Avaya Solution & Interoperability Test Lab

# Application Notes for Configuring SIP Trunks among AudioCodes Mediant 1000 MSBG e-SBC, Avaya Aura® Session Manager, and Avaya Aura® Communication Manager - Issue 1.0

## Abstract

These Application Notes describe a sample configuration for a network that uses Avaya Aura® Session Manager to connect AudioCodes Mediant 1000 MSBG e-SBC and Avaya Aura® Communication Manager using SIP trunks.

The AudioCodes Mediant 1000 MSBG e-SBC is a SIP Session Border Controller (SBC) that manages and protects the flow of SIP signaling and related media across an untrusted IP network. The compliance testing focused on telephony scenarios between an enterprise site, where the AudioCodes Mediant 1000 MSBG e-SBC, Session Manager, and Communication Manager were located, and a second site simulating a service provider service node.

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.

# 1. Introduction

These Application Notes describe a sample configuration for a network that uses Avaya Aura® Session Manager to connect AudioCodes Mediant 1000 MSBG e-SBC and Avaya Aura® Communication Manager using SIP trunks.

The Mediant 1000 MSBG is an all-in-one multi-service access solution for Service Providers offering managed services and distributed Enterprises. This multi-service business gateway is designed to provide converged Voice & Data services for business customers at wire speed, while maintaining SLA parameters for voice quality.

The compliance testing focused on telephony scenarios between an enterprise site, where the AudioCodes Mediant 1000 MSBG e-SBC, Session Manager, and Communication Manager were located, and a second site simulating a service provider service node.

# 2. General Test Approach and Test Results

The general test approach was to make calls between the main enterprise site and the 2nd site simulating a service provider service node using various codec settings and exercising common telephony features.

## 2.1. Interoperability Compliance Testing

The compliance testing focused on interoperability between AudioCodes Mediant 1000 MSBG e-SBC and Session Manager / Communication Manager by making calls between the enterprise site and a second site simulating a service provider service node that were connected through the Mediant 1000 MSBG e-SBC using direct SIP trunks. The following functions and features were tested:

- Calls from both SIP and non-SIP endpoints between sites.
- G.711MU and G.729AB codec support.
- Proper recognition of DTMF transmissions by navigating voicemail menus.
- Proper operation of voicemail with message waiting indicators (MWI).
- PBX features including Multiple Call Appearances, Hold, Transfer, and Conference.
- Extended telephony features using Communication Manager Feature Name Extensions (FNE) such as Call Forwarding, Call Park, Call Pickup, Automatic Redial, Automatic Call Back, and Send All Calls.
- Proper system recovery after a Mediant 1000 MSBG e-SBC restart and/or re-establishment of broken IP connectivity.

## 2.2. Test Results

The AudioCodes Mediant 1000 MSBG e-SBC passed compliance testing.

## 2.3. Support

For technical support on the AudioCodes Mediant 1000 MSBG e-SBC, visit their online support at http://www.audiocodes.com/support.

# 3. Reference Configuration

**Figure 1** illustrates the test configuration. The test configuration shows two sites connected via a SIP trunk across an untrusted IP network: the main enterprise site and a second site that simulates a service provider service node. The AudioCodes Mediant 1000 MSBG e-SBC Session Border Controller (SBC) is at the edge of the main site. The public side of the Mediant 1000 MSBG e-SBC is connected to the untrusted network and the private side is connected to the trusted corporate LAN.

All SIP traffic between two sites flows through the Mediant 1000 MSBG e-SBC. In this manner, the Mediant 1000 MSBG e-SBC can protect the main site's infrastructure from any SIP-based attacks. The voice communication across the untrusted network uses SIP over TCP and RTP for the media streams.

Also connected to the LAN at the main site are:

- An Avaya S8300D Server running Avaya Aura® Communication Manager in an Avaya G450 Media Gateway. Avaya Aura® Communication Manager Messaging is also running on the Avaya S8300D Server to provide voice mail functionality.
- A Dell™ PowerEdge™ R610 Server running Avaya Aura® System Manager. System Manager provides management functions for Session Manager.
- An HP ProLiant DL360 G7 Server running Avaya Aura® Session Manager that provides SIP registrar and proxy server functions for SIP endpoints in the enterprise IP telephony network.

The Session Manager connects the Mediant 1000 MSBG e-SBC and Communication Manager using SIP trunks. Endpoints include both SIP and non-SIP endpoints. An ISDN-PRI trunk connects the media gateway to the PSTN.

The 2$^{nd}$ site (shown as a cloud), simulates a service provider service node, and also comprises of a Communication Manager, System Manager, and Session Manager, with both SIP and non-SIP endpoints.

The SIP endpoints located at both sites are registered to the local Session Manager.

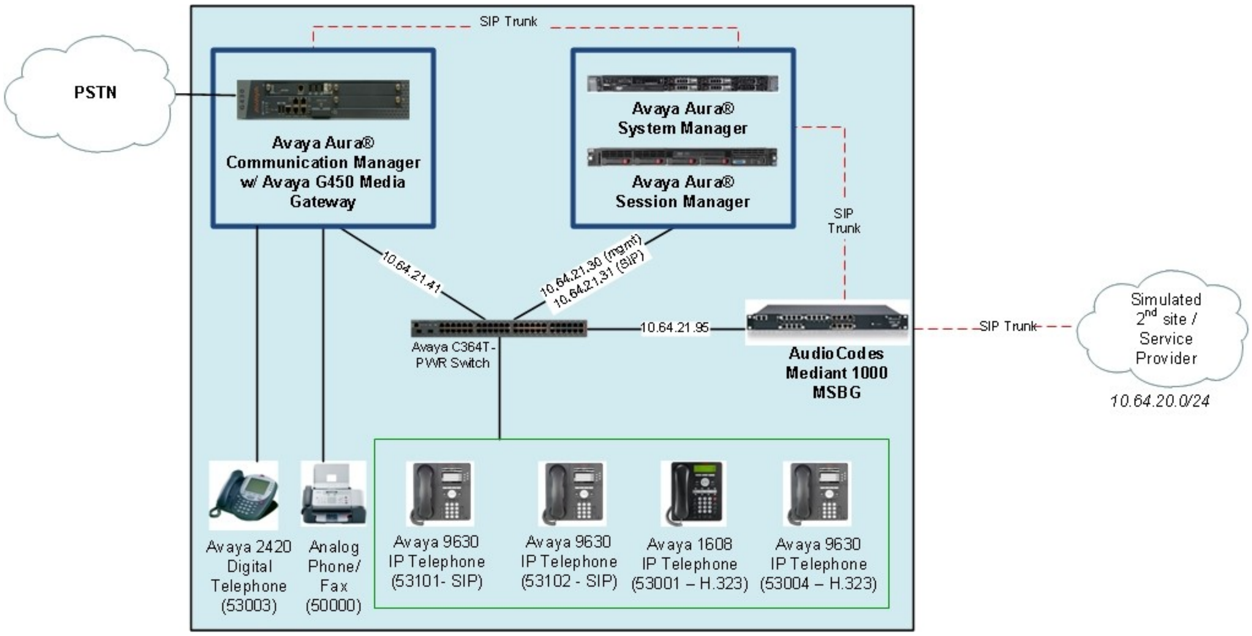



**Figure 1: AudioCodes Mediant 1000 MSBG e-SBC SIP Trunking Test Configuration**

# 4. Equipment and Software Validated

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

| Equipment | Software |
|---|---|
| Avaya S8300D Server with a Avaya G450 Media Gateway | Avaya Aura® Communication Manager 6.0.1, R016x.00.1.510.1, Patch 18621<br>(Avaya Aura® System Platform: 6.0.2.1.5) |
| Dell™ PowerEdge™ R610 Server | Avaya Aura® System Manager: 6.1.0 (Build No. – 6.1.0.4.5072-6.1.4.11)<br>(Avaya Aura® System Platform: 6.0.2.1.5) |
| HP ProLiant DL360 G7 Server | Avaya Aura® Session Manager 6.1.0 (Build No. – 6.1.0.0.42003-6.1.0.610012) |
| Avaya 9600 Series IP Telephones<br>• H.323<br>• SIP | <br>3.1. Service Pack 1<br>2.6.4 |
| Fax Machine | - |
| AudioCodes Mediant 1000 MSBG e-SBC | 6.2<br>• MSBG based software: M1000_MSBG_SIP_F6.20A.014.003.zip<br>• Firmware load: M1000_MSBG_SIP_F6.20A.014.003.zip |

# 5. Configure Communication Manager

This section describes the Communication Manager configuration at the main enterprise site to support the network shown in **Figure 1**. It is assumed the procedures necessary to support SIP and connectivity to Session Manager have been performed as described in **[2]** and **[3]**; however, some of the configuration is shown in this section and the next section as a reference.

The configuration of Communication Manager was performed using the System Access Terminal (SAT). The screens in this section display the Communication Manager configuration that was administered and already in place prior to the start of compliance testing. After the completion of the configuration, a **save translation** command was performed to make the changes permanent.

| Step | Description |
|---|---|
| 1. | **System Capacities**<br>On **Page 2** of the **display system-parameters customer-options** form, verify that the **Maximum Administered SIP Trunks** is sufficient for the combination of trunks to AudioCodes and any other SIP trunking entities. Be aware that for each call between a non-SIP endpoint at the enterprise site and Audio Codes, one SIP trunk is used for the duration of the call. An Avaya SIP endpoint uses two SIP trunks for the duration of the call. |

```
display system-parameters customer-options                      Page   2 of  11
                                OPTIONAL FEATURES

IP PORT CAPACITIES                                            USED
                     Maximum Administered H.323 Trunks: 12000 22
           Maximum Concurrently Registered IP Stations: 18000 3
             Maximum Administered Remote Office Trunks: 12000 0
Maximum Concurrently Registered Remote Office Stations: 18000 0
              Maximum Concurrently Registered IP eCons: 414   0
  Max Concur Registered Unauthenticated H.323 Stations: 100   0
                        Maximum Video Capable Stations: 18000 0
                   Maximum Video Capable IP Softphones: 18000 1
                       Maximum Administered SIP Trunks: 24000 20
  Maximum Administered Ad-hoc Video Conferencing Ports: 24000 0
   Maximum Number of DS1 Boards with Echo Cancellation: 522   0
                             Maximum TN2501 VAL Boards: 128   0
                     Maximum Media Gateway VAL Sources: 250   0
           Maximum TN2602 Boards with 80 VoIP Channels: 128   0
          Maximum TN2602 Boards with 320 VoIP Channels: 128   0
   Maximum Number of Expanded Meet-me Conference Ports: 300   0

        (NOTE: You must logoff & login to effect the permission changes.)
```

| Step | Description |
|---|---|
| 2. | **IP network region**<br>All equipment at the main site were located in a single IP network region (IP network region 1) using the parameters described below. Use the **display ip-network-region** command to view these settings. The example below shows the values used during compliance testing.<br><br>▪ **Authoritative Domain**: ***avaya.com***<br>This field was configured to match the domain name configured on Session Manager. The domain will appear in the "From" header of SIP messages originating from this IP region.<br>▪ **Name**: Any descriptive name may be used (if desired).<br>▪ **Intra-region IP-IP Direct Audio**: ***yes***<br>**Inter-region IP-IP Direct Audio**: ***yes***<br>By default, IP-IP direct audio (media shuffling) is enabled to allow audio traffic to be sent directly between IP endpoints without using media resources in the Avaya Media Gateway. Shuffling can be further restricted at the trunk level on the **Signaling Group** form.<br>▪ **Codec Set**: ***1***<br>The codec set contains the list of codecs available for calls within this IP network region. |
| | <pre>display ip-network-region 1                                    Page   1 of  20<br>                               IP NETWORK REGION<br>  Region: 1<br>Location:         **Authoritative Domain: avaya.com**<br>    **Name:**<br>MEDIA PARAMETERS                **Intra-region IP-IP Direct Audio: yes**<br>      **Codec Set: 1**            **Inter-region IP-IP Direct Audio: yes**<br>   UDP Port Min: 2048                      IP Audio Hairpinning? n<br>   UDP Port Max: 3329<br>DIFFSERV/TOS PARAMETERS<br> Call Control PHB Value: 46<br>        Audio PHB Value: 46<br>        Video PHB Value: 26<br>802.1P/Q PARAMETERS<br> Call Control 802.1p Priority: 6<br>        Audio 802.1p Priority: 6<br>        Video 802.1p Priority: 5      AUDIO RESOURCE RESERVATION PARAMETERS<br>H.323 IP ENDPOINTS                                      RSVP Enabled? n<br>  H.323 Link Bounce Recovery? y<br> Idle Traffic Interval (sec): 20<br>   Keep-Alive Interval (sec): 5<br>            Keep-Alive Count: 5</pre> |
| | |

| Step | Description |
|---|---|
| 3. | **Codecs**<br>IP codec set 1 was used during compliance testing. Multiple codecs were listed in priority order to allow the codec used by a specific call to be negotiated during call establishment. The example below shows the values used during compliance testing. It should be noted that when testing the use of each individual codec, only the single codec under test was included in the list. |
| | <pre>display ip-codec-set 1                                         Page   1 of   2<br><br>                          IP Codec Set<br><br>    Codec Set: 1<br><br>    Audio        Silence      Frames   Packet<br>    Codec        Suppression  Per Pkt  Size(ms)<br> 1: G.711MU           n         2        20<br> 2: G.729AB           n         2        20<br> 3:<br> 4:<br> 5:<br> 6:<br> 7:</pre> |
| 4. | **Node Names**<br>Use the **change node-names ip** command to create a node name for the IP address of Session Manager. Enter a descriptive name in the **Name** column and the IP address assigned to Session Manager in the **IP address** column. |
| | <pre>change node-names ip                                           Page   1 of   2<br>                                  IP NODE NAMES<br>    Name              IP Address<br>CM_20_40            10.64.20.40<br>SM_20_31            10.64.20.31<br>SM_21_31            10.64.21.31<br>default             0.0.0.0<br>msgserver           10.64.21.41<br>procr               10.64.21.41<br>procr6              ::</pre> |

| Step | Description |
| --- | --- |
| 5. | **Signaling Group**<br>Signaling group 1 was used for the signaling group associated with the SIP trunk group between Communication Manager and Session Manager. Signaling group 1 was configured using the parameters highlighted below.<br>▪ **Near-end Node Name**: ***procr*** This node name maps to the IP address of the Avaya S8300D Server. Node names are defined using the **change node-names ip** command.<br>▪ **Far-end Node Name**: ***SM_21_31*** This node name maps to the IP address of Session Manager.<br>▪ **Far-end Network Region**: ***1*** This defines the IP network region which contains Session Manager.<br>▪ **Far-end Domain**: ***avaya.com*** This domain is sent in the "To" header of SIP messages of calls using this signaling group.<br>▪ **Direct IP-IP Audio Connections**: ***y*** This field must be set to ***y*** to enable media shuffling on the SIP trunk. |

```
display signaling-group 1
                                SIGNALING GROUP

 Group Number: 1              Group Type: sip
  IMS Enabled? n        Transport Method: tls
        Q-SIP? n                                               SIP Enabled LSP? n
     IP Video? n                                     Enforce SIPS URI for SRTP? y
  Peer Detection Enabled? y  Peer Server: SM



   Near-end Node Name: procr                  Far-end Node Name: SM_21_31
 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? y                      Initial IP-IP Direct Media? n
H.323 Station Outgoing Direct Media? n               Alternate Route Timer(sec): 6
```

| Step | Description |
|---|---|
| 6. | **Trunk Group**<br>Trunk group 1 was used for the SIP trunk group between Communication Manager and Session Manager. Trunk group 1 was configured using the parameters highlighted below.<br>▪ **Group Type:** ***sip*** This field sets the type of the trunk group.<br>▪ **TAC:** ***101*** Enter an valid value consistent with the Communication Manager dial plan<br>▪ **Member Assignment Method:** ***auto*** Set to Auto.<br>▪ **Signaling Group**: ***1*** This field is set to the signaling group shown in the previous step.<br>▪ **Number of Members:** ***10*** This field represents the number of trunk group members in the SIP trunk group. It determines how many simultaneous SIP calls can be supported by the configuration. Each SIP call between two SIP endpoints (whether internal or external) requires two SIP trunks for the duration of the call. Thus, a call from a SIP telephone to another SIP telephone will use two SIP trunks. A call between a non-SIP telephone and a SIP telephone will only use one trunk. |

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

Group Number: 1                   Group Type: sip           CDR Reports: y
  Group Name: to SM_21_31                COR: 1        TN: 1        TAC: 101
   Direction: two-way       Outgoing Display? n
 Dial Access? n                                    Night Service:
Queue Length: 0
Service Type: tie                  Auth Code? n
                                              Member Assignment Method: auto
                                                       Signaling Group: 1
                                                     Number of Members: 10
```

| Step | Description |
|---|---|
| | **Trunk Group – continued**<br>On **Page 3**:<br>▪ The **Numbering Format** field was set to ***unk-pvt***. This field specifies the format of the calling party number sent to the far-end.<br>▪ The default values may be retained for the other fields. |
| | <pre>display trunk-group 1                                            Page   3 of  21<br>TRUNK FEATURES<br>          ACA Assignment? n            Measured: none<br>                                                     Maintenance Tests? y<br><br><br><br>                     **Numbering Format: unk-pvt**<br>                                           UUI Treatment: service-provider<br><br>                                            Replace Restricted Numbers? n<br>                                           Replace Unavailable Numbers? n<br><br><br>                                Modify Tandem Calling Number: no<br><br><br><br><br> Show ANSWERED BY on Display? y</pre> |
| 7. | **Private Numbering**<br>Private Numbering defines the calling party number to be sent to the far-end. In the example shown below, all calls originating from a 5-digit extension beginning with 5 and routed across any trunk group will be sent as a 5 digit calling number. The calling party number is sent to the far-end in the SIP "From" header. |
| | <pre>display private-numbering 0                                      Page   1 of   2<br>                           NUMBERING - PRIVATE FORMAT<br><br>Ext Ext              Trk        Private          Total<br>Len Code             Grp(s)     Prefix           Len<br> **5**  **5**                                             **5**     Total Administered: 1<br>                                                          Maximum Entries: 540</pre> |

| Step | Description |
|---|---|
| 8. | **Automatic Alternate Routing**<br>Automatic Alternate Routing (AAR) was used to route calls to Session Manager. In the example shown, dialed numbers that begin with 3 and are 5 digits long use route pattern 1. Route pattern 1 routes calls to the trunk group defined in **Step 6**. |

```
display aar analysis 3                                        Page   1 of   2
                             AAR DIGIT ANALYSIS TABLE
                                   Location: all            Percent Full: 1

          Dialed           Total     Route    Call   Node   ANI
          String          Min  Max  Pattern   Type   Num    Reqd
    3                     5    5     1        aar           n
    4                     7    7     999      aar           n
    531                   5    5     1        unku          n
    532                   5    5     1        unku          n
    59997                 5    5     99       aar           n
```

| Step | Description |
|---|---|
| 9. | **Route Pattern**<br>Route pattern 1 was used for calls destined for the 2nd site through Session Manager and the Mediant 1000 MSBG e-SBC. Route pattern 1 was configured using the parameters highlighted below.<br>▪ **Pattern Name**: Any descriptive name.<br>▪ **Grp No**: ***1*** This field is set to the trunk group number defined in **Step 6**.<br>▪ **FRL**: ***0*** This field sets the Facility Restriction Level of the trunk. It must be set to an appropriate level to allow authorized users to access the trunk. The level of 0 is the least restrictive. |

```
display route-pattern 1                                       Page   1 of   3
                    Pattern Number: 1   Pattern Name: to SM_21_31
                             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                    0                                       n   user
 2:                                                                   n   user
 3:                                                                   n   user
 4:                                                                   n   user
 5:                                                                   n   user
 6:                                                                   n   user

     BCC VALUE  TSC CA-TSC    ITC BCIE Service/Feature PARM  No. Numbering LAR
    0 1 2 M 4 W     Request                                 Dgts Format
                                                         Subaddress
 1: y y y y y n  n            rest                                lev0-pvt  none
 2: y y y y y n  n            rest                                          none
 3: y y y y y n  n            rest                                          none
 4: y y y y y n  n            rest                                          none
 5: y y y y y n  n            rest                                          none
 6: y y y y y n  n            rest                                          none
```

# 6. Configure Session Manager

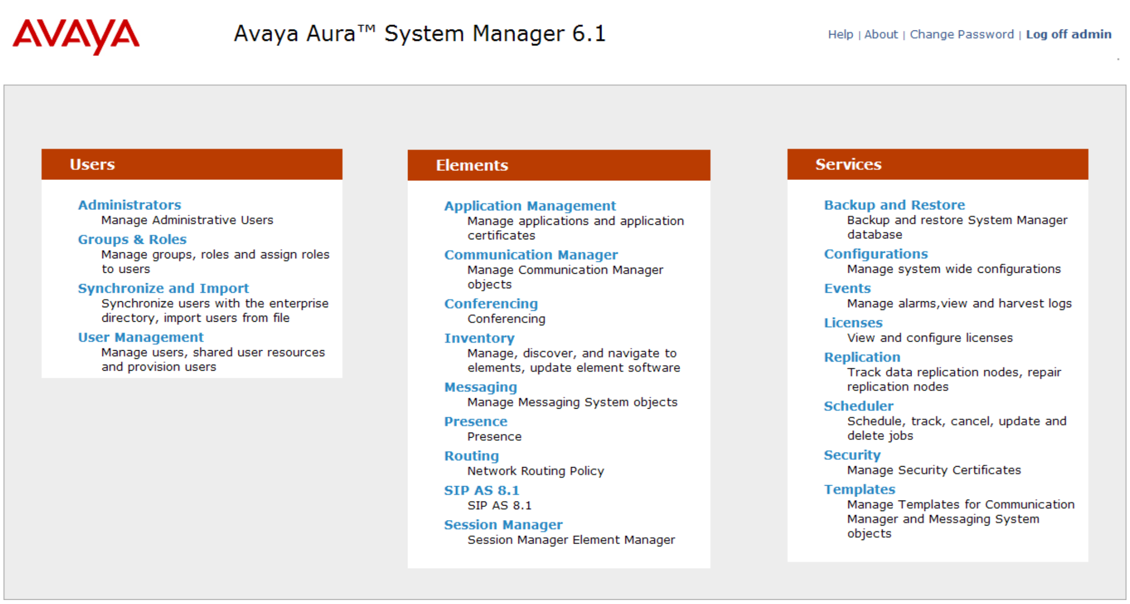
This section provides the procedures for configuring Session Manager as provisioned in the reference configuration. All provisioning for Session Manager is performed via the System Manager web interface.

The following sections assume that Session Manager and System Manager have been installed and that network connectivity exists between the two platforms.

The Session Manager server contains an SM-100 security module that provides the network interface for all inbound and outbound SIP signaling and media transport to all provisioned SIP entities. During compliance testing, the IP address assigned to the SM-100 interface is 10.64.21.31 as specified in **Figure 1**. The Session Manager server also has a separate network interface used for connectivity to System Manager for provisioning Session Manager. The IP address assigned to the Session Manager management interface is 10.64.21.30. The SM-100 interface and the management interface were both connected to the same IP network. If desired, the SM-100 interface can be configured to use a different network than the management interface.

The procedures described in this section include configurations in the following areas:

- **SIP domain**
- Logical/physical **Locations** that can be occupied by SIP Entities
- **SIP Entities** corresponding to the SIP telephony systems (including Communication Manager and Session Border Controller) and Session Manager itself
- **Entity Links** which define the SIP trunk parameters used by Session Manager when routing calls to/from SIP Entities
- **Time Ranges** during which routing policies are active
- **Routing Policies** which control call routing between the SIP Entities
- **Dial Patterns** which govern to which SIP Entity a call is routed

| | |
|---|---|
| 1. | **Login**<br>Access the Session Manager administration web interface by entering https://<*ip-addr*>/network-login/ as the URL in an Internet browser, where <*ip-addr*> is the IP address of the System Manager server.<br><br>Log in with the appropriate credentials. The main page for the administrative interface is shown below.<br> |

2. **Add SIP Domain**

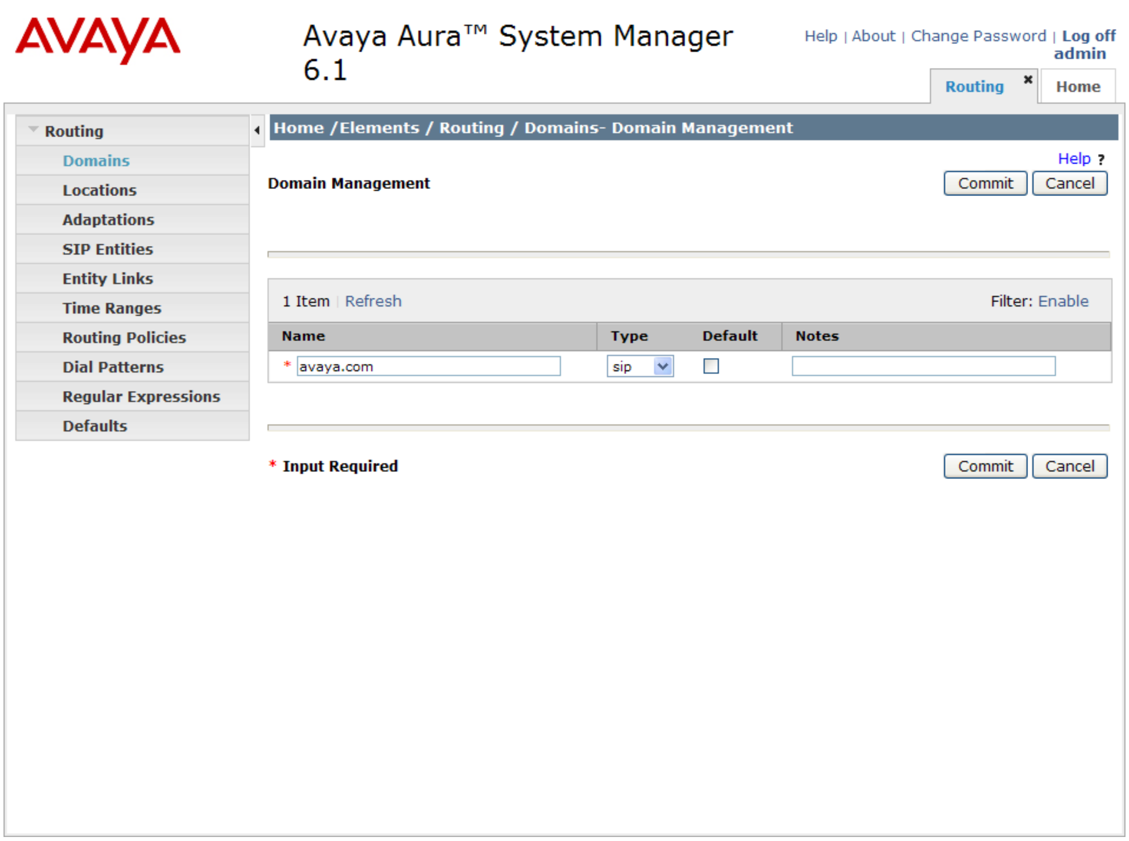
The **Routing** menu contains all the configuration tasks listed at the beginning of this section.

During compliance testing, one SIP Domain was configured on each Session Manager since all SIP entities were located within the same authoritative domain.

Navigate to **Routing→Domains**, and click the **New** button (not shown) to add the SIP domain with

- **Name**: ***avaya.com*** (as set in **Section 5, Step 2**)
- **Notes**: Optional descriptive text.

Click **Commit** to save the configuration.



3. **Add Location**

Locations identify logical and/or physical locations where SIP entities reside. Only one Location was configured at each site for compliance testing.

Navigate to **Routing→Locations** and click the **New** button (not shown) to add the Location.

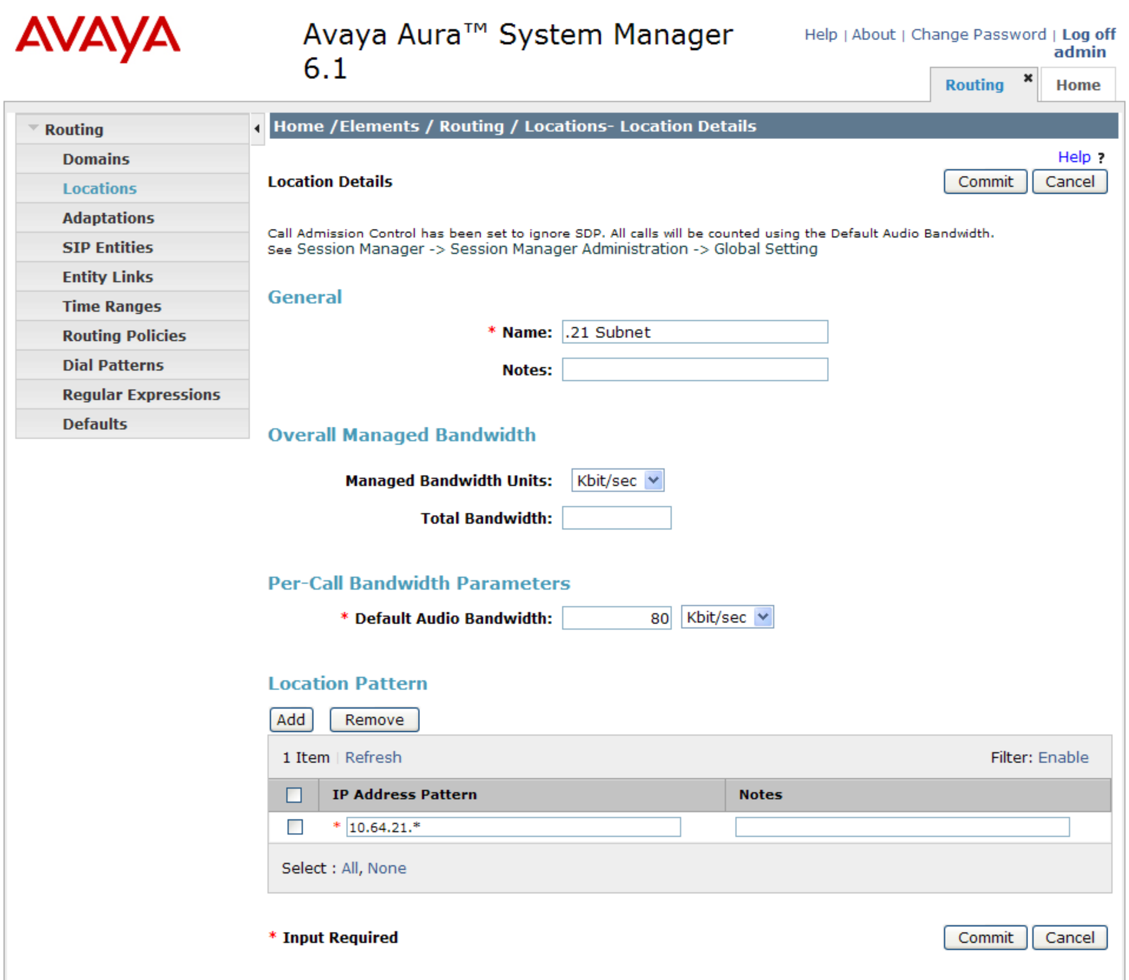
Under **General**:

- **Name**: A descriptive name.
- **Notes**: Optional descriptive text.

Under **Location Pattern**, click the **Add** button to add a new line:

- **IP Address Pattern**: ***10.64.21.****
- **Notes**: Optional descriptive text.

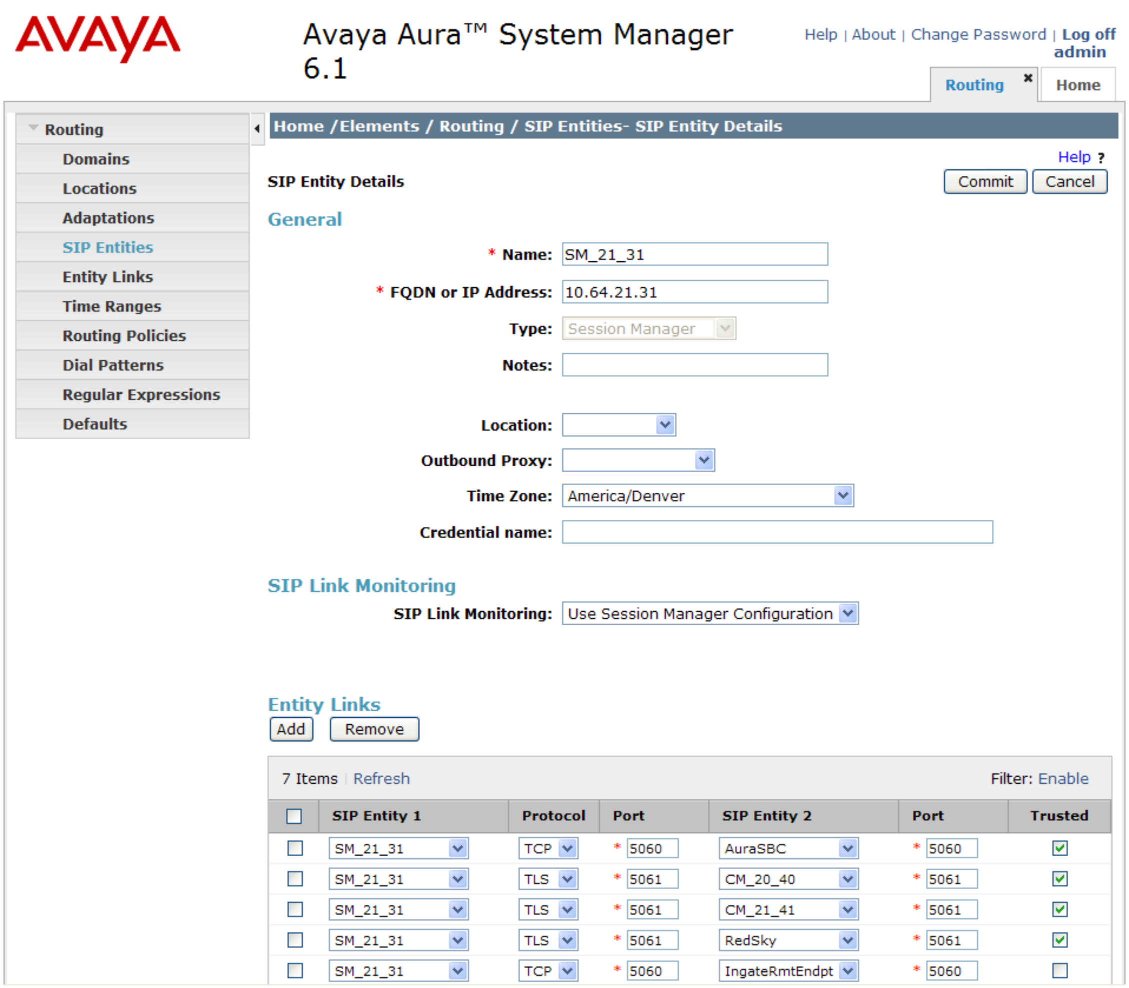
Click **Commit** to save the configuration.

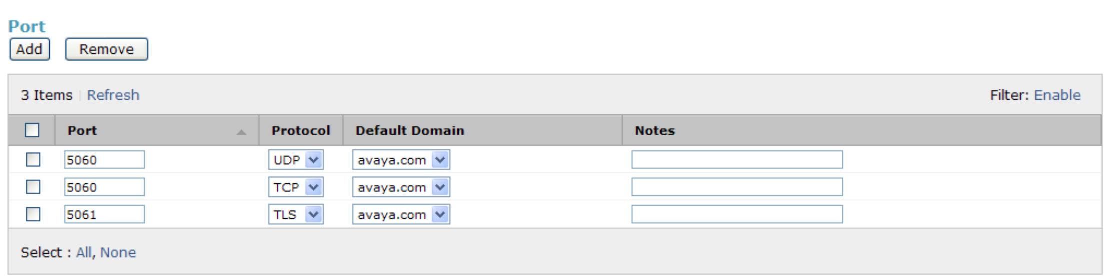


| | |
|---|---|
| 4. | **Add SIP Entities**<br>A SIP Entity must be added for Session Manager and for each SIP-based telephony system supported by it using SIP trunks. During compliance testing, a SIP Entity was added for the Session Manager, Communication Manager, and the AudioCodes Mediant 1000 MSBG e-SBC.<br><br>Navigate to **Routing→SIP Entities**, and click the **New** button (not shown) to add a SIP Entity. The configuration details for the SIP Entity defined for Session Manager are as follows:<br><br>Under **General**:<br>• **Name**: A descriptive name.<br>• **FQDN or IP Address**: ***10.64.21.31*** as specified in **Figure 1**. This is the IP address assigned to the SM-100 security module installed in the Session Manager.<br>• **Type**: select ***Session Manager.***<br><br>Under **Port**, click **Add**, then edit the fields in the resulting new row as shown below:<br>• **Port**: ***5060***. This is the port number on which the system listens for SIP requests.<br>• **Protocol**: ***UDP***. UDP was used between Session Manager and AudioCodes during compliance testing. These steps were repeated to add **Port *5061*** and **Protocol *TLS*** for communication between Session Manager and Communication Manager.<br>• **Default Domain**: Select the SIP Domain created in **Step 2**.<br><br>Default settings can be used for the remaining fields. Click **Commit** to save the SIP Entity definition. |

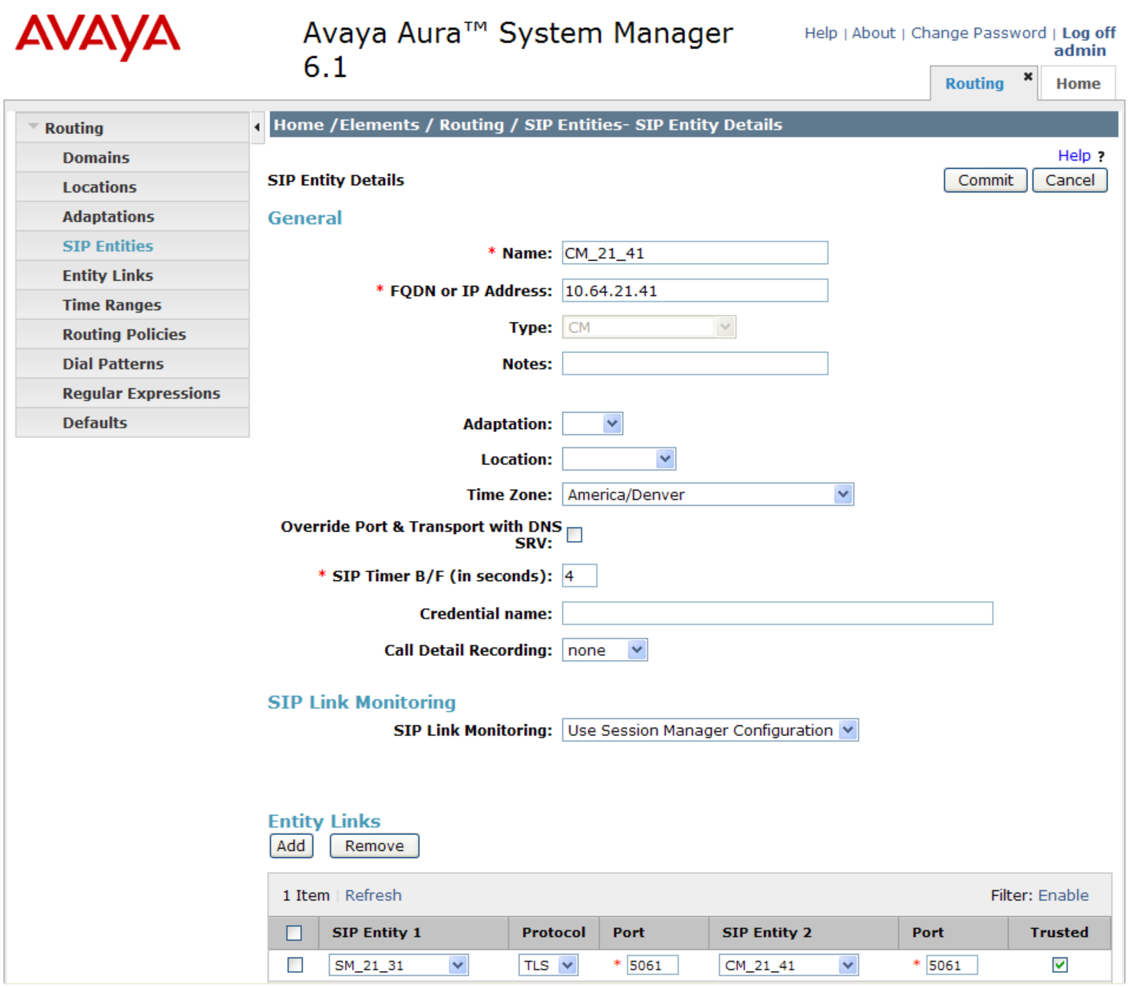
## Add SIP Entities (continued) – Session Manager

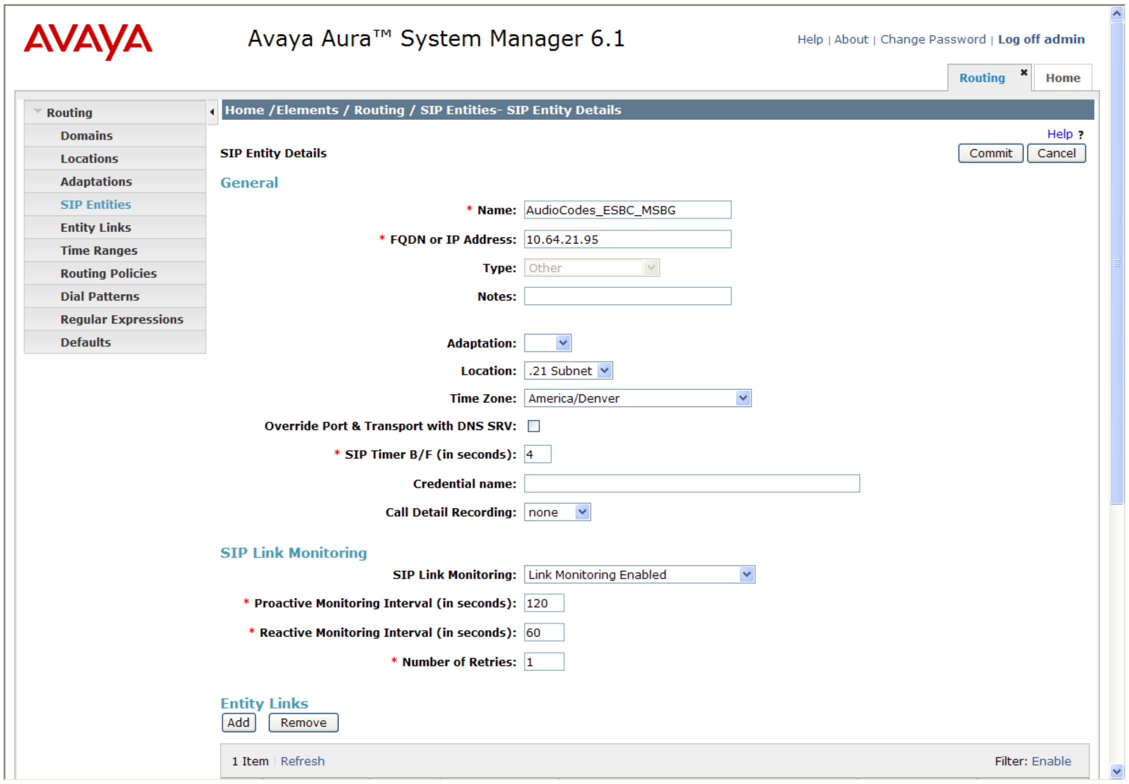
The screens below show the SIP Entity configuration details for the Session Manager.



**Add SIP Entities (continued) – Communication Manager**
The screen below shows the SIP Entity configuration details for the Communication Manager. Note the ***CM*** selection for **Type**.



| | |
|---|---|
| | **Add SIP Entities (continued) – AudioCodes Mediant 1000 MSBG e-SBC**<br>The screen below shows the SIP Entity configuration details for the AudioCodes Mediant 1000 MSBG e-SBC. Note the ***Other*** selection for **Type**.<br> |

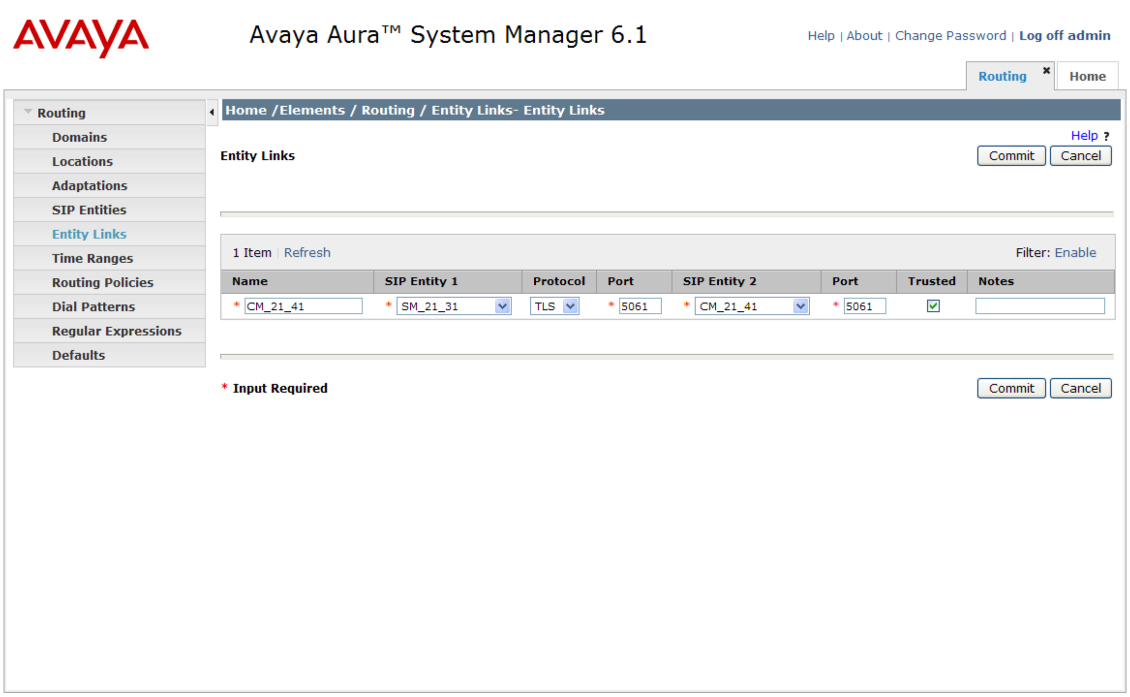
5. **Add Entity Links**

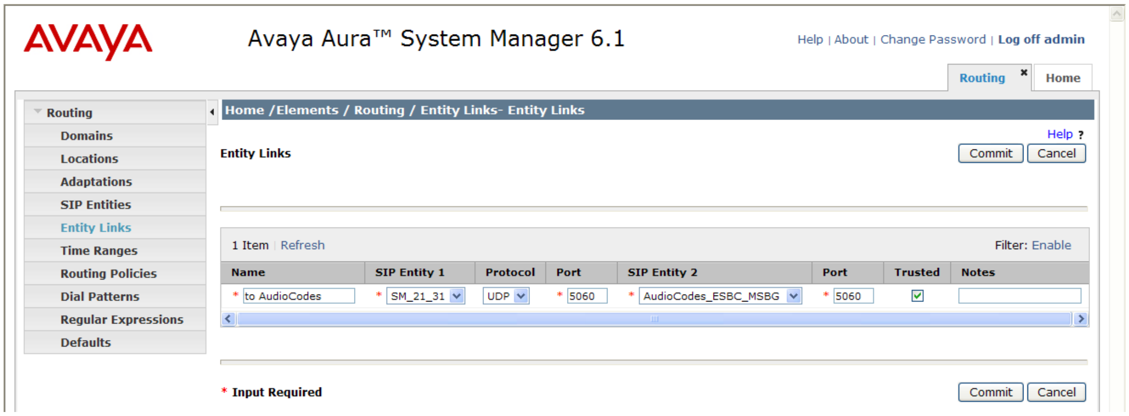
A SIP trunk between Session Manager and a telephony system is described by an Entity link. Two Entity Links were created: one between Session Manager and Communication Manger; the other between Session Manager and AudioCodes Mediant 1000 MSBG e-SBC.

Navigate to **Routing→Entity Links**, and click the **New** button (not shown) to add a new Entity Link. The screen below shows the configuration details for the Entity Link connecting Session Manager to Communication Manager.

- **Name**: A descriptive name.
- **SIP Entity 1**: Select the Session Manager SIP Entity.
- **Port**: ***5061***. This is the port number to which the other system sends SIP requests.
- **SIP Entity 2**: Select the Communication Manager SIP Entity.
- **Port**: ***5061***. This is the port number on which the other system receives SIP requests.
- **Trusted**: Check this box.
- **Protocol**: Select ***TLS*** as the transport protocol.
- **Notes**: Optional descriptive text.

Click **Commit** to save the configuration.



| | |
|---|---|
| | **Add Entity Links (continued)**<br>The Entity Link for connecting Session Manager to AudioCodes Mediant 1000 MSBG e-SBC was similarly defined as shown in the screen below, using the UDP protocol and port 5060.<br> |

| | |
|---|---|
| 6. | **Add Time Ranges** |

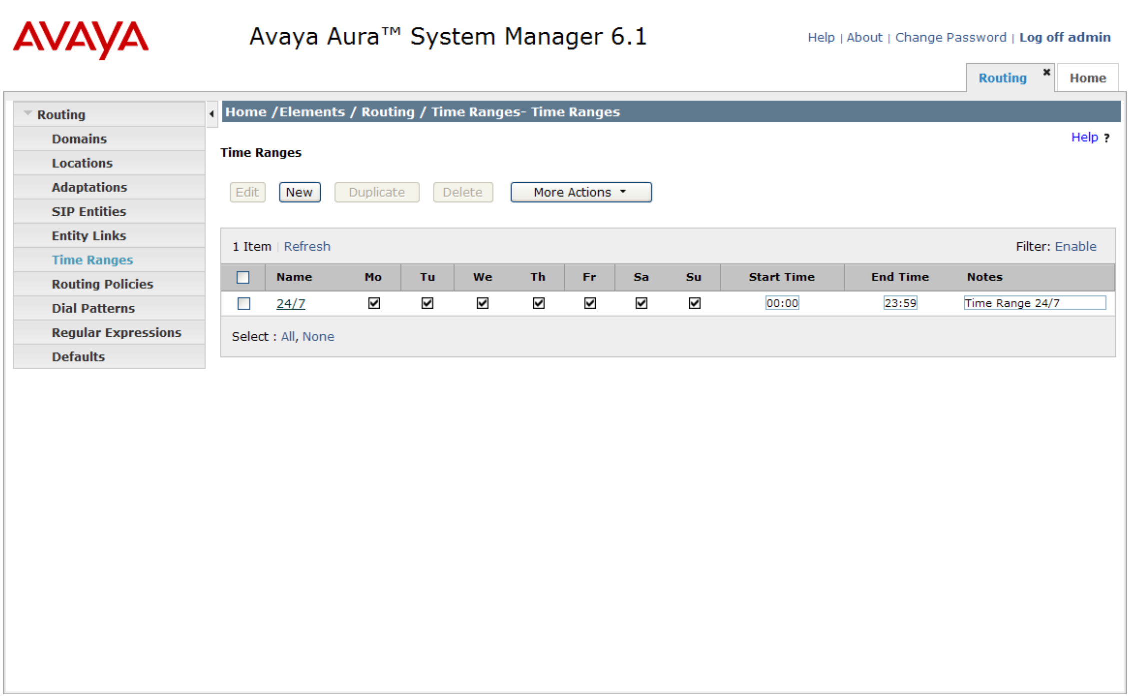
**Add Time Ranges**

Before adding routing policies (configured in next step), time ranges must be defined during which the policies will be active. One Time Range was defined that would allow routing to occur at anytime.

Navigate to **Routing→Time Ranges**, and click the **New** button to add a new Time Range:

- **Name**: A descriptive name.
- **Mo** through **Su**: Check the box under each of these headings.
- **Start Time**: Enter ***00:00***.
- **End Time**: Enter ***23:59***.

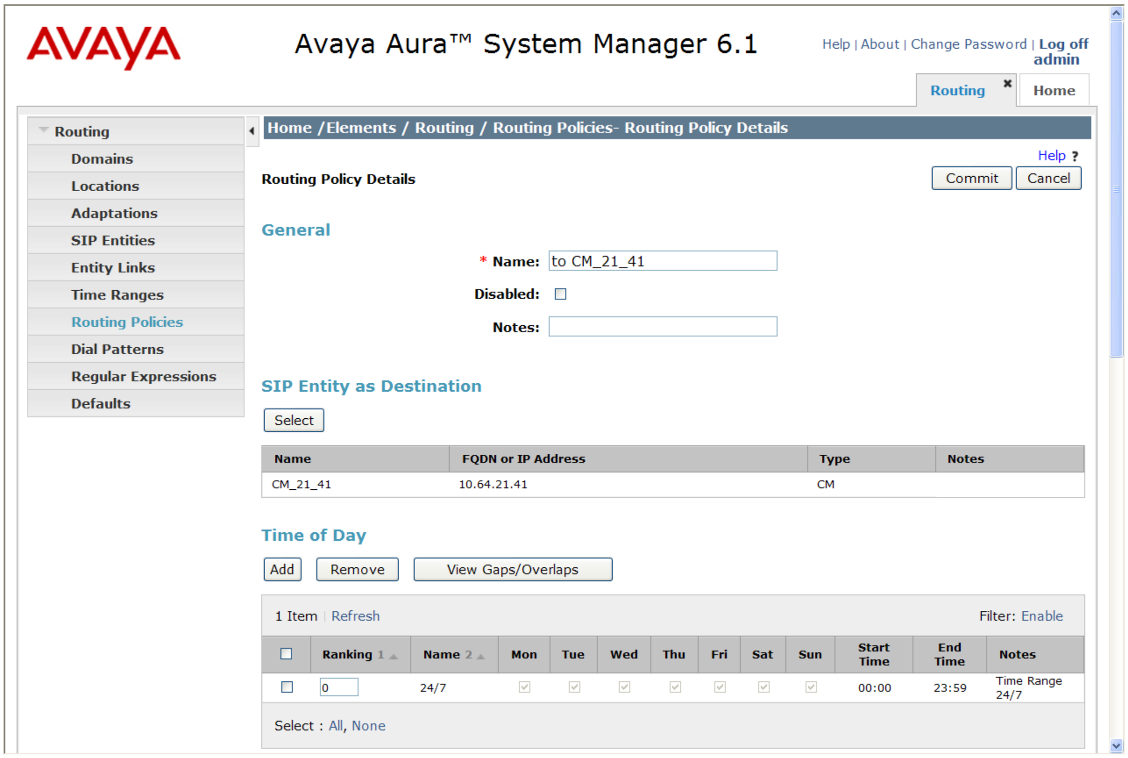
Click **Commit** to save this time range. The screen below shows the configured Time Range.

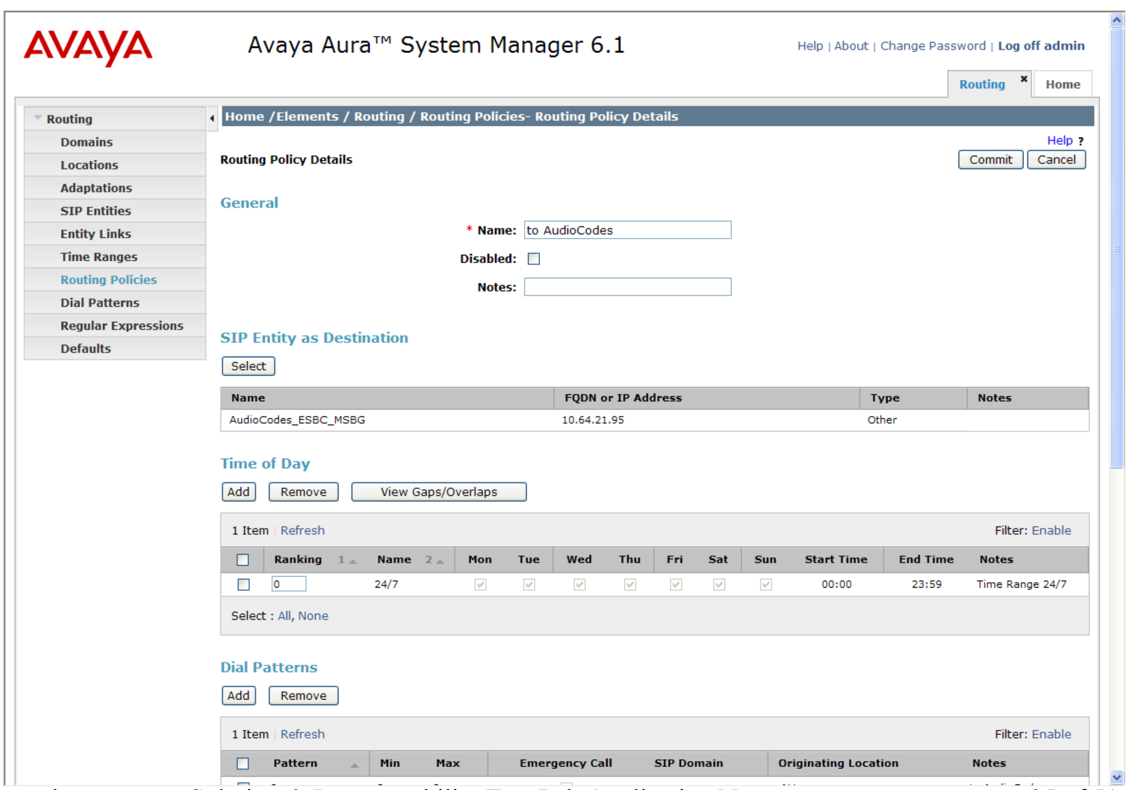


| | |
|---|---|
| 7. | **Add Routing Policies**<br>Routing policies describe the conditions under which calls will be routed to the SIP Entities connected to the Session Manager. Two routing policies were added – one for routing calls to Communication Manager, and the other for routing calls to AudioCodes Mediant 1000 MSBG e-SBC.<br><br>Navigate to **Routing→Routing Policies**, and click the **New** button (not shown) to add a new Routing Policy.<br>Under **General**:<br>• **Name**: A descriptive name.<br>• **Notes**: Optional descriptive text.<br><br>Under **SIP Entity as Destination**<br>Click **Select** to select the appropriate SIP Entity to which the routing policy applies (not shown).<br><br>Under **Time of Day**<br>Click **Add** to select the Time Range configured in the previous step (not shown).<br><br>Default settings can be used for the remaining fields. Click **Commit** to save the configuration. |

## Add Routing Policies (continued)

The screens below show the configuration details for the two Routing Policies used during compliance testing.

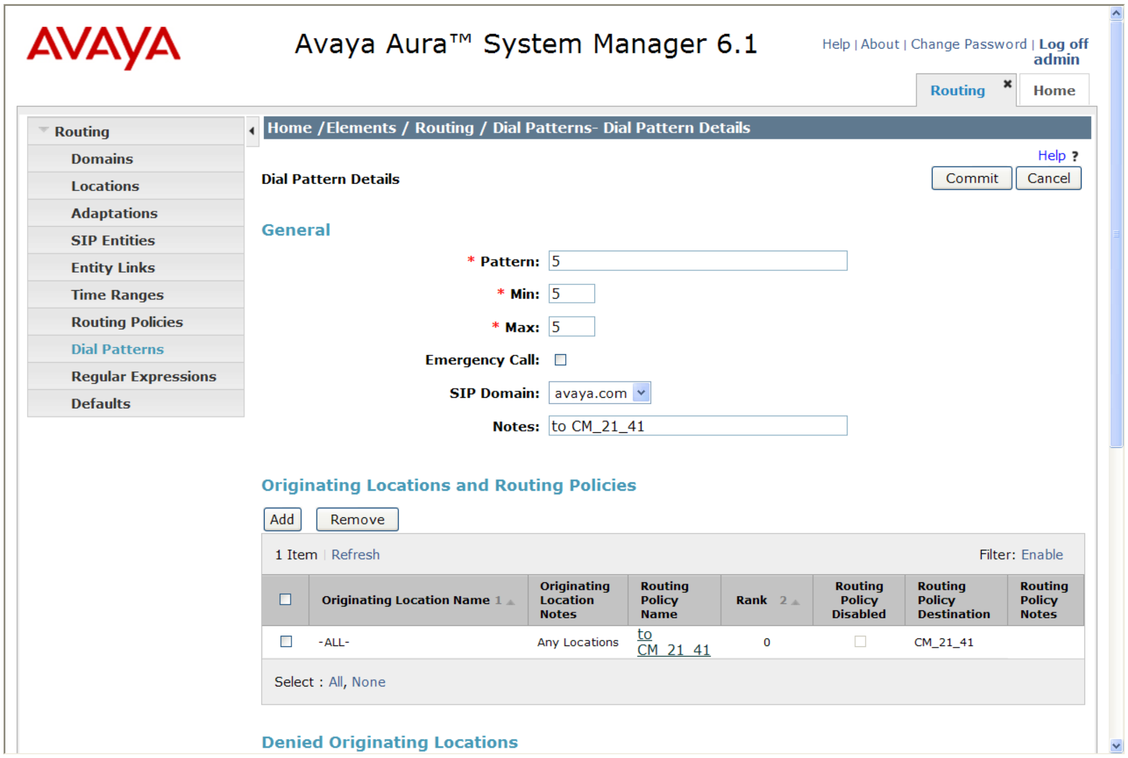


| | |
|---|---|
| 8. | **Add Dial Patterns**<br>Dial Patterns define digit strings to be matched against dialed numbers for directing calls to the appropriate SIP Entities. 5-digit extensions beginning with "5" resided on Communication Manager at the main enterprise site. 5-digit extensions beginning with "3" should were routed to AudioCodes Mediant 1000 MSBG e-SBC for onward routing to the 2nd site. Therefore two Dial Patterns were created accordingly.<br><br>Navigate to **Routing→Dial Patterns**, click the **New** button (not shown) to add a new Dial Pattern.<br><br>Under **General**:<br>• **Pattern**: Dialed number or prefix.<br>• **Min**: Minimum length of dialed number.<br>• **Max**: Maximum length of dialed number.<br>• **SIP Domain**: Select the SIP Domain created in **Step 2** (or select **–ALL–** to be less restrictive).<br>• **Notes**: Optional descriptive text.<br><br>Under **Originating Locations and Routing Policies**<br>Click **Add** to select the appropriate originating Location and Routing Policy from the list (not shown).<br><br>Under **Time of Day**<br>Click **Add** to select the time range configured in **Step 6**.<br><br>Default settings can be used for the remaining fields. Click **Commit** to save the configuration. |

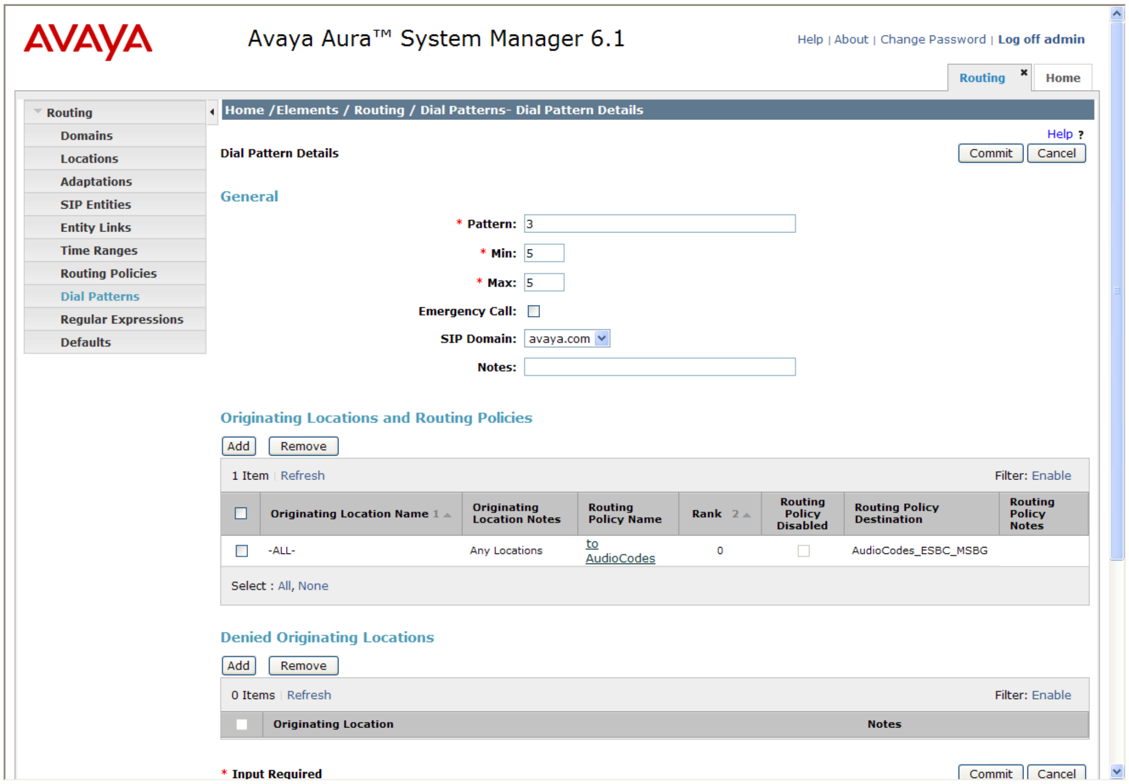
**Add Dial Patterns (continued)**

The screen below shows the configuration details for the Dialed Pattern defined for routing calls to Communication Manager at the main enterprise site.



**Add Dial Patterns (continued)**

The screen below shows the configuration details for the Dialed Pattern defined for routing calls to AudioCodes Mediant 1000 MSBG e-SBC (for onward routing to the 2nd site simulating a service provider service node).



# 7. Configure AudioCodes Mediant 1000 MSBG e-SBC

This section provides the procedures for configuring the AudioCodes Mediant 1000 MSBG e-SBC. It is assumed that proper knowledge of the AudioCodes MSBG e-SBC usage, configuration, support in general is understood, and the craft person has experience with the product platform. The following information is derived from the product manuals and is referenced only as a general guide. Configuration of the e-SBC will vary for each specific customer environment; however, AudioCodes has provided screenshots (and called-out specific fields on each screen with “arrows”), to show the configuration used during compliance testing.

All of the configuration shown in this section can be completed using the AudioCodes Mediant 1000 MSBG e-SBC web interface. From a browser, enter the IP address of the e-SBC and log in with the appropriate credentials.

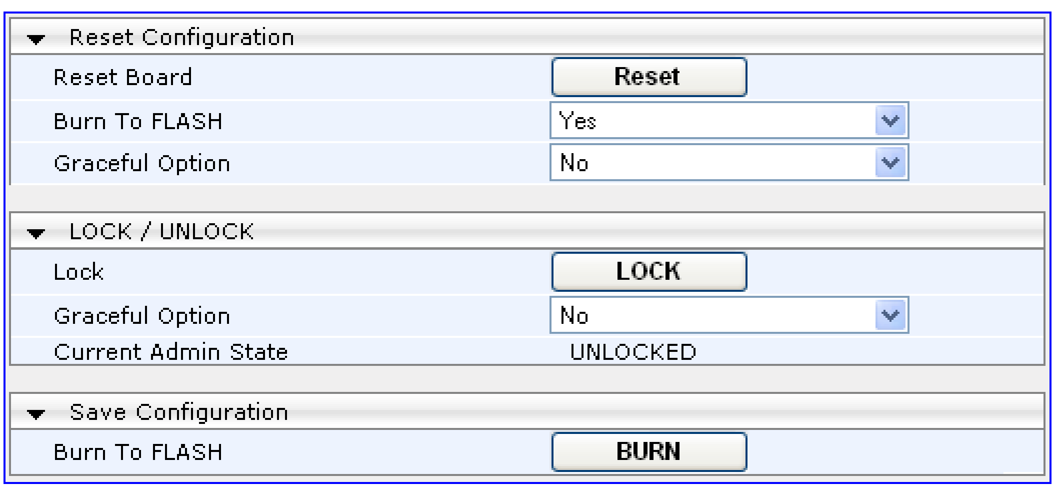
## 7.1. Configure Data and IP Routing Network Parameters

Ensure the IP Data Routing is set properly for support of routing for each network that is intended to interwork (these details are not shown, but they can be found in the installation manual with regards to the WAN interface setting and routing, as well as LAN side settings).

Once the administration is completed for the data segment, submit, Burn to Flash, and restart the device. Navigate 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 Burn to flash will save the configuration and will allow the unit to recover from future resets in the configuration saved.

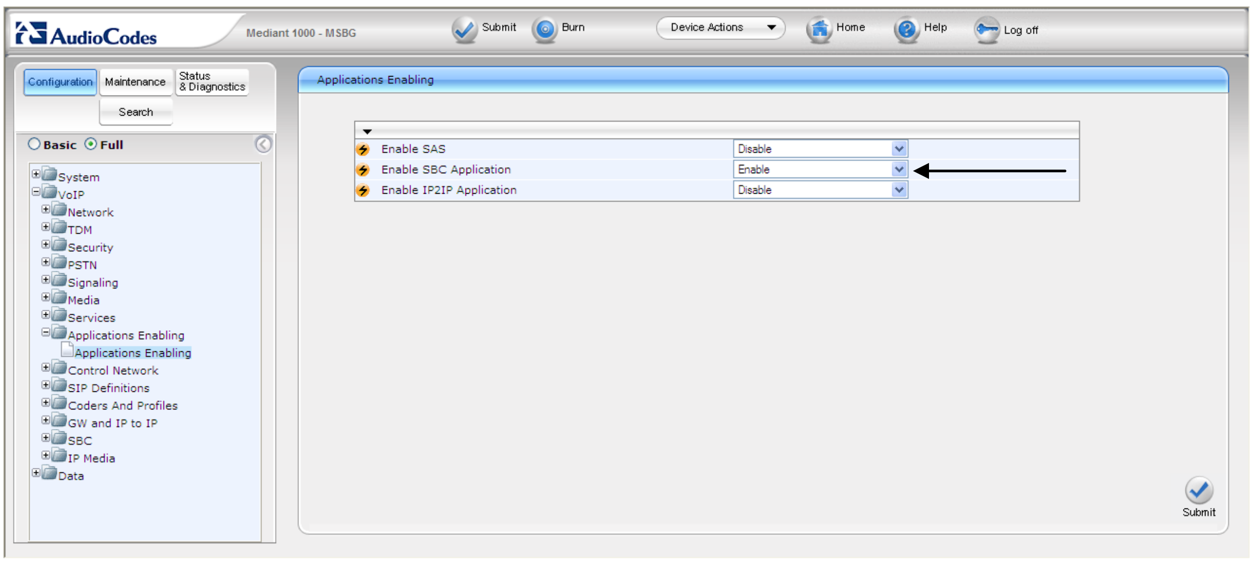
The device's new configuration (i.e., global IP address) is saved (burned) to the flash memory and the device performs a reset. The Web interface session terminates, as it's no longer accessible using the blade's private IP address.



## 7.2. Enable SBC functionality

Open the **Applications** page (**Configuration** tab > **VoIP** menu > **Applications Enabling**) to configure the SBC functionality.

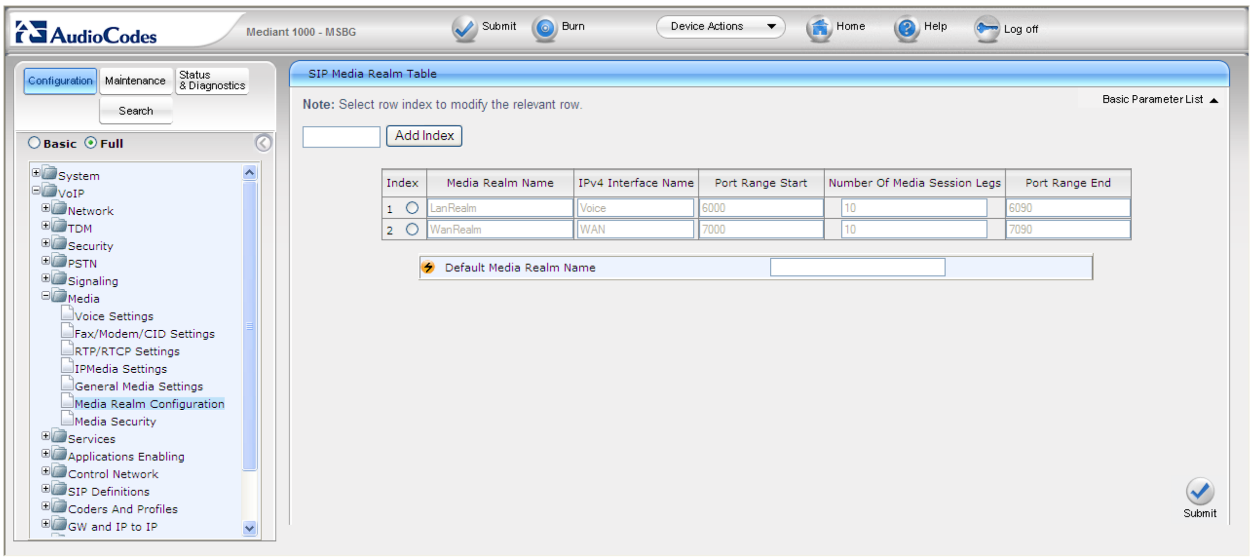
- Configure the parameter **Enable SBC Application** to **Enable**.
- Click the **Submit** button to save changes.
- Save the changes to flash memory. This is performed by selecting the **Burn** button at the top of the page. This is referred to as, "Saving Configuration", and will be referenced as such throughout this document.
- Notice the "Lightning Bolt" . All items marked with this symbol require a reset to take effect. Reset the device as noted previously in **Section 7.1**. Once the device is reset with the SBC application enabled, a submenu within VoIP menu will appear.



## 7.3. Configure Media Realm

Open the **Media Realm Configuration** page (**Configuration** tab > **VoIP** menu > **Media** submenu > **Media Realm Configuration** submenu) to configure the Media Realm settings.

- Configure the parameters as required. In the configuration used for compliance testing, only the **LanRealm** was used. The **Port Range Start** field indicates first RTP port of the range defined on the SBC. After the desired **Number of Media Session Legs** is entered, the SBC automatically populates the **Port Range End**.
- Click the **Submit** button to save changes.
- Save the changes to flash memory, refer to "Saving Configuration" as shown in **Section 7.2**.



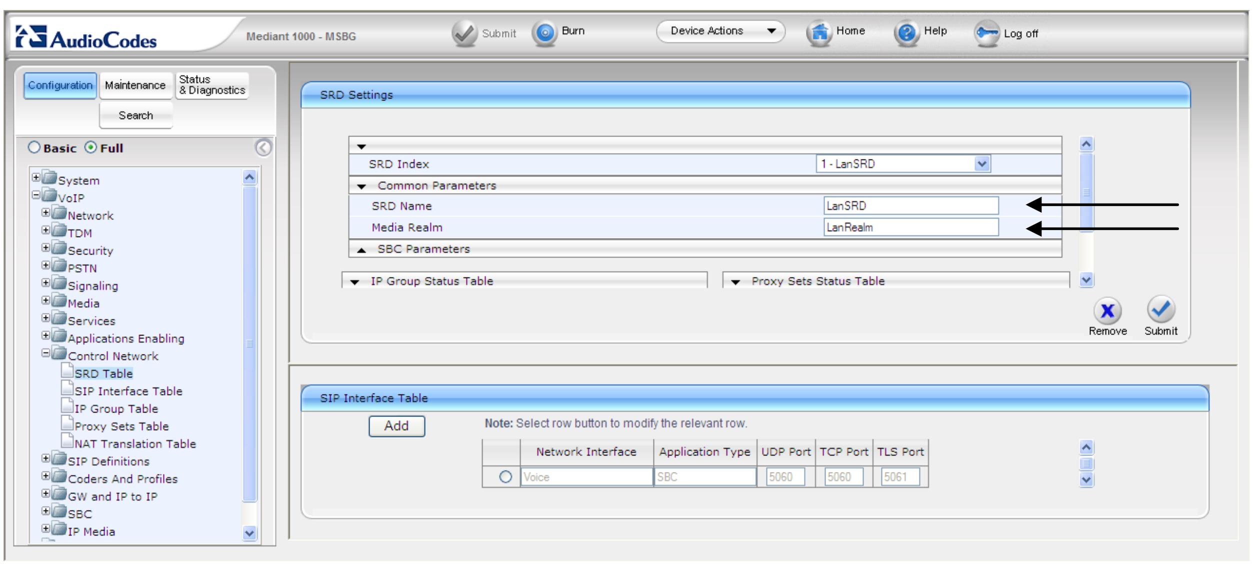
## 7.4. Configure SRD Table

Open the **SRD Table** page (**Configuration** tab > **VoIP** menu > **Control Network** submenu > **SRD Table** submenu) to configure the device's Signaling Routing Domain (SRD) table. An SRD is configured with a unique name and assigned a Media Realm (defined in **Section 7.3**). Once configured, SRDs can be used to do the following:

- Associate the SRD with a SIP Interface, IP Group, and Proxy Set
- Define the SRD as a destination IP-to-IP routing rule

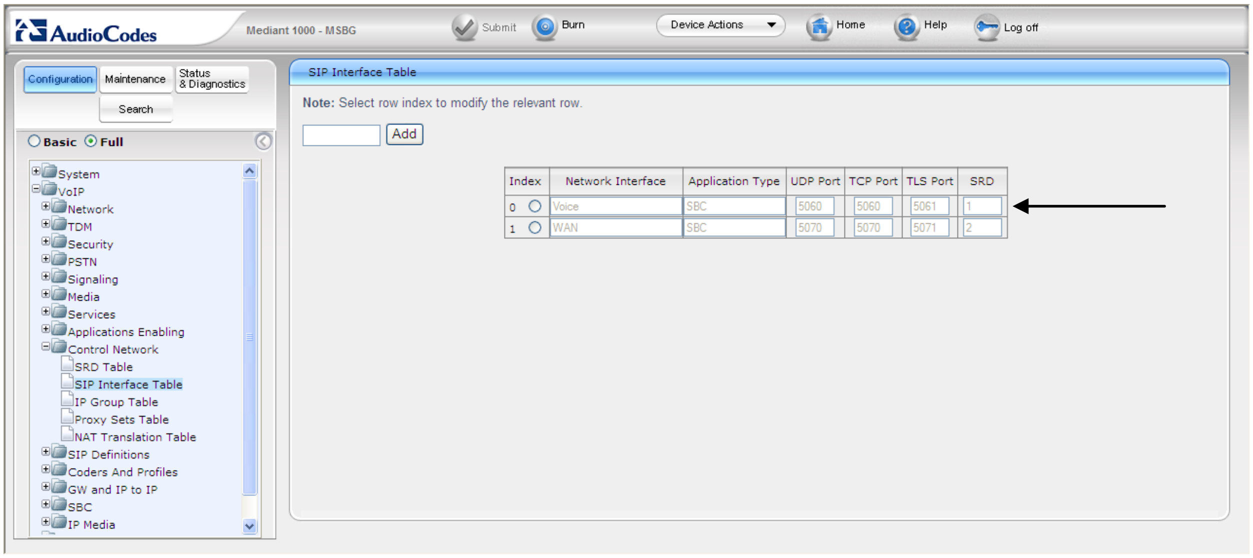
Therefore, an SRD is a set of definitions, together creating multiple, virtual multi-service IP gateways. Typically, one SRD is defined for each group of SIP User Agents (e.g. proxies, IP phones, application servers, gateways, soft switches, etc.) that communicate with each other. This provides these entities with VoIP services that reside on the same Layer-3 network (which must be able to communicate without traversing NAT devices and must not have overlapping IP addresses). Routing from one SRD to another is possible, whereby each routing destination (IP Group or destination address) indicates the SRD to which it belongs.

- Select an index that is unused.
- Configure the parameters as required. During compliance testing, **SRD Index** 1 (**LanSRD**) was mapped to **LanRealm**.
- Click the **Submit** button to save changes.
- Save the changes to flash memory, refer to "Saving Configuration" as shown in **Section 7.2**.
- Repeat the process for the required SRD(s).
- Ensure that there is a unique SRD name which is bound to a Media Realm created previously.



## 7.5. Configure SIP Interfaces

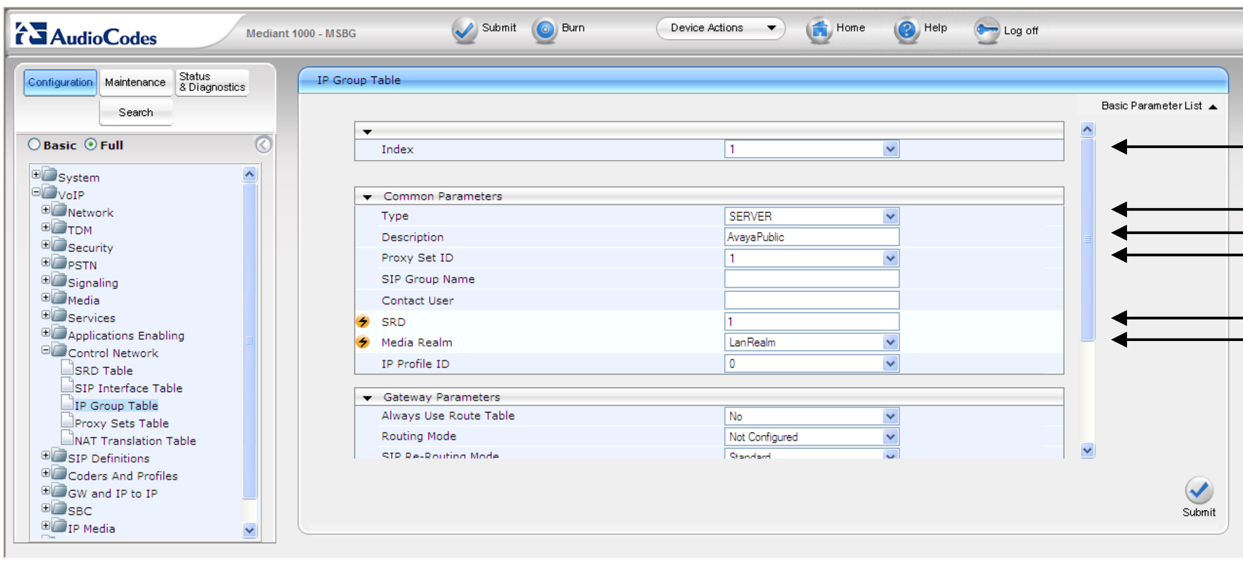
Create an interface in the **SIP Interface Table**. Ensure the **Network Interface** name used for the new index matches the name used in the initial settings for IP Settings, in this case **Voice**. This is the interface for the SBC Application. The SIP Interface table below states that the **Network Interface** known as **Voice** is being utilized by the SBC application. It also states that port **5060** should be used for both **UDP** and TCP, and port 5061 should be used for TLS. Note, port 5060 and UDP was utilized during compliance testing for communication between Session Manager and the SBC, as defined in the Entity Link configuration in **Section 6**, **Step 5**. Finally, the table below states the **Voice** network interface is bound to **SRD 1**.

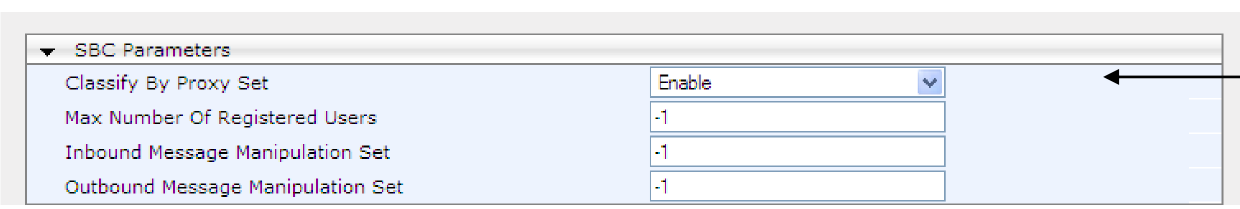


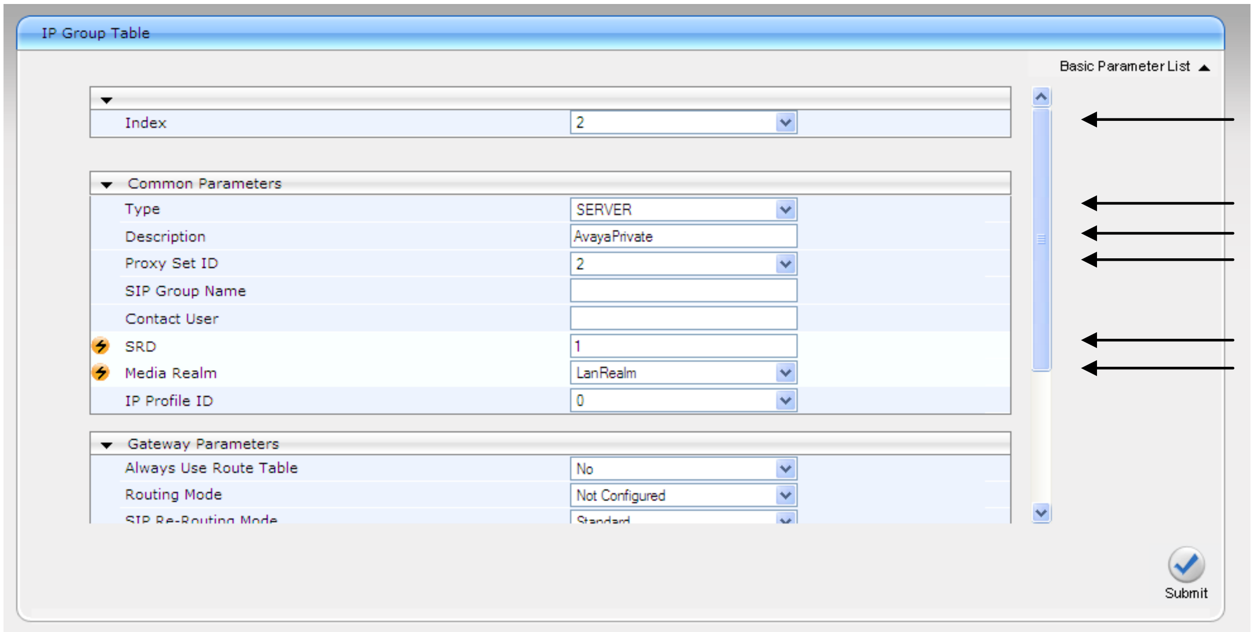
## 7.6. Configure the IP Group Table Settings

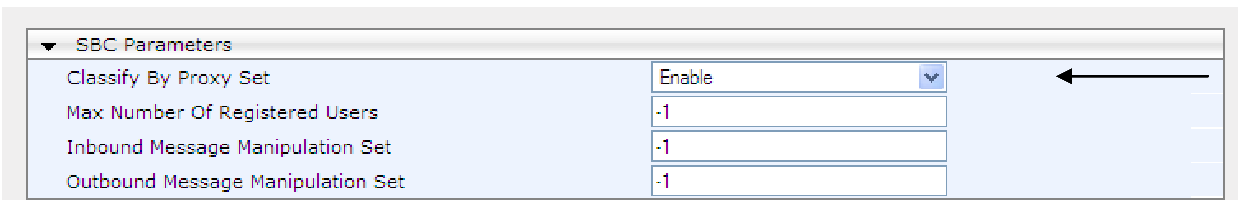
Open the **IP Group Table** page (**Configuration** tab > **VoIP** menu > **Control Network** submenu > **IP Group Table**) to configure the IP Group(s) and their respective parameters.

- Configure an unused IP Group index and assign appropriate parameters as required. During compliance testing, two indices were created representing the public interface (**AvayaPublic**) and the private interface (**AvayaPrivate**) on the SBC. Both indices used the **LanRealm** and **SRD 1** defined in **Sections 7.3** and **7.4**, respectively.
- Click the **Submit** button to save changes.
- Repeat previous two steps for the required amount of routes needed.
- To save the changes to flash memory, refer to "Saving Configuration" as shown in **Section 7.2**.




IP Group Table
Basic Parameter List
Index
2
Common Parameters
Type
SERVER
Description
AvayaPrivate
Proxy Set ID
2
SIP Group Name
Contact User
SRD
1
Media Realm
LanRealm
IP Profile ID
0
Gateway Parameters
Always Use Route Table
No
Routing Mode
Not Configured
Submit

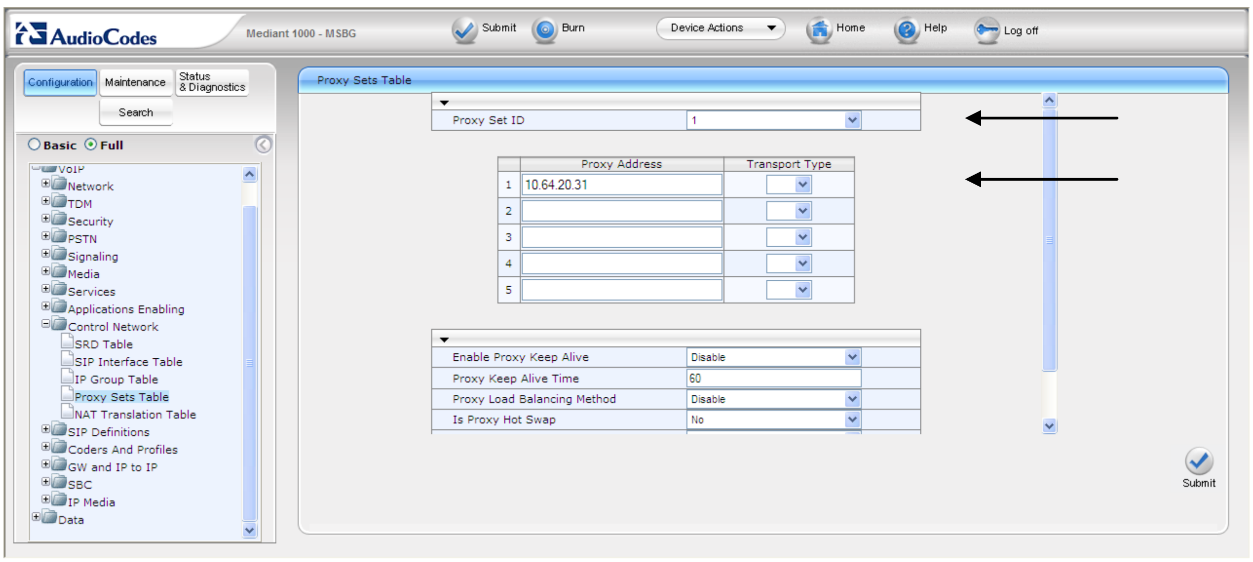

SBC Parameters
Classify By Proxy Set
Enable
Max Number Of Registered Users
-1
Inbound Message Manipulation Set
-1
Outbound Message Manipulation Set
-1

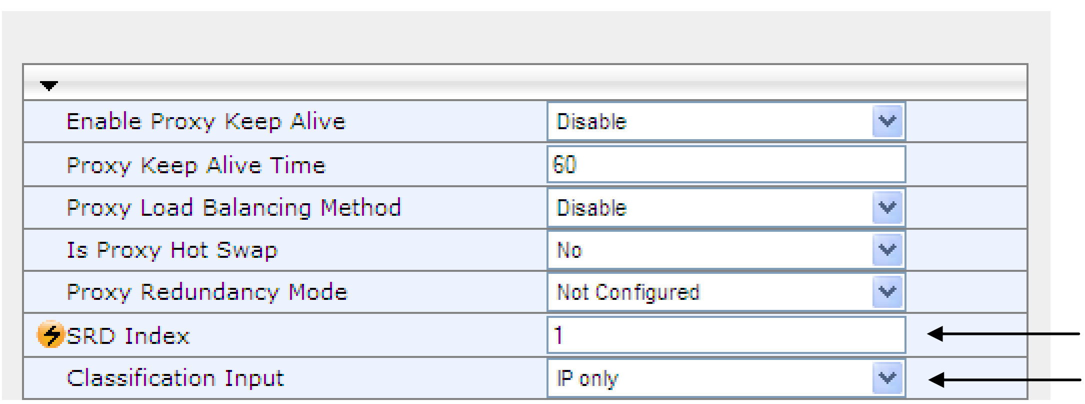
## 7.7. Configure Proxy Set Indices

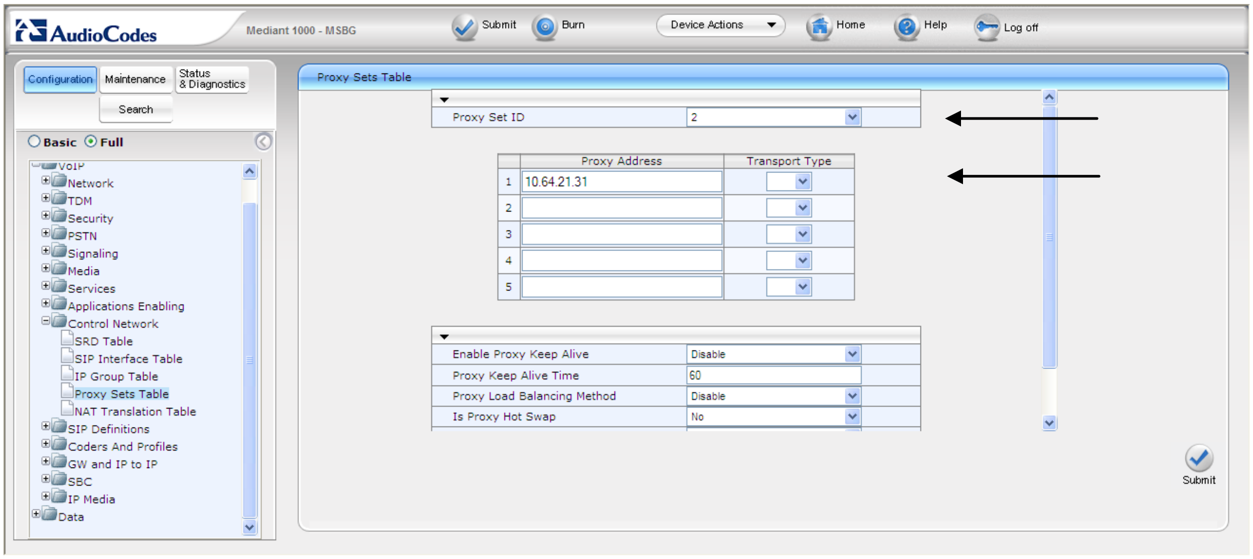
The use of Proxy Set index is utilized for identifying the specific Proxy (or set of proxy devices) for a respective IP Group Index (reference **Section 7.6** as an example: IP Group 1 is serviced by IP Proxy Set 1). Configure an unused Proxy Set Index and identify the IP address of the proxy for which calls will be routed. Do this for each unique IP group.

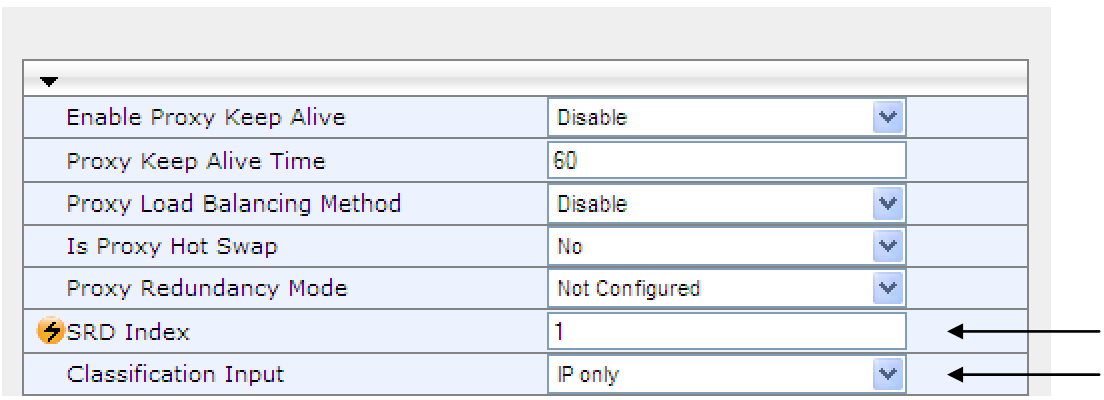
Open the **IP Group Table** page (**Configuration** tab > **VoIP** menu > **Control Network** submenu > **Proxy Sets Table**) to configure the Proxy Set(s) and their respective parameters:

- Configure an unused IP Group index and assign its appropriate parameters as required. (Note: 10.64.21.31 is the IP address of Session Manager and the Enterprise site. 10.64.20.31 is the IP address of Session Manager at the simulated 2nd site)
- Click the **Submit** button to save changes.
- Repeat previous two steps for the required amount of routes needed.
- To save the changes to flash memory, refer to "Saving Configuration" as shown in **Section 7.2**.



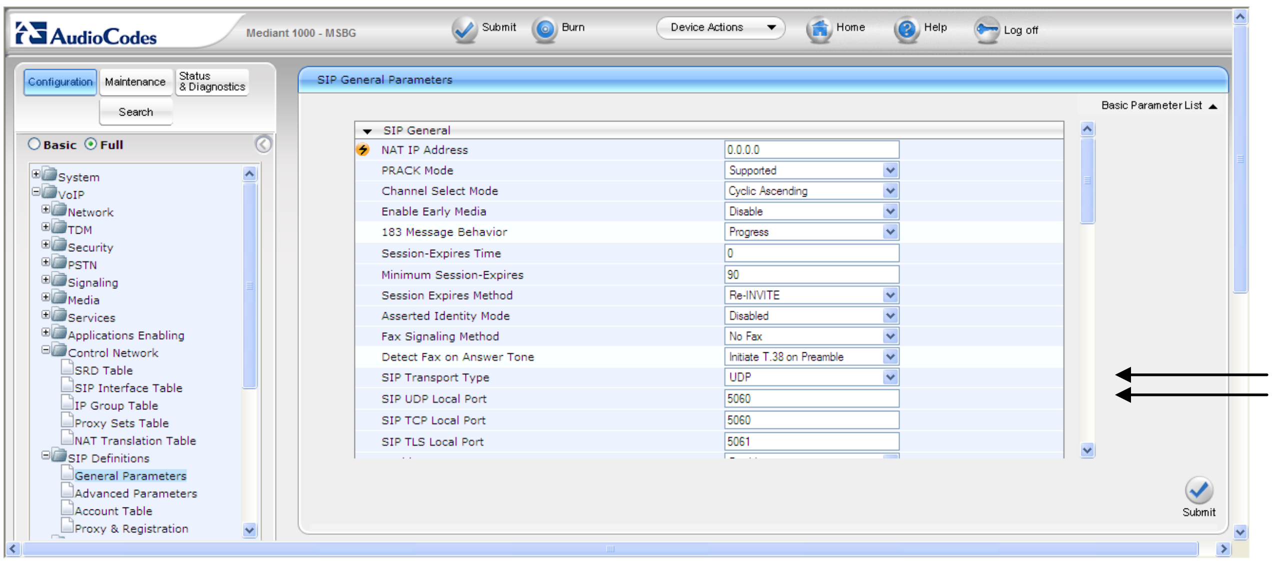

AudioCodes
Mediant 1000 - MSBG
Submit
Burn
Device Actions
Home
Help
Log off
Configuration
Maintenance
Status & Diagnostics
Search
Basic
Full
VoIP
Network
TDM
Security
PSTN
Signaling
Media
Services
Applications Enabling
Control Network
SRD Table
SIP Interface Table
IP Group Table
Proxy Sets Table
NAT Translation Table
SIP Definitions
Coders And Profiles
GW and IP to IP
SBC
IP Media
Data
Proxy Sets Table
Proxy Set ID
2
Proxy Address
Transport Type
10.64.21.31
Enable Proxy Keep Alive
Disable
Proxy Keep Alive Time
60
Proxy Load Balancing Method
Disable
Is Proxy Hot Swap
No
Submit

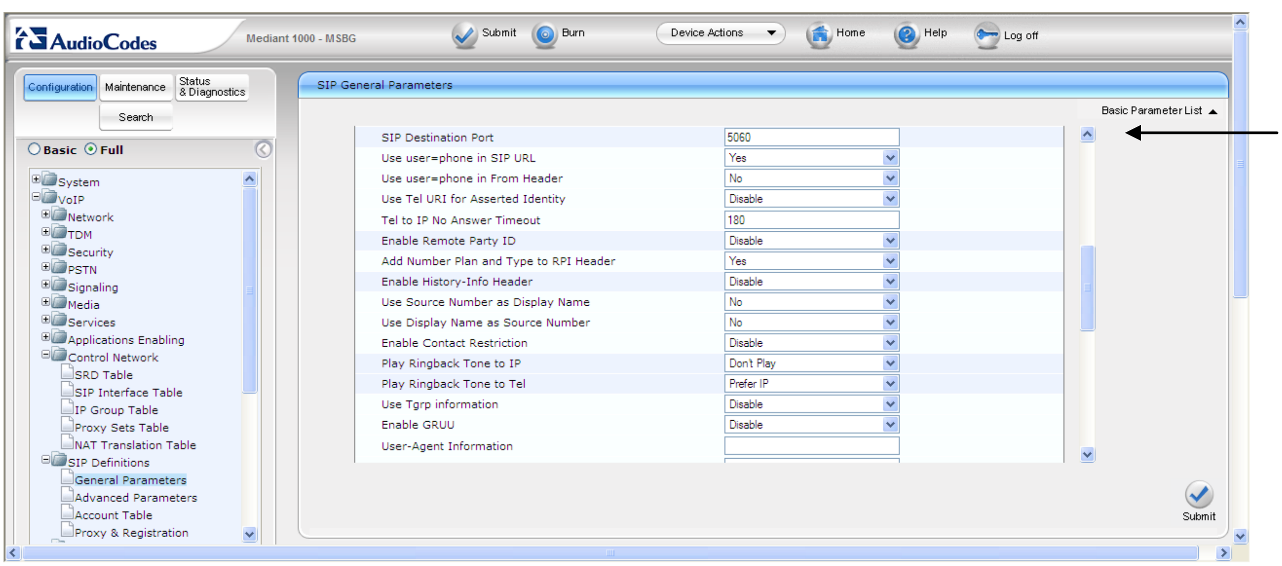

Enable Proxy Keep Alive
Disable
Proxy Keep Alive Time
60
Proxy Load Balancing Method
Disable
Is Proxy Hot Swap
No
Proxy Redundancy Mode
Not Configured
SRD Index
1
Classification Input
IP only

## 7.8. Configure SIP General Parameters

Open the **SIP General Parameters** page (**Configuration** tab > **VoIP** menu > **SIP Definitions** submenu > **General Parameters**) o configure the general SIP protocol parameters.

- Configure the parameters as required. (Note: Transport protocol UDP and Port 5060 were used for communication with Session Manager. See the Entity Link defined in **Section 6**, **Step 5**).
- Click the **Submit** button to save changes.
- To save the changes to flash memory, refer to “Saving Configuration” as shown in **Section 7.2**.

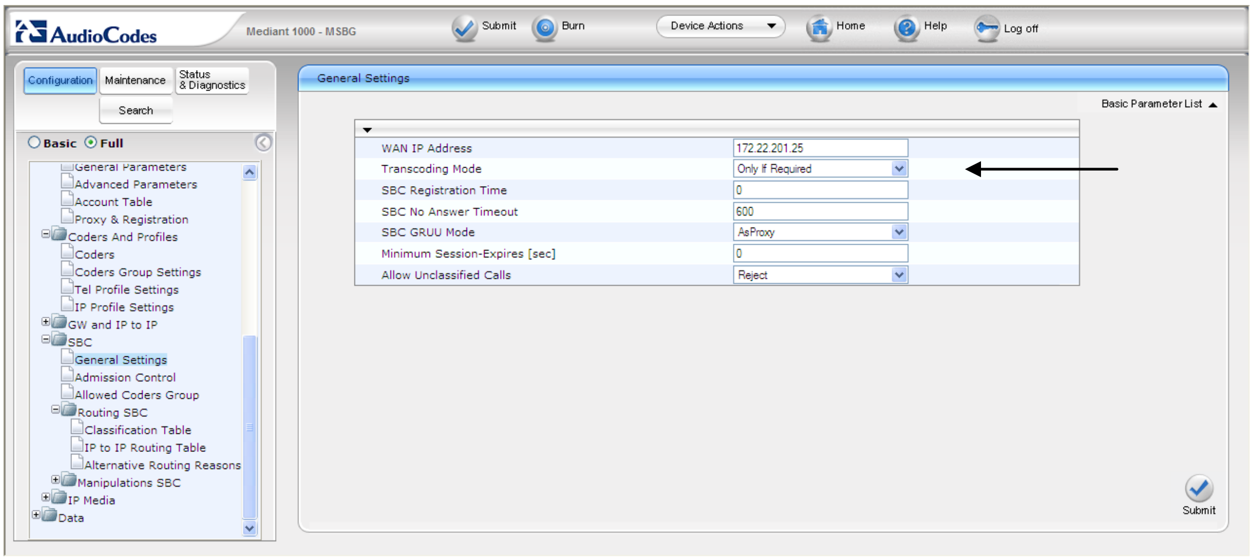


## 7.9. Configure General Settings

Open the **General Settings** page (**Configuration** tab > **VoIP** menu > **SBC** submenu > **General Settings**) to configure the general SBC parameters.

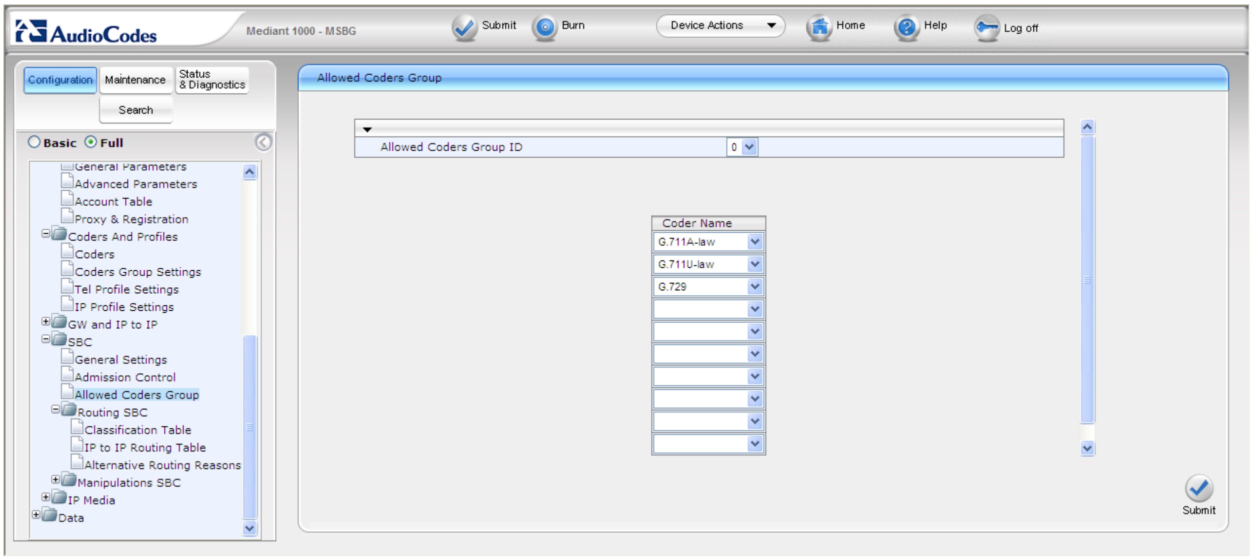
- Configure the parameters as required. Note, the **WAN IP Address** below was not used for the compliance tested configuration.
- Allowing of Unclassified calls is optional. All calls were classified by IP Group Index.
- Click the **Submit** button to save changes.
- To save the changes to flash memory, refer to "Saving Configuration" as shown in **Section 7.2**.



## 7.10. Configure Coders

Open the **Coders** page for the SBC application (**Configuration** tab > **VoIP** menu > **SBC** submenu > **Allowed Coders Group**) to configure the device's SBC Allowed coders.

- From the **Coder Name** drop-down list, select the required coder. (Note: G.711A-law, G.711U-law, and G.729 were compliance tested)
- Repeat steps for the next optional coders.
- Click the **Submit** button to save changes.
- To save the changes to flash memory, refer to "Saving Configuration" as shown in **Section 7.2**.



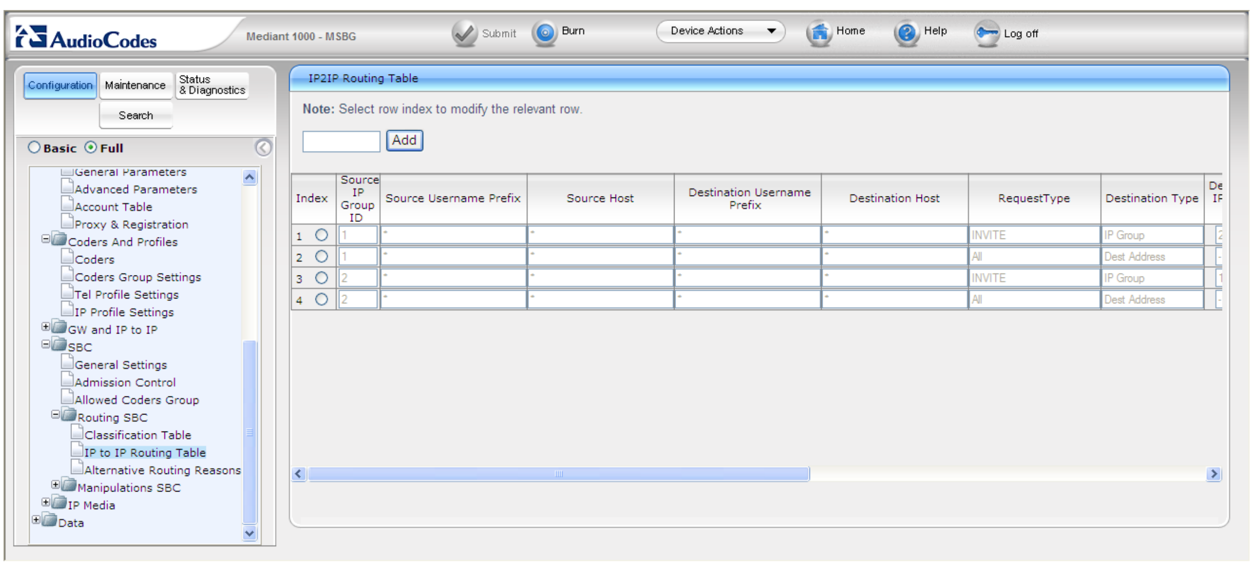
## 7.11. Configure IP to IP Routing Table

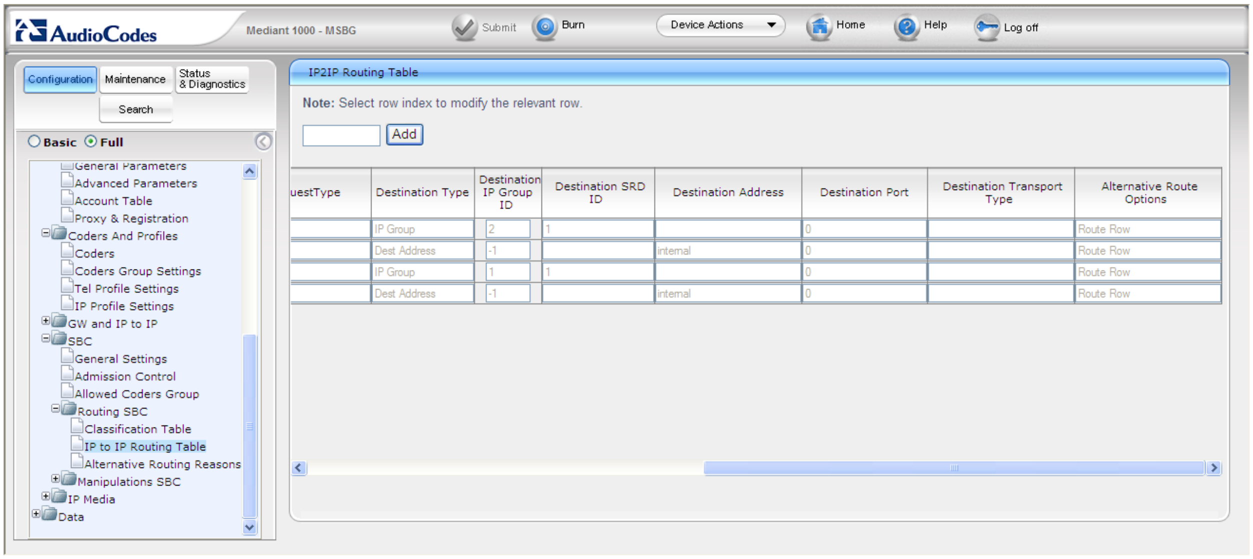
Open the **IP to IP Routing Table** page (**Configuration** tab > **VoIP** menu > **SBC** submenu > **Routing SBC** submenu > **IP to IP Routing Table**) to configure IP2IP routing rules.

The figures below shows the following configured outbound IP routing rules:

- **Rule 1:** If the incoming message originates from Source IP Group “1” and is associated with a call (Invite) then the call will be routed to a Destination IP Group of “2” and an SRD of “1”.
- **Rule 2:** If the incoming message is not associated with a call, and originates from Source IP Group “1”, then terminate the message to the internal device. This is set to enable the Avaya method of Heartbeat interworking for the product to return a 200 OK rather than send the received “Options” message to the terminating route.
- **Rule 3:** If the incoming message originates from Source IP Group “2” and is associated with a call (Invite) then the call will be routed to a Destination IP Group of “1” and an SRD of “1”.
- **Rule 4:** If the incoming message is not associated with a call, and originates from Source IP Group “2”, then terminate the message to the internal device. This is set to enable the Avaya method of Heartbeat interworking for the product to return a 200 OK rather than send the received “Options” message to the terminating route.

  - 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 changes.
  - To save the changes to flash memory, refer to “Saving Configuration” as shown in **Section 7.2**.

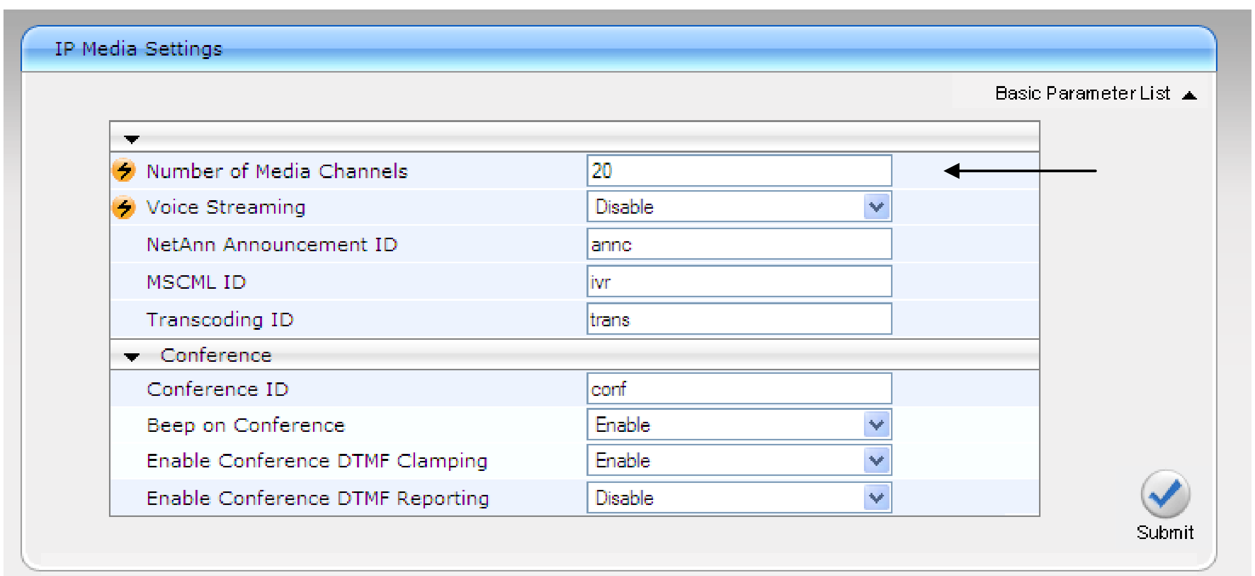





## 7.12. Configure IP Media Settings

Open the **IP Media Settings** page (**Configuration** tab > **VoIP** menu > **IP Media** submenu > **IP Media Settings**) to configure the IP Media Settings.

- Configure the IP Media Settings according to the required amount of supported sessions**.**
- Click the **Submit** button to save changes.
- To save the changes to the flash memory, refer to "Saving Configuration" as shown in **Section 7.2**.
- Reset the device to ensure the media resources are properly reserved.

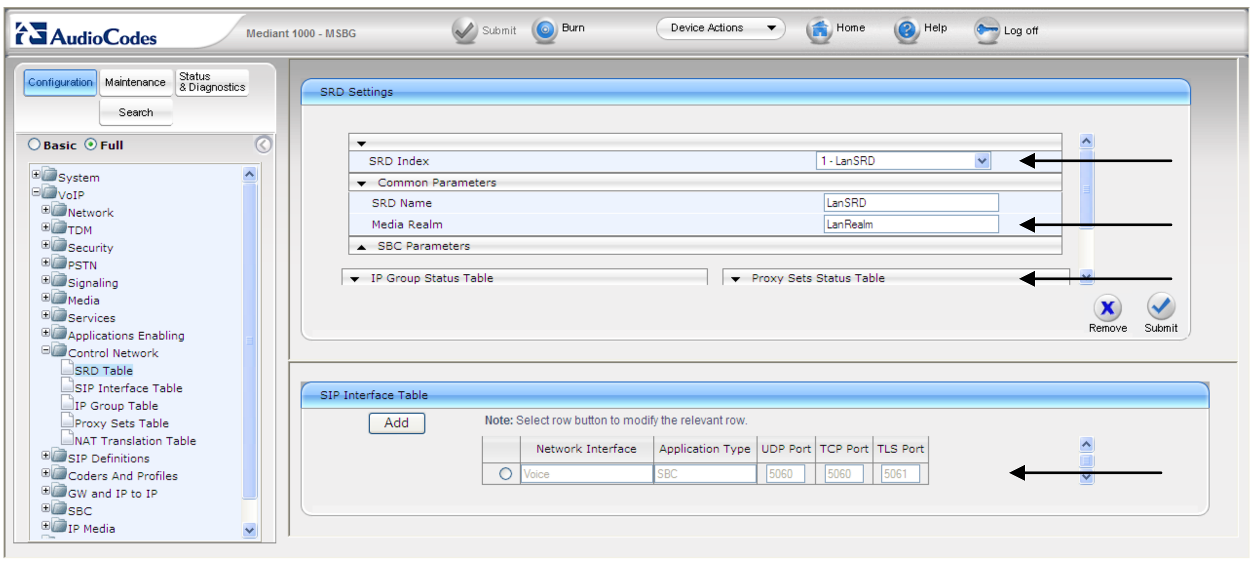


## 7.13. Configure SRD Table

Open the **SRD Table** page (**Configuration** tab > **VoIP** menu > **Control Network submenu > SRD Table** submenu) to view and confirm the device's intended SRD tables and respective routing interdependencies:

- Select the index that was created earlier.
- Insure the configured parameters are set as required.
- Click the IP Group Status and Proxy Sets Status sections to expand.
- Ensure the entries match that of the data previously entered.

Ensure the Network Interface name used for the new index matches the name used in the initial settings for IP Settings. This is the interface for the SBC Application.




**IP Group Status Table**

| Index | Type | Description | Proxy set ID | SIP group name | IP profile ID |
|---|---|---|---|---|---|
| 1 | SERVER | AvayaPublic | 1 | | 0 |
| 2 | SERVER | AvayaPrivate | 2 | | 0 |

- If Heart beating is required by the device, ensure that the value is set accordingly in the Proxy Set Indices.
- Ensure that there is a unique SRD name which is bound to a Media Realm created previously.

Proxy Sets Status Table

| Index | Enable Proxy Keep Alive |
|---|---|
| 1 | Disable |
| 2 | Disable |

## 8. Verification Steps

The following steps may be used to verify the configuration:

- Using System Manager, navigate to **Session Manager→System Status→SIP Entity Monitoring**, and click on the appropriate SIP Entities to verify that the Entity Links to the Mediant 1000 MSBG e-SBC and Communication Manager are up.
- From the Communication Manager SAT, use the **status signaling-group *x*** command to verify that the SIP signaling group is in-service (where ***x*** is the signaling group number associated with the trunk between Communication Manager and Session Manager).
- From the Communication Manager SAT, use the **status trunk-group *y*** command to verify that the SIP trunk group is in-service (where ***y*** is the trunk group number for the trunk between Communication Manager and Session Manager).
- Verify that calls can be placed from both SIP and non-SIP endpoints between sites.

## 9. Conclusion

The AudioCodes Mediant 1000 MSBG e-SBC passed compliance testing. These Application Notes describe the procedures required to configure the AudioCodes Mediant 1000 MSBG e-SBC to interoperate with Session Manager and Communication Manager to support the network shown in **Figure 1** where Session Manger connects the Mediant 1000 MSBG e-SBC to Communication Manager using SIP trunking interface.

## 10. Additional References

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

[1] *Avaya Aura™ Communication Manager Feature Description and Implementation*, Doc # 555-245-205, August 2010.
[2] *Administering Avaya Aura™ Communication Manager*, Doc # 03-300509, August 2010.
[3] *Administering Avaya Aura™ Session Manager*, Doc # 03-603324, December 2010.
[4] *Installing and Configuring Avaya Aura™ Session Manager*, Doc # 03-603472, January 2011.

Product documentation for the AudioCodes Mediant 1000 MSBG e-SBC can be found at http://www.audiocodes.com/support.

[5] *LTRT-26901_SIP_CPE_Release_Notes_Ver._6.2.pdf*
[6] *LTRT-52306_SIP_CPE_Product_Reference_Manual_Ver_6.2.pdf*
[7] *LTRT-83508_Mediant_1000_SIP_Installation_Manual_Ver._6.2.pdf*
[8] *LTRT-83307_Mediant_600_and_Mediant_1000_SIP_User's_Manual_v6.2.pdf*

# 11. Appendix – AudioCodes .ini file

For completeness, the AudioCodes Mediant 1000 MSBG e-SBC ini configuration file (with its appropriate parameters) that was used during compliance testing is shown below:

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

[SYSTEM Params]

SyslogServerIP = 10.64.21.100
EnableSyslog = 1
PM_VEDSPUtil = '1,43,48,15'

[BSP Params]

PCMLawSelect = 3
RoutingTableDestinationsColumn = 128.0.0.0
RoutingTableDestinationPrefixLensColumn = 1
RoutingTableGatewaysColumn = 10.33.0.1
WANIPAddress = 172.22.201.25
WanInterfaceName = 'GigabitEthernet 0/0'

[Analog Params]

FXSLoopCharacteristicsFilename = 'M1K13-1-fxs16khz.dat'

[ControlProtocols Params]

AdminStateLockControl = 0

[MGCP Params]


[MEGACO Params]

EP_Num_0 = 0
EP_Num_1 = 1
```

```
EP_Num_2 = 1
EP_Num_3 = 0
EP_Num_4 = 0

[PSTN Params]


[SS7 Params]


[Voice Engine Params]

CNGDetectorMode = 1
CallProgressTonesFilename = 'usa_tones_13.dat'

[WEB Params]

LogoWidth = '145'
HTTPSCipherString = 'RC4:EXP'
WanMgmtHttpPort = 80

[SIP Params]

GWDEBUGLEVEL = 5
FAXCNGMODE = 1
ENABLESBCAPPLICATION = 1
SBCMAXFORWARDSLIMIT = 70

[SCTP Params]


[VXML Params]


[IPsec Params]


[Audio Staging Params]


[SNMP Params]


;
;  *** TABLE InterfaceTable ***
;
;
```

```
[ InterfaceTable ]
FORMAT InterfaceTable_Index = InterfaceTable_ApplicationTypes, InterfaceTable_InterfaceMode,
InterfaceTable_IPAddress, InterfaceTable_PrefixLength, InterfaceTable_Gateway,
InterfaceTable_VlanID, InterfaceTable_InterfaceName;
InterfaceTable 0 = 6, 10, 10.64.21.95, 24, 10.64.21.1, 1, Voice;
InterfaceTable 15 = 11, 10, 10.64.2.60, 16, 10.64.1.1, 1, Data;

[ \InterfaceTable ]

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

;
;  *** TABLE CpMediaRealm ***
;
;

[ CpMediaRealm ]
FORMAT CpMediaRealm_Index = CpMediaRealm_MediaRealmName, CpMediaRealm_IPv4IF,
CpMediaRealm_IPv6IF, CpMediaRealm_PortRangeStart, CpMediaRealm_MediaSessionLeg,
CpMediaRealm_PortRangeEnd;
CpMediaRealm 1 = LanRealm, Voice, , 6000, 10, 6090;
CpMediaRealm 2 = WanRealm, WAN, , 7000, 10, 7090;

[ \CpMediaRealm ]

;
;  *** TABLE ProxyIp ***
;
;

[ ProxyIp ]
FORMAT ProxyIp_Index = ProxyIp_IpAddress, ProxyIp_TransportType, ProxyIp_ProxySetId;
ProxyIp 0 = 10.64.20.31, -1, 1;
ProxyIp 1 = 10.64.21.31, -1, 2;
ProxyIp 2 = 172.22.201.21, -1, 3;

[ \ProxyIp ]

;
;  *** TABLE IpProfile ***
;
;

[ IpProfile ]
```

FORMAT IpProfile_Index = IpProfile_ProfileName, IpProfile_IpPreference, IpProfile_CodersGroupID, IpProfile_IsFaxUsed, IpProfile_JitterBufMinDelay, IpProfile_JitterBufOptFactor, IpProfile_IPDiffServ, IpProfile_SigIPDiffServ, IpProfile_SCE, IpProfile_RTPRedundancyDepth, IpProfile_RemoteBaseUDPPort, IpProfile_CNGmode, IpProfile_VxxTransportType, IpProfile_NSEMode, IpProfile_IsDTMFUsed, IpProfile_PlayRBTone2IP, IpProfile_EnableEarlyMedia, IpProfile_ProgressIndicator2IP, IpProfile_EnableEchoCanceller, IpProfile_CopyDest2RedirectNumber, IpProfile_MediaSecurityBehaviour, IpProfile_CallLimit, IpProfile_DisconnectOnBrokenConnection, IpProfile_FirstTxDtmfOption, IpProfile_SecondTxDtmfOption, IpProfile_RxDTMFOption, IpProfile_EnableHold, IpProfile_InputGain, IpProfile_VoiceVolume, IpProfile_AddIEInSetup, IpProfile_SBCExtensionCodersGroupID, IpProfile_MediaIPVersionPreference, IpProfile_TranscodingMode, IpProfile_SBCAllowedCodersGroupID, IpProfile_SBCAllowedCodersMode, IpProfile_SBCMediaSecurityBehaviour, IpProfile_SBCRFC2833Behavior, IpProfile_SBCAlternativeDTMFMethod, IpProfile_SBCAssertIdentity, IpProfile_AMDSensitivityParameterSuit, IpProfile_AMDSensitivityLevel, IpProfile_AMDMaxGreetingTime, IpProfile_AMDMaxPostSilenceGreetingTime, IpProfile_SBCDiversionMode, IpProfile_SBCHistoryInfoMode;
IpProfile 1 = , 1, 0, 1, 10, 10, 46, 40, 0, 0, 0, 0, 2, 0, 0, 0, 0, -1, 1, 0, 0, -1, 1, 4, -1, 1, 1, 0, 0, , -1, 0, 0, -1, 0, 0, 0, 0, -1, 0, 8, 300, 400, -1, -1;

[ \IpProfile ]

;
;  *** TABLE ProxySet ***
;
;

[ ProxySet ]
FORMAT ProxySet_Index = ProxySet_EnableProxyKeepAlive, ProxySet_ProxyKeepAliveTime, ProxySet_ProxyLoadBalancingMethod, ProxySet_IsProxyHotSwap, ProxySet_SRD, ProxySet_ClassificationInput, ProxySet_ProxyRedundancyMode;
ProxySet 0 = 0, 60, 0, 0, 0, 0, -1;
ProxySet 1 = 0, 60, 0, 0, 1, 0, -1;
ProxySet 2 = 0, 60, 0, 0, 1, 0, -1;
ProxySet 3 = 0, 60, 0, 0, 2, 0, -1;

[ \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_ContactName;
IPGroup 1 = 0, AvayaPublic, 1, , , 0, -1, 0, 0, -1, 1, LanRealm, 1, 0, -1, -1, -1, ;
IPGroup 2 = 0, AvayaPrivate, 2, , , 0, -1, 0, 0, -1, 1, LanRealm, 1, 0, -1, -1, -1, ;
IPGroup 3 = 0, WANMP21, 3, , , 0, -1, 0, 0, -1, 2, WanRealm, 1, 0, -1, -1, -1, ;

[ \IPGroup ]

;
;  *** TABLE IP2IPRouting ***
;
;

[ IP2IPRouting ]
FORMAT IP2IPRouting_Index = IP2IPRouting_SrcIPGroupID, IP2IPRouting_SrcUsernamePrefix,
IP2IPRouting_SrcHost, IP2IPRouting_DestUsernamePrefix, IP2IPRouting_DestHost,
IP2IPRouting_RequestType, IP2IPRouting_DestType, IP2IPRouting_DestIPGroupID,
IP2IPRouting_DestSRDID, IP2IPRouting_DestAddress, IP2IPRouting_DestPort,
IP2IPRouting_DestTransportType, IP2IPRouting_AltRouteOptions;
IP2IPRouting 1 = 1, *, *, *, *, 1, 0, 2, 1, , 0, -1, 0;
IP2IPRouting 2 = 1, *, *, *, *, 0, 1, -1, -1, internal, 0, -1, 0;
IP2IPRouting 3 = 2, *, *, *, *, 1, 0, 1, 1, , 0, -1, 0;
IP2IPRouting 4 = 2, *, *, *, *, 0, 1, -1, -1, internal, 0, -1, 0;
IP2IPRouting 5 = 3, *, *, 30x, *, 0, 0, 1, 1, , 0, -1, 0;
IP2IPRouting 6 = 3, *, *, 50x, *, 0, 0, 2, 1, , 0, -1, 0;

[ \IP2IPRouting ]

;
;  *** TABLE SIPInterface ***
;
;

[ SIPInterface ]
FORMAT SIPInterface_Index = SIPInterface_NetworkInterface, SIPInterface_ApplicationType,
SIPInterface_UDPPort, SIPInterface_TCPPort, SIPInterface_TLSPort, SIPInterface_SRD;
SIPInterface 0 = Voice, 2, 5060, 5060, 5061, 1;
SIPInterface 1 = WAN, 2, 5070, 5070, 5071, 2;

[ \SIPInterface ]

;
;  *** TABLE SRD ***
;
;

[ SRD ]
```

FORMAT SRD_Index = SRD_Name, SRD_MediaRealm, SRD_IntraSRDMediaAnchoring, SRD_BlockUnRegUsers, SRD_MaxNumOfRegUsers, SRD_EnableUnAuthenticatedRegistrations;
SRD 1 = LanSRD, LanRealm, 0, 0, -1, 1;
SRD 2 = WanSRD, WanRealm, 0, 0, -1, 1;

[ \SRD ]

;
; *** 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 ]

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.