Avaya Solution & Interoperability Test Lab

Application Notes for Integrating the Research In Motion BlackBerry® Mobile Voice System with Avaya Aura® Communication Manager and Avaya Aura® Session Manager using a SIP Trunk – Issue 1.0

Abstract

These Application Notes describe configuring the Research In Motion BlackBerry® Mobile Voice System solution using Avaya Aura® Communication Manager, Avaya Aura® Session Manager, Avaya H.323 IP Telephones and AudioCodes Mediant 1000 with SIP trunking in a Fixed Mobile Convergence (FMC) VoIP solution.

The Research In Motion BlackBerry® Mobile Voice System solution extends the enterprise PBX functionality to mobile devices. This allows end users to be accessible when out of the office as well as to leverage wireless LAN networks to improve wireless coverage, reduce costs and provide the ability to seamlessly move calls from the Wi-Fi network to the mobile network and vice-versa.

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

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1 Introduction

These Application Notes describe configuring the Research in Motion (RIM) BlackBerry® Mobile Voice System (MVS) solution using Avaya Aura® Communication Manager, Avaya Aura® Session Manager, Avaya H.323 IP Telephones and AudioCodes Mediant 1000 with SIP trunking in a Fixed Mobile Convergence (FMC) VoIP solution.

The RIM MVS solution extends the enterprise PBX functionality to mobile devices. This allows end users to be accessible when out of the office as well as to leverage wireless LAN (WLAN) networks to improve wireless coverage, reduce costs and provide the ability to seamlessly move calls from the Wi-Fi network to the mobile network and vice-versa.

2 General Test Approach and Test Results

The general test approach was to make mobile originated and mobile terminated calls route through the Avaya telephony infrastructure. The configuration shown in **Figure 1** was used to exercise the features and functionality listed in **Section 2.1**.

2.1 Interoperability Compliance Testing

All functional test cases were performed manually. Testing entailed verifying different types of Avaya system features interacting with the RIM MVS solution. Tests were performed focusing on the following:

- Mobile originated calls routed through the Avaya telephony infrastructure terminating to a desk phone, mobile device or PSTN
- Mobile terminated calls routed through the Avaya telephony infrastructure
- Seamlessly move calls from the Wi-Fi network to the mobile network and vice-versa.
- Desktop originated calls routed to mobile devices
- DTMF digit support for voicemail and conference calls
- Abbreviated Dialing
- Call Forwarding
- Call Hold /Resume
- Shared Line Appearance
- Transfer
- Move Call To Desk

2.2 Test Results

The RIM MVS solution successfully completed all test cases and passed compliance testing for the features identified in **Section 2.1**. The RIM MVS solution was able to route inbound/outbound calls from/to the Avaya telephony infrastructure with all services tested.

The following observations were made:

1. During transfers and hold/resume for DNIS pooling there was a noticeable delay in the BB device screen refresh. This resulted in a delay in audio between 10-20 seconds. It is expected that the cause of this delay was due to poor cellular coverage for the data channel between the BB device and the MVS server. Also DTMF tones are heard during this time.
2. On occasion transfer testing involving the PSTN resulted in one-way audio. This was likely due to problems in the PSTN network. Tests were retried in a simulated PSTN environment and were successful.
3. In certain failure cases of call transfer the DSP resources on the AudioCodes gateway may become unresponsive. By setting the "MaxCallDuration" field in the AudioCodes configuration SIP Definitions→Advanced Parameters→MaxCallDuration, an administrator can allow the AudioCodes gateway to automatically release and clean up the resources.

2.3 Support

Use the BlackBerry Technical Support Subscription to engage RIM for technical support using one of the following options:

- Online ticket submission: Visit the BlackBerry Expert Support Center at www.blackberry.com/besc
- Telephone: +1 877-255-2377 (North America Toll-free) or +1 519-888-6181 (International)

3 Reference Configuration

- One Avaya S8300D Server with a Avaya G450 Media Gateway running Communication Manager
- Communication Manager Messaging
- Avaya 2400 Series Digital Telephone
- Avaya 9600 Series IP Telephones running Avaya one-X® Deskphone Edition
- RIM BlackBerry® MVS Server
- RIM BlackBerry® Enterprise Server
- RIM BlackBerry® phones running the MVS Client software
- AudioCodes Mediant 1000
- L2/L3 switch
- DHCP/TFTP/IAS Server
- Microsoft Exchange Server

In **Figure 1**, two SIP trunks are used to connect the Avaya and RIM infrastructures. The Avaya Aura® Session Manager connects both the Avaya Aura® Communication Manager and the AudioCodes Mediant 1000. The Mediant 1000 serves as a gateway between the Avaya Aura® systems and the Blackberry® MVS. The configuration includes four Blackberry devices. Three are associated to desk phones and one is standalone. The standalone mobile device still requires a station extension provisioned on Communication Manager but no physical phone will be logged into that extension.

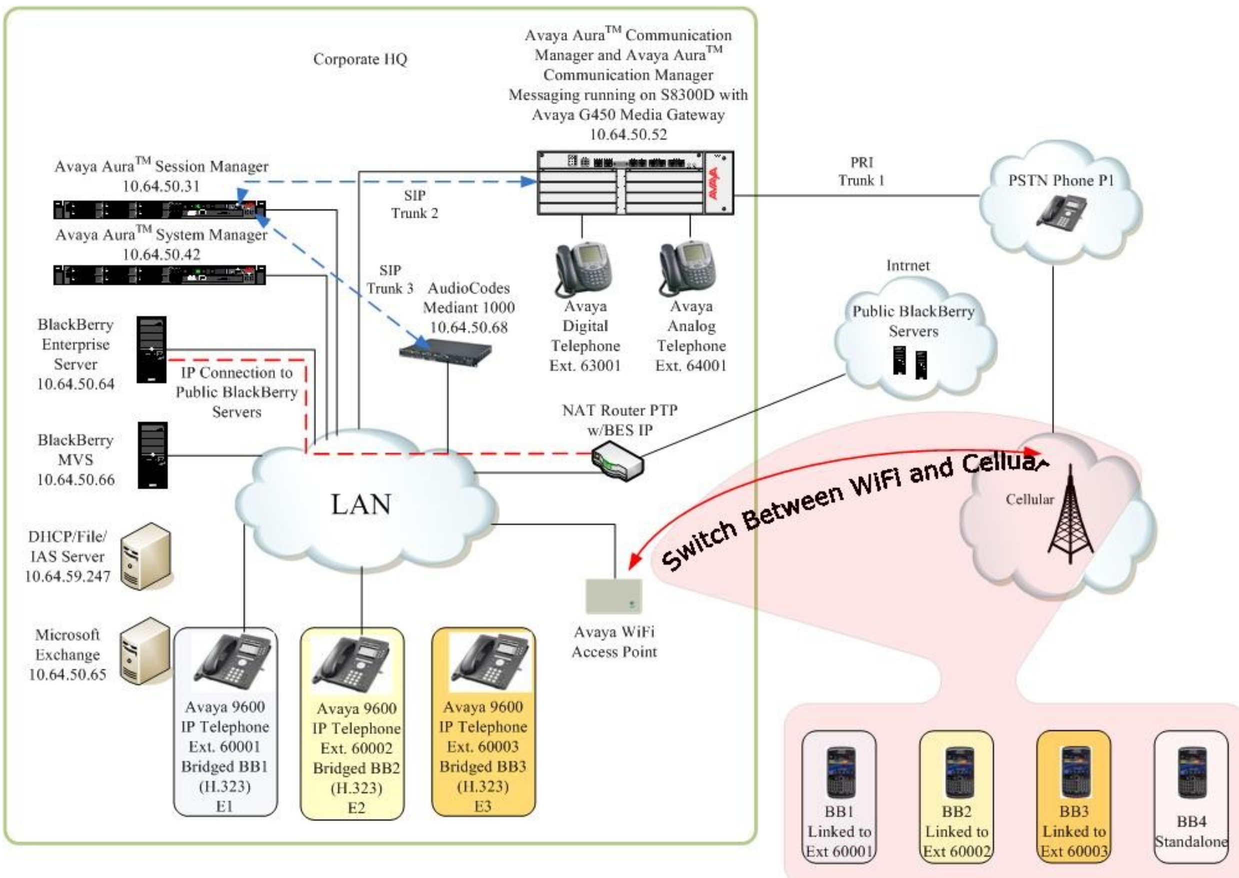


Figure 1: Avaya/RIM MVS Solution Diagram

On the MVS, each of the Blackberry® devices can be provisioned to use one of two calling methods: device-initiated calling or PBX-initiated calling. Each call involving the Blackberry device is comprised of two call legs that connect to the MVS server and is joined together by the MVS.

The following is a high-level, highly simplified description of the call flow of both an inbound and outbound call using the two calling methods. For a complete detailed description of the call flows, refer to RIM MVS documentation in **Section 11**.

Inbound call (Device-initiated calling)

- PSTN caller calls the enterprise DID number assigned to the desk phone/Blackberry® device pair. The call arrives at Avaya Aura® Communication Manager on trunk 1.
- Avaya Aura® Communication Manager rings the desk phone (if it exists) and also sends the call out trunk 2 to the Avaya Aura® Session Manager
- Avaya Aura® Session Manger sends the call out trunk 3 to the MVS.
- The MVS exchanges information with the MVS client on the Blackberry® device via a data channel which includes the DNIS call-back number assigned by the enterprise.
- This exchange causes the Blackberry® device to place a call across the wireless network/PSTN to the DNIS call-back number. This number is provisioned in **Section 8.3.1**.
- This call arrives at Avaya Aura® Communication Manager on trunk 1. The call is directed to trunk 2 to reach Avaya Aura® Session Manager then trunk 3 to the MVS.
- The MVS exchanges information with the MVS client to indicate the call has been received.
- The Blackberry® device rings and the user answers.
- The MVS joins the two call legs together.

Inbound call (PBX-initiated calling)

- PSTN caller calls the enterprise DID number assigned to the desk phone/Blackberry® device pair. The call arrives at Avaya Aura® Communication Manager on trunk 1.
- Avaya Aura® Communication Manager rings the desk phone (if it exists) and also sends the call out trunk 2 to the Avaya Aura® Session Manager
- Avaya Aura® Session Manager sends the call out trunk 3 to reach the MVS.
- The MVS places a call to the Blackberry® device using its mobile number.
- This call arrives at the Avaya Aura® Session Manager on trunk 3 and is sent to the Avaya Aura® Communication Manager on trunk 2.
- The call is directed to trunk 1 to reach the PSTN/wireless network.
- The Blackberry® device rings and the user answers. The caller ID is a number assigned by the enterprise. The caller ID is a number assigned by the enterprise and provisioned in **Section 8.3.2**.
- The MVS exchanges information with the MVS client on the Blackberry® device.
- The MVS joins the two call legs together.

Outbound call (Device-initiated calling)

- Blackberry® device user dials a PSTN number that is not assigned to the enterprise.
- The MVS client on the Blackberry® device exchanges information with the MVS via a data channel which includes the DNIS call-back number assigned by the enterprise.
- This exchange causes the Blackberry® device to place a call across the wireless network/PSTN to the DNIS call-back number.
- This call arrives at Avaya Aura® Communication Manager on trunk 1. The call is directed to trunk 2 and sent to the Avaya Aura® Session Manager.
- Avaya Aura® Session Manager sends the call out trunk 3 to reach the MVS.
- The MVS answers the call and exchanges information with the MVS client.

- The MVS places a call to the external number.
- This call arrives at Avaya Aura® Session Manager on trunk 3 and is sent to Avaya Aura® Communication Manager on trunk 2. The call is directed to trunk 1 to reach the PSTN.
- The recipient answers the call at the external phone.
- The MVS exchanges information with the MVS client and then joins the two call legs together.

Outbound call (PBX-initiated calling)

- Blackberry® device user dials a PSTN number that is not assigned to the enterprise.
- The MVS client on the Blackberry® device exchanges information with the MVS via a data channel which includes the external number dialed.
- The MVS places a call to the Blackberry® device using its mobile number.
- This call arrives at Avaya Aura® Session Manager on trunk 3 and is sent to Avaya Aura® Communication Manager on trunk 2. The call is directed to trunk 1 to reach the PSTN/wireless network.
- The Blackberry® device rings and the user answers. The caller ID is a number assigned by the enterprise.
- The MVS exchanges information with the MVS client on the Blackberry® device.
- The MVS places a call to the external number.
- This call arrives at Avaya Aura® Session Manager on trunk 3 and is sent to Avaya Aura® Communication Manager on trunk 2. The call is directed to trunk 1 to reach the PSTN/wireless network.
- The recipient answers the call at the external phone.
- The MVS exchanges information with the MVS client and then joins the two call legs together.

4 Equipment and Software Validated

The following equipment and software were used in this compliance test.

Equipment	Software/Firmware
Avaya PBX Products	
Avaya S8300D Server running Avaya Aura® Communication Manager	Avaya Aura® Communication Manager 6.0.1 with SP5.0.1(Patch 19303)
Avaya G450 Media Gateway	
MGP	HW 2 FW 31.20.0
MM710 T1 Module	HW 5 FW 22
MM711 Analog Module	HW 23 FW 73
MM712 DCP Media Module	HW 7 FW 14
MP80 VoIP-DSP	HW 6 FW 67
Avaya Aura® Session Manager	
Avaya Aura® Session Manager, HP Proliant DL360 G7	6.1 with SP5

<table>
<tr><th colspan="2">Equipment</th><th colspan="2">Software/Firmware</th></tr>
<tr><td colspan="2">Avaya Aura® System Manager, HP Proliant DL360 G7</td><td colspan="2">6.1 with SP5</td></tr>
<tr><td colspan="4">Avaya Messaging (Voice Mail) Products</td></tr>
<tr><td colspan="2">Avaya Aura® Communication Manager Messaging (CMM)</td><td colspan="2">6.0</td></tr>
<tr><td colspan="4">Avaya Telephony Sets</td></tr>
<tr><td colspan="2">Avaya 9600 Series IP Telephones</td><td colspan="2">(H.323 3.1SP2)</td></tr>
<tr><td colspan="2">Avaya 2420 Digital Telephone</td><td colspan="2">-</td></tr>
<tr><td colspan="2">Analog Telephone</td><td colspan="2">-</td></tr>
<tr><td colspan="4">RIM Products</td></tr>
<tr><td colspan="2">BlackBerry Enterprise Server</td><td colspan="2">BES 5.0.3</td></tr>
<tr><td colspan="2">BlackBerry MVS</td><td colspan="2">MVS 5.1.1</td></tr>
<tr><td colspan="4">BlackBerry Devices</td></tr>
<tr><th>Device</th><th>Carrier</th><th>Bundle</th><th>Type</th></tr>
<tr><td>9670</td><td>Sprint</td><td>HH6.0</td><td>CDMA</td></tr>
<tr><td>9810</td><td>AT&T</td><td>HH7.0</td><td>GSM</td></tr>
<tr><td>9800</td><td>AT&T</td><td>HH6.0</td><td>GSM</td></tr>
<tr><td>9850</td><td>Verizon</td><td>HH7.0</td><td>CDMA</td></tr>
<tr><td colspan="4">AudioCodes Products</td></tr>
<tr><td colspan="2">Audio Codes Gateway 1000</td><td colspan="2">6.40A.011.008</td></tr>
<tr><td colspan="4">Microsoft products</td></tr>
<tr><td colspan="2">Microsoft Windows 2008-SP2 Server</td><td colspan="2">Microsoft Windows 2008 Server</td></tr>
</table>

5 Configure Avaya Aura® Communication Manager

This section describes the steps required for Communication Manager to support the configuration in **Figure 1**. The following pages provide step-by-step instructions on how to administer parameters specific to the RIM MVS solution only. The assumption is that the appropriate license and authentication files have been installed on the servers and that login and password credentials are available and that the reader has a basic understanding of the administration of Communication Manager. It is assumed that all other connections (e.g., PSTN or LAN) are configured and will not be covered in this document. The reader will need access to the System Access Terminal (SAT). For detailed information on the installation, maintenance, and configuration of Communication Manager, please consult **Section 11** (**[1]**).

5.1 System Parameters Customer Options

The steps in this section verify that there is enough off-PBX Telephone capacity on Communication Manager to support the configuration in **Figure 1**.

Using the SAT, verify that there is enough Off-PBX Telephones (OPS for Off-PBX Stations) capacity on the **System-Parameters Customer-Options** form to support the configuration. The license file installed on the system controls the number of allowable OPS stations. If enough capacity is not available, contact an authorized Avaya sales representative.

Issue the command **display system-parameters customer-options** to display the active licensed features and their related capacities. Go to **Page 1** to ensure that the **Maximum Off-PBX Telephones – OPS** value is equal to or greater than the number of endpoints projected in the configuration.

```
display system-parameters customer-options                    Page   1 of  11
                               OPTIONAL FEATURES

     G3 Version: V16                                Software Package: Enterprise
       Location: 2                                   System ID (SID): 1
       Platform: 28                                  Module ID (MID): 1

                                                                 USED
                               Platform Maximum Ports: 6400  54
                                     Maximum Stations: 2400  9
                             Maximum XMOBILE Stations: 2400  0
                   Maximum Off-PBX Telephones - EC500: 9600  5
                   Maximum Off-PBX Telephones -   OPS: 9600  5
                   Maximum Off-PBX Telephones - PBFMC: 9600  0
                   Maximum Off-PBX Telephones - PVFMC: 9600  0
                   Maximum Off-PBX Telephones - SCCAN: 0     0
                        Maximum Survivable Processors: 313   0
```

Automatic Route Selection (ARS) will be used to route calls to the PSTN trunk. On **Page 3**, verify that **ARS** is set to **y**.

```
display system-parameters customer-options                      Page   3 of  11
                                OPTIONAL FEATURES

    Abbreviated Dialing Enhanced List? y          Audible Message Waiting? y
        Access Security Gateway (ASG)? n              Authorization Codes? y
        Analog Trunk Incoming Call ID? y                      CAS Branch? n
 A/D Grp/Sys List Dialing Start at 01? y                        CAS Main? n
Answer Supervision by Call Classifier? y               Change COR by FAC? n
                                 ARS? y  Computer Telephony Adjunct Links? y
                ARS/AAR Partitioning? y   Cvg Of Calls Redirected Off-net? y
          ARS/AAR Dialing without FAC? y                     DCS (Basic)? y
          ASAI Link Core Capabilities? n               DCS Call Coverage? y
          ASAI Link Plus Capabilities? n              DCS with Rerouting? y
       Async. Transfer Mode (ATM) PNC? n
  Async. Transfer Mode (ATM) Trunking? n    Digital Loss Plan Modification? y
              ATM WAN Spare Processor? n                        DS1 MSP? y
                                ATMS? y           DS1 Echo Cancellation? y
                  Attendant Vectoring? y
```

5.2 Dial Plan and Access Codes

The dial plan defines what digit strings are defined as extensions and access codes. Feature access codes (fac) can be used to invoke specific PBX features.

Use the **display dialplan analysis** command to display the dial plan. Verify the dial strings that represent extensions and which are configured as a fac or dac. This information will be used in subsequent steps and sections.

```
display 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
   6           5   ext
   8           1   fac
   9           1   fac
   *           4   dac
```

Use the "**change feature-access-codes**" command to assign feature access codes for **AAR** and **ARS** (if not already assigned) that is consistent with the existing dial plan. .

```
change feature-access-codes                                        Page   1 of  10
                               FEATURE ACCESS CODE (FAC)
         Abbreviated Dialing List1 Access Code:
         Abbreviated Dialing List2 Access Code:
         Abbreviated Dialing List3 Access Code:
Abbreviated Dial - Prgm Group List Access Code:
                      Announcement Access Code:
                       Answer Back Access Code:
                         Attendant Access Code:
      Auto Alternate Routing (AAR) Access Code: 8
    Auto Route Selection (ARS) - Access Code 1: 9     Access Code 2:
                 Automatic Callback Activation:        Deactivation:
Call Forwarding Activation Busy/DA:        All:        Deactivation:
   Call Forwarding Enhanced Status:        Act:        Deactivation:
                         Call Park Access Code:
                       Call Pickup Access Code:
CAS Remote Hold/Answer Hold-Unhold Access Code:
                  CDR Account Code Access Code:
                        Change COR Access Code:
                   Change Coverage Access Code:
            Conditional Call Extend Activation:        Deactivation:
                   Contact Closure   Open Code:          Close Code:
```

5.3 Configure IP Node Names

In the **IP Node Names** form, assign the name and IP address of Session Manager. This is used to terminate the SIP trunk with the Session Manager. The names will be used in the signaling group configuration configured later.

```
change node-names ip                                            Page   1 of   2
                                  IP NODE NAMES
    Name              IP Address
default             0.0.0.0
msgserver           10.64.50.52
procr               10.64.50.52
procr6              ::
sm5031              10.64.50.31

( 5  of 5    administered node-names were displayed )
Use 'list node-names' command to see all the administered node-names
Use 'change node-names ip xxx' to change a node-name 'xxx' or add a node-name
```

5.4 Configure IP Network Region

In the **IP Network Region** form, the **Authoritative Domain** field is configured to match the domain name configured on Session Manager. In this configuration, the domain name is *avaya.com*. By default, **IP-IP Direct Audio** (shuffling) is enabled to allow audio traffic to be sent directly between IP endpoints without using media resources in the Avaya G450 Media Gateway. The **IP Network Region** form also specifies the **IP Codec Set** to be used for Desk Phone calls. This IP codec set is used when its corresponding network region (i.e., IP Network Region '1') is specified in the SIP signaling groups. Accept the default values for the other fields.

```
change ip-network-region 1                                      Page   1 of  20
                               IP NETWORK REGION
  Region: 1
Location: 1       Authoritative Domain: avaya.com
    Name:
MEDIA PARAMETERS                Intra-region IP-IP Direct Audio: yes
      Codec Set: 1              Inter-region IP-IP Direct Audio: yes
   UDP Port Min: 49152                       IP Audio Hairpinning? y
   UDP Port Max: 65535
DIFFSERV/TOS PARAMETERS
 Call Control PHB Value: 46
        Audio PHB Value: 46
        Video PHB Value: 26
802.1P/Q PARAMETERS
 Call Control 802.1p Priority: 6
        Audio 802.1p Priority: 6
        Video 802.1p Priority: 5      AUDIO RESOURCE RESERVATION PARAMETERS
H.323 IP ENDPOINTS                                       RSVP Enabled? n
  H.323 Link Bounce Recovery? y
 Idle Traffic Interval (sec): 20
   Keep-Alive Interval (sec): 5
            Keep-Alive Count: 5
```

5.5 Configure IP Codec Set

In the **IP Codec Set** form, select the audio codec type supported for Desk Phone calls. The form is accessed via the **change ip-codec-set 1** command. Note that IP codec set '1' was specified in IP Network Region '1'. The default settings of the **ip-codec-set** form are shown below.

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

                          IP Codec Set

    Codec Set: 1

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


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

5.6 Configure SIP Trunk to Avaya Aura® Session Manager

Add a signaling group for calls that need to reach the BlackBerry® MVS. Prior to configuring a SIP trunk group for communication with Session Manager, a SIP signaling group must be configured. Configure the Signaling Group form as shown below:

- Set the **Group Type** field to *sip*.
- Set the **Transport Method** to *tcp*.
- Specify the Communication Manager (procr) and the Session Manager as the two end-points of the signaling group in the **Near-end Node Name** field and the **Far-end Node Name** field, respectively. These field values were configured in the **IP Node Names** form shown in **Section 5.3**.
- Ensure that the recommended TCP port value of *5060* is configured in the **Near-end Listen Port** and the **Far-end Listen Port** fields. If the **Far-end Network Region** field is configured, the codec for the call will be selected from the IP codec set assigned to that network region.
- Enter the domain name in the **Far-end Domain** field. In this configuration, the domain name is *avaya.com*.
- If calls to the **BlackBerry® MVS** are to be shuffled, then the **Direct IP-IP Audio Connections** field must be set to *y*.
- The **DTMF over IP** field is set to the default value of *rtp-payload*. Avaya Communication Manager supports DTMF transmission using RFC 2833.

- The default values for the other fields may be used.

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

 Group Number: 10            Group Type: sip
  IMS Enabled? n       Transport Method: tcp
        Q-SIP? n                                            SIP Enabled LSP? n
     IP Video? n                                   Enforce SIPS URI for SRTP? y
  Peer Detection Enabled? y  Peer Server: Others



   Near-end Node Name: procr                  Far-end Node Name: sm5031
 Near-end Listen Port: 5060                 Far-end Listen Port: 5060
                                         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
```

Configure the **Trunk Group** form shown below for outgoing calls to **BlackBerry® MVS**. Set the **Group Type** field to *sip*, set the **Service Type** field to *tie*, specify the signaling group associated with this trunk group in the **Signaling Group** field, and specify the **Number of Members** supported by this SIP trunk group.

```
add trunk-group 2                                               Page   1 of  21
                                TRUNK GROUP

Group Number: 2                   Group Type: sip           CDR Reports: y
  Group Name: To sm5032                  COR: 1        TN: 1        TAC: *002
   Direction: two-way       Outgoing Display? n
 Dial Access? n                                   Night Service:
Queue Length: 0
Service Type: tie                  Auth Code? n
                                              Member Assignment Method: auto
                                                       Signaling Group: 2
                                                     Number of Members: 10
```

On **Page 3** of the trunk group form, set the **Numbering Format** field to *private*. This field specifies the format type of the calling party number sent to **BlackBerry® MVS**. The specific calling party number format is specified in the **Numbering- Private Format** form.

```
add trunk-group 2                                                Page   3 of  21
TRUNK FEATURES
          ACA Assignment? n            Measured: none
                                                           Maintenance Tests? y



                     Numbering Format: private
                                                UUI Treatment: service-provider

                                                 Replace Restricted Numbers? n
                                                Replace Unavailable Numbers? n


                                Modify Tandem Calling Number: no




 Show ANSWERED BY on Display? y
```

5.7 Configure Route Pattern

A route pattern is configured to use the trunk defined in **Section 5.6**. The route pattern can also be configured to perform digit manipulation on outgoing calls if necessary. Calls destined for the Mediant 1000 will be routed via the Session Manager using the route pattern defined below.

When configuring a route pattern, use the **change route-pattern x** command, where **x** is an available route pattern number. For the compliance test, route pattern 3 was selected. Set the parameters as shown below.

- For the **Pattern Name**, enter a descriptive name.
- Set the **Grp No** to the trunk group number created in **Section 5.6**.
- Set the **FRL** (Facility Restriction Level) to a value that allows all users access to the trunk that need to use it. The value of **0** is the least restrictive. This is the value used for the compliance test.
- Default values may be used for all other fields.

```
change route-pattern 3                                             Page   1 of   3
                    Pattern Number: 3   Pattern Name: To sm5031
                             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: 2    0                    0                                            n   user
 2:                                                                        n   user
 3:                                                                        n   user
 4:                                                                        n   user
 5:                                                                        n   user
 6:                                                                        n   user

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

5.8 Configure Automatic Alternate Routing

Automatic Alternate Routing (AAR) is used to route the calls to the Mediant 1000 via the Session Manager in order to reach the mobile devices.

When creating entries in the AAR DIGIT ANALYSIS TABLE, use the **change aar analysis x** command, where **x** is the first digit in the dialed string to be entered. Create an entry to reach the mobile user extensions supported by the configuration in **Figure 1**. The extensions are reached using the aar table entry “6”. In addition, a DNIS call-back number must be assigned to the MVS server from the pool of DID numbers owned by the enterprise. This number must also be routed to the trunk connected to the Avaya Aura® Session Manager. In the example below the DNIS call-back number is **13035383606** and the aar table entry “130” is used for routing. When creating the entries, enter the parameters as defined below.

- For the **Dialed String**, enter the mobile extension or the DNIS call-back number.
- Set the **Total Min** and **Total Max** fields to the number length.
- Set the **Route Pattern** to the route pattern defined in **Section 5.7** that directs calls to the trunk connected to the Avaya Aura® Session Manager.
- Set the **Call Type** to *aar*.

```
change aar analysis 6                                           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
    6                     5    5     3        aar          n
```

```
change aar analysis 130                                         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
    130                   11   11    3        aar          n
```

5.9 Incoming Call Treatment for the PSTN Trunk

Inbound calls from the PSTN are routed using incoming call treatment associated with the PSTN trunk.

Incoming call treatment is used to match on an incoming number and then perform digit manipulation to properly route the call to an internal extension or route it to a trunk via AAR. Use the **change inc-call-handling-trmt trunk-group 1** command to create the entries in the example below. Trunk group 1 is used because this is the trunk group connected to the PSTN as shown in **Figure 1**. The DID numbers 5381250, 5381619 and 5383613 are associated with the internal extensions 60003, 60002 and 60001 respectively. By deleting all digits of these numbers and inserting the internal extension, the inbound DID is converted to an internal extension. The DID number 5383606 is the DNIS call-back number so it is pre-pended with 81303. The digit 8 routes the call to AAR for further processing while the digits 1303 are inserted for dialing. The parameters in the table are defined as follows:

- Set the **Service/Feature** to *public-ntwrk*.
- Set the **Number Len** to the length of the incoming number to match on.
- Set the **Number Digits** to the incoming number or prefix to match on.
- Set the **Del** field to the number of digits to delete from the beginning of the number.
- Set the **Insert** field to the digits to be inserted at the beginning of the number.

```
change inc-call-handling-trmt trunk-group 1                     Page   1 of   3
                       INCOMING CALL HANDLING TREATMENT
 Service/      Number   Number      Del Insert            Per Call Night
 Feature        Len      Digits                           CPN/BN   Serv
 public-ntwrk    7  5381250          all 60003
 public-ntwrk    7  5381619          all 60002
 public-ntwrk    7  5383606              81303
 public-ntwrk    7  5383613          all 60001
```

5.10 Stations and Off-PBX Station Mapping For Mobile Devices

Each mobile device will be associated with a station extension configured on Communication Manager. The station extension may represent a physical desk phone or may be an extension with no phone logged in to it. In the case of the compliance test, all three extensions 60001, 60002, and 60003 were configured on Communication Manager. (See **Figure 1)**.

To associate a mobile device to each of these station extensions requires an off-pbx station mapping as shown below.

In general, a mobile device will be associated with an existing desk phone for which the Communication Manager station extension will already be configured. However, in the case of mobile devices that are not associated with a physical phone then a station must be added.

Use the **add station 60001** command to create the station for this user.

```
add station 60001                                              Page   1 of   5
                                     STATION

Extension: 60001                         Lock Messages? n              BCC: 0
     Type: 9620                          Security Code: 123456          TN: 1
     Port: IP                          Coverage Path 1: 99             COR: 1
     Name: Station 60009               Coverage Path 2:                COS: 1
                                       Hunt-to Station:
STATION OPTIONS
                                           Time of Day Lock Table:
              Loss Group: 19         Personalized Ringing Pattern: 1
                                                 Message Lamp Ext: 60001
            Speakerphone: 2-way               Mute Button Enabled? y
        Display Language: english
 Survivable GK Node Name:
          Survivable COR: internal              Media Complex Ext:
   Survivable Trunk Dest? y                          IP SoftPhone? n

                                                         IP Video? n
                              Short/Prefixed Registration Allowed: default

                                               Customizable Labels? y
```

On **Page 4** under BUTTON ASSIGNMENTS, add an **ec500** button. This step needs to be completed for all extensions associated with mobile users, both existing extensions and new ones.

```
add station 60001                                               Page   4 of   5
                                     STATION
 SITE DATA
       Room:                                            Headset? n
       Jack:                                            Speaker? n
      Cable:                                           Mounting: d
      Floor:                                        Cord Length: 0
   Building:                                          Set Color:

ABBREVIATED DIALING
     List1:                    List2:                    List3:



BUTTON ASSIGNMENTS
 1: call-appr                            4: ec500      Timer? n
 2: call-appr                            5:
 3: call-appr                            6:


    voice-mail
```

To create the mapping between a desktop extension and a mobile device, use the **add off-pbx-telephone station-mapping x** command, where **x** is the desktop extension to be mapped. Multiple station extensions can be added at the same time. Enter the parameters as described below.

- Enter the desktop extension for the **Station Extension**.
- Enter *EC500* for the **Application**.
- Enter the mobile extension for the **Phone Number**. These are the digits that will be sent to the Mediant 1000 via the Session Manager.
- Enter *aar* for **Trunk Selection**. This instructs Communication Manager to use the AAR tables to determine how to route this call.
- Enter an off-pbx-telephone configuration set to use with this call. This configuration set is defined in the next step.

```
add off-pbx-telephone station-mapping                           Page   1 of   3
                  STATIONS WITH OFF-PBX TELEPHONE INTEGRATION

 Station        Application Dial   CC  Phone Number     Trunk       Config  Dual
 Extension                  Prefix                      Selection   Set     Mode
 60001            EC500         -      60001             aar        5
-
```

It is important to note that the ANI (Originating) number of the incoming call must match the **Phone Number** field in the off-pbx-telephone station-mapping in order for the EC500 call to be initiated.

The off-pbx-telephone configuration set defines certain parameters applicable to the applications defined on the off-pbx-telephone station-mapping form. To define a configuration set, use the **change off-pbx-telephone configuration-set x** command, where **x** is an available configuration-set number. On **Page 1**of the form, configure the following for use with this solution.

- **Configuration Set Description** – Enter a meaningful name/description.
- **Calling Number Verification?** – Set to *n*.

```
change off-pbx-telephone configuration-set 5                    Page   1 of   1



                                     CONFIGURATION SET: 5

                         Configuration Set Description: RIM
                                  Calling Number Style: network
                                   CDR for Origination: phone-number
                    CDR for Calls to EC500 Destination? y
                           Fast Connect on Origination? n
                          Post Connect Dialing Options: dtmf
                         Cellular Voice Mail Detection: none
                                        Barge-in Tone? n
                           Calling Number Verification? n
             Call Appearance Selection for Origination: primary-first
                                     Confirmed Answer? n

 Use Shared Voice Connections for Second Call Answered? n
Use Shared Voice Connections for Second Call Initiated? n
```

6 Configure Avaya Aura® Session Manager

This section provides the procedures for configuring Session Manager. The procedures include adding the following items:

- SIP domain
- Logical/physical Locations that can be occupied by SIP Entities
- SIP Entities corresponding to Session Manager, Communication Manager, and AudioCodes Mediant 1000
- Entity Links, which define the SIP trunk parameters used by Session Manager when routing calls to/from SIP Entities
- Routing Policies, which control call routing between the SIP Entities
- Dial Patterns, which govern to which SIP Entity a call is routed
- Session Manager, corresponding to the Avaya Aura® Session Manager Server to be managed by Avaya Aura® System Manager

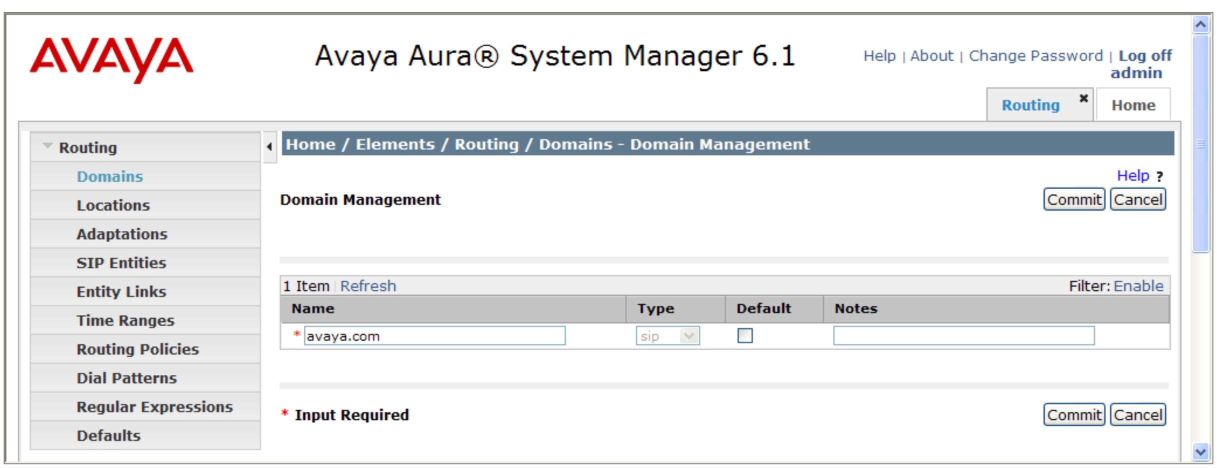
Configuration is accomplished by accessing the browser-based GUI of Avaya Aura® System Manager using the URL "https://<*ip-address*>/SMGR", where <*ip-address*> is the IP address of System Manager. Log in with the appropriate credentials.

6.1 Specify SIP Domain

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

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

Click **Commit**.



6.2 Add Locations

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

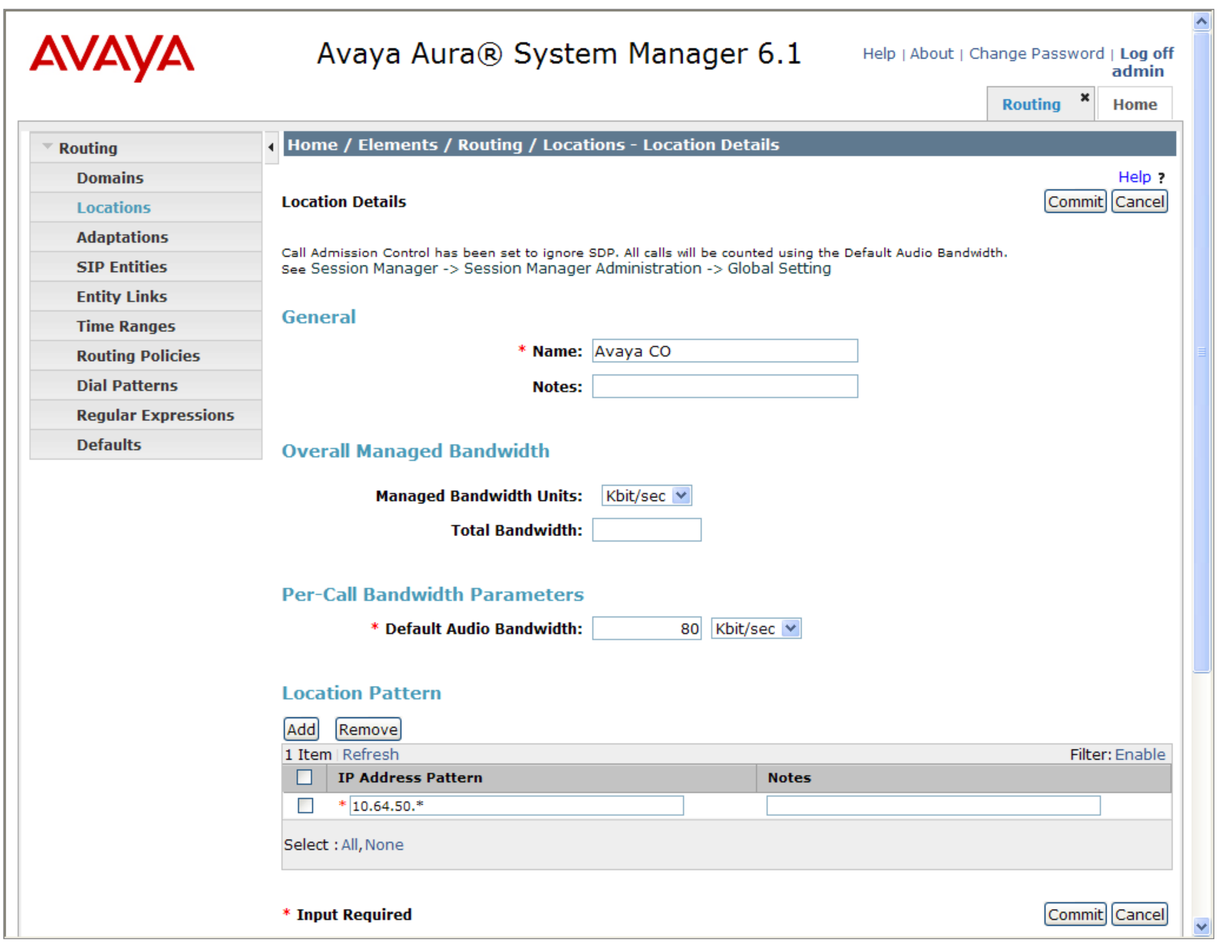
Under *General*:

- **Name:** A descriptive name.
- **Notes:** Descriptive text (optional).

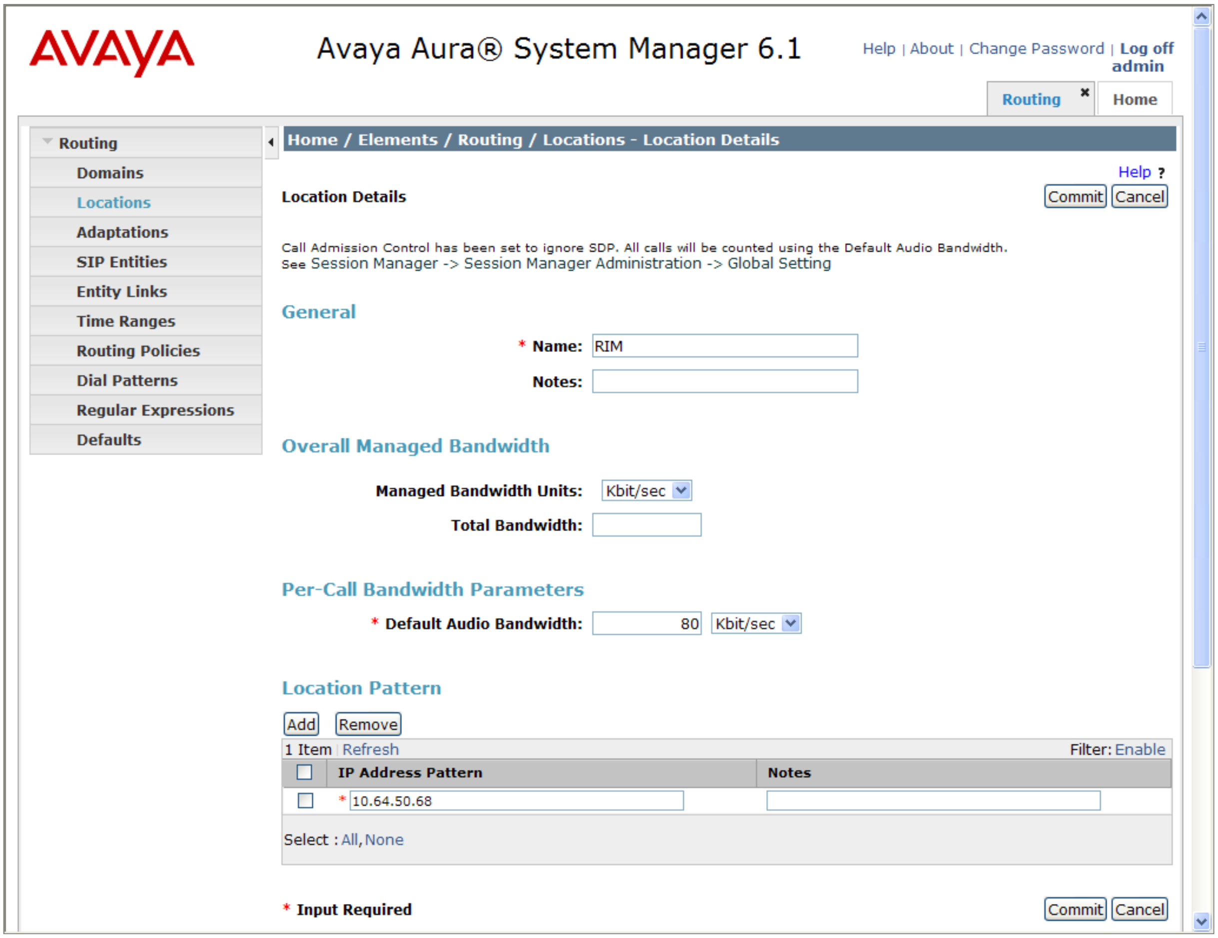
Under *Location Pattern*:

- **IP Address Pattern:** A pattern used to logically identify the location.
- **Notes:** Descriptive text (optional).

The screen below shows the addition of the *Avaya CO* location, where Communication Manager and Session Manager reside. Click **Commit** to save the Location definition.



The screen below shows the addition of the *RIM* location, which uses the AudioCodes at IP Address 10.64.50.68. Other BlackBerry® servers are accessible via this device. Click **Commit** to save the Location definition.



6.3 Add SIP Entities

In the sample configuration, a SIP Entity is added for Session Manager, the Communication Manager, and the AudioCodes Mediant 1000.

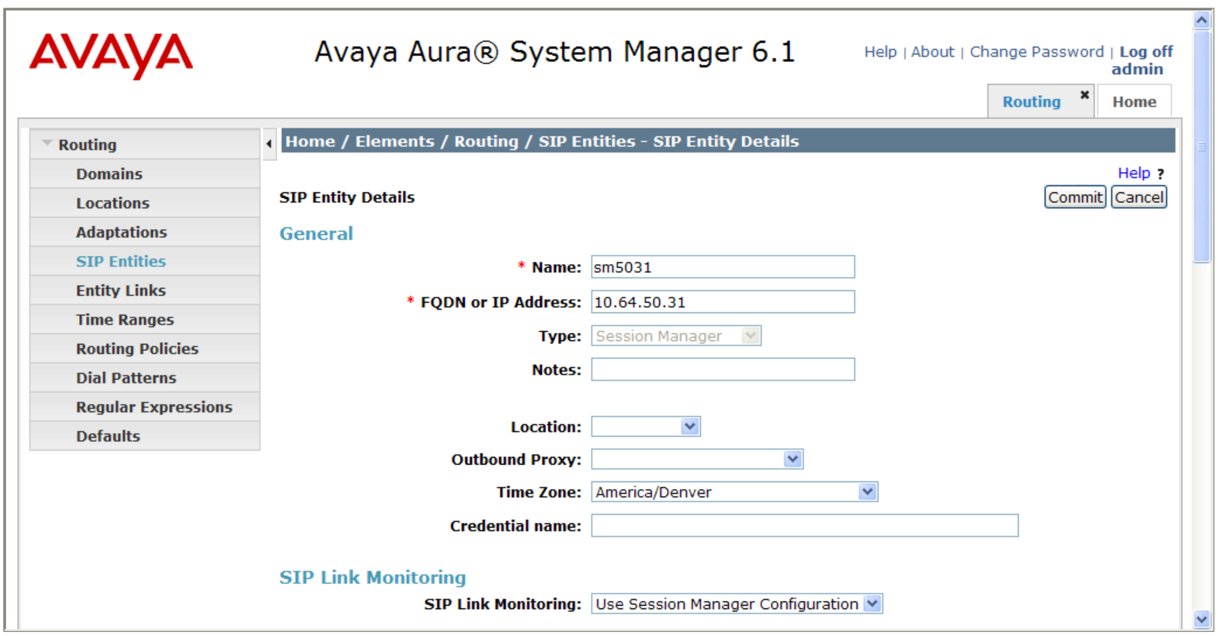
6.3.1 Avaya Aura® Session Manager

A SIP Entity must be added for Session Manager. To add a SIP Entity, select **SIP Entities** on the left and click on the **New** button (not shown) on the right. The following screen is displayed. Fill in the following:

Under *General*:

- **Name:** A descriptive name.
- **FQDN or IP Address:** IP address of the signaling interface on Session Manager.
- **Type:** Select *Session Manager*.
- **Location:** Select one of the locations defined previously.
- **Time Zone:** Time zone for this location.

Defaults may be used for the remaining fields. Click **Commit** to save each SIP Entity definition.



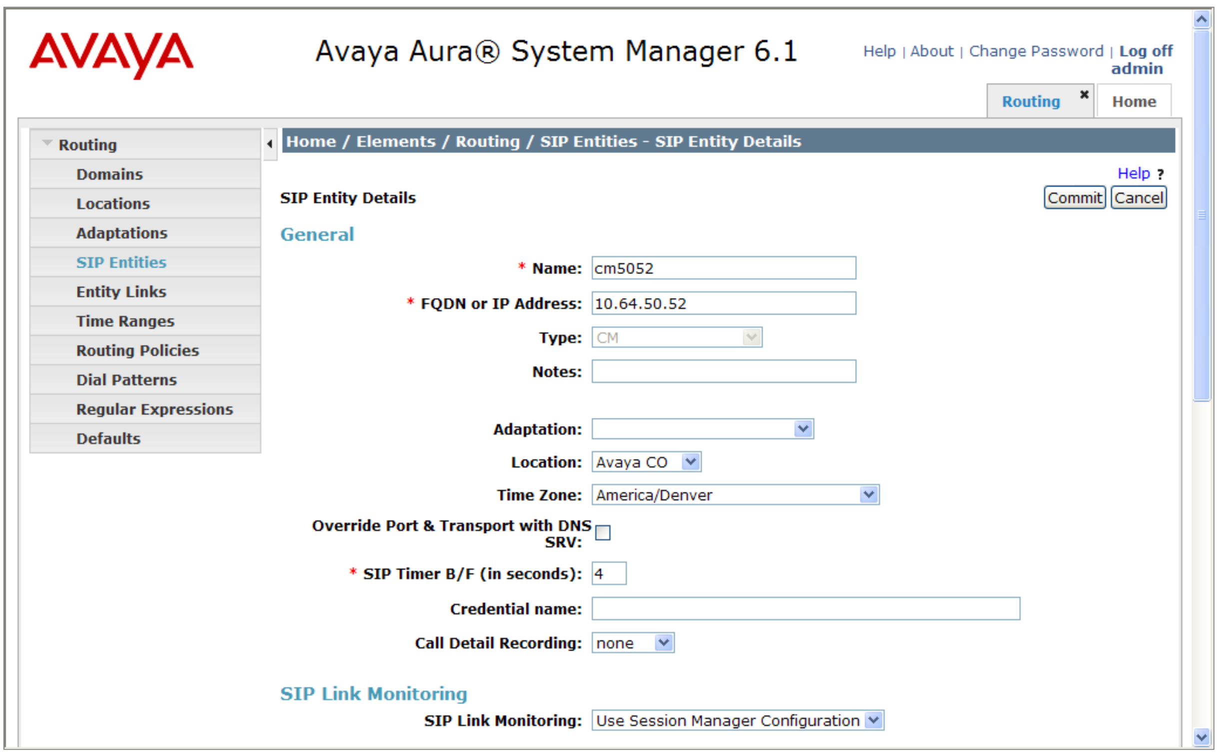
6.3.2 Avaya Aura® Communication Manager

A SIP Entity must be added for Communication Manager. To add a SIP Entity, select **SIP Entities** on the left and click on the **New** button (not shown) on the right. The following screen is displayed. Fill in the following:

Under *General*:

- **Name:** A descriptive name.
- **FQDN or IP Address:** IP address of the signaling interface (e.g., S8300D board) in the G450 telephony system.
- **Type:** Select *CM*.
- **Location:** Select one of the locations defined previously.
- **Time Zone:** Time zone for this location.

Defaults may be used for the remaining fields. Click **Commit** to save each SIP Entity definition.



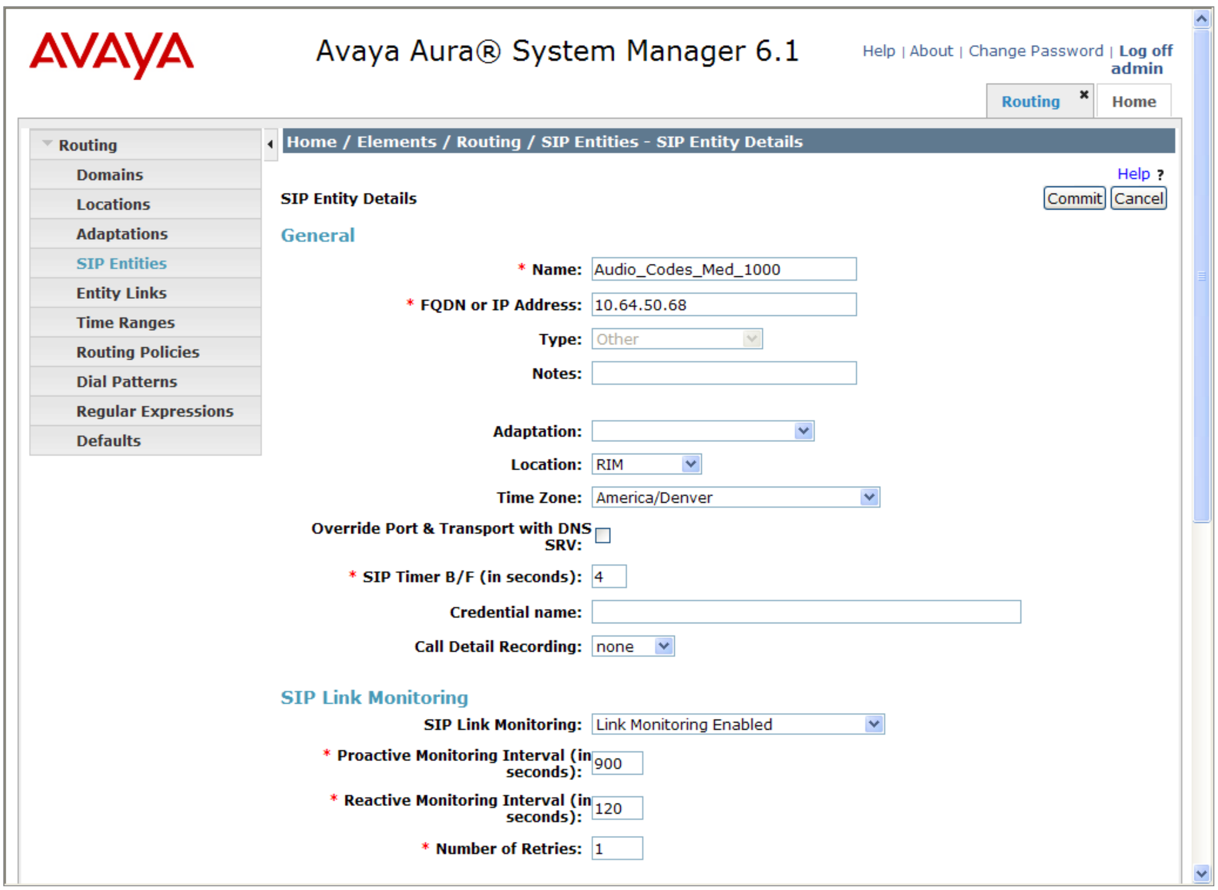
6.3.3 AudioCodes Mediant 1000

A SIP Entity must be added for the AudioCodes Mediant 1000. To add a SIP Entity, select **SIP Entities** on the left and click on the **New** button (not shown) on the right. The following screen is displayed. Fill in the following:

Under *General*:

- **Name:** A descriptive name.
- **FQDN or IP Address:** AudioCodes IP address.
- **Type:** Select *Other*.
- **Location:** Select one of the locations defined previously.
- **Time Zone:** Time zone for this location.

Defaults may be used for the remaining fields. Click **Commit** to save each SIP Entity definition.



6.4 Add Entity Links

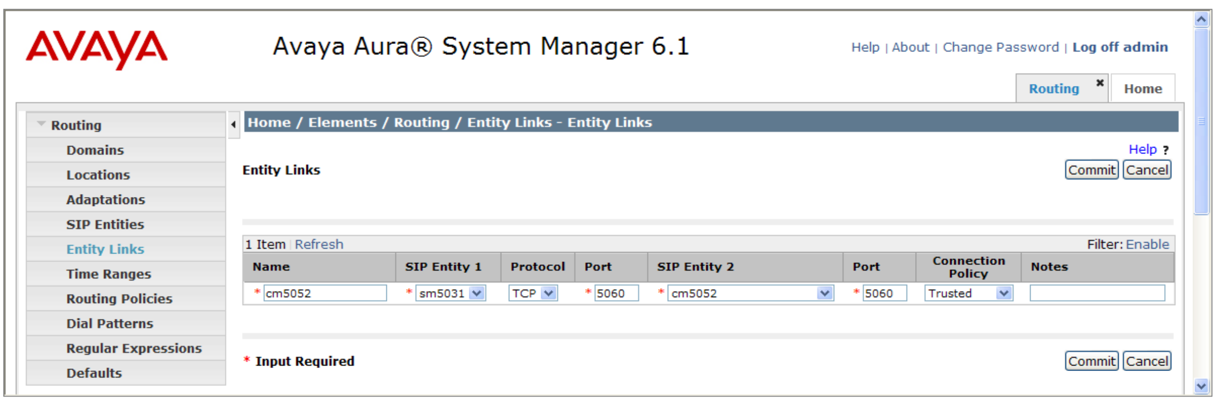
The SIP trunk from Session Manager to Communication Manager and the AudioCodes Mediant 1000 are described by Entity Links. To add an Entity Link, select **Entity Links** on the left and click on the **New** button (not shown) on the right. Fill in the following fields in the new row that is displayed:

- **Name:** A descriptive name.
- **SIP Entity 1:** Select the Session Manager.
- **Protocol:** Select *TCP* as the transport protocol.
- **Port:** Port number to which the other system sends SIP Requests (e.g., *5060* for TCP).
- **SIP Entity 2:** Select SIP Entity 2.
- **Port:** Port number to which the other system sends SIP Requests (e.g., *5060* for TCP).
- **Connection Policy:** Select *Trusted*.

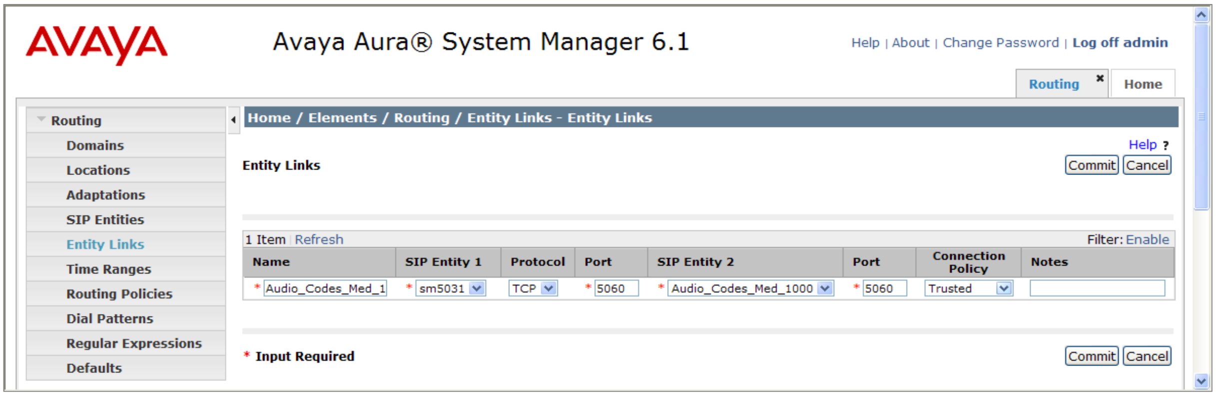
Repeat configuration for Communication Manager and the AudioCodes Mediant 1000.

The following screens display the configuration of each Entity Link. The first entity link is for the connection between Session Manager and Communication Manager and the second entity link is for the connection between Session Manager and the AudioCodes Mediant 1000.

Session Manager ←→ Communication Manager



Session Manager ←→ AudioCodes Mediant 1000



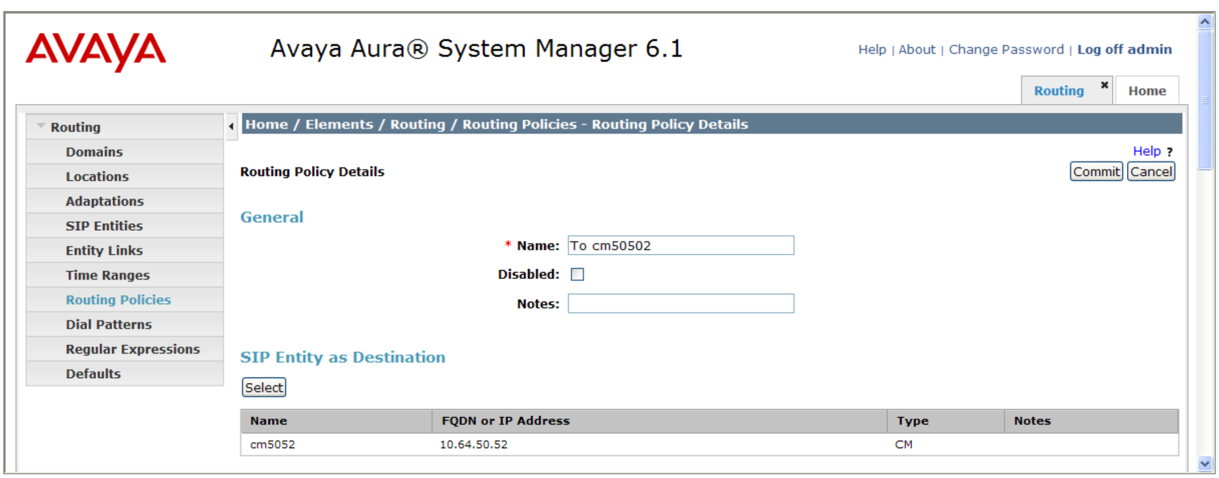
6.5 Add Routing Policies

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

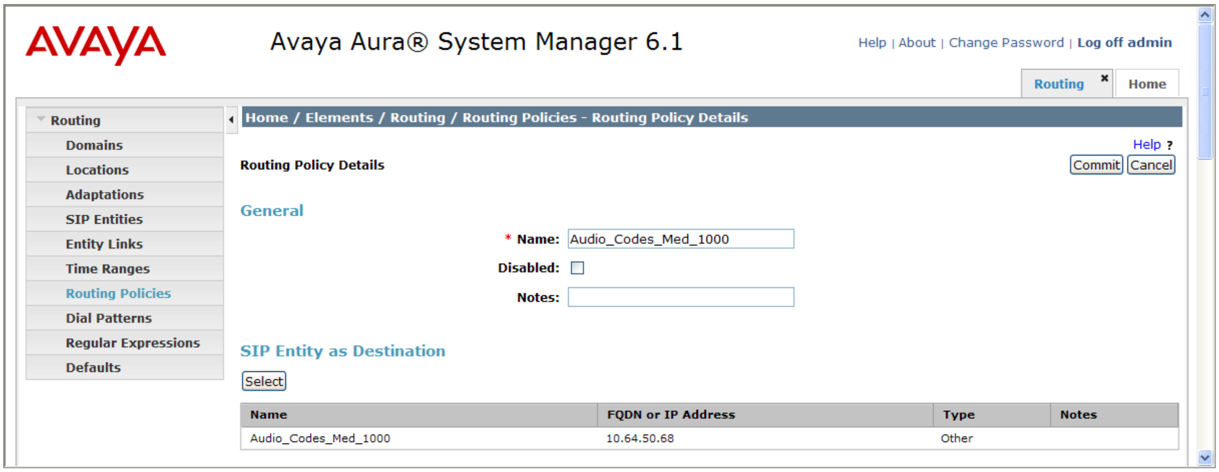
Under *General*:
Enter a descriptive name in **Name**.

Under *SIP Entity as Destination*:
Click **Select**, and then select the appropriate SIP entity to which this routing policy applies.

Defaults can be used for the remaining fields. Click **Commit** to save each Routing Policy definition. The following screen shows the Routing Policy for Communication Manager.



The following screen shows the Routing Policy for the RIM systems.



6.6 Add Dial Patterns

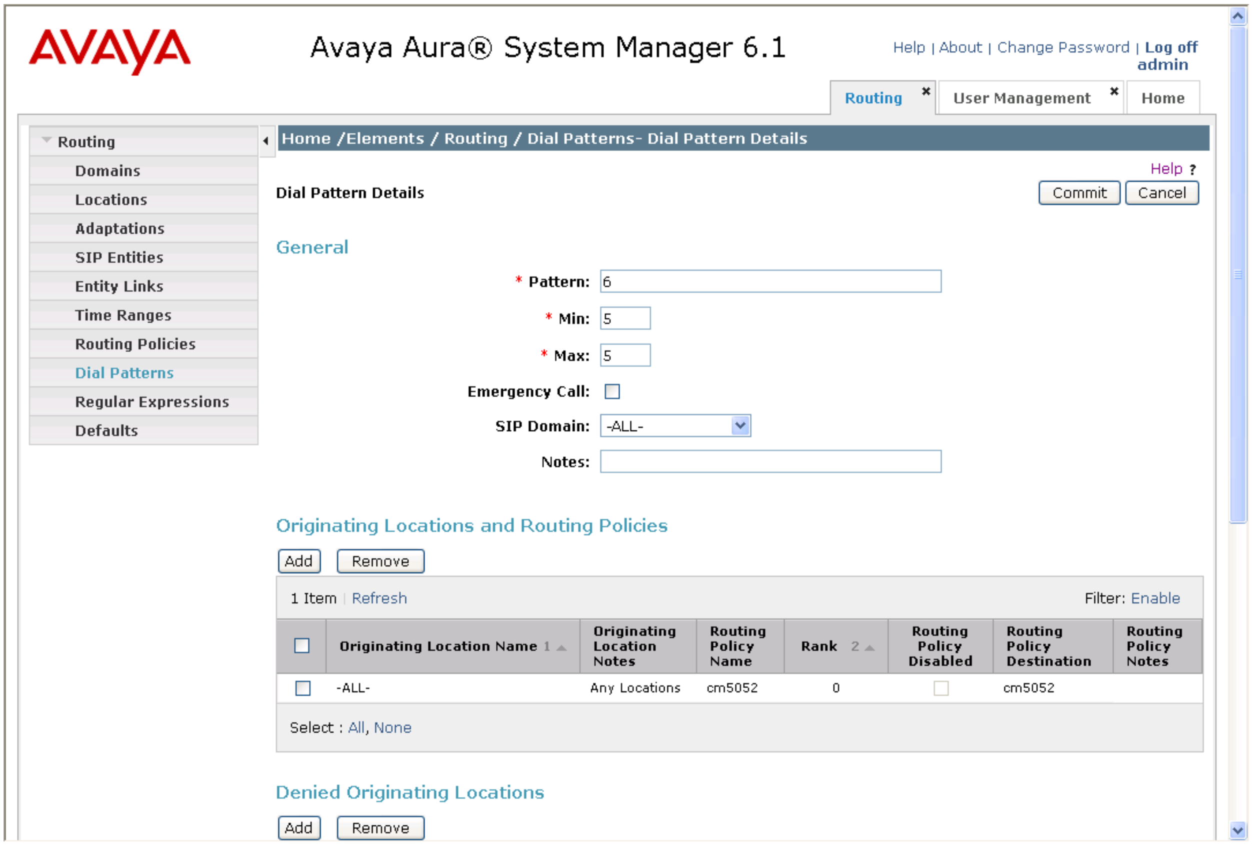
Dial patterns must be defined that will direct calls to the appropriate SIP Entity. In the sample configuration, 5-digit extensions beginning with "6" reside on Communication Manager. To add a dial pattern, select **Dial Patterns** on the left and click on the **New** button (not shown) on the right. Fill in the following:

Under *General*:

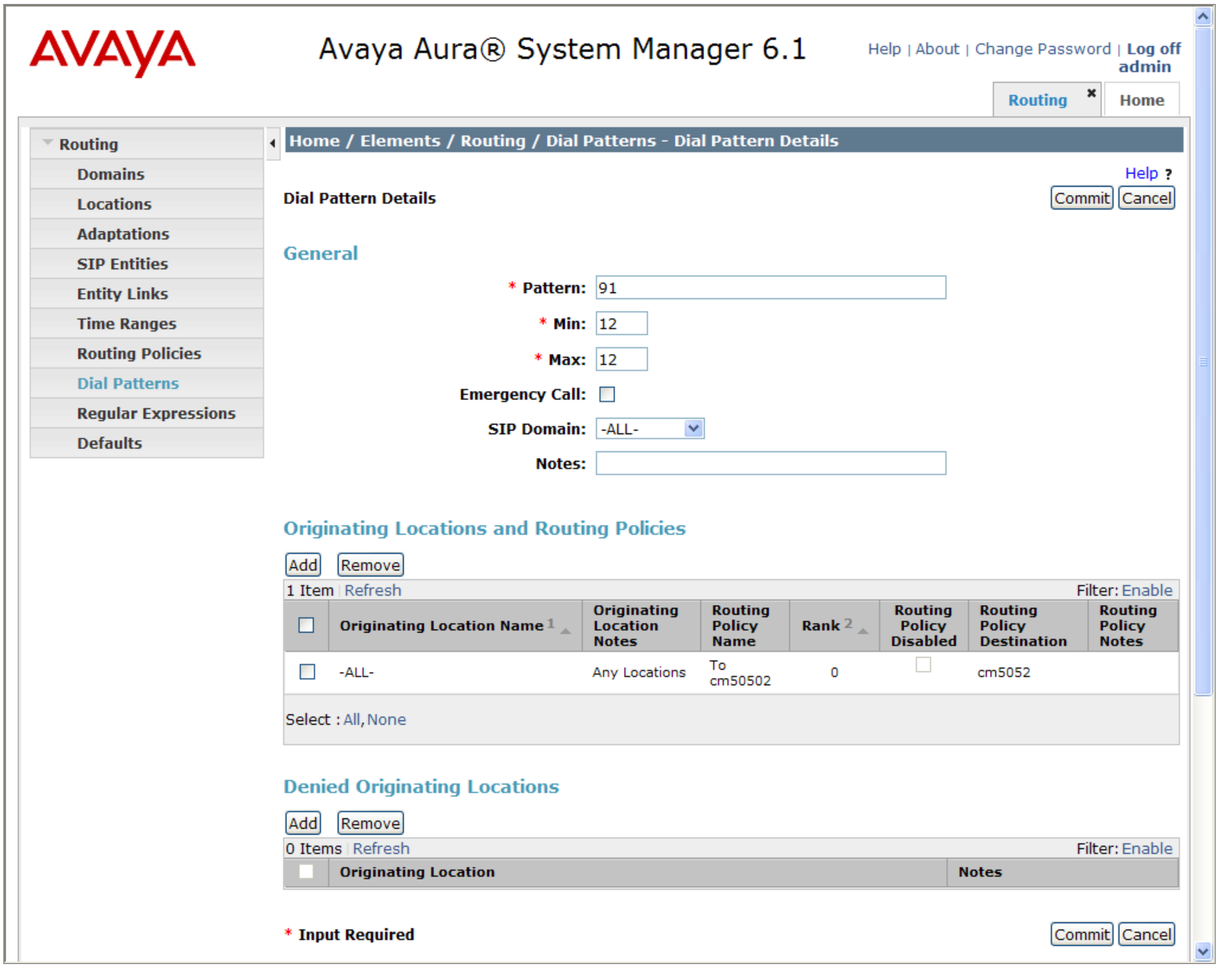
- **Pattern:** Dialed number or prefix.
- **Min** Minimum length of dialed number.
- **Max** Maximum length of dialed number.
- **SIP Domain** SIP domain of dial pattern.
- **Notes** Comment on purpose of dial pattern.

Under *Originating Locations and Routing Policies*:
Click **Add**, and then select the appropriate location and routing policy from the list.

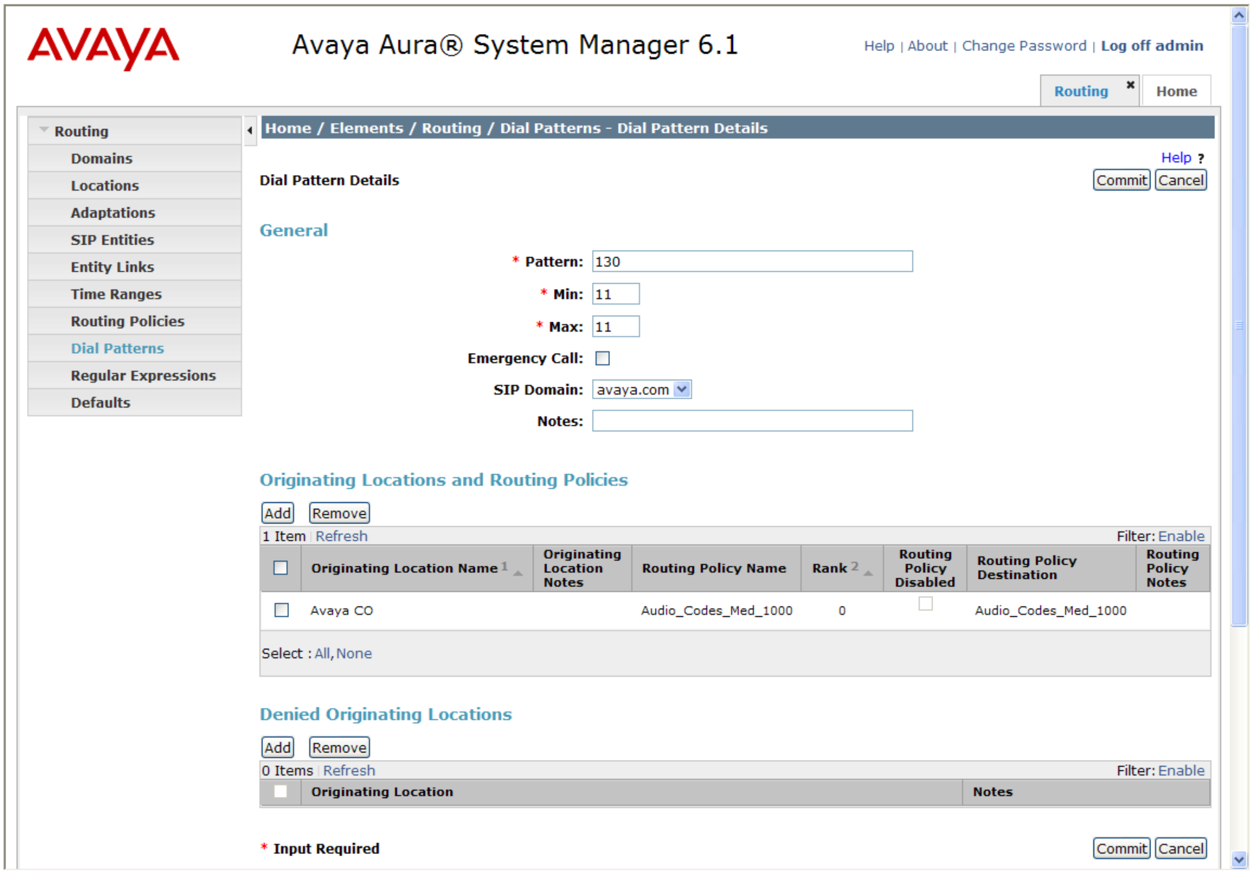
Default values can be used for the remaining fields. Click **Commit** to save this dial pattern. The following screen shows the dial pattern definitions for local extensions on Communication Manager.


AVAYA
Avaya Aura® System Manager 6.1
Help | About | Change Password | Log off admin
Routing
User Management
Home
Routing
Domains
Locations
Adaptations
SIP Entities
Entity Links
Time Ranges
Routing Policies
Dial Patterns
Regular Expressions
Defaults
Home /Elements / Routing / Dial Patterns- Dial Pattern Details
Help ?
Dial Pattern Details
Commit
Cancel
General
* Pattern: 6
* Min: 5
* Max: 5
Emergency Call:
SIP Domain: -ALL-
Notes:
Originating Locations and Routing Policies
Add
Remove
1 Item | Refresh
Filter: Enable
Originating Location Name
Originating Location Notes
Routing Policy Name
Rank
Routing Policy Disabled
Routing Policy Destination
Routing Policy Notes
-ALL-
Any Locations
cm5052
0
cm5052
Select : All, None
Denied Originating Locations
Add
Remove

The following screen shows the dial pattern definition for reaching the PSTN via Communication Manager.



The following screen shows the dial pattern definition that allows calls destined for the DNIS call-back number to be passed through to the RIM Servers.



Example:
DNIS call-back number is 13035383606 and the table entry "130" is used for routing.

7 Configure AudioCodes Mediant 1000 VoIP Media Gateway

This section provides the procedures for configuring the AudioCodes Mediant 1000 VoIP Media Gateway as part of the RIM MVS solution. It is assumed that the Mediant 1000 has been properly installed with the initial configuration following Mediant 1000 standard installation procedures.

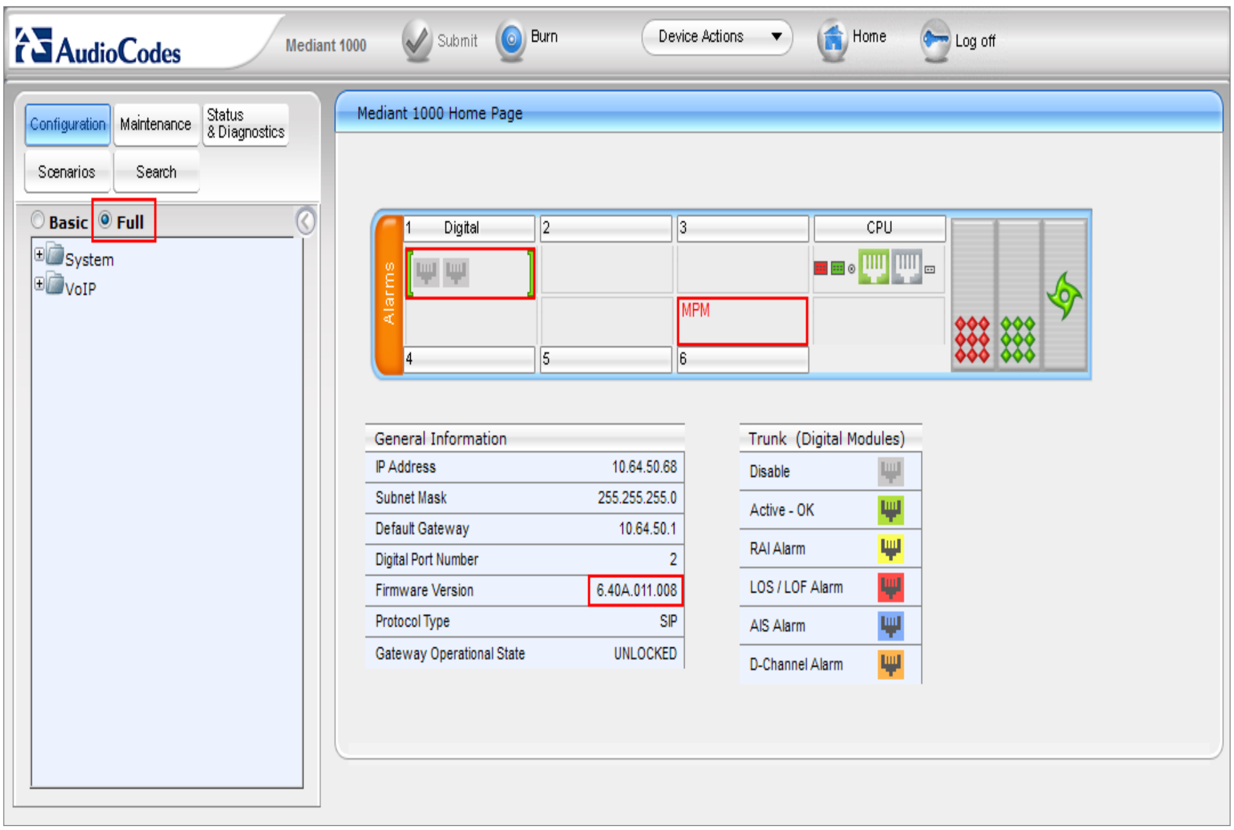
The Mediant 1000 configuration procedures include the following areas:

- System Settings
    - Application Settings
    - Syslog Settings
- VoIP Settings
    - TDM
    - SIP Definitions
    - Network

- Applications Enabling
- Media
- Control Network
- Coders and Profile
- GW and IP to IP

The configuration of the Mediant 1000 is performed via a Web browser. To access the device, enter the IP address of the gateway as the URL, then log in with the proper credentials. The main Mediant 1000 screen after login is shown below.

Note: *The AudioCodes GW may display "Trunk", "MPM" or both.*



7.1 System Settings

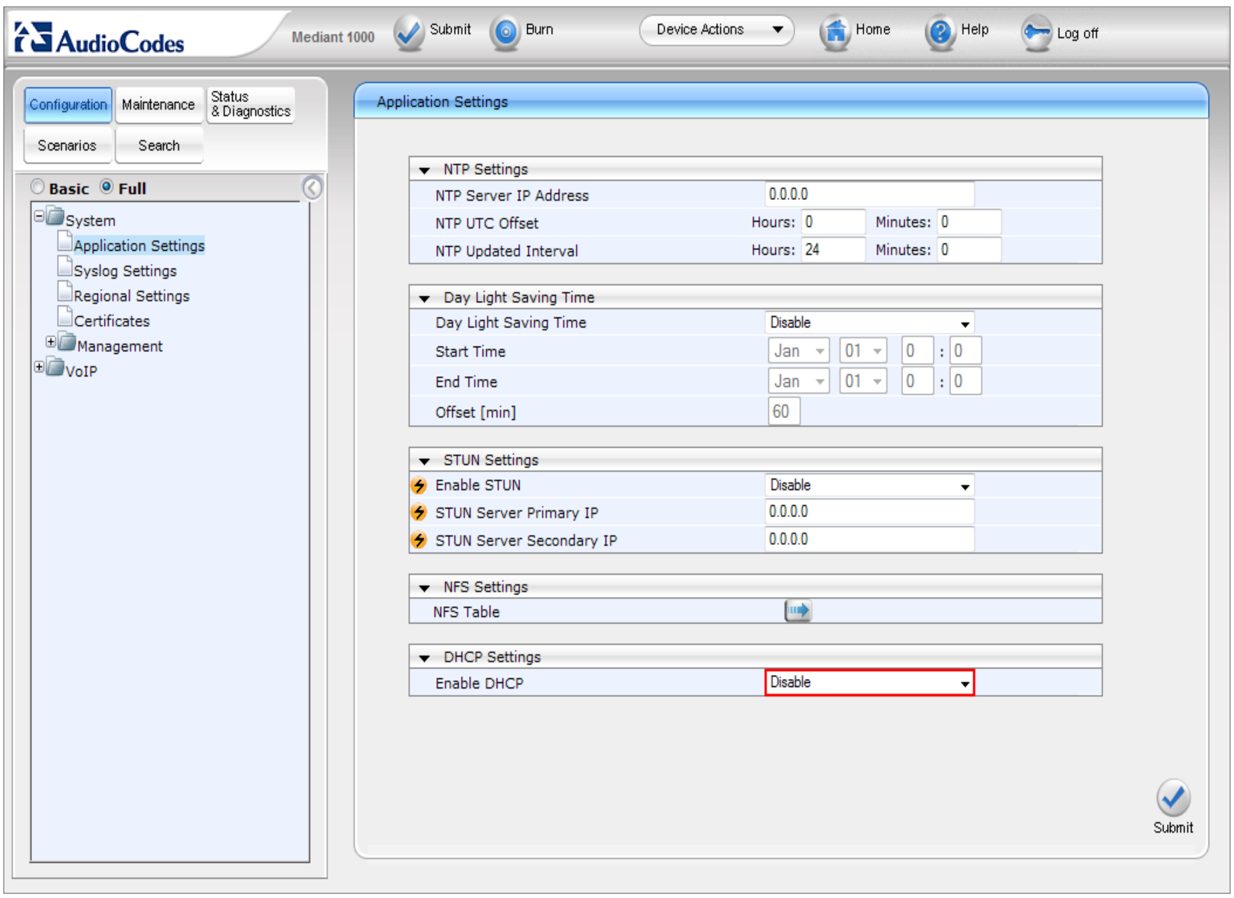
The system settings that were configured during installation can be viewed by navigating the **System** tree in the left pane. If necessary, changes can be made to the settings on these pages followed by clicking the **Submit** icon button at the bottom of the screen. For compliance testing, DHCP client was disabled, and Syslog was enabled.

7.1.1 Application Settings

7.1.1.1 Disable DHCP Client

Navigate to **System → Application Settings**. Configure the parameters as described below.

- For the **Enable DHCP** field, select ***Disable***.



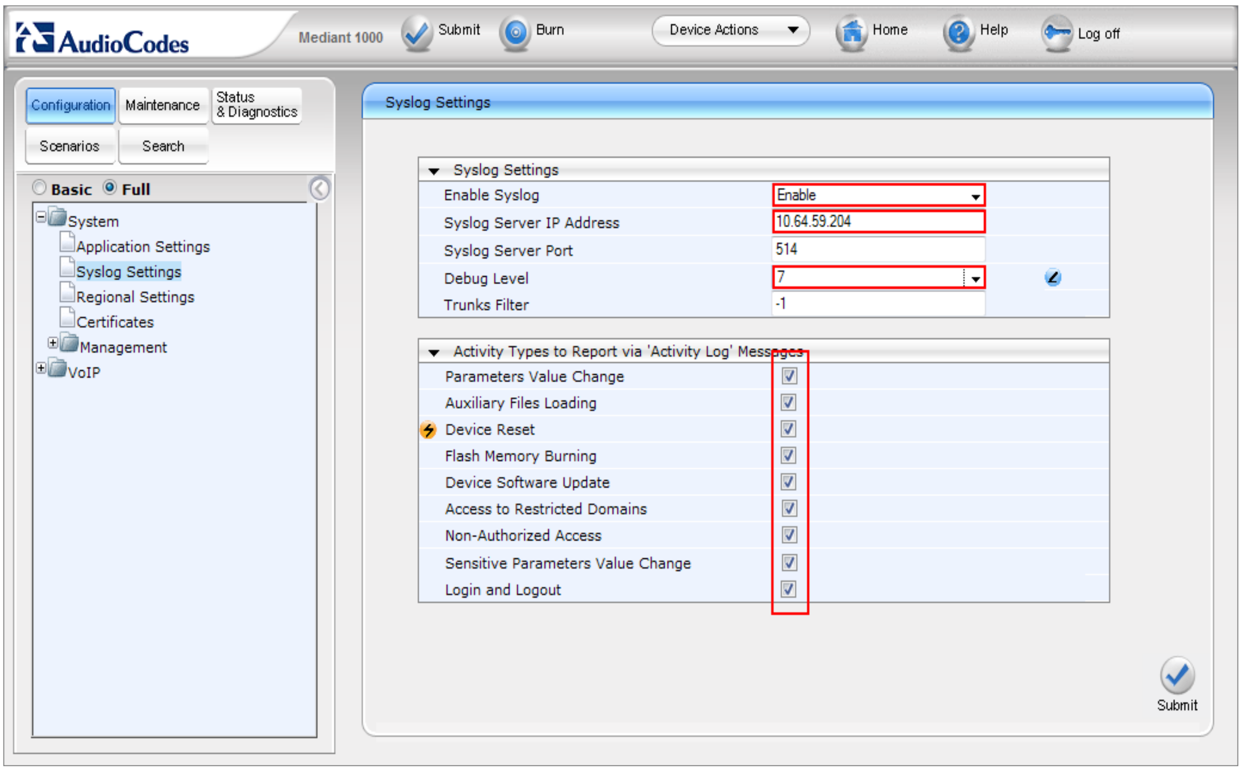
7.1.2 Syslog Settings

7.1.2.1 Enable Syslog

Navigate to **System → Syslog Settings**. Configure the parameters as described below.

- For the **Enable Syslog** field, select ***Enable***. Enabling Syslog is strongly recommended.

Note: Syslog server is required.



7.2 VoIP Settings

The VoIP settings that were configured during installation can be viewed by navigating the **VoIP** tree in the left pane. If necessary, changes can be made to the settings on these pages followed by clicking the **Submit** icon button at the bottom of the screen.

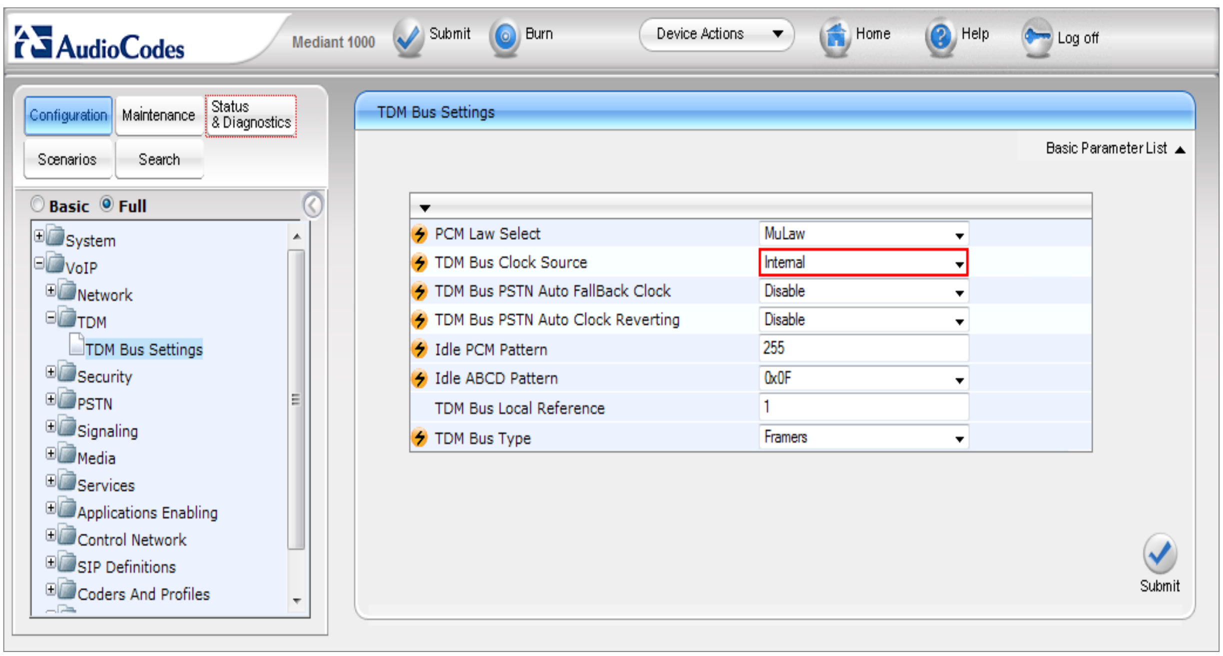
7.2.1 TDM

7.2.2 TDM Bus Settings

Navigate to **VoIP → TDM → TDM Bus Settings**. Configure the parameters as described below.

- For the **TDM Bus Clock Source** field, select ***Internal***.

Note: In the absence of TDM trunks, it is recommended that the TDM Bus Clock Source be configured as Internal.



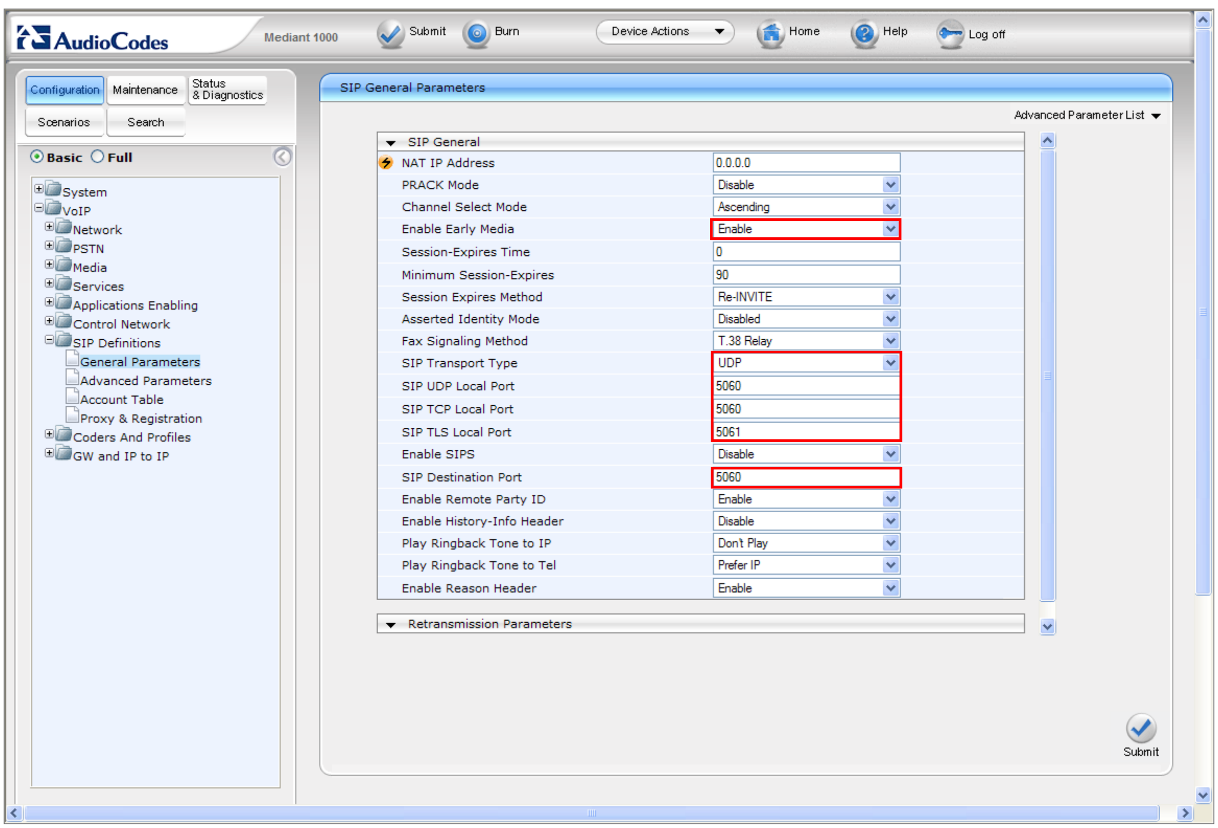
7.3 SIP Definitions

7.3.1 SIP General Parameters

Navigate to **VoIP → SIP Definitions → General Parameters**. Configure the parameters as described below.

- For the **Enable Early Media** field, select ***Enable***.
- Select ***UDP*** for the **SIP Transport Type** field.
- Verify the correct port numbers are set for **SIP UDP Local Port** (*5060*), **SIP TCP Local Port** (*5060*), **SIP TLS Local Port** (*5061*), **SIP Destination Port** (*5060*).

Default values may be retained for all other fields.

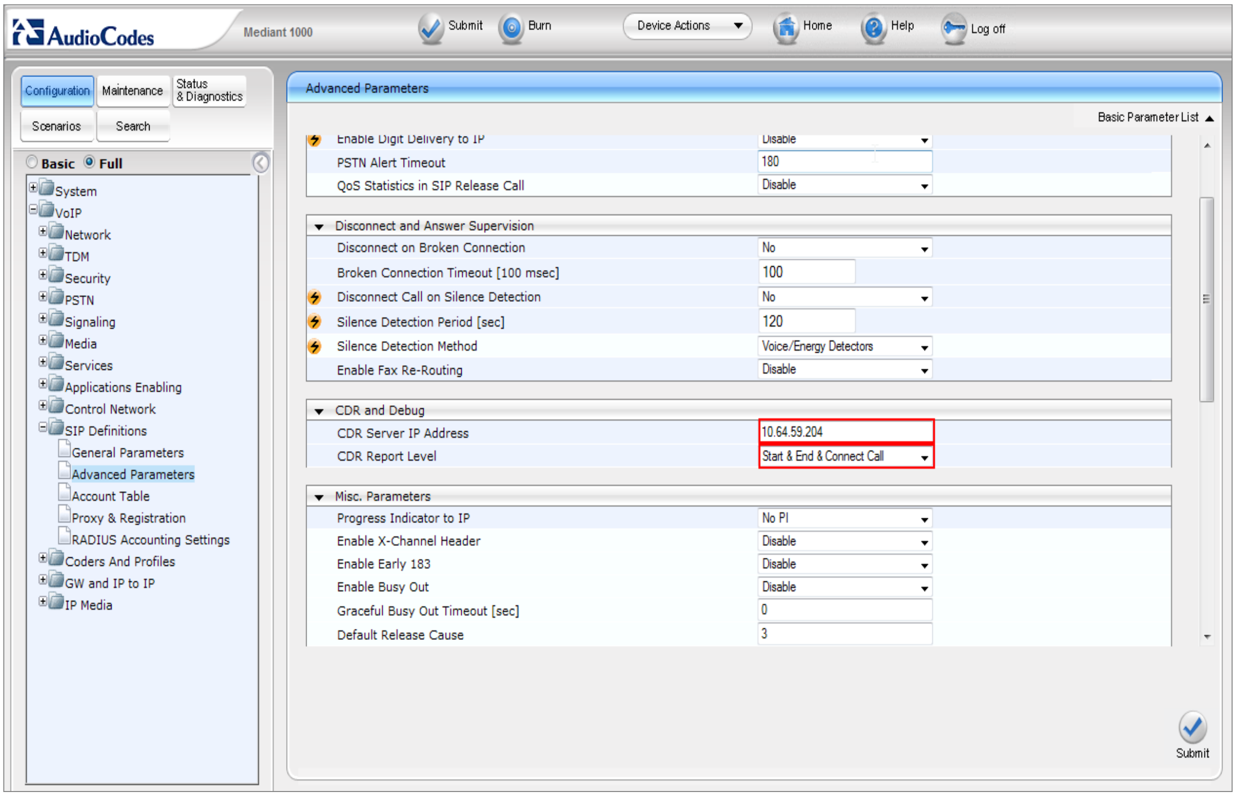


7.3.2 SIP Advanced Parameters

Navigate to **VoIP → SIP Definitions → Advanced Parameters**. Configure the parameters as described below.

- For the **CDR Server IP Address** field, enter the Server IP Address. This could be the syslog server.
- For the **CDR Report Level** field, select ***Start & End & Connect Call***.

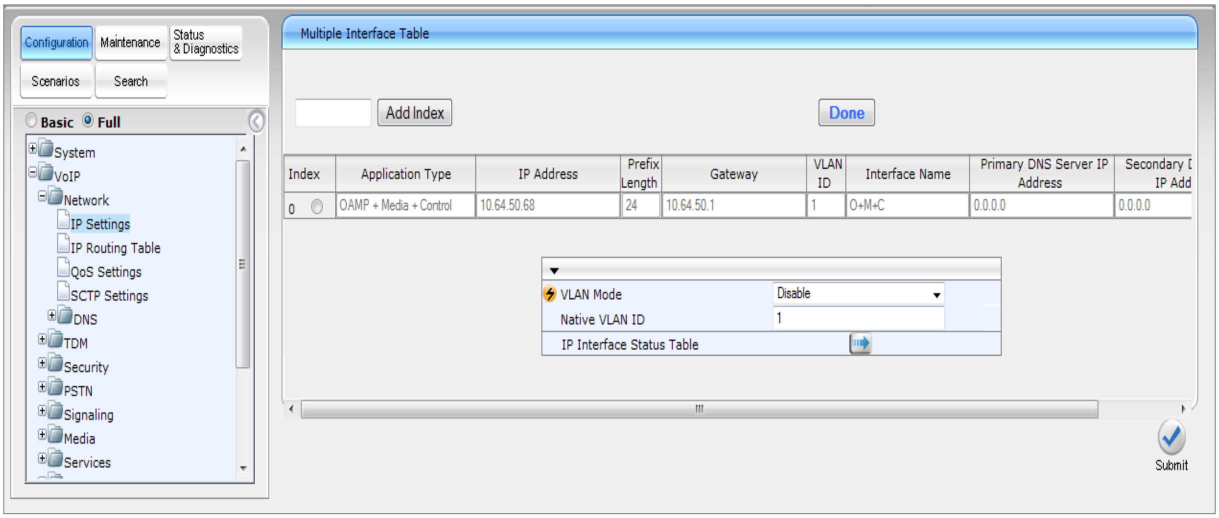
Default values may be retained for all other fields.



7.4 Network

7.4.1 IP Settings

The network settings that were configured during installation can be viewed by navigating to **VoIP →Network → IP Settings** in the left pane. If necessary, changes can be made to the settings on this page followed by clicking the **Submit** icon button at the bottom of the screen. For compliance testing, the **IP Address**, **Subnet Mask** and **Default Gateway Address** were set to values consistent with the test configuration shown in **Figure 1**.



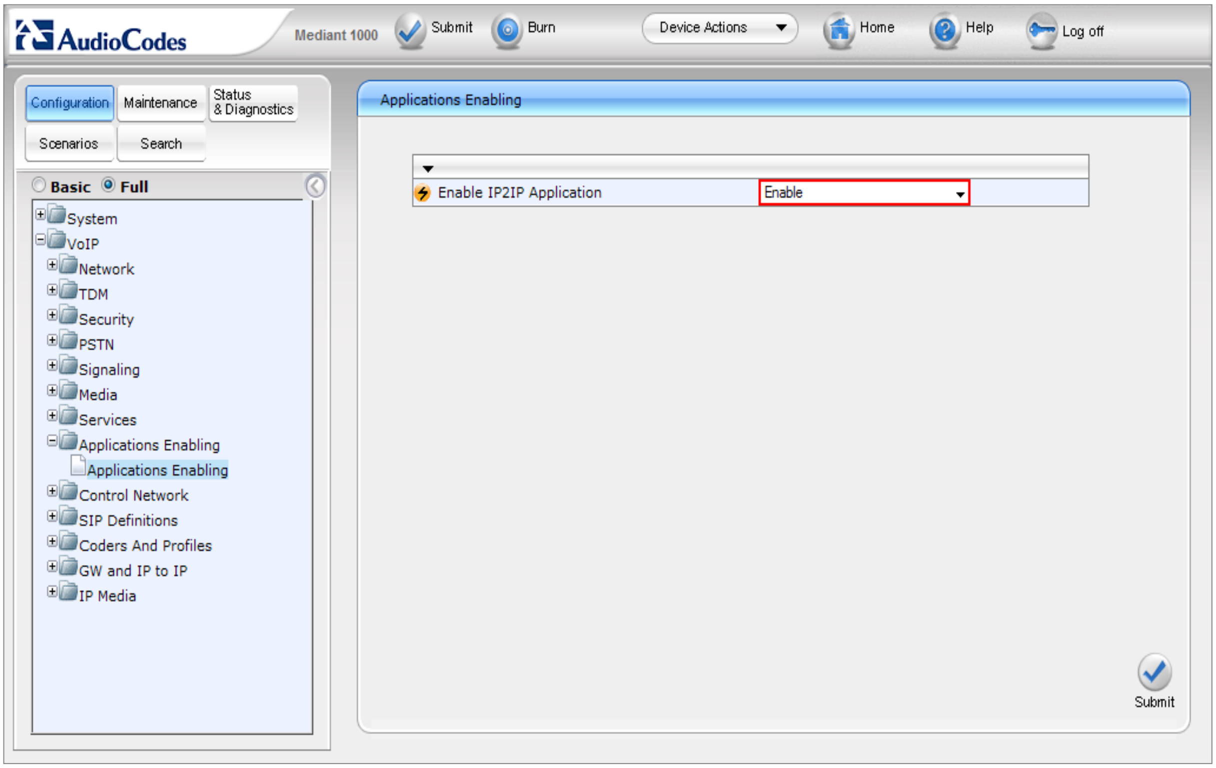
7.5 Applications Enabling

7.5.1.1 Applications Enabling

Navigate to **VoIP → Applications Enabling → Applications Enabling**. If this option is not available in the menu, please contact your re-seller or AudioCodes to obtain proper licensing. Configure the parameters as described below.

- For the **Enable IP2IP Application** field, select ***Enable***.

Note: Requires system reset



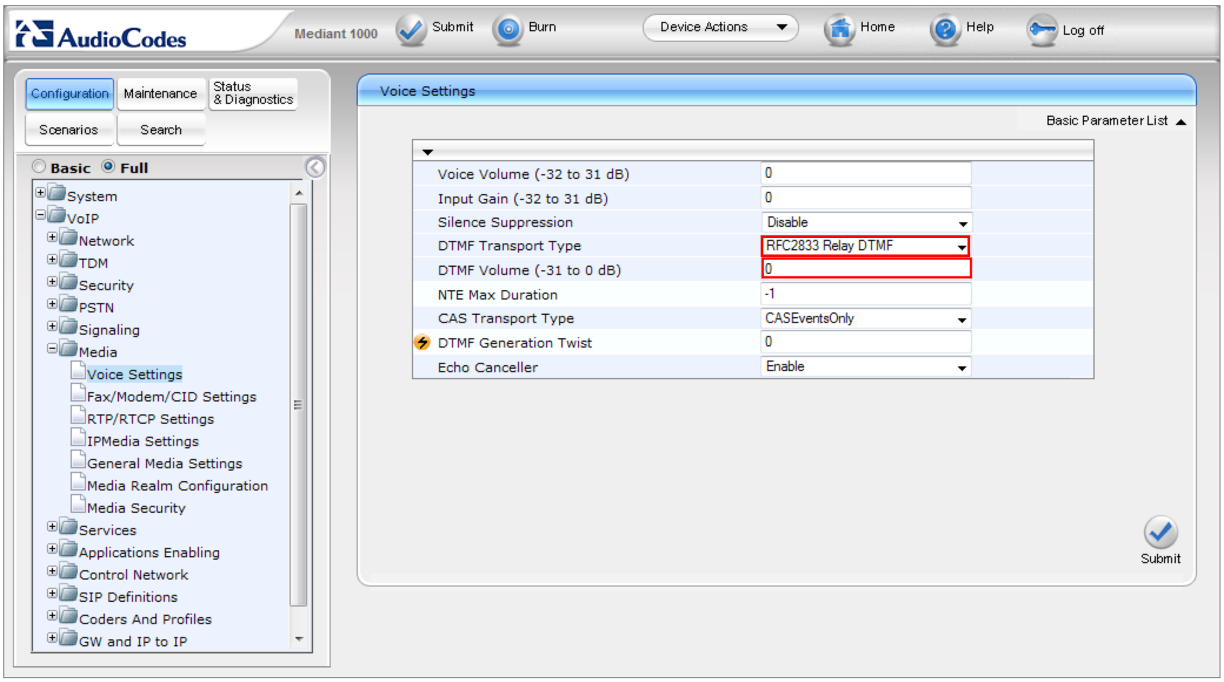
7.6 Media

7.6.1 Voice Settings

Navigate to **VoIP → Media → Voice Settings**. Configure the parameters as described below.

- For the **DTMF Transport Type** field, select ***RFC2833 Relay DTMF***.
- For the **DTMF Volume** field enter ***0***.

Default values may be retained for all other fields.

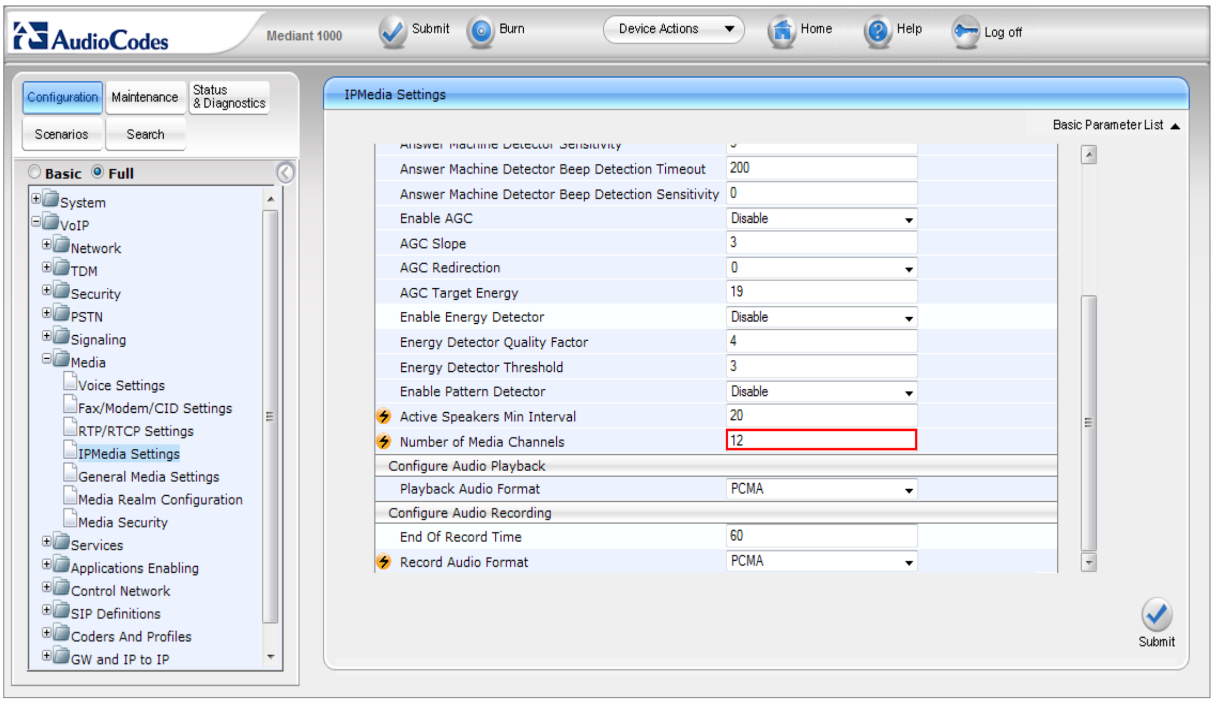


IP Media Settings

Navigate to **VoIP → Media → IPMedia Settings**. Configure the parameters as described below.

- For the **Number of Media Channels** field, enter the number of licensed channels you have for this device. For compliance testing there were ***12*** channels.

Default values may be retained for all other fields.
Note: Four channels are required for each MVS call.



7.7 Control Network

7.7.1 Proxy Sets Table

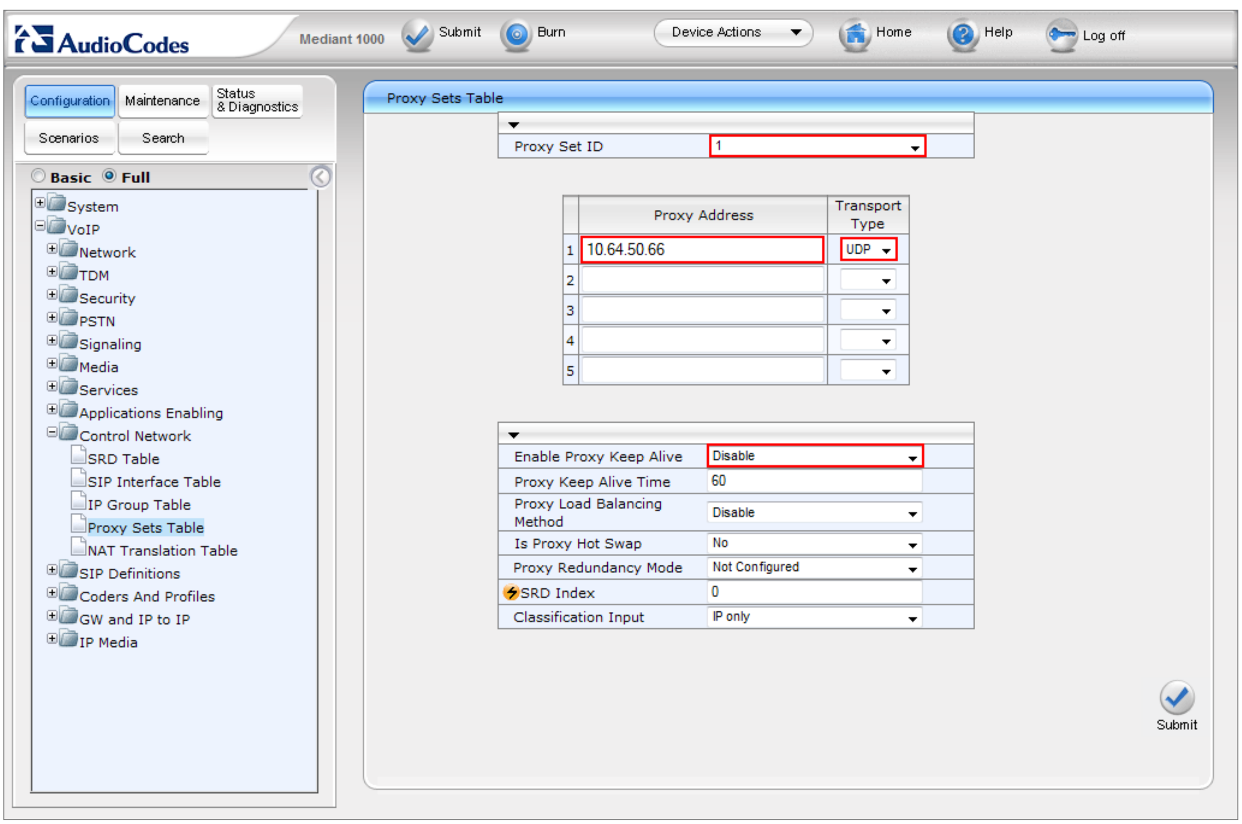
7.7.1.1 Proxy Sets Table BlackBerry® MVS

Navigate to **VoIP → Control Network → Proxy Sets Table** to configure proxy parameters for connecting to the BlackBerry® MVS.

Configure the parameters as described below.

- For the **Proxy Set ID** field select an **ID**. (Number 1 was used for the BlackBerry® MVS.)
- Enter the IP Address of the BlackBerry® MVS in the **Proxy Address** field. Select **UDP** for the **Transport Type**.
- For the **Enable Proxy Keep Alive** field select, **Disable**.

Default values may be retained for all other fields.



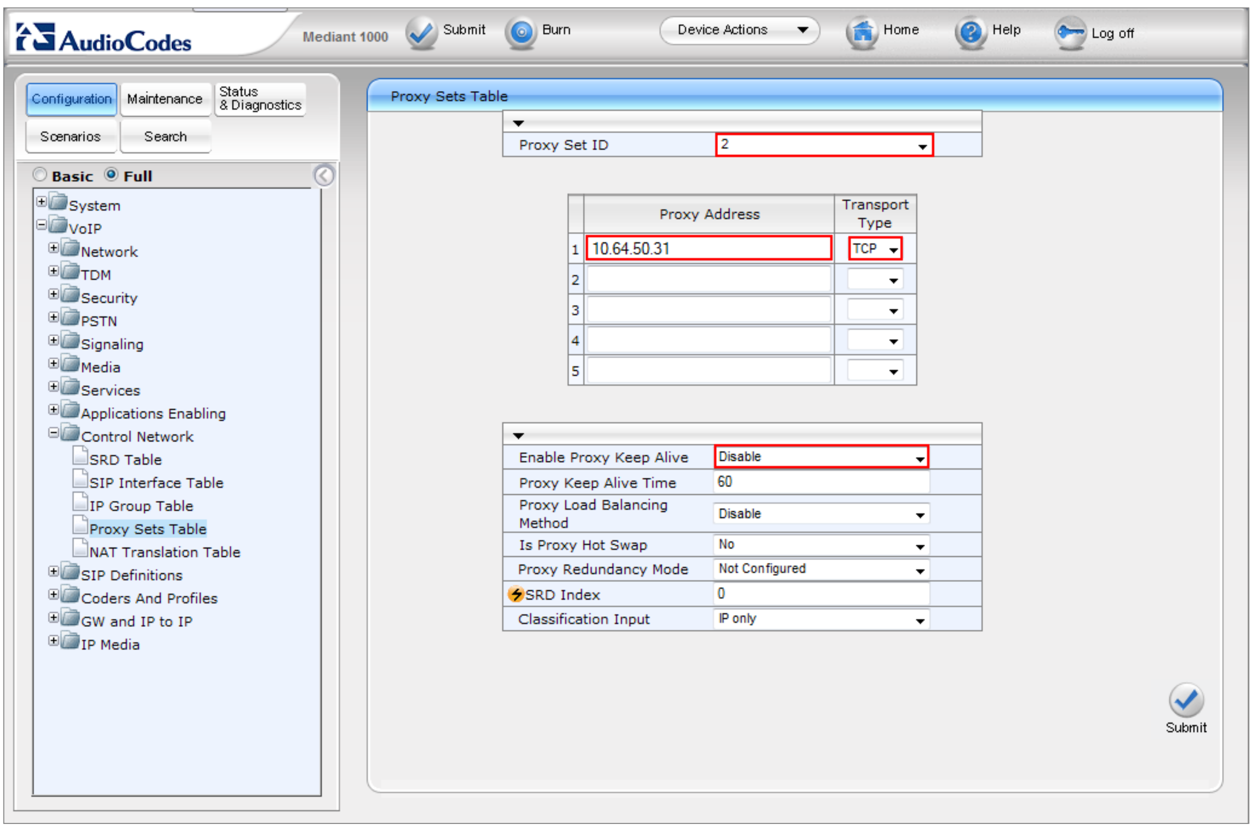
7.7.1.2 Proxy Sets Table Avaya Aura® Session Manager

Navigate to **VoIP → Control Network → Proxy Sets Table** to configure proxy parameters for connecting to the Avaya Aura® Session Manager.

Configure the parameters as described below.

- For the **Proxy Set ID** field select an **ID**. (Number 2 was used for the Avaya Aura® Session Manager.)
- Enter the IP Address of the Avaya Aura® Session Manager in the **Proxy Address** field. Select **TCP** for the **Transport Type**.

For the **Enable Proxy Keep Alive** field select, **Disable**.



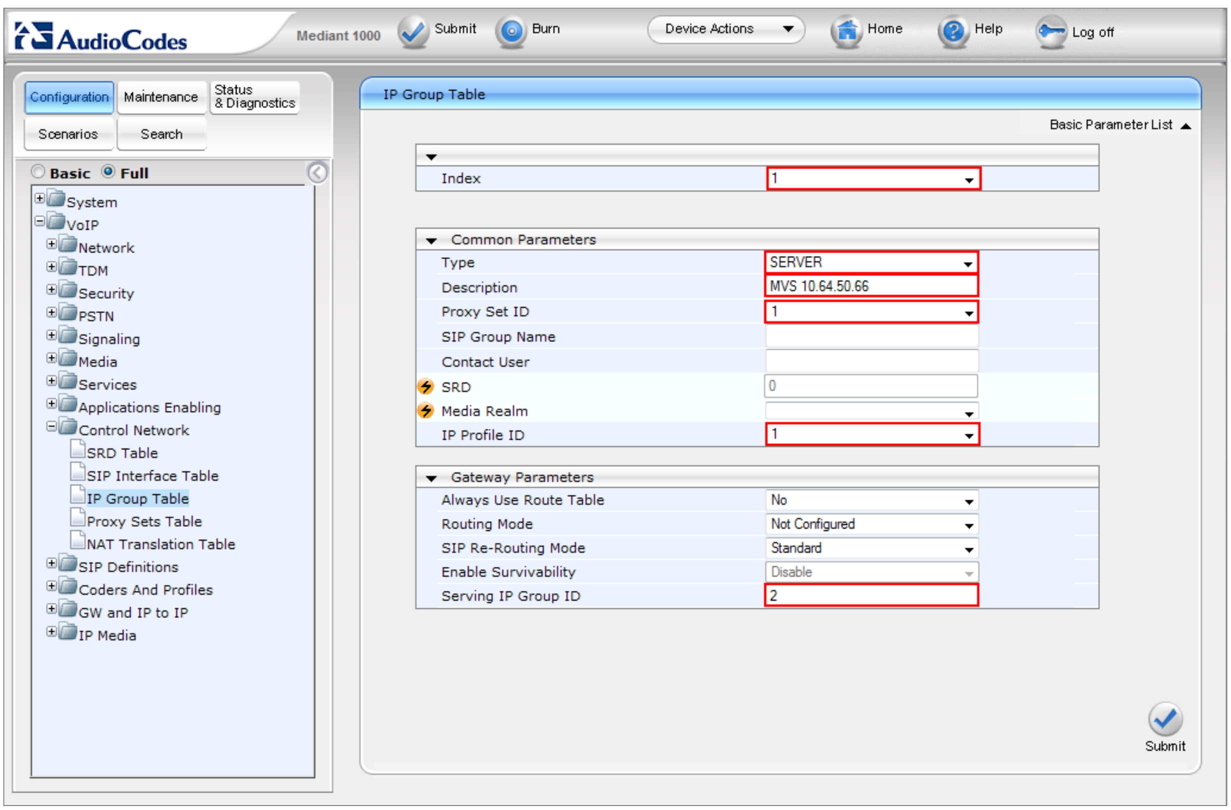
7.7.2 IP Group Table

Navigate to **Control Network → IP Group Table** to configure session with the **MVS Server**.

Configure the following parameters.

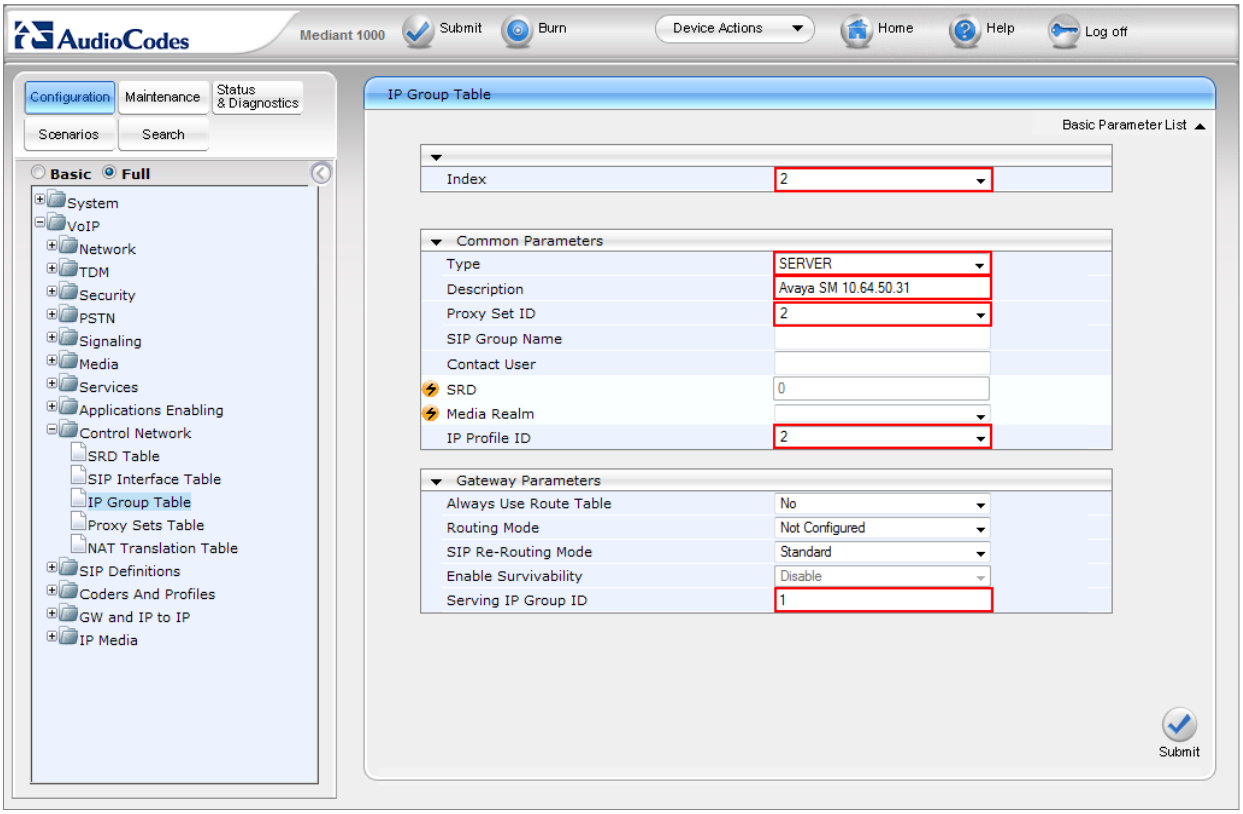
- For the **Index** field, select an ***Index*** number.
- For the **Type** field, select a ***SERVER***.
- For the **Description** field, enter a meaningful description. This is an informational parameter only.
- For the **Proxy Set ID** field, select the number that corresponds to the **Proxy Sets Table** configured in **Section 7.7.1.**
- For the **IP Profile ID** field, select profile number 1.
- For the **Serving IP Group ID** field, enter the number **2**.

Default values may be retained for all other fields. Click the **Submit** button at the bottom of the screen.



Repeat the configuration for the Avaya Aura® Session Manager.

Note: *In this configuration Index 1 uses Proxy Set ID 1, and IP Profile 1, while the Gateway Parameters refer to Serving Group ID 2. Serving Group ID is configured is Section 7.9.2.2*

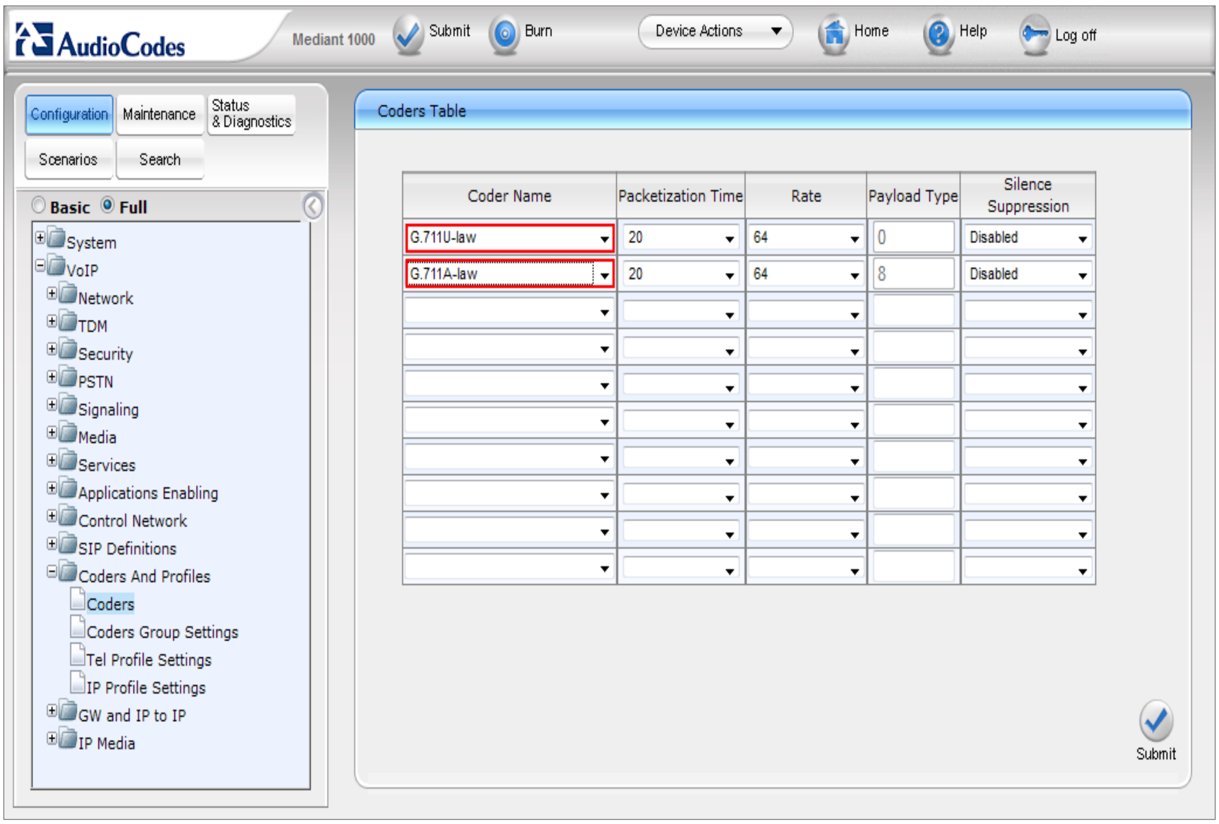

AudioCodes
Mediant 1000
Submit
Burn
Device Actions
Home
Help
Log off
Configuration
Maintenance
Status & Diagnostics
Scenarios
Search
Basic
Full
System
VoIP
Network
TDM
Security
PSTN
Signaling
Media
Services
Applications Enabling
Control Network
SRD Table
SIP Interface Table
IP Group Table
Proxy Sets Table
NAT Translation Table
SIP Definitions
Coders And Profiles
GW and IP to IP
IP Media
IP Group Table
Basic Parameter List
Index 2
Common Parameters
Type SERVER
Description Avaya SM 10.64.50.31
Proxy Set ID 2
SIP Group Name
Contact User
SRD 0
Media Realm
IP Profile ID 2
Gateway Parameters
Always Use Route Table No
Routing Mode Not Configured
SIP Re-Routing Mode Standard
Enable Survivability Disable
Serving IP Group ID 1
Submit

7.8 Coders and Profiles

7.8.1 Coders

Navigate to **VoIP → Coders and Profiles → Coders**. In the screen below, select the list of preferred codecs to be used by the Mediant 1000 with the most preferred codec at the top and working downward to the least preferred.

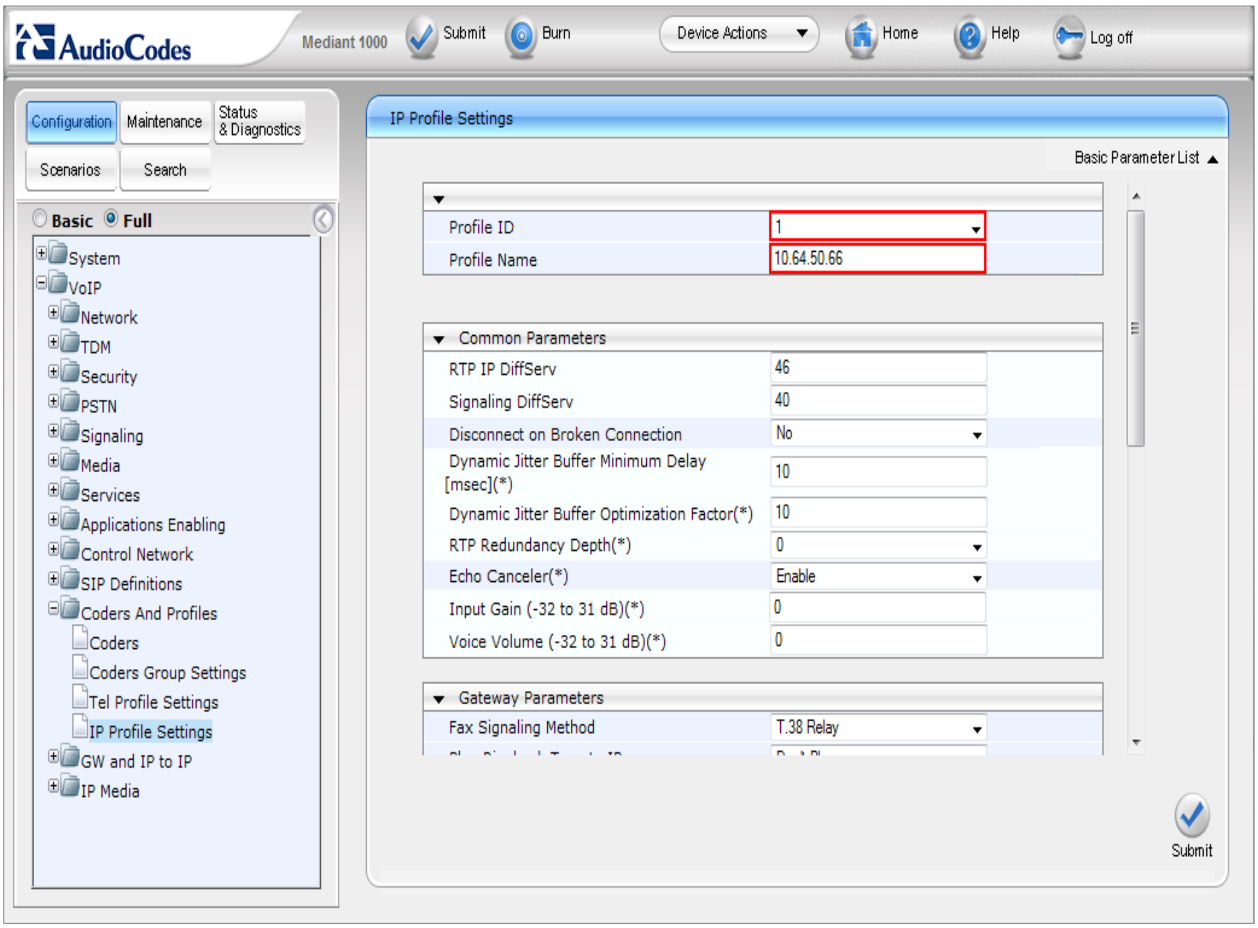
The AudioCodes Mediant 1000 supports multiple codecs, however, during compliance testing ***G.711U-law*** was selected as the most preferred codec. Default values were retained for all other fields.

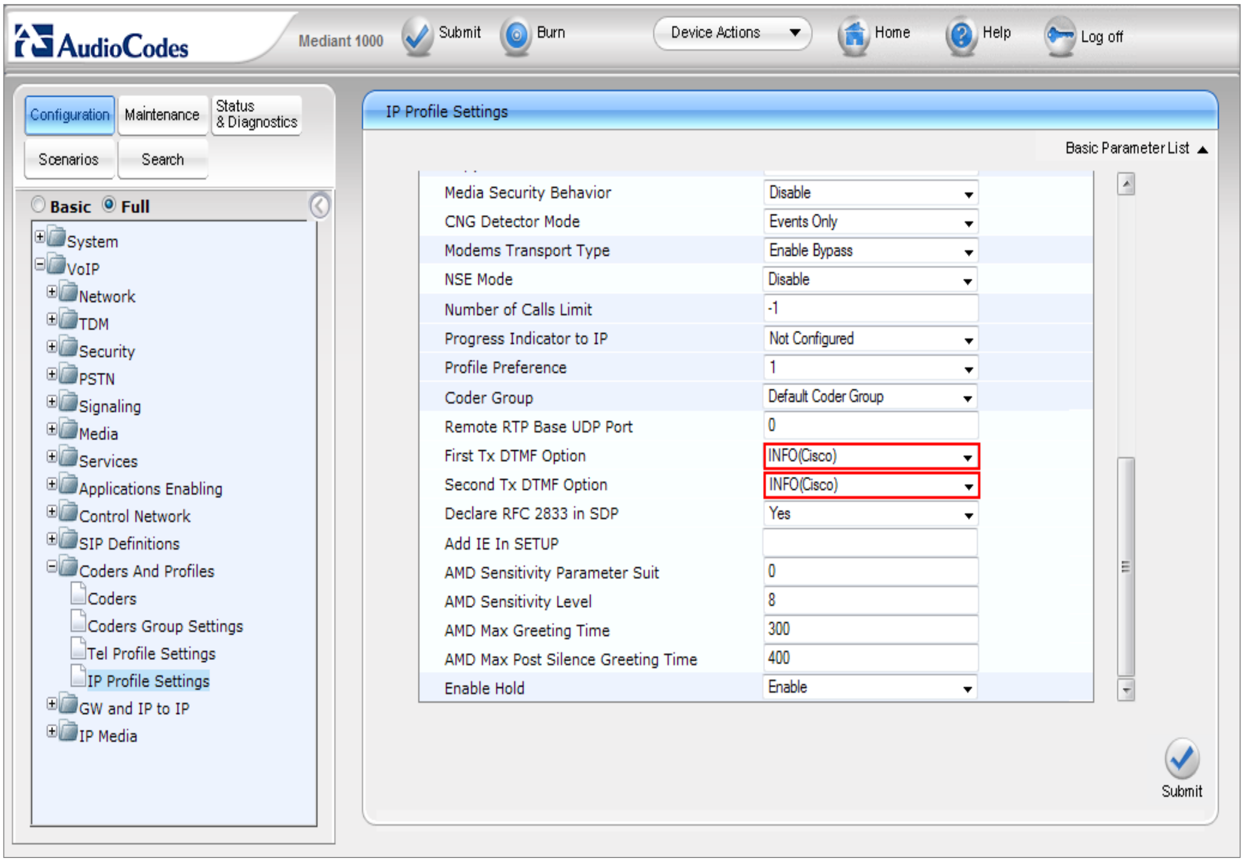


7.8.2 IP Profiles

Navigate to **VoIP → Coders and Profiles → IP Profile Settings**. The following parameters are used between the AudioCodes Mediant 100 and the BlackBerry® MVS Server. In the screen below, select a **Profile ID** and enter a **Profile Name.** Scroll down the page and configure the **First** and **Second Tx DTMF Options** as **INFO(Cisco)**.

Default values were retained for all other fields.

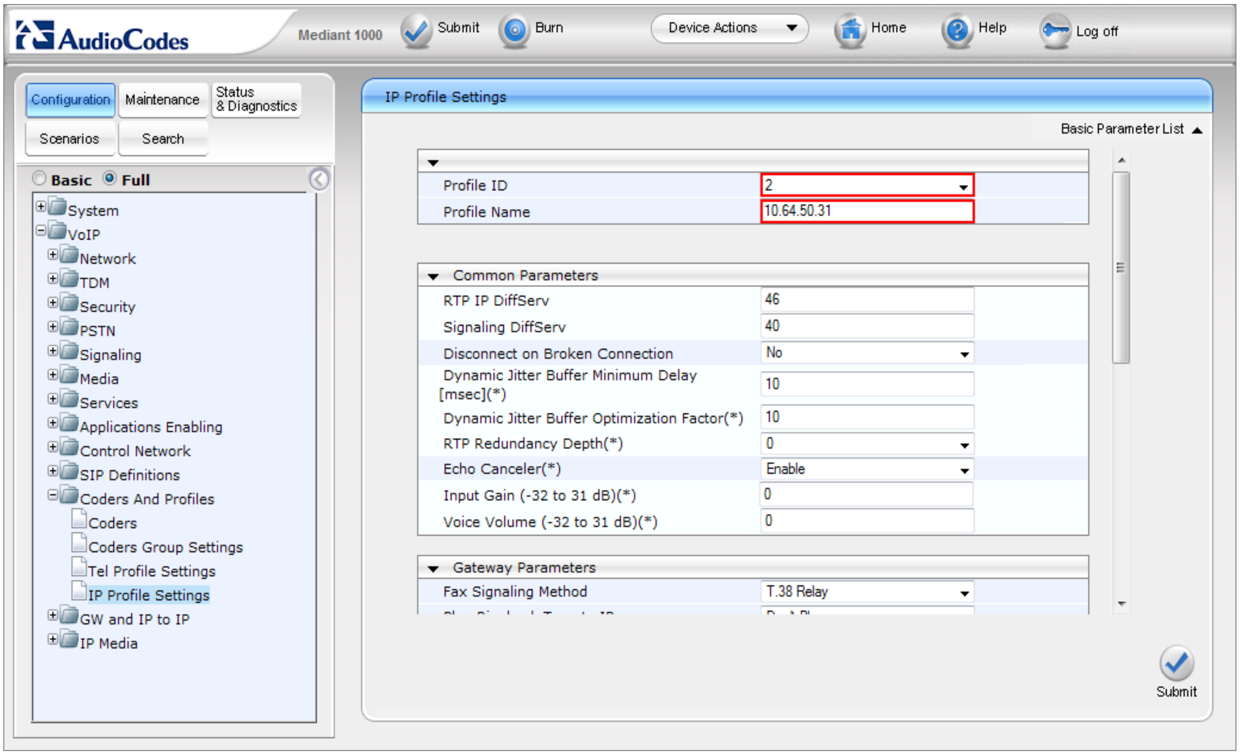


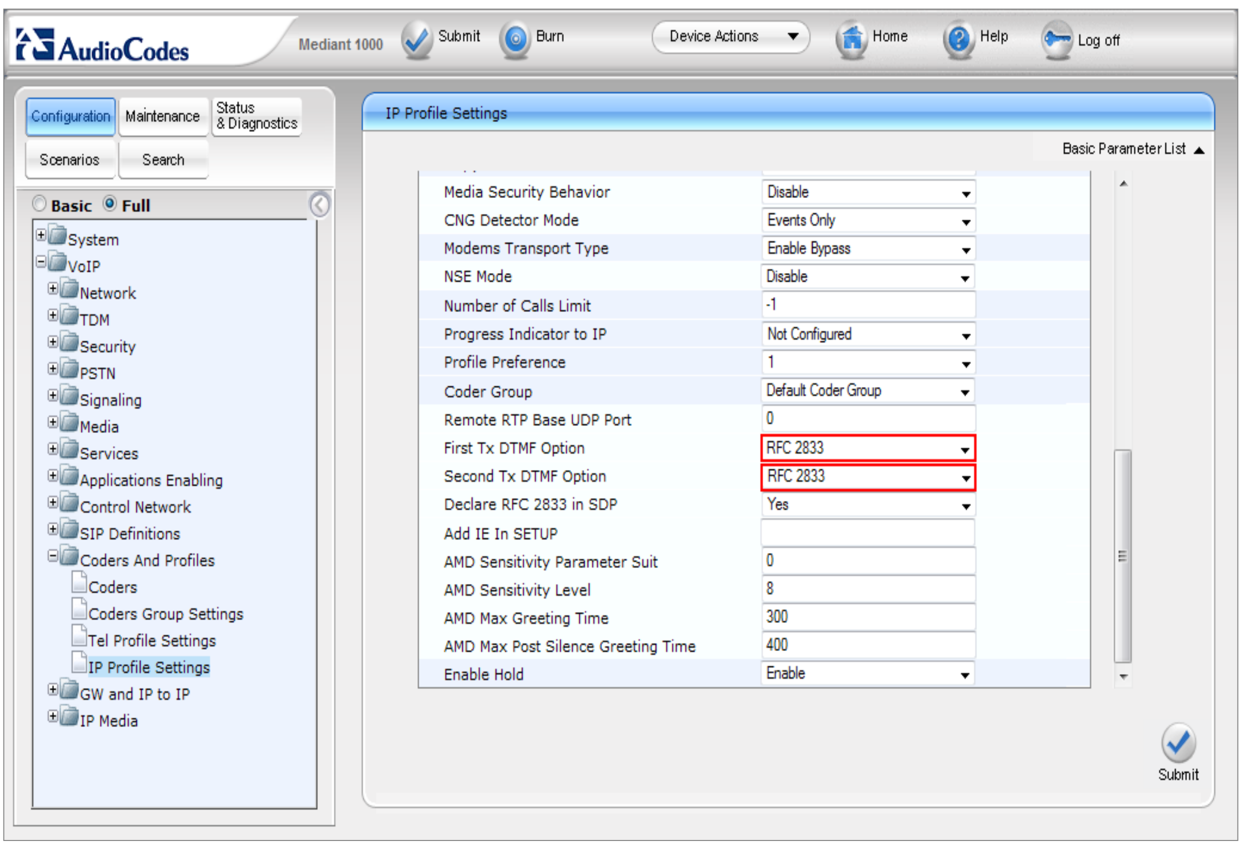


Repeat the configuration for the Avaya Aura® Session Manager.

Navigate to **VoIP → Coders and Profiles → IP Profile Settings**. The following parameters are used between the AudioCodes Mediant 100 and the Avaya Aura® Session Manager. In the screen below, select a **Profile ID** and enter a **Profile Name.** Scroll down the page and configure the **First** and **Second Tx DTMF Options** as **RFC 2833**.

Default values were retained for all other fields.


AudioCodes
Mediant 1000
Submit
Burn
Device Actions
Home
Help
Log off
Configuration
Maintenance
Status & Diagnostics
Scenarios
Search
Basic
Full
System
VoIP
Network
TDM
Security
PSTN
Signaling
Media
Services
Applications Enabling
Control Network
SIP Definitions
Coders And Profiles
Coders
Coders Group Settings
Tel Profile Settings
IP Profile Settings
GW and IP to IP
IP Media
IP Profile Settings
Basic Parameter List
Profile ID
2
Profile Name
10.64.50.31
Common Parameters
RTP IP DiffServ
46
Signaling DiffServ
40
Disconnect on Broken Connection
No
Dynamic Jitter Buffer Minimum Delay [msec](*)
10
Dynamic Jitter Buffer Optimization Factor(*)
10
RTP Redundancy Depth(*)
0
Echo Canceler(*)
Enable
Input Gain (-32 to 31 dB)(*)
0
Voice Volume (-32 to 31 dB)(*)
0
Gateway Parameters
Fax Signaling Method
T.38 Relay
Submit

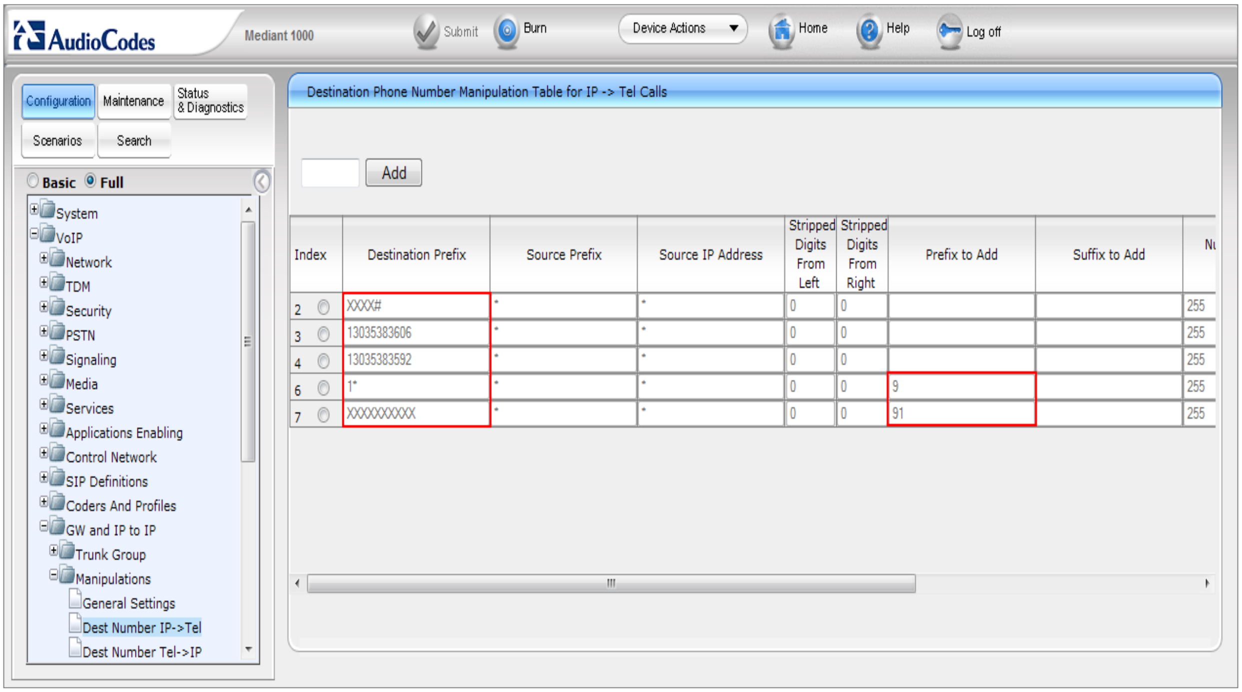

AudioCodes
Mediant 1000
Submit
Burn
Device Actions
Home
Help
Log off
Configuration
Maintenance
Status & Diagnostics
Scenarios
Search
Basic
Full
System
VoIP
Network
TDM
Security
PSTN
Signaling
Media
Services
Applications Enabling
Control Network
SIP Definitions
Coders And Profiles
Coders
Coders Group Settings
Tel Profile Settings
IP Profile Settings
GW and IP to IP
IP Media
IP Profile Settings
Basic Parameter List
Media Security Behavior
Disable
CNG Detector Mode
Events Only
Modems Transport Type
Enable Bypass
NSE Mode
Disable
Number of Calls Limit
-1
Progress Indicator to IP
Not Configured
Profile Preference
1
Coder Group
Default Coder Group
Remote RTP Base UDP Port
0
First Tx DTMF Option
RFC 2833
Second Tx DTMF Option
RFC 2833
Declare RFC 2833 in SDP
Yes
Add IE In SETUP
AMD Sensitivity Parameter Suit
0
AMD Sensitivity Level
8
AMD Max Greeting Time
300
AMD Max Post Silence Greeting Time
400
Enable Hold
Enable
Submit

7.9 GW and IP to IP

7.9.1 Manipulation Tables

7.9.1.1 Dest Number IP to Tel

Navigate to **VoIP → GW and IP to IP → Dest Number IP > Tel.** These configurations are based on the length of the extensions. The following table displays North American examples of using **9** as the trunk access code**.**

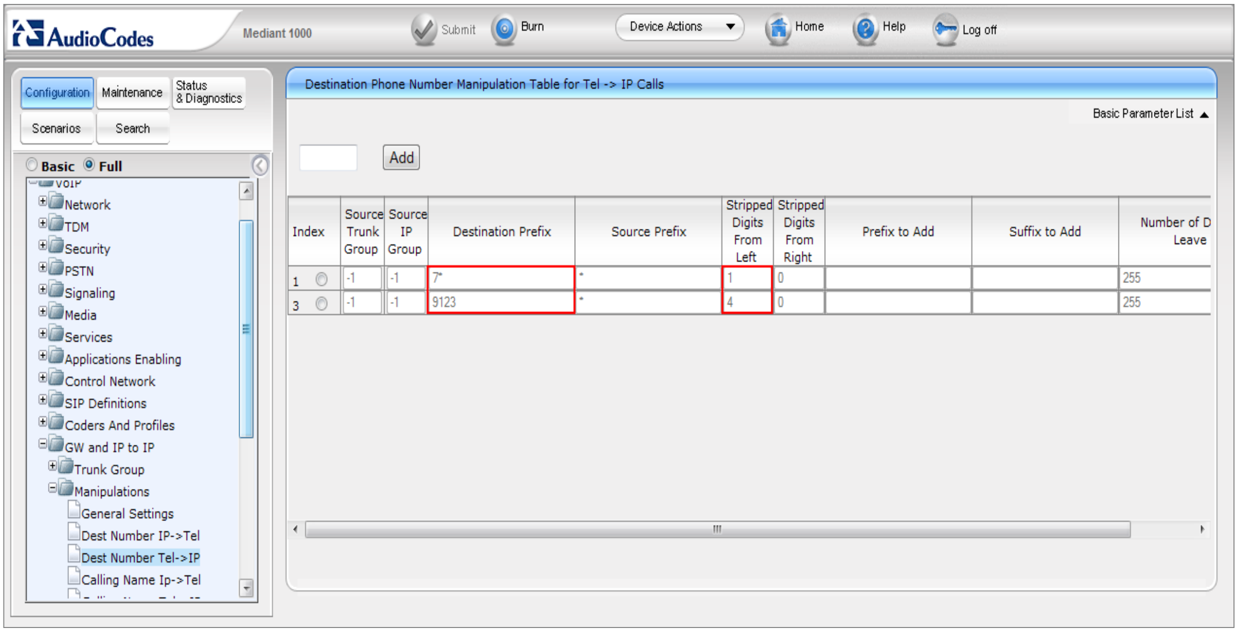


Index	Destination Prefix	Source Prefix	Source IP Address	Stripped Digits From Left	Stripped Digits From Right	Prefix to Add	Suffix to Add	Nu
2	XXXX#	*	*	0	0			255
3	13035383606	*	*	0	0			255
4	13035383592	*	*	0	0			255
6	1*	*	*	0	0	9		255
7	XXXXXXXXXX	*	*	0	0	91		255

Scroll to the right to see the remaining fields.

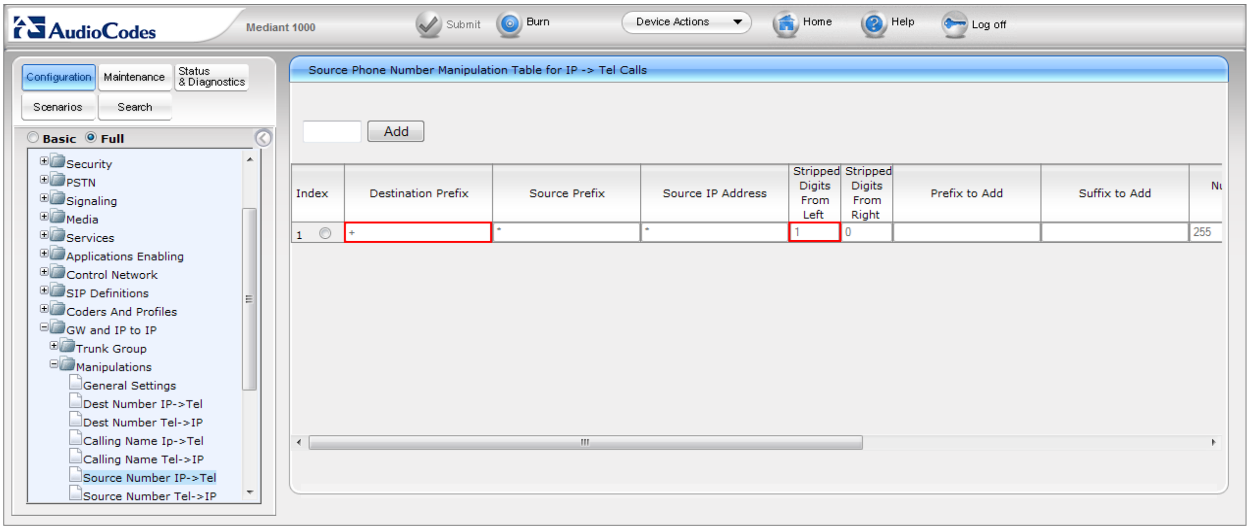
7.9.1.2 Dest Number Tel to IP

Navigate to **VoIP → GW and IP to IP → Dest Number Tel > IP**. This table is used to configure the ANI and DID/DDI numbers that are configured for the BlackBerry® device initiated or PBX initiated calling. (Consult MVS configuration screenshots for more info.)



7.9.1.3 Source Number IP to Tel

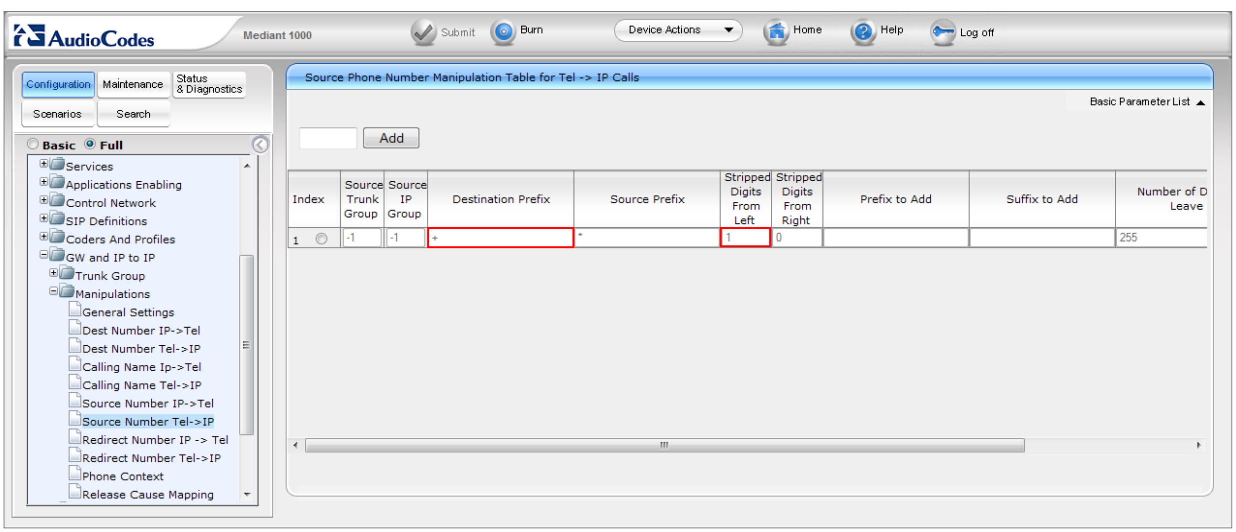
Navigate to **VoIP → GW and IP to IP → Source Number IP > Tel.** This configuration strips a single digit from the left of the calling number to any destination for incoming calls to the Mediant 1000.



Scroll to the right to see the remaining fields.

7.9.1.4 Source Number Tel to IP

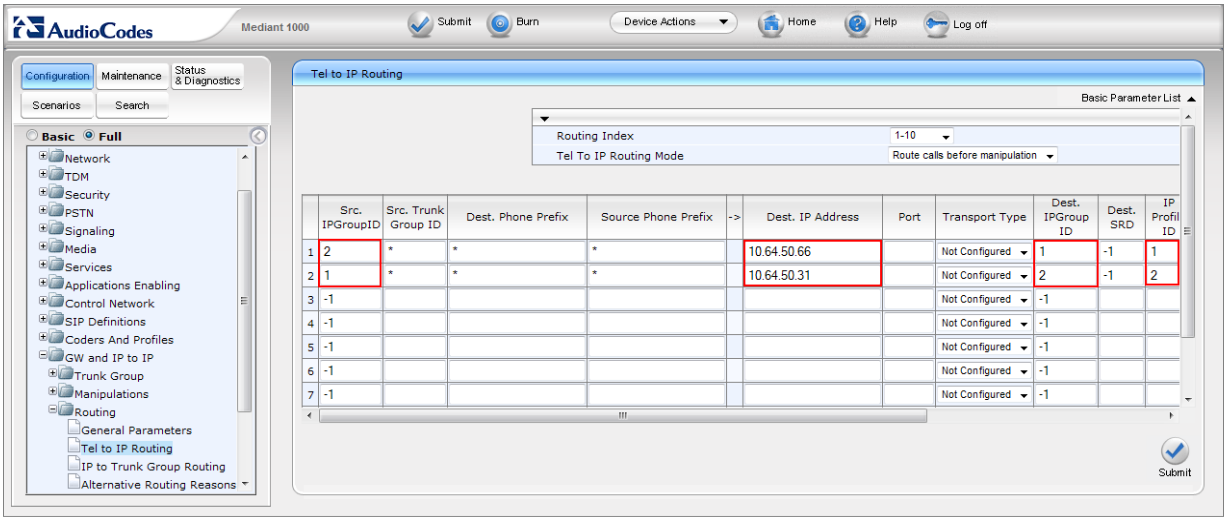
Navigate to **VoIP → GW and IP to IP → Source Number Tel > IP**. This configuration strips a single digit from the left of the calling number to any destination for outgoing calls from the Mediant 1000. The value of "-1" indicates that the field should be ignored for this entry.



7.9.2 Routing

7.9.2.1 Tel to IP Routing

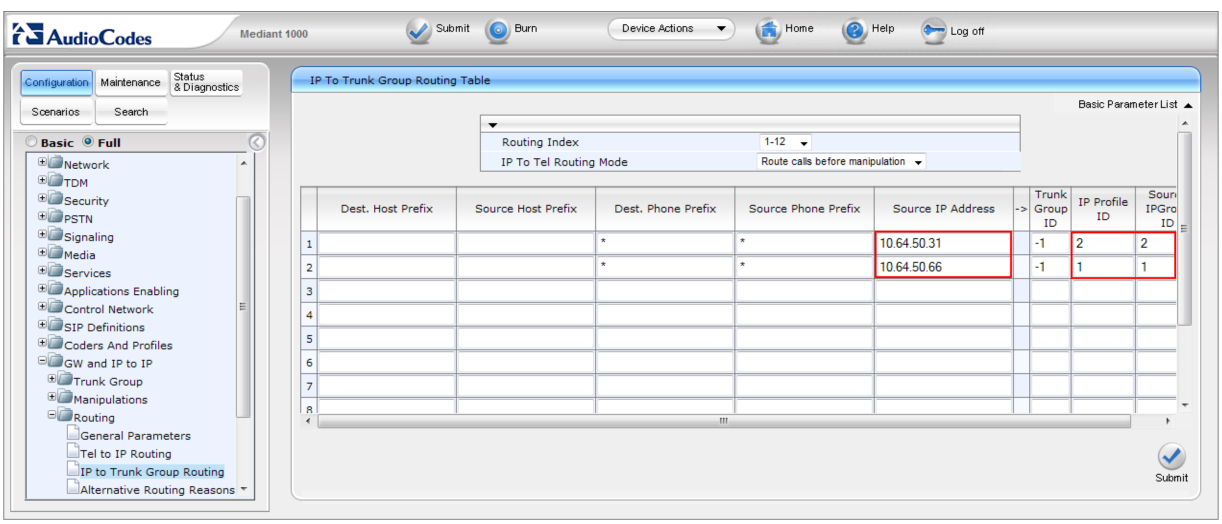
Navigate to **VoIP → GW and IP to IP → Routing → Tel to IP Routing**. Use these settings to routes phone calls to the BlackBerry® MVS Server.



Scroll to the right to see the remaining fields.

7.9.2.2 IP to Trunk Group Routing

Navigate to **VoIP → GW and IP to IP → Routing → IP to Trunk Group Routing**. The settings are used to route the calls to the BlackBerry® MVS Server.



Scroll to the right to see the remaining fields.

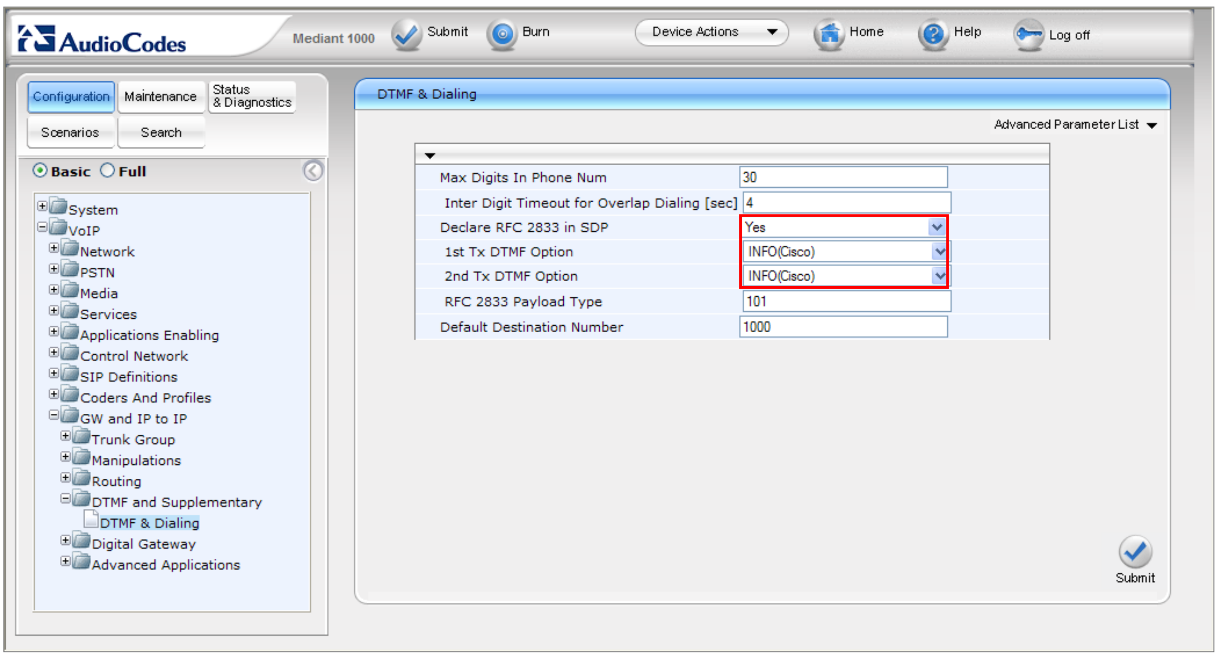
7.9.3 DTMF and Supplementary

7.9.3.1 DTMF and Dialing

Navigate to **VoIP → GW and IP to IP → DTMF and Supplementary → DTMF & Dialing**. Configure the parameters as described below.

- For the **Declare RFC 2833 in SDP** field, select ***Yes***.
- For the **1st Tx DTMF Option** field, select ***INFO (Cisco)***.
- For the **2nd Tx DTMF Option** field, select ***INFO (Cisco)***.

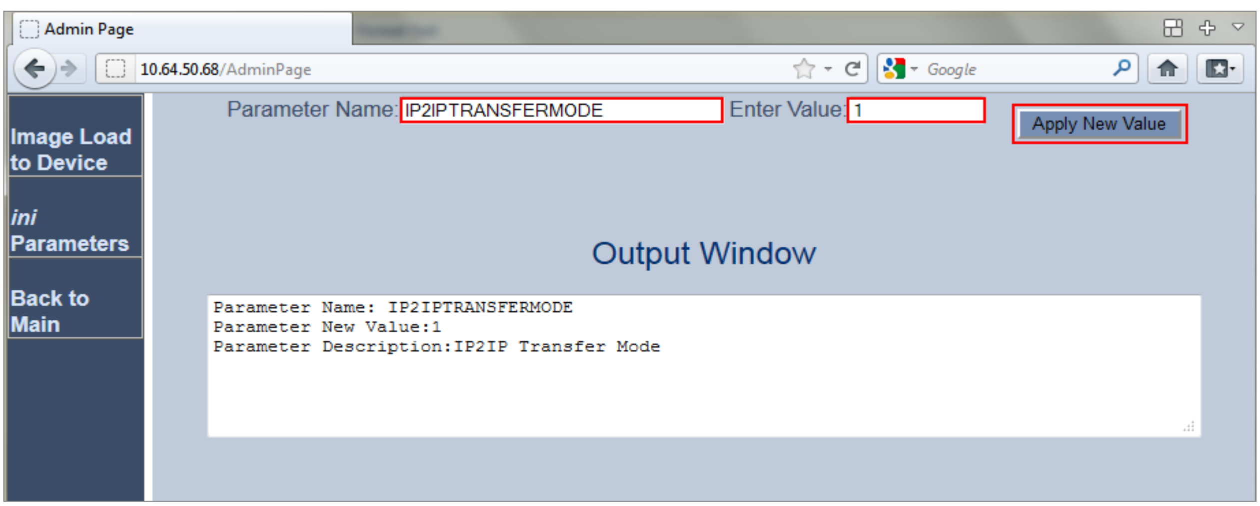
Default values may be retained for all other fields.



7.10 Additional settings for the ini file

In the AudioCodes UI (<IP address of AudioCodes gateway/AdminPage>), add the following entries for supporting DTMF on hold and IP2IP transfers. (The Parameter Names are not case sensitive).

- **PLAYDTMFDURINGHOLD 1**
- **IP2IPTRANSFERMODE 1**

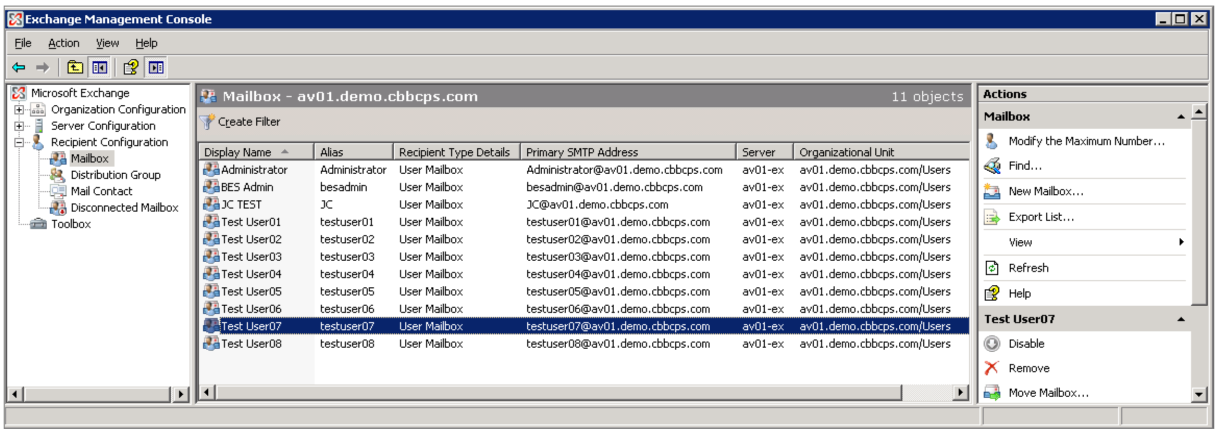


8 Research in Motion Mobile Voice System Configuration

This section describes the configuration of the RIM Mobile Voice System which involves the configuration of the Blackberry® Enterprise Server, and the Blackberry® MVS Server. As part of the compliance test, MVS users were imported from the corporate directory maintained on a Microsoft Exchange Server.

8.1 Microsoft Exchange Server

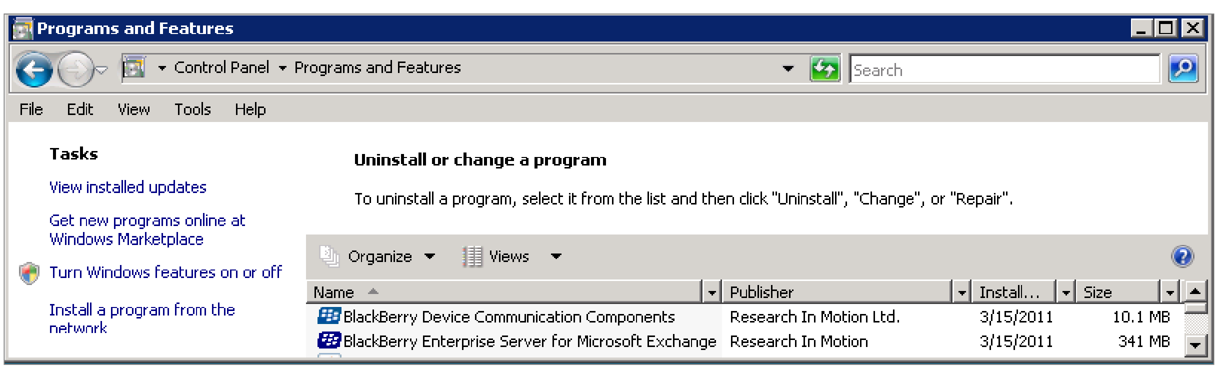
Verify that each user that will be configured on the MVS is also configured as a user within Microsoft Exchange. Log in to the Exchange Management Console. Navigate to **Microsoft Exchange → Recipient Configuration → Mailbox** to view the existing users.



8.2 BlackBerry® Enterprise Server Configuration

8.2.1 Verify Software Version

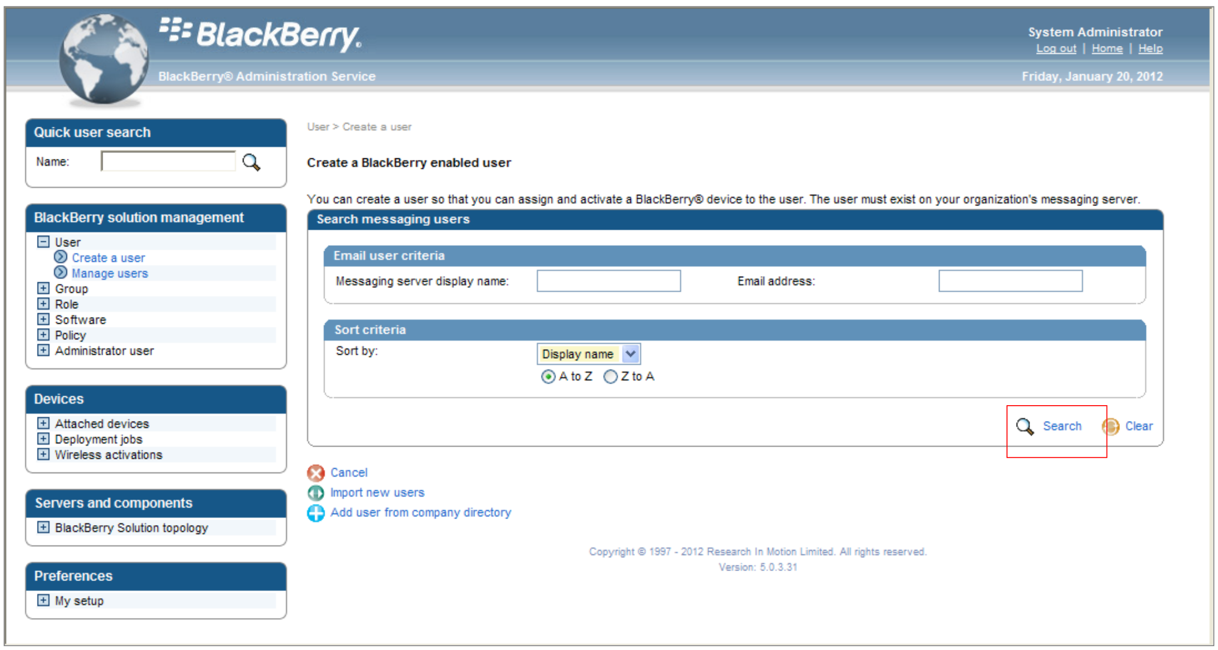
Log in to the Blackberry® Enterprise Server running on Microsoft Windows 2008 Server. From the Windows menus, navigate to **Control Panel → Programs and Features**. A list of installed programs will be displayed. Verify that ***Blackberry Enterprise Server for Microsoft Exchange*** is listed along with the appropriate software version.



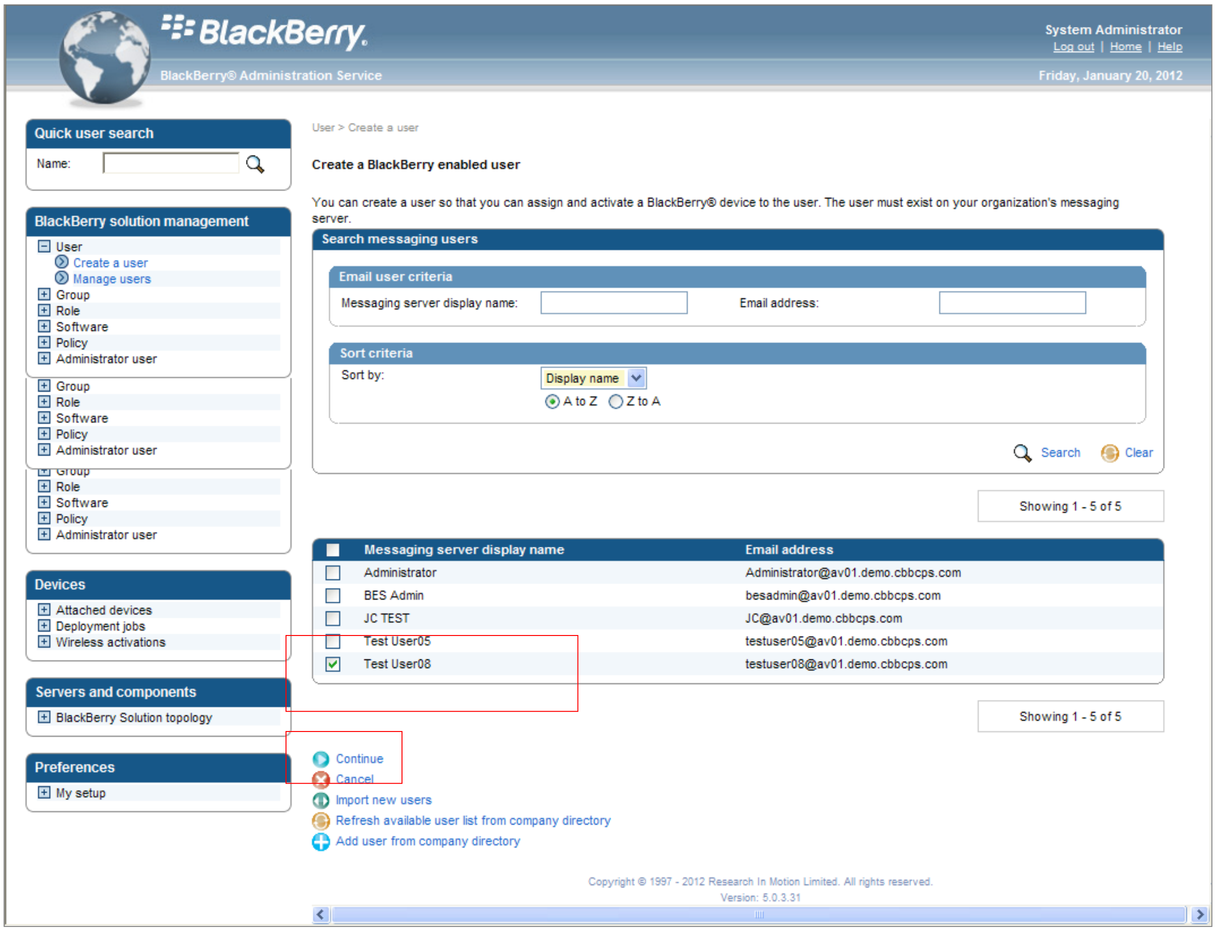
8.2.2 Create Users

A Blackberry® Enterprise Server user account must be created for each user of the MVS. To configure the Blackberry® Enterprise Server, launch the Blackberry® Administration Service by clicking the Blackberry® Administration Service icon on the Windows desktop. Each Blackberry® Administration Service window contains a menu on the left used to access and manage the necessary components.

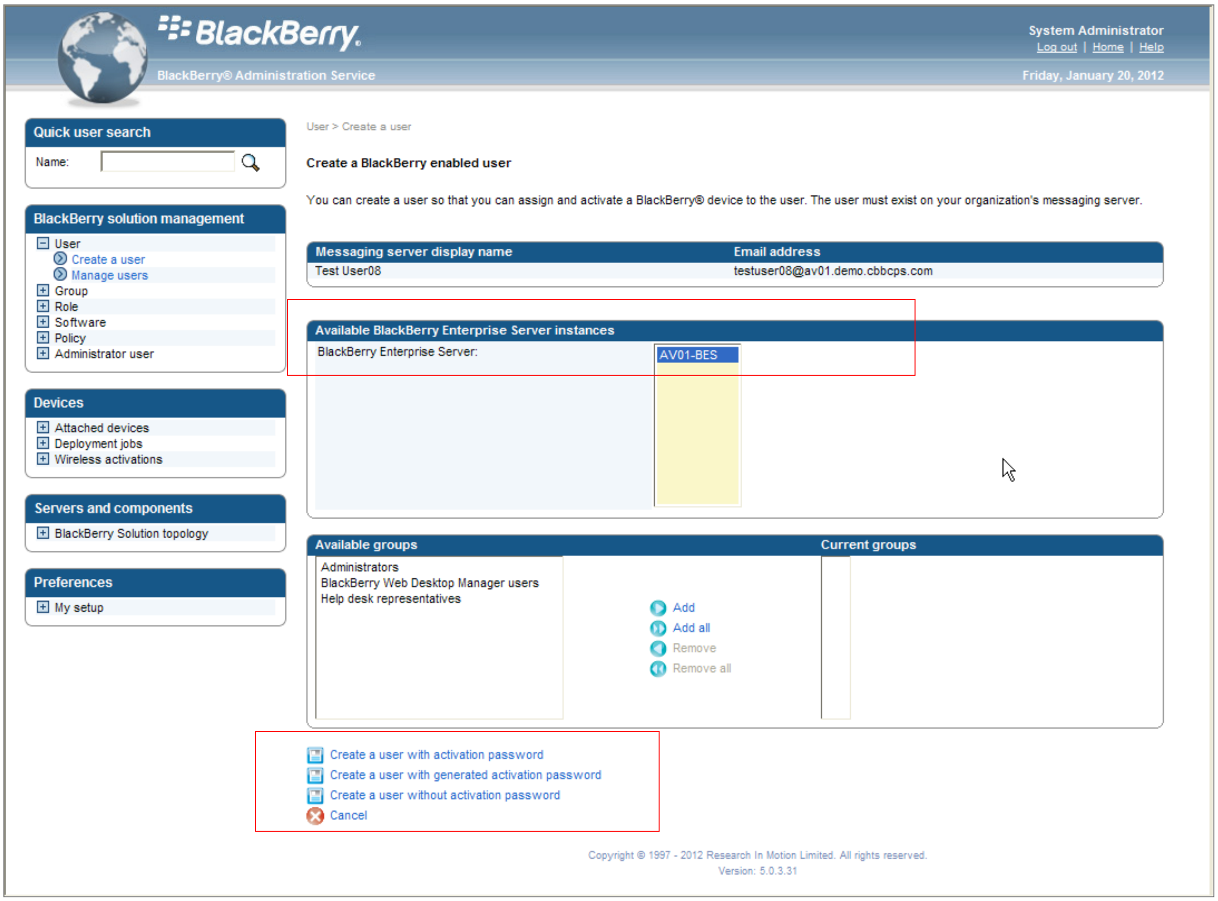
To create a user account, navigate to **Blackberry solution management → Users → Create a user**. The **Create a Blackberry enabled user** screen appears as shown below. The easiest way to add a new user is to add a user from the corporate directory. To do this, click **Search** at the bottom of the right pane.



A list of available users from the corporate directory appears at the bottom of the same page. **Test User01** – **Test User08** shown in **Section 8.1** have already been added, thus only the users named that have not been associated are shown as available users to be added. Select a user to add by clicking the box next to the user name. Click **Continue**.



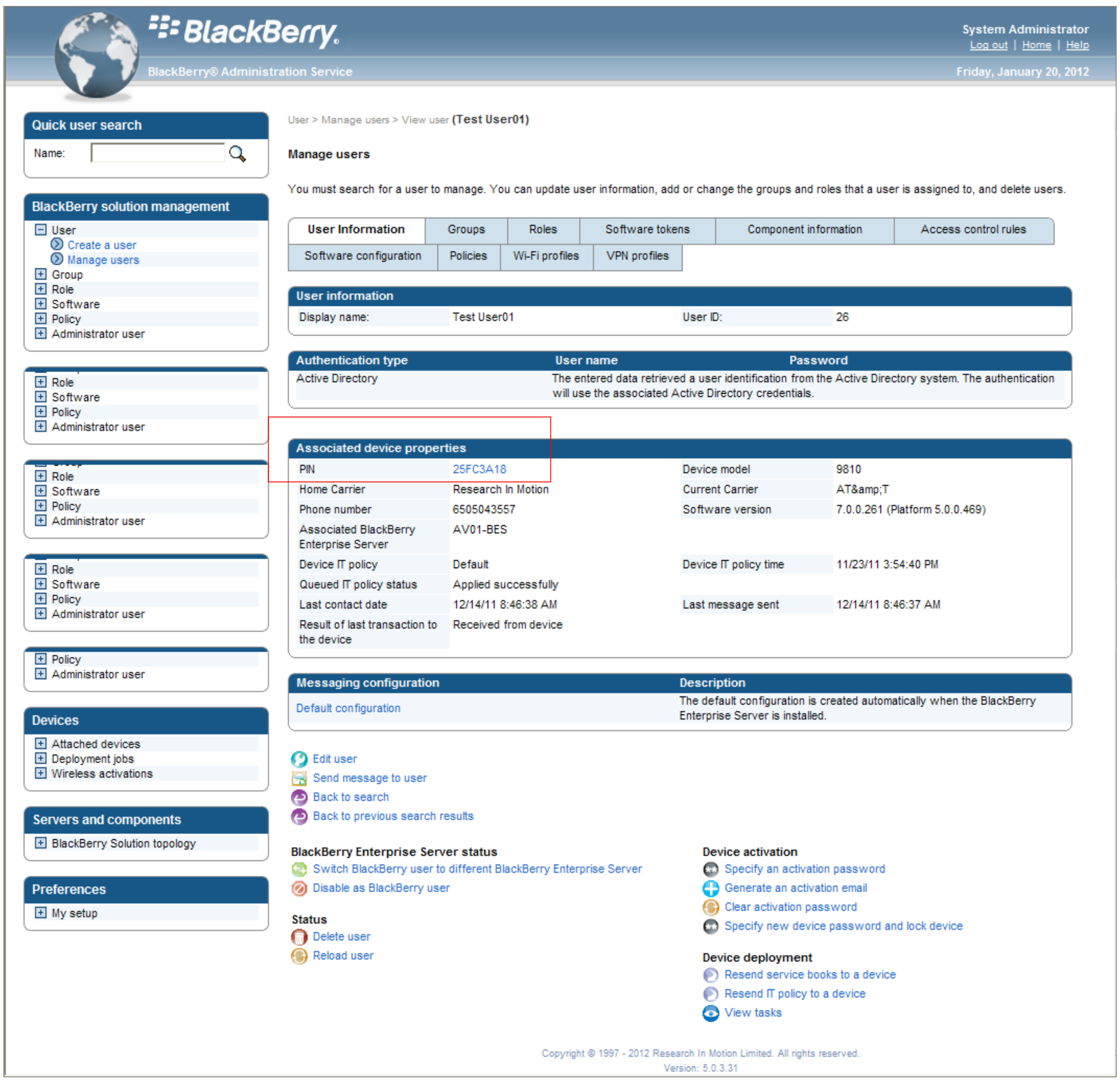
In the next screen, select an available **Blackberry Enterprise Server** to associate with this user from those listed in the middle of the screen. In the case of the compliance test, there was only one server **AV01-BES** to select. Click one of the create options listed at the bottom of the page to complete the creation of the user. Provide a password if prompted for one.



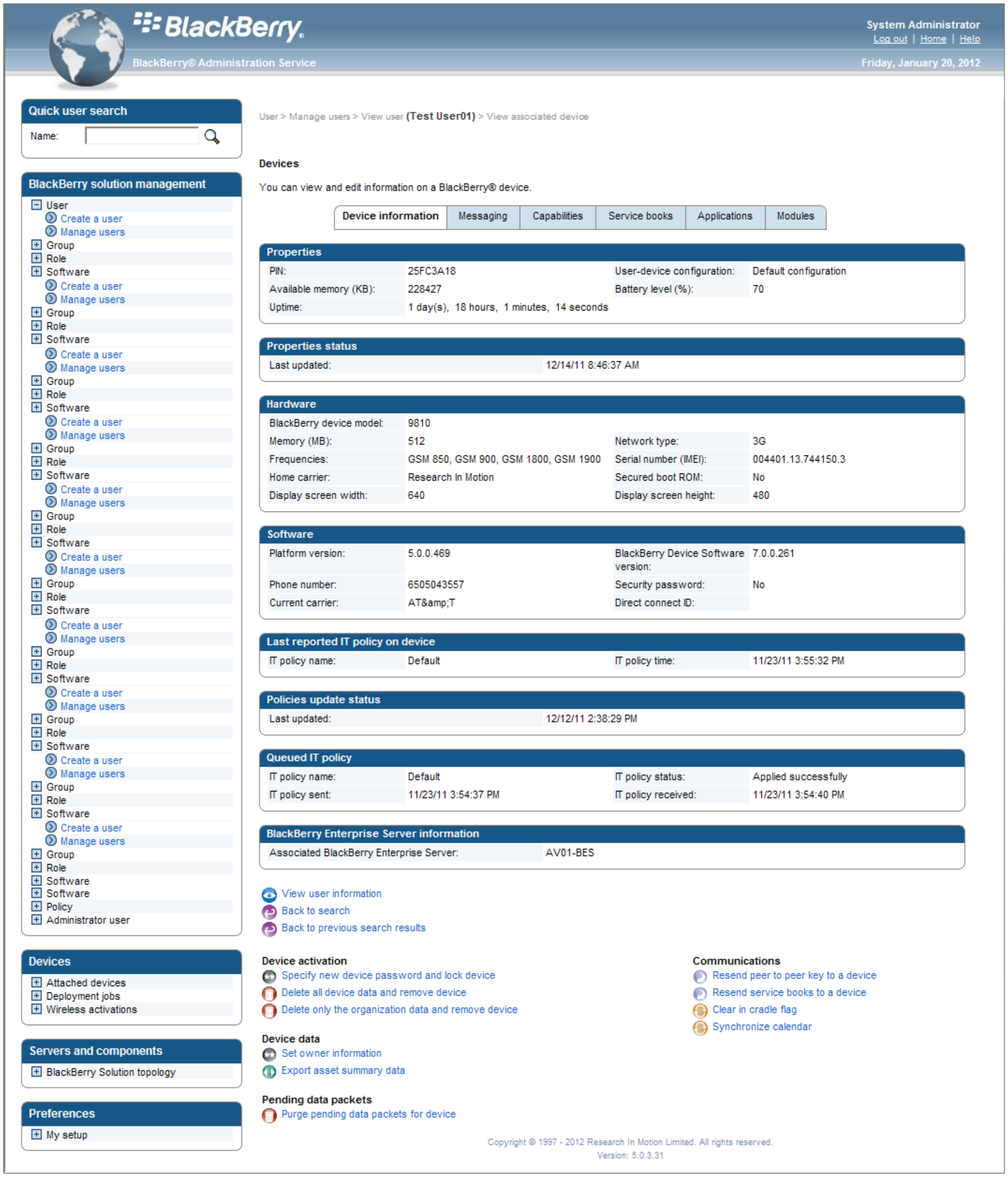
8.2.3 Manage Users

After a user is created, a user account may be modified by navigating to **Blackberry solution management → User → Manage Users**. The resulting screen shows the list of users (not shown). Clicking on one of the users displays the user details. The example below shows the user details for user **Test User01**.

The **Associated device properties** part of the screen is populated when the user logs into the Blackberry® device for the first time. The device details can be viewed by clicking on the **PIN** number in this section.



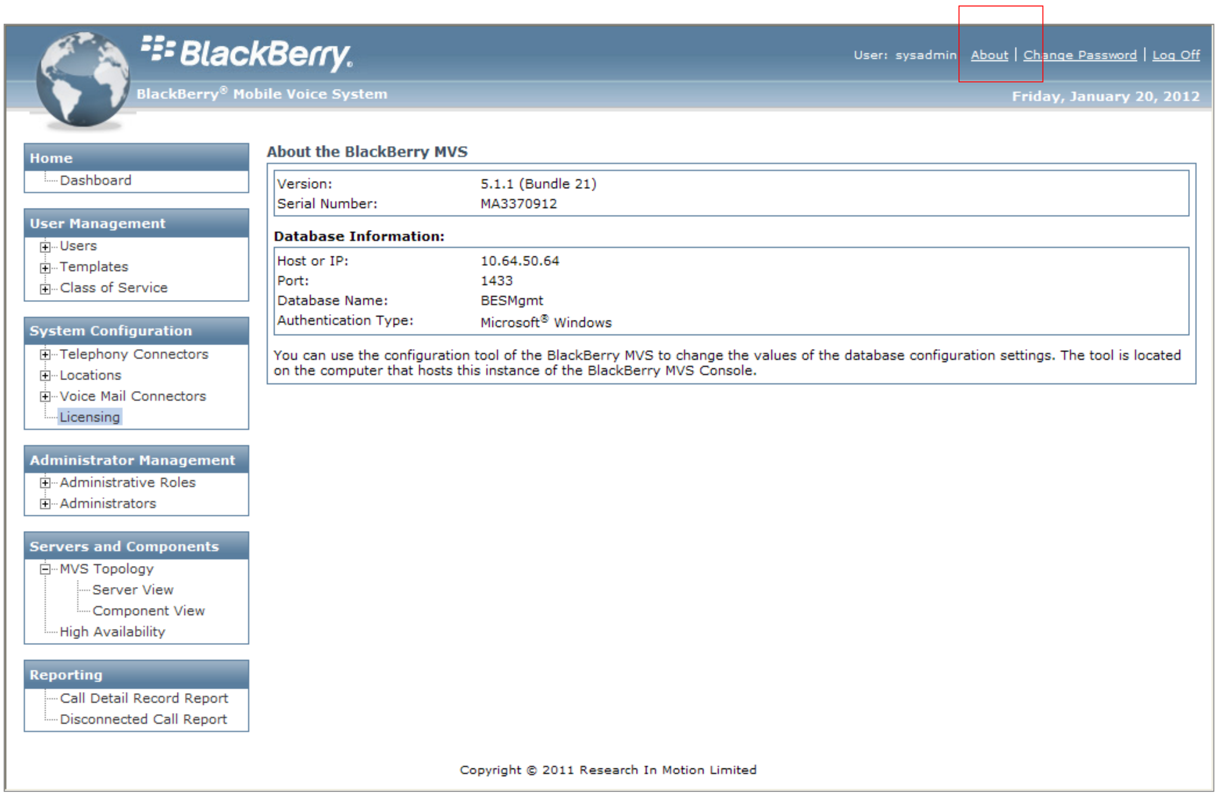
After clicking the **PIN** number, the device details are displayed below.



8.3 BlackBerry® Mobile Voice System Server Configuration

The Blackberry® MVS server is configured from the Blackberry® MVS console. This application can be launched by clicking the Blackberry® MVS console icon on the Windows Desktop. Each Blackberry® MVS console window contains a menu on the left used to access and manage the necessary components as shown below. This menu will be referenced throughout this section and its subsections.

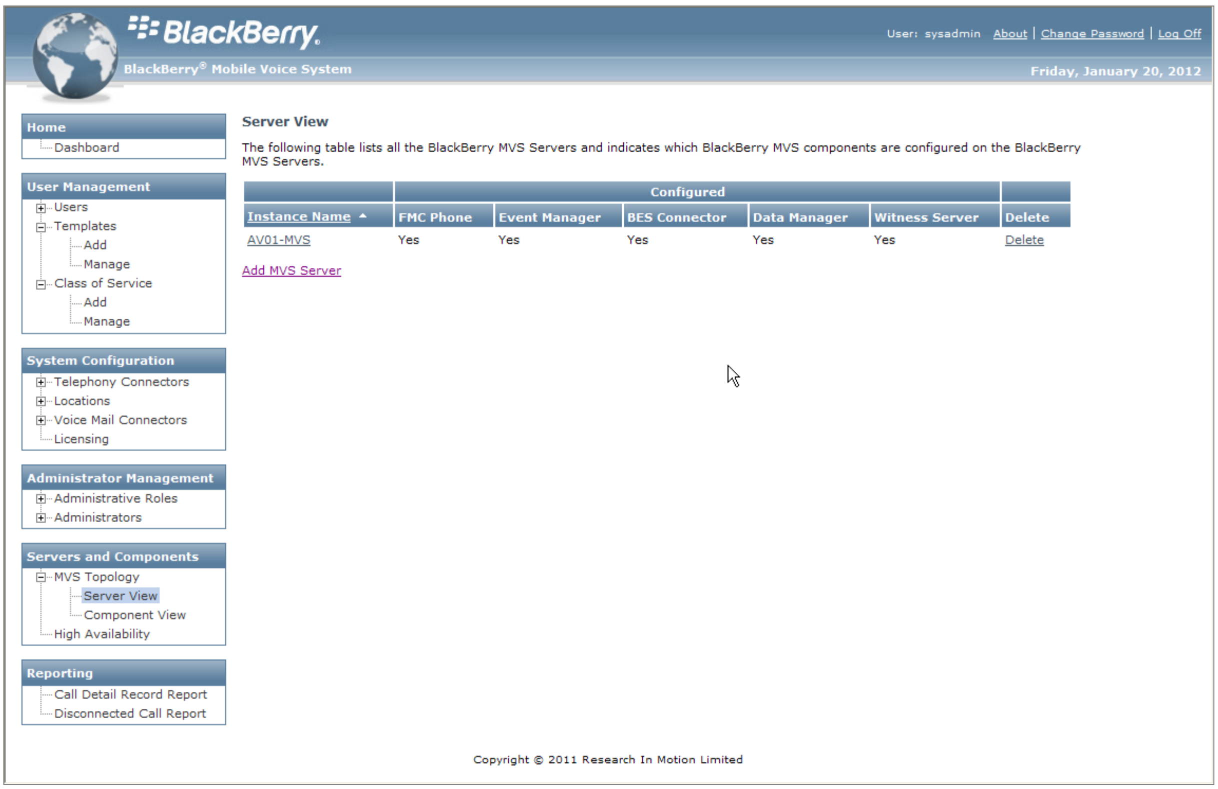
To verify the proper Blackberry® MVS software release, click the **About** link at the top of the page.



8.3.1 Create a Mobile Voice System Server

Create a MVS Session Manager for communication with the Communication Manager. Multiple Blackberry® Enterprise Server instances can be associated with one MVS Session Manager. Only one MVS Session Manager can be installed per MVS server. In the case of the compliance test, a single Blackberry® Enterprise Server was associated with the MVS Session Manager.

To create the MVS Session Manager, navigate to **Servers and Components → MVS Topology → Server View** and Click **Add MVS Server** from the right pane. In the following example instead of configuring a new MVS Server the existing MVS configuration is displayed.



Configure the parameters as described below. After creation, if the Session Manager needs to be modified, it can be edited by navigating to **Servers and Components → MVS Topology → Component View**.

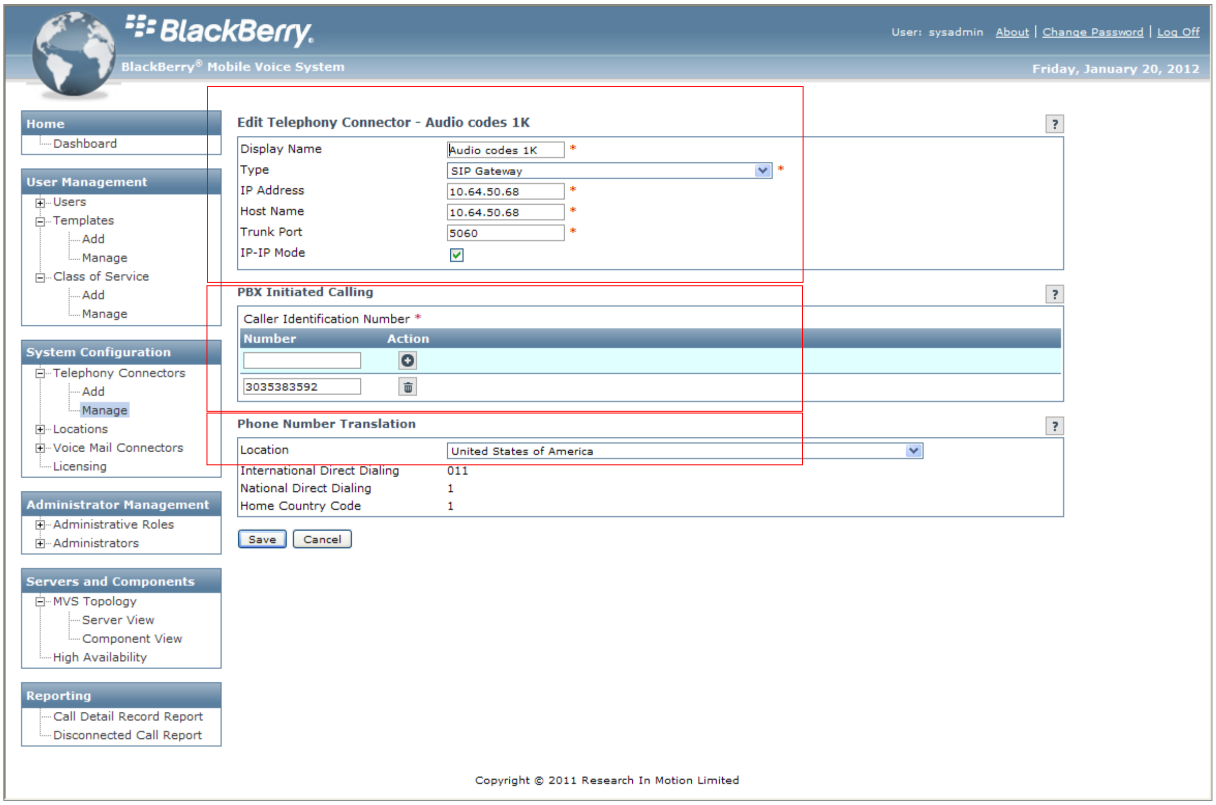
- In the **Instance Name** field, type the instance name that was specified when the MVS Session Manager was installed.
- In the **SIP IP Address** field, type the IP address that was specified when the MVS Session Manager was installed.
- In the **Line Port** field, type the UDP port number that the BlackBerry® device uses for SIP communications that are made on behalf of a specific telephone extension within the organization. The default value for this port is 5060.

- In the **Trunk Port** field, type the UDP port number that the Blackberry® device uses for general SIP communications. The default value for this port is 6060.
- To use BlackBerry® device–initiated calling, in the DID/DDI number for BlackBerry device-initiated calling field, type a PSTN phone number associated with the PBX that the BlackBerry® MVS Client uses to call the MVS Session Manager. The PBX will route this call from the PSTN to the MVS Session Manager. This number must conform to E.164 specifications with a leading plus sign (+), and the number must be unique to this MVS Session Manager.
- In the **BlackBerry Enterprise Servers** section, select the BlackBerry® Enterprise Server to be associated with the MVS Session Manager.
- Click **Save**.

8.3.2 Create a Telephony Connector

Create a telephony connector to use as a SIP gateway to communicate with the Communication Manager. In the case of the compliance test, the telephony connector was the AudioCodes Mediant 1000. To add a telephony connector, navigate to **System Configuration → Telephony Connectors → Add** from the left-hand navigation menu described at the top of **Section 8.3**. Configure the parameters as described below. After creation, if the Telephone Connector needs to be modified, it can be edited by navigating to **System Configuration → Telephony Connectors → Manage**.

- In the **Display Name** field, type a name for the telephony connector.
- In the **Type** list, click the type of telephony connector to use. In the case of the compliance test, ***SIP Gateway*** was selected.
- In the **IP Address** field, type the IP address of the Mediant 1000.
- In the **Host Name** field, type the host name or FQDN of the gateway. The default host name is the IP address.
- In the **Trunk Port** field, type the UDP port number that the gateway listens on for general SIP communications. In the compliance test, this was the listen port that the AudioCodes Mediant 1000 used for SIP communications.
- In the **PBX Initiated Calling** section, in the **Number** field, type an ANI number for the PBX. Click the add icon (+ sign).
- In the **Location** list, click the location of the telephony connector. After clicking location, the BlackBerry® Mobile Voice System adds prefixes for international direct dialing, national direct dialing, and the home country code if the PBX does not remove the plus sign or add the prefixes.
- Click **Save**.

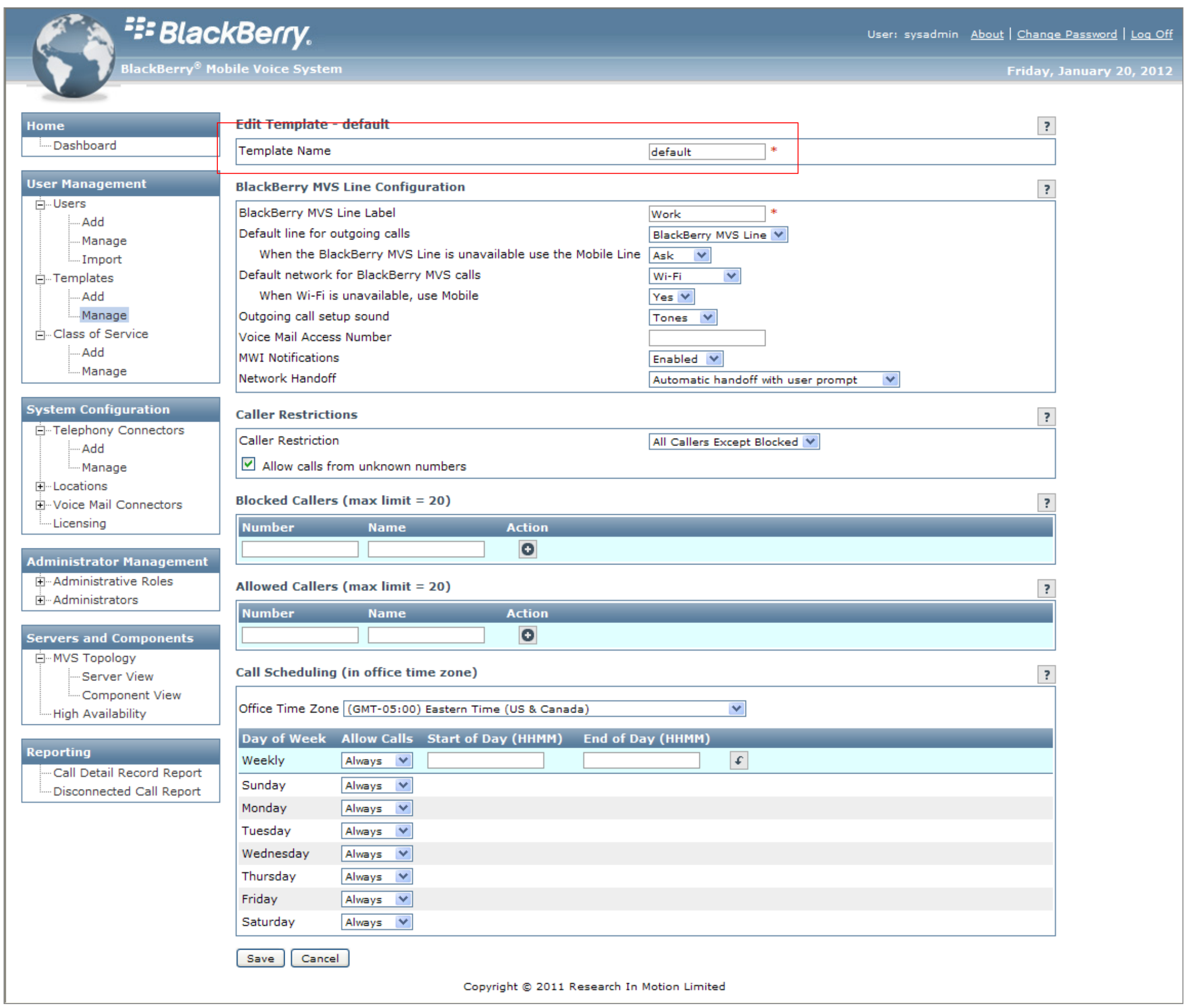

BlackBerry.
BlackBerry® Mobile Voice System
User: sysadmin About | Change Password | Log Off
Friday, January 20, 2012
Home
Dashboard
User Management
Users
Templates
Add
Manage
Class of Service
Add
Manage
System Configuration
Telephony Connectors
Add
Manage
Locations
Voice Mail Connectors
Licensing
Administrator Management
Administrative Roles
Administrators
Servers and Components
MVS Topology
Server View
Component View
High Availability
Reporting
Call Detail Record Report
Disconnected Call Report
Edit Telephony Connector - Audio codes 1K
Display Name
Audio codes 1K
Type
SIP Gateway
IP Address
10.64.50.68
Host Name
10.64.50.68
Trunk Port
5060
IP-IP Mode
PBX Initiated Calling
Caller Identification Number *
Number
Action
3035383592
Phone Number Translation
Location
United States of America
International Direct Dialing
011
National Direct Dialing
1
Home Country Code
1
Save
Cancel
Copyright © 2011 Research In Motion Limited

8.3.3 Create User Account Template

Templates must be carefully planned. The BlackBerry® MVS does not apply future template changes to user accounts that already have the template applied to them.

To add a template, navigate to **User Management → Templates → Add** from the left-hand navigation menu described at the top of **Section 8.3**. Configure the parameters as described below. After creation, if the Template needs to be modified, it can be edited by navigating to **User Management → Templates → Manage**.

- In the **Template Name** field, type a name for the template.
- Set the remaining fields as per the customer's needs. The compliance test used default values for all other fields.
- Click **Save** (shown in the next screen).

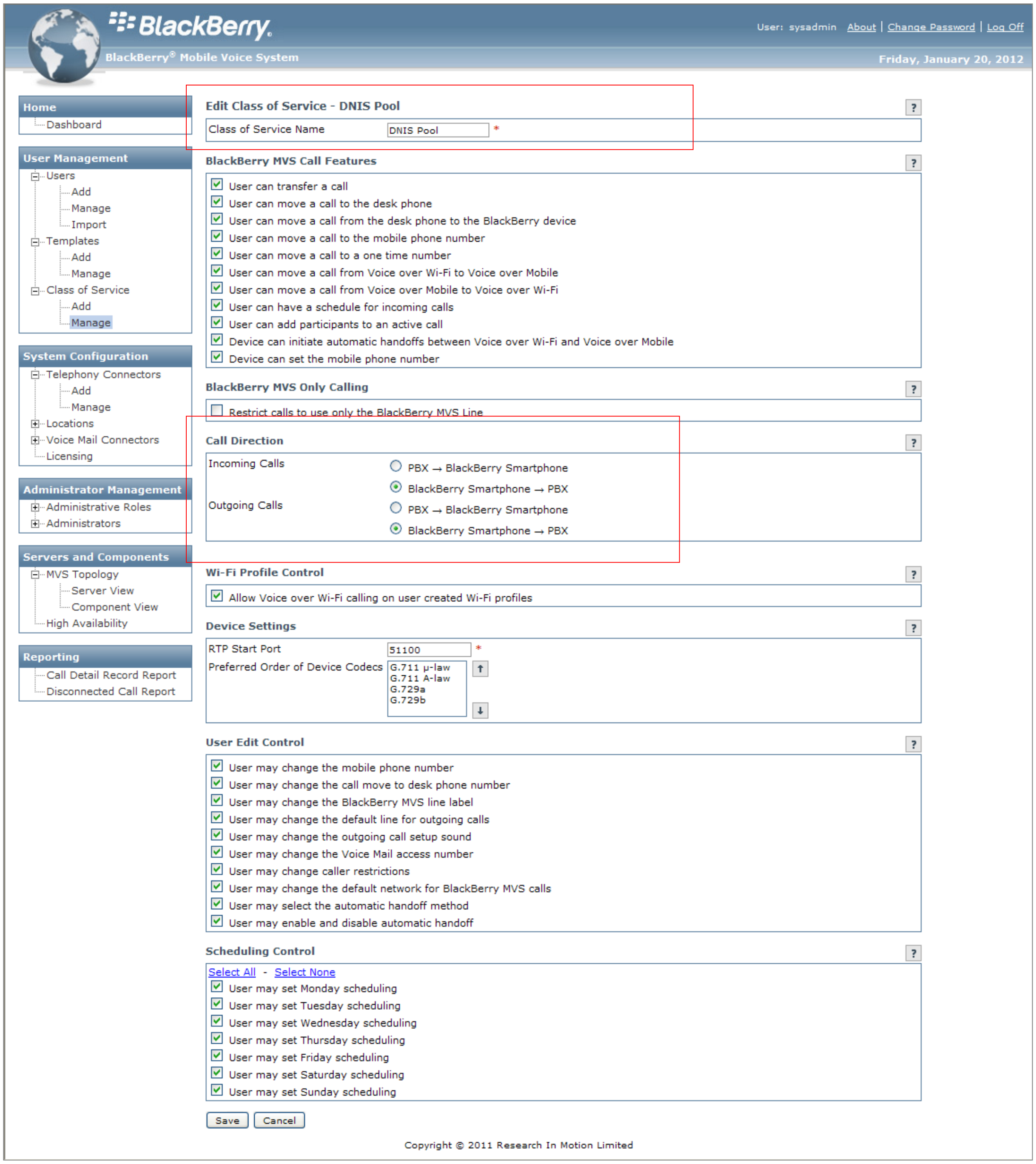


8.3.4 Create Class of Service

To add a template, navigate to **User Management → Class of Service → Add** from the left-hand navigation menu described at the top of **Section 8.3**. Configure the parameters as described below. After creation, if the Class of Service needs to be modified, it can be edited by navigating to **User Management → Class of Service → Manage**.

- In the **Class of Service Name** field, type a name for the class of service. Two class of service pools were created for the compliance test – the DNIS pool (for device-initiated calling) and the ANI pool (for PBX-initiated calling).
- In the **Call Direction** section, for the DNIS pool, select **Blackberry Smartphone → PBX** for both **Incoming Calls** and **Outgoing Calls**. For the ANI pool, select **PBX → Blackberry Smartphone** for both **Incoming Calls** and **Outgoing Calls**.
- Configure the other settings as per the customer's needs. For the compliance test, default values were used for all other parameters
- Click **Save** (as shown in the second and fourth screens).

The example below shows the DNIS pool class of service.


BlackBerry.
BlackBerry® Mobile Voice System
User: sysadmin About | Change Password | Log Off
Friday, January 20, 2012
Home
Dashboard
User Management
Users
Add
Manage
Import
Templates
Add
Manage
Class of Service
Add
Manage
System Configuration
Telephony Connectors
Add
Manage
Locations
Voice Mail Connectors
Licensing
Administrator Management
Administrative Roles
Administrators
Servers and Components
MVS Topology
Server View
Component View
High Availability
Reporting
Call Detail Record Report
Disconnected Call Report
Edit Class of Service - DNIS Pool
Class of Service Name DNIS Pool *
BlackBerry MVS Call Features
User can transfer a call
User can move a call to the desk phone
User can move a call from the desk phone to the BlackBerry device
User can move a call to the mobile phone number
User can move a call to a one time number
User can move a call from Voice over Wi-Fi to Voice over Mobile
User can move a call from Voice over Mobile to Voice over Wi-Fi
User can have a schedule for incoming calls
User can add participants to an active call
Device can initiate automatic handoffs between Voice over Wi-Fi and Voice over Mobile
Device can set the mobile phone number
BlackBerry MVS Only Calling
Restrict calls to use only the BlackBerry MVS Line
Call Direction
Incoming Calls
PBX → BlackBerry Smartphone
BlackBerry Smartphone → PBX
Outgoing Calls
PBX → BlackBerry Smartphone
BlackBerry Smartphone → PBX
Wi-Fi Profile Control
Allow Voice over Wi-Fi calling on user created Wi-Fi profiles
Device Settings
RTP Start Port 51100 *
Preferred Order of Device Codecs
G.711 µ-law
G.711 A-law
G.729a
G.729b
User Edit Control
User may change the mobile phone number
User may change the call move to desk phone number
User may change the BlackBerry MVS line label
User may change the default line for outgoing calls
User may change the outgoing call setup sound
User may change the Voice Mail access number
User may change caller restrictions
User may change the default network for BlackBerry MVS calls
User may select the automatic handoff method
User may enable and disable automatic handoff
Scheduling Control
Select All - Select None
User may set Monday scheduling
User may set Tuesday scheduling
User may set Wednesday scheduling
User may set Thursday scheduling
User may set Friday scheduling
User may set Saturday scheduling
User may set Sunday scheduling
Save
Cancel
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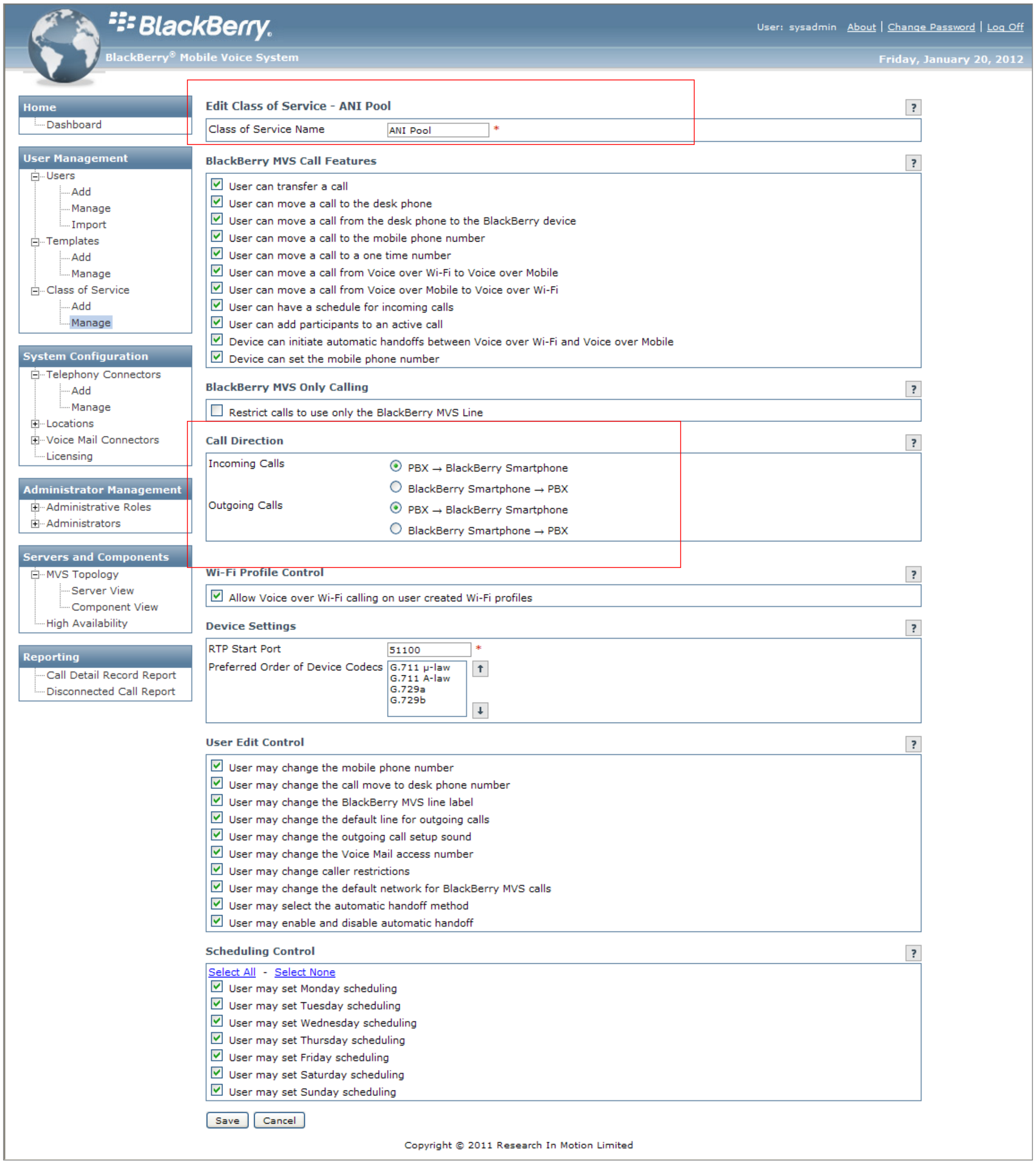
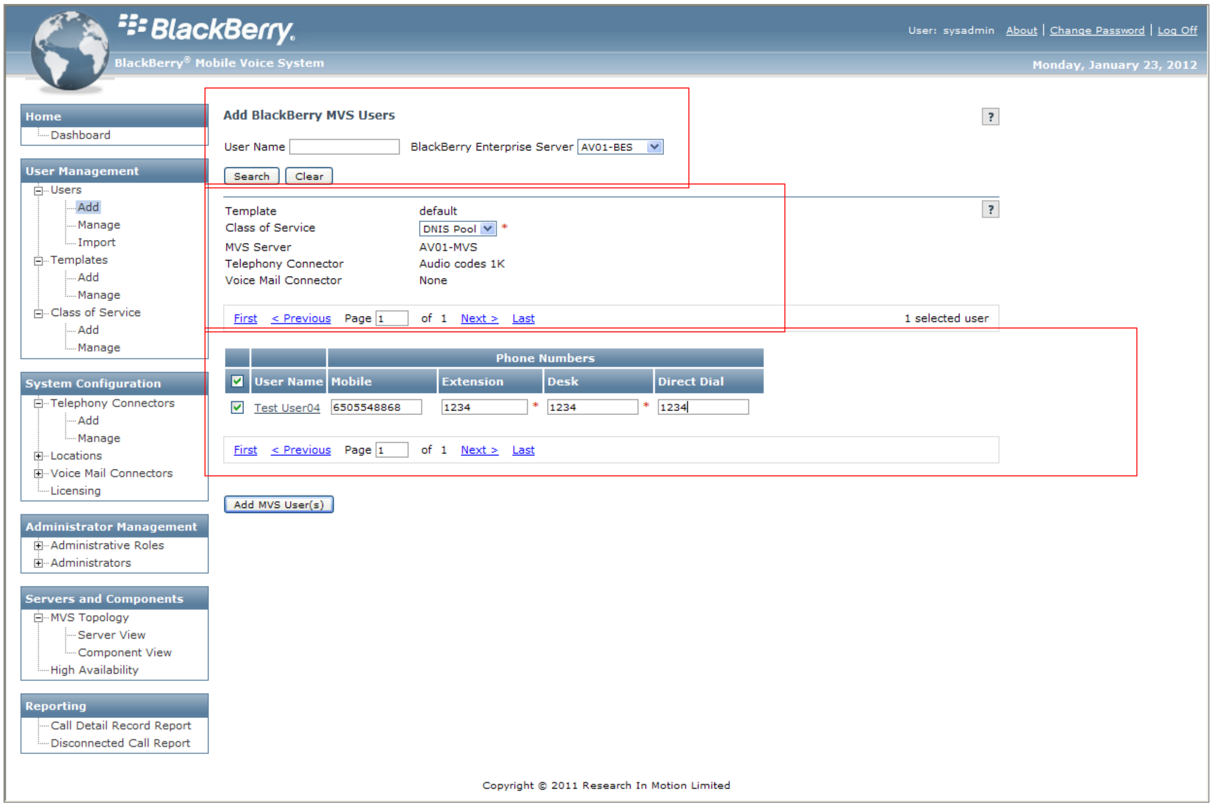
The example below shows the ANI pool class of service.



8.3.5 Add Mobile Voice System Users

To add a MVS user, navigate to **User Management → Add.** Configure the parameters as described below.

- For the **BlackBerry Enterprise Server** field, select the BlackBerry® Enterprise Server from which the users will be imported.
- To see all available BlackBerry® Enterprise Server user accounts, click **Search**.
- For the **Template** field, the **Default** template created in **Section 8.3.3** is automatically selected since only one template has been defined. If more than one template was available, it would be selectable from a drop-down menu.
- In the **Class of Service** drop-down list, select one of the class of service pools created in **Section 8.3.4**. The Class of Service selected depends on whether the mobile device will be using Blackberry® device-initiated calling or if it will be using PBX-initiated calling. **DNIS Pool** is selected for Blackberry® device-initiated calling. **ANI Pool** is selected for PBX-initiated calling.
- In the **Telephony Connector** drop-down list, select the telephony connector created in **Section 8.3.2**.
- Select the check box beside each BlackBerry® Enterprise Server user account that should be added.
- If necessary, configure the settings for the BlackBerry® Enterprise Server user accounts that were selected. Fields that are marked with an asterisk (*) are required.
- Click **Add MVS User(s)**.


BlackBerry.
BlackBerry® Mobile Voice System
User: sysadmin About | Change Password | Log Off
Monday, January 23, 2012
Home
Dashboard
User Management
Users
Add
Manage
Import
Templates
Add
Manage
Class of Service
Add
Manage
System Configuration
Telephony Connectors
Add
Manage
Locations
Voice Mail Connectors
Licensing
Administrator Management
Administrative Roles
Administrators
Servers and Components
MVS Topology
Server View
Component View
High Availability
Reporting
Call Detail Record Report
Disconnected Call Report
Add BlackBerry MVS Users
User Name
BlackBerry Enterprise Server AV01-BES
Search
Clear
Template default
Class of Service DNIS Pool
MVS Server AV01-MVS
Telephony Connector Audio codes 1K
Voice Mail Connector None
First < Previous Page 1 of 1 Next > Last
1 selected user
Phone Numbers
User Name
Mobile
Extension
Desk
Direct Dial
Test User04
6505548868
1234
1234
1234
First < Previous Page 1 of 1 Next > Last
Add MVS User(s)
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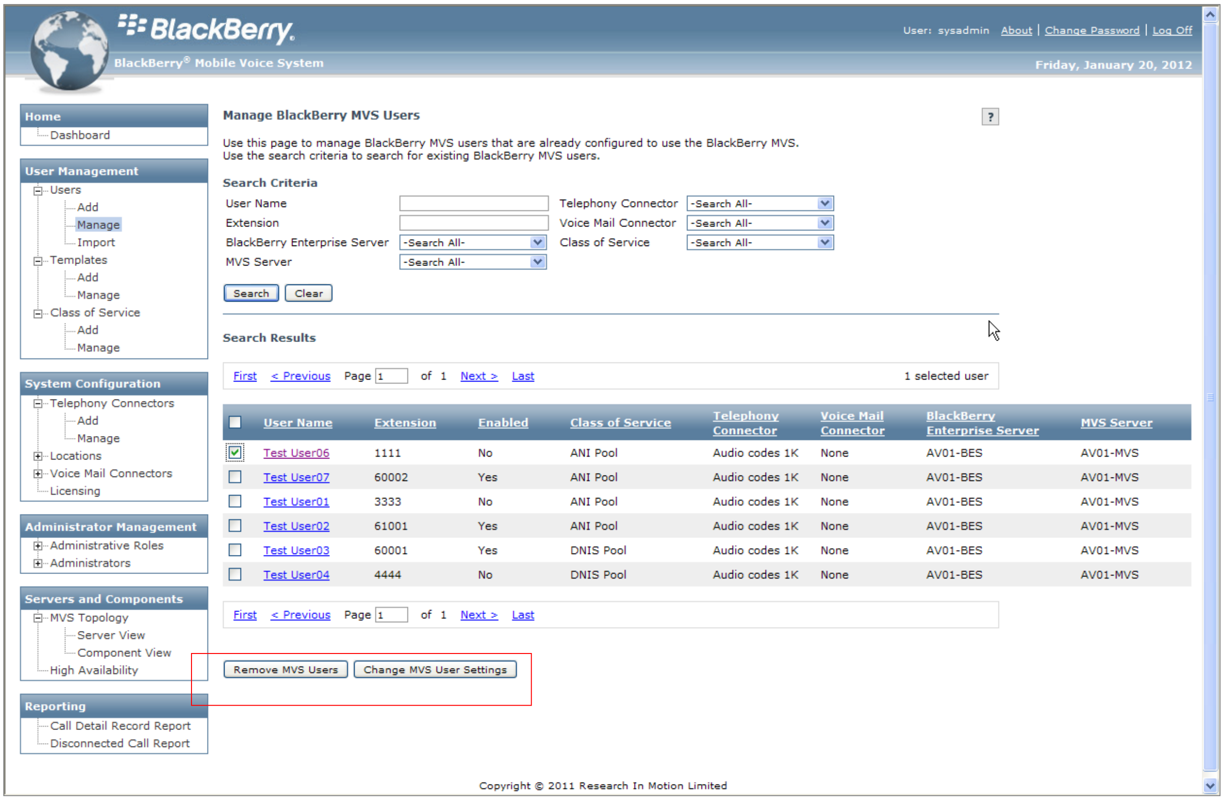
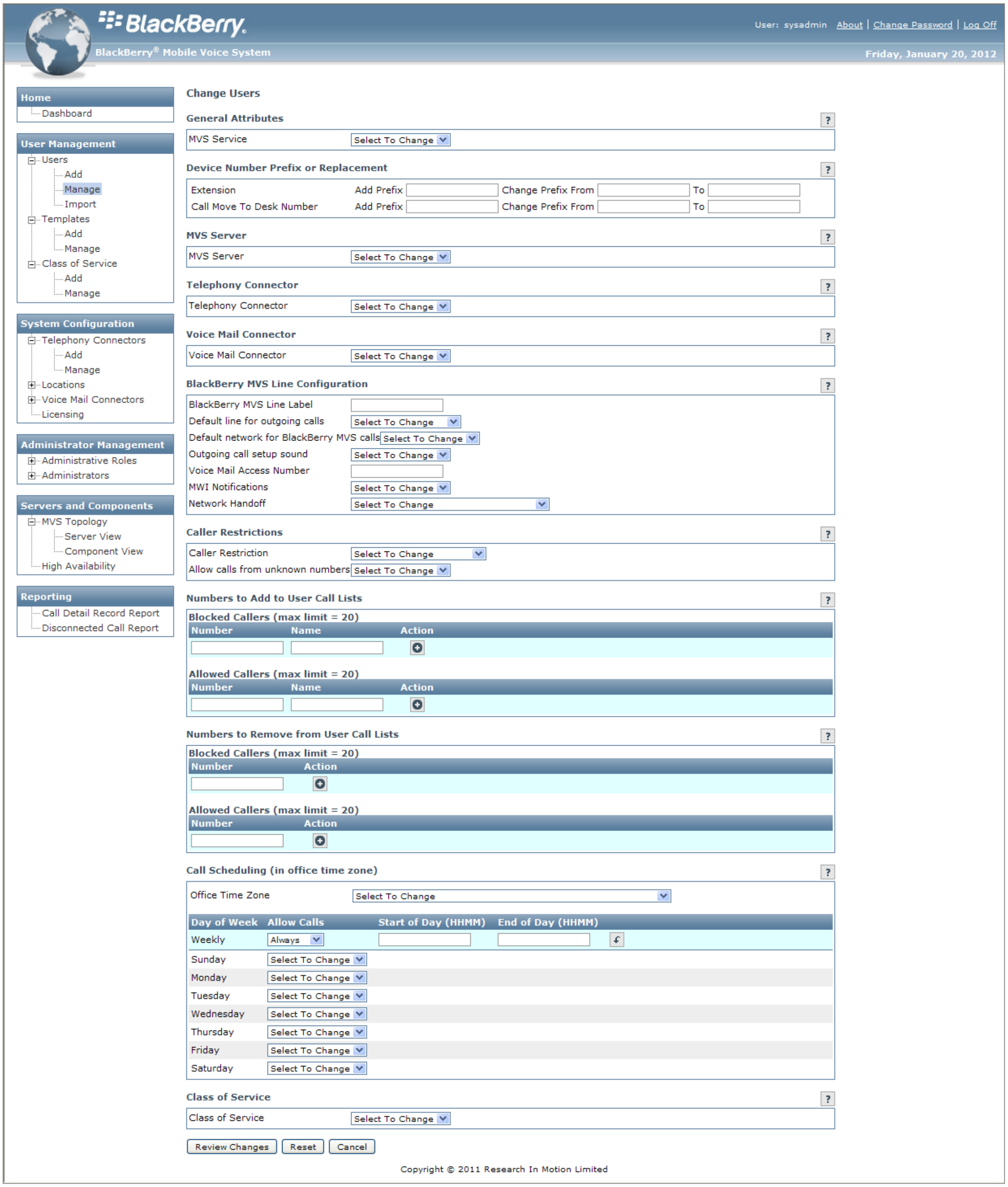
8.3.6 Manage Mobile Voice System Users

After creation, if the user needs to be modified, it can be edited by navigating to **User Management → Users → Manage**. Select the check box next to the **User Name** in the search results that is to be modified. Click **Change MVS User Settings**.



The following screens show all the user parameters that are available using the example of **Test User06**.


BlackBerry
BlackBerry® Mobile Voice System
User: sysadmin About | Change Password | Log Off
Friday, January 20, 2012
Home
Dashboard
User Management
Users
Add
Manage
Import
Templates
Add
Manage
Class of Service
Add
Manage
System Configuration
Telephony Connectors
Add
Manage
Locations
Voice Mail Connectors
Licensing
Administrator Management
Administrative Roles
Administrators
Servers and Components
MVS Topology
Server View
Component View
High Availability
Reporting
Call Detail Record Report
Disconnected Call Report
Change Users
General Attributes
MVS Service Select To Change
Device Number Prefix or Replacement
Extension Add Prefix Change Prefix From To
Call Move To Desk Number Add Prefix Change Prefix From To
MVS Server
MVS Server Select To Change
Telephony Connector
Telephony Connector Select To Change
Voice Mail Connector
Voice Mail Connector Select To Change
BlackBerry MVS Line Configuration
BlackBerry MVS Line Label
Default line for outgoing calls Select To Change
Default network for BlackBerry MVS calls Select To Change
Outgoing call setup sound Select To Change
Voice Mail Access Number
MWI Notifications Select To Change
Network Handoff Select To Change
Caller Restrictions
Caller Restriction Select To Change
Allow calls from unknown numbers Select To Change
Numbers to Add to User Call Lists
Blocked Callers (max limit = 20)
Number Name Action
Allowed Callers (max limit = 20)
Number Name Action
Numbers to Remove from User Call Lists
Blocked Callers (max limit = 20)
Number Action
Allowed Callers (max limit = 20)
Number Action
Call Scheduling (in office time zone)
Office Time Zone Select To Change
Day of Week Allow Calls Start of Day (HHMM) End of Day (HHMM)
Weekly Always
Sunday Select To Change
Monday Select To Change
Tuesday Select To Change
Wednesday Select To Change
Thursday Select To Change
Friday Select To Change
Saturday Select To Change
Class of Service
Class of Service Select To Change
Review Changes Reset Cancel
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9 Verification Steps

The following steps may be used to verify the configuration:

- From the Avaya Communication Manager SAT, use the **status signaling-group** command to verify that the signaling groups configured are in-service.

```
status signaling-group 2
                         STATUS SIGNALING GROUP

        Group ID: 2
      Group Type: sip

     Group State: in-service
```

- From the Avaya Communication Manager SAT, use the **status trunk-group** command to verify that the trunk groups configured are in-service.

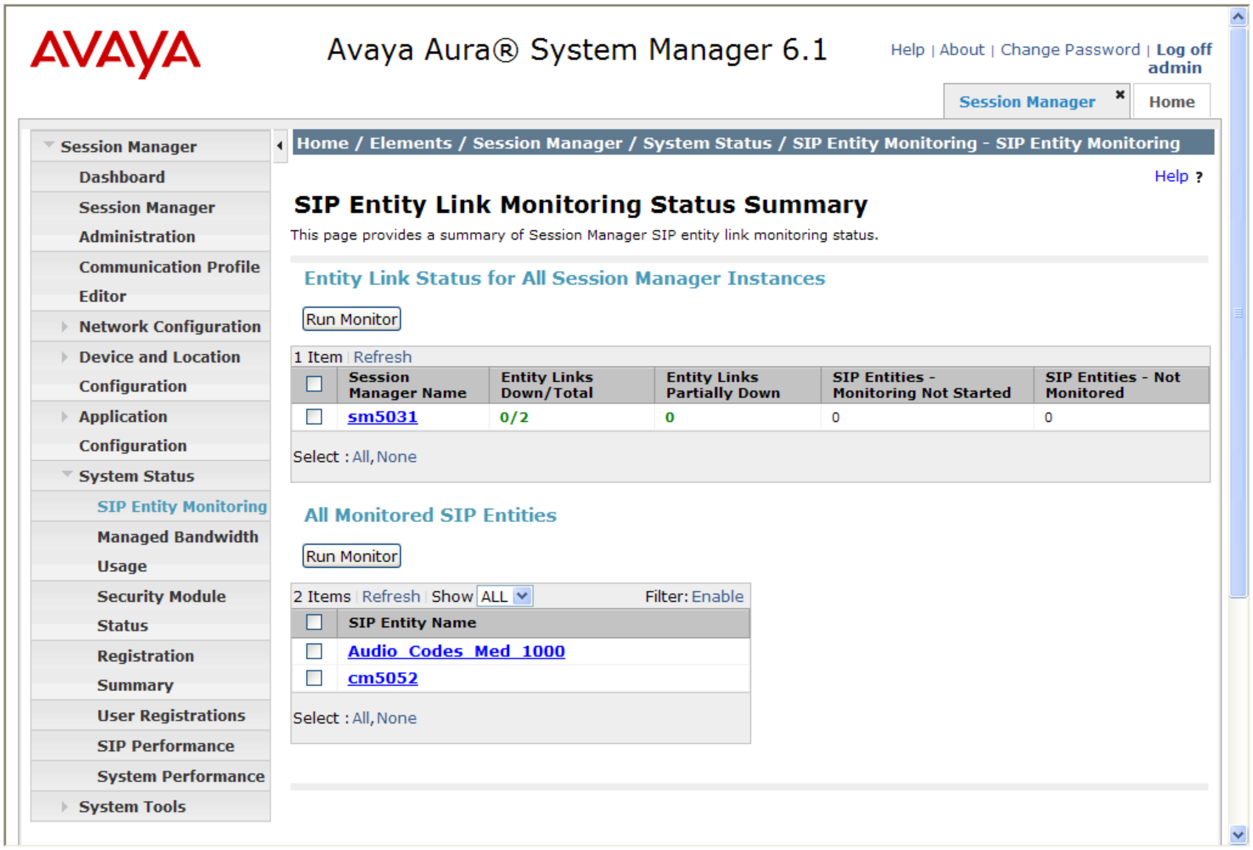
```
status trunk 2

                             TRUNK GROUP STATUS

Member   Port     Service State      Mtce Connected Ports
                                     Busy

0002/001 T00036   in-service/idle    no
0002/002 T00037   in-service/idle    no
0002/003 T00038   in-service/idle    no
0002/004 T00039   in-service/idle    no
0002/005 T00040   in-service/idle    no
0002/006 T00041   in-service/idle    no
0002/007 T00042   in-service/idle    no
0002/008 T00043   in-service/idle    no
0002/009 T00044   in-service/idle    no
0002/010 T00045   in-service/idle    no
```

- From the Avaya System Manager Navigate to → Session Manager → System Status → SIP Entity Monitoring.
- Verify status for each monitored SIP Entity.



- Verify that mobile originated calls routed through the Avaya telephony infrastructure can terminate to a desk phone, mobile device or the PSTN.
- Verify that calls from a desk phone, mobile device or the PSTN routed through the Avaya telephony infrastructure can terminate to a mobile device.

10 Conclusion

These Application Notes describe the configuration steps required for integrating the Research In Motion Mobile Voice System solution into an Avaya telephony infrastructure. For the configuration described in these Application Notes, the Research In Motion Mobile Voice System solution was responsible for bridging landline connectivity to Avaya Aura® Communication Manager and Avaya Aura® Session Manager with the wireless connectivity of the GSM/CMDA network. The functionality of the Avaya/RIM solution was validated via the DevConnect Program at the Avaya Solution and Interoperability Test Lab. All feature functionality test cases passed.

11 Additional References

This section references the documentation relevant to these Application Notes. Additional Avaya product documentation is available at http://support.avaya.com.

[1] *Administering Avaya* Aura® *Communication Manager*, June 2010, Release 6.0, Issue 6.0, Document Number 03-300509, available at http://support.avaya.com.

[2] *Administering Avaya* Aura® *Session Manager*, October 2010, Issue 1.1, Release 6.1, Document Number 03-603324, available at http://support.avaya.com.

[3] *Avaya one-X Deskphone Edition for 9600 Series IP Telephones Administrator Guide Release 3.1,* November 2009, Document Number 16-300698.

[4] *Implementing Avaya Aura® Communication Manager Messaging*, May 2011, Document Number 18-603644.

Product documentation for the RIM MVS solution can be obtained from RIM at the following link:

[5] BlackBerry Mobile Voice System, available at http://docs.blackberry.com/en/admin/categories/?userType=2&category=BlackBerry+Mobile+Voice+System

Product documentation for the AudioCodes Mediant 1000 VoIP Media Gateway can be obtained from AudioCodes at the following link:

[6] AudioCodes Product Documentation and Software, available at http://www.audiocodes.com/downloads

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.