M   Flash size: 32M
;Num of DSP Cores: 126  Num DSP Channels: 2016
;Profile: NONE
;Key features:
;Board Type: TrunkPack 8410
;SS7 Links: MTP2=16 MTP3=16 M2UA=16 M3UA=1
;IP Media: Conf VoicePromptAnnounc(H248.9) CALEA TrunkTesting POC
;DSP Voice features: IpmDetector RTCP-XR AMRPolicyManagement
;Coders: G723 G729 G728 NETCODER GSM-FR GSM-EFR AMR EVRC-QCELP G727 ILBC EVRC-B AMR-WB
G722 H263 H264 MPEG4 EG711
;PSTN FALLBACK Supported
;E1Trunks=84
;T1Trunks=84;Security: IPSEC MediaEncryption StrongEncryption EncryptControlProtocol
;PSTN Protocols: IUA=16
;Channel Type: RTP ATM PCI DspCh=2016 IPMediaDspCh=480
;M3K HA
;Control Protocols: MGCP MEGACO H323 SIP TPNCP SASurvivability IP2IP=100 MSFT
;Default features:
;Coders: G711 G726
;
;----------------------------------------------

[SYSTEM Params]
DNSPriServerIP = 80.179.52.100
SyslogServerIP = 195.189.192.148
EnableSyslog = 1

[BSP Params]
PCMLawSelect = 1
TDMBusSpeed = 3
StorageServerNetworkAddress = 255.255.255.255

[ControlProtocols Params]
AdminStateLockControl = 0
cpRecordCoder = 'PCMA'
[MGCP Params]

[MEGACO Params]
EP_Num_0 = 0
EP_Num_1 = 1
EP_Num_2 = 0
EP_Num_3 = 0
EP_Num_4 = 0

[PSTN Params]
```

```
ProtocolType_0 = 1
ProtocolType_1 = 0
ProtocolType_2 = 0
ProtocolType_3 = 0
ProtocolType_4 = 0
ProtocolType_5 = 0
ProtocolType_6 = 0
ProtocolType_7 = 1
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
ProtocolType_21 = 0
ProtocolType_22 = 0
ProtocolType_23 = 0
ProtocolType_24 = 0
ProtocolType_25 = 0
ProtocolType_26 = 0
ProtocolType_27 = 0
ProtocolType_28 = 0
ProtocolType_29 = 0
ProtocolType_30 = 0
ProtocolType_31 = 0
ProtocolType_32 = 0
ProtocolType_33 = 0
ProtocolType_34 = 0
ProtocolType_35 = 0
ProtocolType_36 = 0
ProtocolType_37 = 0
ProtocolType_38 = 0
ProtocolType_39 = 0
ProtocolType_40 = 0
ProtocolType_41 = 0
ProtocolType_42 = 0
ProtocolType_43 = 0
ProtocolType_44 = 0
ProtocolType_45 = 0
ProtocolType_46 = 0
ProtocolType_47 = 0
ProtocolType_48 = 0
ProtocolType_49 = 0
ProtocolType_50 = 0
ProtocolType_51 = 0
ProtocolType_52 = 0
ProtocolType_53 = 0
ProtocolType_54 = 0
ProtocolType_55 = 0
ProtocolType_56 = 0
ProtocolType_57 = 0
ProtocolType_58 = 0
ProtocolType_59 = 0
ProtocolType_60 = 0
ProtocolType_61 = 0
```

```
ProtocolType_62 = 0
ClockMaster_0 = 1
ClockMaster_1 = 0
ClockMaster_2 = 0
ClockMaster_3 = 0
ClockMaster_4 = 0
ClockMaster_5 = 0
ClockMaster_6 = 0
ClockMaster_7 = 0
ClockMaster_8 = 0
ClockMaster_9 = 0
ClockMaster_10 = 0
ClockMaster_11 = 0
ClockMaster_12 = 0
ClockMaster_13 = 0
ClockMaster_14 = 0
ClockMaster_15 = 0
ClockMaster_16 = 0
ClockMaster_17 = 0
ClockMaster_18 = 0
ClockMaster_19 = 0
ClockMaster_20 = 0
ClockMaster_21 = 0
ClockMaster_22 = 0
ClockMaster_23 = 0
ClockMaster_24 = 0
ClockMaster_25 = 0
ClockMaster_26 = 0
ClockMaster_27 = 0
ClockMaster_28 = 0
ClockMaster_29 = 0
ClockMaster_30 = 0
ClockMaster_31 = 0
ClockMaster_32 = 0
ClockMaster_33 = 0
ClockMaster_34 = 0
ClockMaster_35 = 0
ClockMaster_36 = 0
ClockMaster_37 = 0
ClockMaster_38 = 0
ClockMaster_39 = 0
ClockMaster_40 = 0
ClockMaster_41 = 0
ClockMaster_42 = 0
ClockMaster_43 = 0
ClockMaster_44 = 0
ClockMaster_45 = 0
ClockMaster_46 = 0
ClockMaster_47 = 0
ClockMaster_48 = 0
ClockMaster_49 = 0
ClockMaster_50 = 0
ClockMaster_51 = 0
ClockMaster_52 = 0
ClockMaster_53 = 0
ClockMaster_54 = 0
ClockMaster_55 = 0
ClockMaster_56 = 0
ClockMaster_57 = 0
ClockMaster_58 = 0
ClockMaster_59 = 0
ClockMaster_60 = 0
```

```
ClockMaster_61 = 0
ClockMaster_62 = 0
TerminationSide_0 = 1
TerminationSide_1 = 0
TerminationSide_2 = 0
TerminationSide_3 = 0
TerminationSide_4 = 0
TerminationSide_5 = 0
TerminationSide_6 = 0
TerminationSide_7 = 1
TerminationSide_8 = 0
TerminationSide_9 = 0
TerminationSide_10 = 0
TerminationSide_11 = 0
TerminationSide_12 = 0
TerminationSide_13 = 0
TerminationSide_14 = 0
TerminationSide_15 = 0
TerminationSide_16 = 0
TerminationSide_17 = 0
TerminationSide_18 = 0
TerminationSide_19 = 0
TerminationSide_20 = 0
TerminationSide_21 = 0
TerminationSide_22 = 0
TerminationSide_23 = 0
TerminationSide_24 = 0
TerminationSide_25 = 0
TerminationSide_26 = 0
TerminationSide_27 = 0
TerminationSide_28 = 0
TerminationSide_29 = 0
TerminationSide_30 = 0
TerminationSide_31 = 0
TerminationSide_32 = 0
TerminationSide_33 = 0
TerminationSide_34 = 0
TerminationSide_35 = 0
TerminationSide_36 = 0
TerminationSide_37 = 0
TerminationSide_38 = 0
TerminationSide_39 = 0
TerminationSide_40 = 0
TerminationSide_41 = 0
TerminationSide_42 = 0
TerminationSide_43 = 0
TerminationSide_44 = 0
TerminationSide_45 = 0
TerminationSide_46 = 0
TerminationSide_47 = 0
TerminationSide_48 = 0
TerminationSide_49 = 0
TerminationSide_50 = 0
TerminationSide_51 = 0
TerminationSide_52 = 0
TerminationSide_53 = 0
TerminationSide_54 = 0
TerminationSide_55 = 0
TerminationSide_56 = 0
TerminationSide_57 = 0
TerminationSide_58 = 0
TerminationSide_59 = 0
```

```
TerminationSide_60 = 0
TerminationSide_61 = 0
TerminationSide_62 = 0
FramingMethod_0 = c
FramingMethod_1 = 0
FramingMethod_2 = 0
FramingMethod_3 = 0
FramingMethod_4 = 0
FramingMethod_5 = 0
FramingMethod_6 = 0
FramingMethod_7 = c
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
FramingMethod_21 = 0
FramingMethod_22 = 0
FramingMethod_23 = 0
FramingMethod_24 = 0
FramingMethod_25 = 0
FramingMethod_26 = 0
FramingMethod_27 = 0
FramingMethod_28 = 0
FramingMethod_29 = 0
FramingMethod_30 = 0
FramingMethod_31 = 0
FramingMethod_32 = 0
FramingMethod_33 = 0
FramingMethod_34 = 0
FramingMethod_35 = 0
FramingMethod_36 = 0
FramingMethod_37 = 0
FramingMethod_38 = 0
FramingMethod_39 = 0
FramingMethod_40 = 0
FramingMethod_41 = 0
FramingMethod_42 = 0
FramingMethod_43 = 0
FramingMethod_44 = 0
FramingMethod_45 = 0
FramingMethod_46 = 0
FramingMethod_47 = 0
FramingMethod_48 = 0
FramingMethod_49 = 0
FramingMethod_50 = 0
FramingMethod_51 = 0
FramingMethod_52 = 0
FramingMethod_53 = 0
FramingMethod_54 = 0
FramingMethod_55 = 0
FramingMethod_56 = 0
FramingMethod_57 = 0
FramingMethod_58 = 0
```

```
FramingMethod_59 = 0
FramingMethod_60 = 0
FramingMethod_61 = 0
FramingMethod_62 = 0
LineCode_0 = 2
LineCode_1 = 0
LineCode_2 = 0
LineCode_3 = 0
LineCode_4 = 0
LineCode_5 = 0
LineCode_6 = 0
LineCode_7 = 2
LineCode_8 = 0
LineCode_9 = 0
LineCode_10 = 0
LineCode_11 = 0
LineCode_12 = 0
LineCode_13 = 0
LineCode_14 = 0
LineCode_15 = 0
LineCode_16 = 0
LineCode_17 = 0
LineCode_18 = 0
LineCode_19 = 0
LineCode_20 = 0
LineCode_21 = 0
LineCode_22 = 0
LineCode_23 = 0
LineCode_24 = 0
LineCode_25 = 0
LineCode_26 = 0
LineCode_27 = 0
LineCode_28 = 0
LineCode_29 = 0
LineCode_30 = 0
LineCode_31 = 0
LineCode_32 = 0
LineCode_33 = 0
LineCode_34 = 0
LineCode_35 = 0
LineCode_36 = 0
LineCode_37 = 0
LineCode_38 = 0
LineCode_39 = 0
LineCode_40 = 0
LineCode_41 = 0
LineCode_42 = 0
LineCode_43 = 0
LineCode_44 = 0
LineCode_45 = 0
LineCode_46 = 0
LineCode_47 = 0
LineCode_48 = 0
LineCode_49 = 0
LineCode_50 = 0
LineCode_51 = 0
LineCode_52 = 0
LineCode_53 = 0
LineCode_54 = 0
LineCode_55 = 0
LineCode_56 = 0
LineCode_57 = 0
```

```
LineCode_58 = 0
LineCode_59 = 0
LineCode_60 = 0
LineCode_61 = 0
LineCode_62 = 0
CASProtocolEnable = 0
[SS7 Params]
[Voice Engine Params]
CallProgressTonesFilename = 'usa_tones_1221.dat'
DisableRTCPRandomize = 1
DTMFDetectorSensitivity = 1
SRTPTxPacketMKISize = 1
[WEB Params]
LogoWidth = '145'
HTTPSCipherString = 'ALL'

[SIP Params]
PLAYRBTONE2IP = 1
MEDIACHANNELS = 60
PLAYRBTONE2TEL = 1
USESIPURIFORDIVERSIONHEADER = 1
CHANNELSELECTMODE = 1
GWDEBUGLEVEL = 5
ENABLEEARLYMEDIA = 1
SIPGATEWAYNAME = '195.189.192.138'
DISCONNECTONBROKENCONNECTION = 0
ISFAXUSED = 1
HOLDFORMAT = 1
SIPTRANSPORTTYPE = 1
TLSLOCALSIPPORT = 5064
LOCALISDNRBSOURCE = 1
MEDIASECURITYBEHAVIOUR = 1
USEDIGITFORSPECIALDTMF = 1
FAXCNGMODE = 1
DIGITALOOSBEHAVIORFORTRUNK_0 = 0
DIGITALOOSBEHAVIORFORTRUNK_1 = -1
DIGITALOOSBEHAVIORFORTRUNK_2 = -1
DIGITALOOSBEHAVIORFORTRUNK_3 = -1
DIGITALOOSBEHAVIORFORTRUNK_4 = -1
DIGITALOOSBEHAVIORFORTRUNK_5 = -1
DIGITALOOSBEHAVIORFORTRUNK_6 = -1
DIGITALOOSBEHAVIORFORTRUNK_7 = 0
DIGITALOOSBEHAVIORFORTRUNK_8 = -1
DIGITALOOSBEHAVIORFORTRUNK_9 = -1
DIGITALOOSBEHAVIORFORTRUNK_10 = -1
DIGITALOOSBEHAVIORFORTRUNK_11 = -1
DIGITALOOSBEHAVIORFORTRUNK_12 = -1
DIGITALOOSBEHAVIORFORTRUNK_13 = -1
DIGITALOOSBEHAVIORFORTRUNK_14 = -1
DIGITALOOSBEHAVIORFORTRUNK_15 = -1
DIGITALOOSBEHAVIORFORTRUNK_16 = -1
DIGITALOOSBEHAVIORFORTRUNK_17 = -1
DIGITALOOSBEHAVIORFORTRUNK_18 = -1
DIGITALOOSBEHAVIORFORTRUNK_19 = -1
DIGITALOOSBEHAVIORFORTRUNK_20 = -1
DIGITALOOSBEHAVIORFORTRUNK_21 = -1
DIGITALOOSBEHAVIORFORTRUNK_22 = -1
DIGITALOOSBEHAVIORFORTRUNK_23 = -1
DIGITALOOSBEHAVIORFORTRUNK_24 = -1
DIGITALOOSBEHAVIORFORTRUNK_25 = -1
DIGITALOOSBEHAVIORFORTRUNK_26 = -1
DIGITALOOSBEHAVIORFORTRUNK_27 = -1
```

```
DIGITALOOSBEHAVIORFORTRUNK_28 = -1
DIGITALOOSBEHAVIORFORTRUNK_29 = -1
DIGITALOOSBEHAVIORFORTRUNK_30 = -1
DIGITALOOSBEHAVIORFORTRUNK_31 = -1
DIGITALOOSBEHAVIORFORTRUNK_32 = -1
DIGITALOOSBEHAVIORFORTRUNK_33 = -1
DIGITALOOSBEHAVIORFORTRUNK_34 = -1
DIGITALOOSBEHAVIORFORTRUNK_35 = -1
DIGITALOOSBEHAVIORFORTRUNK_36 = -1
DIGITALOOSBEHAVIORFORTRUNK_37 = -1
DIGITALOOSBEHAVIORFORTRUNK_38 = -1
DIGITALOOSBEHAVIORFORTRUNK_39 = -1
DIGITALOOSBEHAVIORFORTRUNK_40 = -1
DIGITALOOSBEHAVIORFORTRUNK_41 = -1
DIGITALOOSBEHAVIORFORTRUNK_42 = -1
DIGITALOOSBEHAVIORFORTRUNK_43 = -1
DIGITALOOSBEHAVIORFORTRUNK_44 = -1
DIGITALOOSBEHAVIORFORTRUNK_45 = -1
DIGITALOOSBEHAVIORFORTRUNK_46 = -1
DIGITALOOSBEHAVIORFORTRUNK_47 = -1
DIGITALOOSBEHAVIORFORTRUNK_48 = -1
DIGITALOOSBEHAVIORFORTRUNK_49 = -1
DIGITALOOSBEHAVIORFORTRUNK_50 = -1
DIGITALOOSBEHAVIORFORTRUNK_51 = -1
DIGITALOOSBEHAVIORFORTRUNK_52 = -1
DIGITALOOSBEHAVIORFORTRUNK_53 = -1
DIGITALOOSBEHAVIORFORTRUNK_54 = -1
DIGITALOOSBEHAVIORFORTRUNK_55 = -1
DIGITALOOSBEHAVIORFORTRUNK_56 = -1
DIGITALOOSBEHAVIORFORTRUNK_57 = -1
DIGITALOOSBEHAVIORFORTRUNK_58 = -1
DIGITALOOSBEHAVIORFORTRUNK_59 = -1
DIGITALOOSBEHAVIORFORTRUNK_60 = -1
DIGITALOOSBEHAVIORFORTRUNK_61 = -1
DIGITALOOSBEHAVIORFORTRUNK_62 = -1

[SCTP Params]


[VXML Params]

[IPsec Params]

[Audio Staging Params]

[SNMP Params]

[Video Params]
;
;  *** TABLE DspTemplates ***
; This table contains hidden elements and will not be exposed.
; This table exists on board and will be saved during restarts
;
;
;  *** TABLE PREFIX ***
;
;
[ PREFIX ]
FORMAT PREFIX_Index = PREFIX_DestinationPrefix, PREFIX_DestAddress,
PREFIX_SourcePrefix, PREFIX_ProfileId, PREFIX_MeteringCode, PREFIX_DestPort,
PREFIX_SrcIPGroupID, PREFIX_DestHostPrefix, PREFIX_DestIPGroupID,
PREFIX_SrcHostPrefix, PREFIX_TransportType, PREFIX_SrcTrunkGroupID;
```

```
PREFIX 0 = *, 193.120.221.154, *, 0, 255, 0, -1, , -1, , -1, 5;
[ \PREFIX ]
;
;  *** TABLE CoderName ***
; This table contains hidden elements and will not be exposed.
; This table exists on board and will be saved during restarts
;
;
;  *** TABLE TrunkGroup ***
;
;
[ TrunkGroup ]
FORMAT TrunkGroup_Index = TrunkGroup_TrunkGroupNum, TrunkGroup_FirstTrunkId,
TrunkGroup_FirstBChannel, TrunkGroup_LastBChannel, TrunkGroup_FirstPhoneNumber,
TrunkGroup_ProfileId, TrunkGroup_LastTrunkId, TrunkGroup_Module;
TrunkGroup 0 = 5, 7, 1, 31, 1000, 0, 7, 255;
[ \TrunkGroup ]
;
;  *** TABLE NumberMapIp2Tel ***
;
;
[ NumberMapIp2Tel ]
FORMAT NumberMapIp2Tel_Index = NumberMapIp2Tel_DestinationPrefix,
NumberMapIp2Tel_SourcePrefix, NumberMapIp2Tel_SourceAddress,
NumberMapIp2Tel_NumberType, NumberMapIp2Tel_NumberPlan,
NumberMapIp2Tel_RemoveFromLeft, NumberMapIp2Tel_RemoveFromRight,
NumberMapIp2Tel_LeaveFromRight, NumberMapIp2Tel_Prefix2Add,
NumberMapIp2Tel_Suffix2Add, NumberMapIp2Tel_IsPresentationRestricted,
NumberMapIp2Tel_SrcTrunkGroupID, NumberMapIp2Tel_SrcIPGroupID;
NumberMapIp2Tel 1 = *, *, *, 255, 255, 0, 0, 255, , , 255, -1, -1;
[ \NumberMapIp2Tel ]
;
;  *** TABLE NumberMapTel2Ip ***
;
;
[ NumberMapTel2Ip ]
FORMAT NumberMapTel2Ip_Index = NumberMapTel2Ip_DestinationPrefix,
NumberMapTel2Ip_SourcePrefix, NumberMapTel2Ip_SourceAddress,
NumberMapTel2Ip_NumberType, NumberMapTel2Ip_NumberPlan,
NumberMapTel2Ip_RemoveFromLeft, NumberMapTel2Ip_RemoveFromRight,
NumberMapTel2Ip_LeaveFromRight, NumberMapTel2Ip_Prefix2Add,
NumberMapTel2Ip_Suffix2Add, NumberMapTel2Ip_IsPresentationRestricted,
NumberMapTel2Ip_SrcTrunkGroupID, NumberMapTel2Ip_SrcIPGroupID;
NumberMapTel2Ip 0 = +, 44*, *, 255, 255, 0, 0, 255, , , 255, -1, -1;
NumberMapTel2Ip 1 = , 44*, *, 255, 255, 0, 0, 255, +1, , 255, -1, -1;
[ \NumberMapTel2Ip ]
;
;  *** TABLE SourceNumberMapIp2Tel ***
;
;
[ SourceNumberMapIp2Tel ]
FORMAT SourceNumberMapIp2Tel_Index = SourceNumberMapIp2Tel_DestinationPrefix,
SourceNumberMapIp2Tel_SourcePrefix, SourceNumberMapIp2Tel_SourceAddress,
SourceNumberMapIp2Tel_NumberType, SourceNumberMapIp2Tel_NumberPlan,
SourceNumberMapIp2Tel_RemoveFromLeft, SourceNumberMapIp2Tel_RemoveFromRight,
SourceNumberMapIp2Tel_LeaveFromRight, SourceNumberMapIp2Tel_Prefix2Add,
SourceNumberMapIp2Tel_Suffix2Add, SourceNumberMapIp2Tel_IsPresentationRestricted,
SourceNumberMapIp2Tel_SrcTrunkGroupID, SourceNumberMapIp2Tel_SrcIPGroupID;
SourceNumberMapIp2Tel 0 = *, +1, *, 255, 255, 2, 0, 255, , , 255, -1, -1;
SourceNumberMapIp2Tel 1 = *, +, *, 255, 255, 1, 0, 255, , , 255, -1, -1;
[ \SourceNumberMapIp2Tel ]
;
```

```
;  *** TABLE PstnPrefix ***
;
;
[ PstnPrefix ]
FORMAT PstnPrefix_Index = PstnPrefix_DestPrefix, PstnPrefix_TrunkGroupId,
PstnPrefix_SourcePrefix, PstnPrefix_SourceAddress, PstnPrefix_ProfileId,
PstnPrefix_SrcIPGroupID, PstnPrefix_DestHostPrefix, PstnPrefix_SrcHostPrefix;
PstnPrefix 0 = *, 5, *, , 0, -1, , ;
[ \PstnPrefix ]
;
;  *** TABLE CauseMapIsdn2Sip ***
;
;
[ CauseMapIsdn2Sip ]
FORMAT CauseMapIsdn2Sip_Index = CauseMapIsdn2Sip_IsdnReleaseCause,
CauseMapIsdn2Sip_SipResponse;
CauseMapIsdn2Sip 0 = 28, 404;
[ \CauseMapIsdn2Sip ]
;
;  *** TABLE ProxyIp ***
;
;
[ ProxyIp ]
FORMAT ProxyIp_Index = ProxyIp_IpAddress, ProxyIp_TransportType, ProxyIp_ProxySetId;
ProxyIp 2 = 195.189.192.142, 0, 3;
[ \ProxyIp ]
;
;  *** TABLE TxDtmfOption ***
;
;
[ TxDtmfOption ]
FORMAT TxDtmfOption_Index = TxDtmfOption_Type;
TxDtmfOption 0 = 4;
[ \TxDtmfOption ]
;
;  *** TABLE ProxySet ***
;
;
[ ProxySet ]
FORMAT ProxySet_Index = ProxySet_EnableProxyKeepAlive, ProxySet_ProxyKeepAliveTime,
ProxySet_ProxyLoadBalancingMethod, ProxySet_IsProxyHotSwap, ProxySet_SRD,
ProxySet_ClassificationInput;
ProxySet 0 = 0, 60, 0, 0, 0, 0;
ProxySet 3 = 0, 60, 0, 0, 0, 0;
[ \ProxySet ]
;
;  *** TABLE IPGroup ***
;
;
[ IPGroup ]
FORMAT IPGroup_Index = IPGroup_Type, IPGroup_Description, IPGroup_ProxySetId,
IPGroup_SIPGroupName, IPGroup_ContactUser, IPGroup_EnableSurvivability,
IPGroup_ServingIPGroup, IPGroup_SipReRoutingMode, IPGroup_AlwaysUseRouteTable,
IPGroup_RoutingMode, IPGroup_SRD, IPGroup_MediaRealm, IPGroup_ClassifyByProxySet,
IPGroup_ProfileId, IPGroup_MaxNumOfRegUsers, IPGroup_InboundManSet,
IPGroup_OutboundManSet;
IPGroup 1 = 0, , -1, , , 0, -1, 0, 0, -1, 0, , 1, 0, -1, -1, -1;
IPGroup 2 = 0, , -1, , , 0, -1, 0, 0, -1, 0, , 1, 0, -1, -1, -1;
[ \IPGroup ]
;
;  *** TABLE CodersGroup0 ***
;
```

```
;
[ CodersGroup0 ]
FORMAT CodersGroup0_Index = CodersGroup0_Name, CodersGroup0_pTime, CodersGroup0_rate,
CodersGroup0_PayloadType, CodersGroup0_Sce;
CodersGroup0 0 = g711Alaw64k, 20, 0, -1, 0;
CodersGroup0 1 = g711Ulaw64k, 20, 0, -1, 0;
CodersGroup0 2 = g729, 20, 0, -1, 0;
[ \CodersGroup0 ]
;
;  *** TABLE CodersGroup1 ***
;
;
[ CodersGroup1 ]
FORMAT CodersGroup1_Index = CodersGroup1_Name, CodersGroup1_pTime, CodersGroup1_rate,
CodersGroup1_PayloadType, CodersGroup1_Sce;
CodersGroup1 0 = g711Alaw64k, 20, 0, -1, 0;
[ \CodersGroup1 ]
;
;  *** TABLE CodersGroup2 ***
;
;
[ CodersGroup2 ]
FORMAT CodersGroup2_Index = CodersGroup2_Name, CodersGroup2_pTime, CodersGroup2_rate,
CodersGroup2_PayloadType, CodersGroup2_Sce;
CodersGroup2 0 = g729, 20, 0, -1, 0;
[ \CodersGroup2 ]
;
;  *** TABLE InterfaceTable ***
;
;
[ InterfaceTable ]
FORMAT InterfaceTable_Index = InterfaceTable_ApplicationTypes,
InterfaceTable_InterfaceMode, InterfaceTable_IPAddress, InterfaceTable_PrefixLength,
InterfaceTable_Gateway, InterfaceTable_VlanID, InterfaceTable_InterfaceName;
InterfaceTable 0 = 6, 10, 195.189.192.150, 24, 195.189.192.129, 1, O+M+C;
[ \InterfaceTable ]
```

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.