Avaya Solution & Interoperability Test Lab

Application Notes for Integrating the Research In Motion BlackBerry® Mobile Voice System with Avaya Aura® Communication Manager using a QSIG 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 H.323 IP Telephones and AudioCodes Mediant 1000 with QSIG 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 H.323 IP Telephones and AudioCodes Mediant 1000 with QSIG 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 originating and mobile terminating 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 Forward All
- Call Hold /Resume
- Shared Line Appearance
- Transfer
- Move Call To Desk

2.2 Test Results

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

The following observations were made:

1. When the data network is very slow, occasionally the caller id name is not displayed.

2. When the data network is very slow, the talk path back to the end user is sometimes delayed one to two seconds.

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 S8300 Server with a Avaya G450 Media Gateway running Communication Manager
- Communication Manager Messaging
- One Avaya 2400 Series Digital Telephone
- Avaya 9600 Series IP Telephones running Avaya one-X® Deskphone Edition
- One RIM BlackBerry® MVS Server
- One RIM BlackBerry® Enterprise Server
- RIM BlackBerry® phones running the MVS Client software.
- AudioCodes Mediant 1000
- One router
- One corporate DHCP/TFTP/IAS Server.
- One Microsoft Exchange Server.

In **Figure 1**, the Communication Manager has two trunks. The first is trunk 56 which goes to the PSTN. The second is trunk 58 which is a T1 ISDN-PRI QSIG trunk that connects to the Mediant 1000. The Mediant 1000 serves as a gateway between the QSIG trunk on one side to a SIP trunk on the other side which connects to the Blackberry® MVS. The configuration includes three Blackberry devices. Two 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.

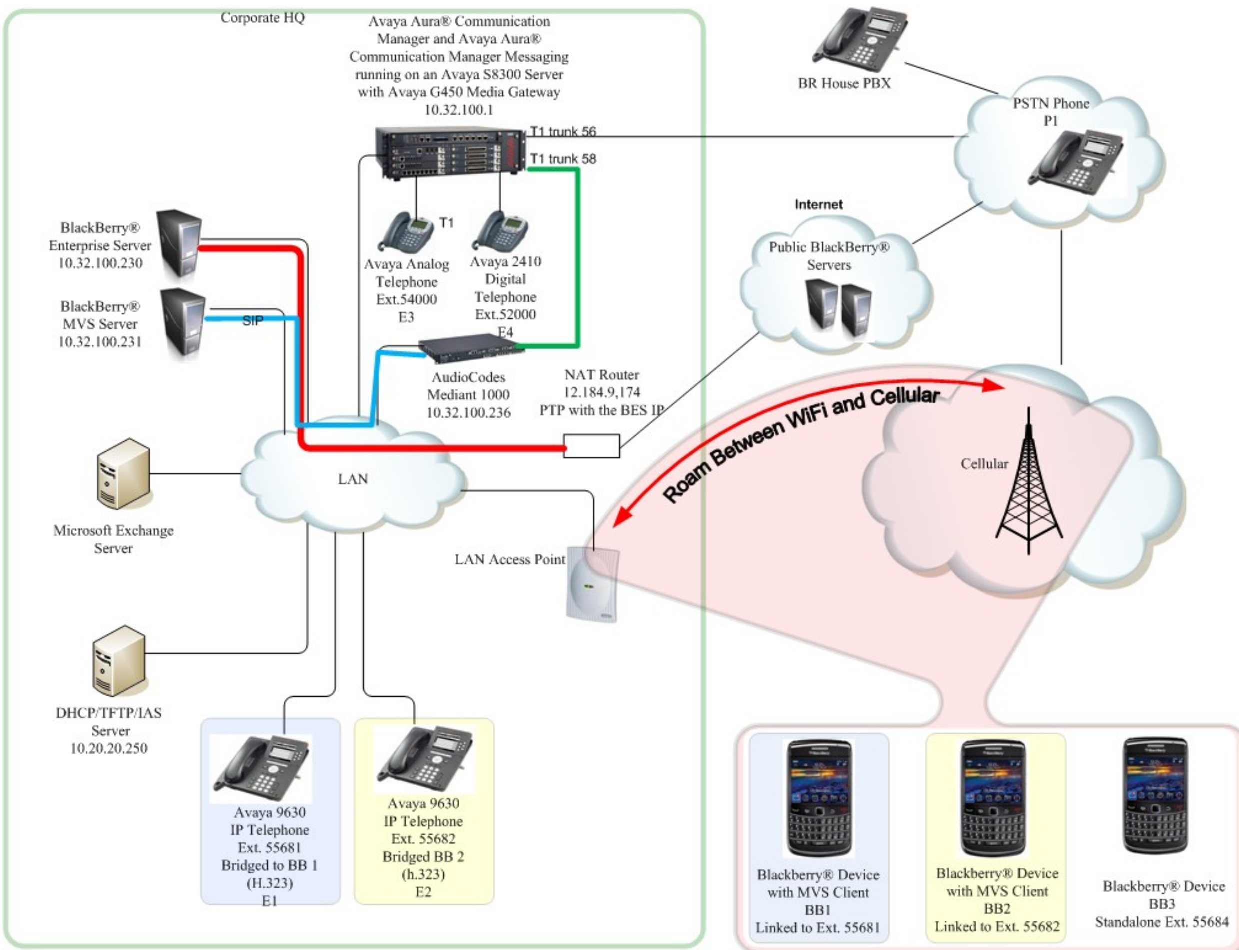


Figure 1: Avaya/RIM MVS Solution Diagram

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 10**.

Inbound call (Device-initiated calling)

- PSTN caller calls the enterprise DID number assigned to the desk phone/Blackberry® device pair. The call arrives at Communication Manager on trunk 56.
- Communication Manager rings the desk phone (if it exists) and also sends the call out trunk 58 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 number is provisioned in **Section 7.3.1**.
- 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 Communication Manager on trunk 56. The call is directed to trunk 58 to reach 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 Communication Manager on trunk 56.
- Communication Manager rings the desk phone (if it exists) and also sends the call out trunk 58 to the MVS.
- The MVS places a call to the Blackberry® device using its mobile number.
- This call arrives at Communication Manager on trunk 58. The call is directed to trunk 56 to reach the PSTN/wireless network.
- The Blackberry® device rings and the user answers. The caller ID is a number assigned by the enterprise and provisioned in **Section 7.3.2**.
- 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 Communication Manager on trunk 56. The call is directed to trunk 58 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 Communication Manager on trunk 58. The call is directed to trunk 56 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 Communication Manager on trunk 58. The call is directed to trunk 56 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 Communication Manager on trunk 58. The call is directed to trunk 56 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

Equipment	Software/Firmware
Avaya PBX Products	
Avaya S8300 Server running Avaya Aura® Communication Manager	Avaya Aura® Communication Manager 6.0.1 Service Pack 0.01 (00.1.510.1-18621)
Avaya G450 Media Gateway (Corporate Site) MGP MM712 DCP Media Module	30 .13 .2 HW9
Avaya Messaging (Voice Mail) Products	
Avaya Aura® Communication Manager Messaging (CMM)	6.0.1
Avaya Telephony Sets	
Avaya 96xx Series IP Telephones	(H.323 3.1.1)
Avaya 2410 Digital Telephone	5.0
RIM Products	
BlackBerry® Enterprise Server (running on Microsoft Windows 2008 Server)	BES 5.0
BlackBerry® MVS (running on Microsoft Windows 2008 Server)	MVS 5.0.2 GA bundle
AudioCodes Products	
AudioCodes Mediant 1000	*6.00A.037*
Microsoft products	
DHCP/HTTP/TFTP Server	Microsoft Windows 2003 Server
Microsoft Exchange (running on Microsoft Windows 2008 Server)	Microsoft Exchange 2007

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 10** (**[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) 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.

Step	Description
1.	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. (screen below)

```
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  147
                                     Maximum Stations: 2400  48
                             Maximum XMOBILE Stations: 2400  0
                   Maximum Off-PBX Telephones - EC500: 9600  3
                   Maximum Off-PBX Telephones -   OPS: 9600  35
                   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
```

Step	Description

2.	Automatic Route Selection (ARS) will be used to route calls to the PSTN trunk. On **Page 3**, verify that **ARS** is set to **y**.
	
	<pre>display system-parameters customer-options Page 3 of 11
	                                OPTIONAL FEATURES
	
	    Abbreviated Dialing Enhanced List? y Audible Message Waiting? y
	        Access Security Gateway (ASG)? y 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? y DCS Call Coverage? y
	          ASAI Link Plus Capabilities? y 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</pre>

3.	The trunk between Communication Manager and the Mediant 1000 is a T1 ISDN-PRI QSIG trunk. On **Page 4**, verify that **ISDN-PRI** is set to **y**. In addition, EC500 will be used to associate desk extensions to mobile extension so set **Enhanced EC500** to **y**.
	
	<pre>display system-parameters customer-options Page 4 of 11
	                                OPTIONAL FEATURES
	
	   Emergency Access to Attendant? y IP Stations? y
	           Enable 'dadmin' Login? y
	           Enhanced Conferencing? y ISDN Feature Plus? n
	                  Enhanced EC500? y ISDN/SIP Network Call Redirection? y
	    Enterprise Survivable Server? n ISDN-BRI Trunks? y
	       Enterprise Wide Licensing? n **ISDN-PRI? y**
	              ESS Administration? y Local Survivable Processor? n
	          Extended Cvg/Fwd Admin? y Malicious Call Trace? y
	     External Device Alarm Admin? y Media Encryption Over IP? n
	  Five Port Networks Max Per MCC? n Mode Code for Centralized Voice Mail? n
	                Flexible Billing? n
	   Forced Entry of Account Codes? y Multifrequency Signaling? y
	      Global Call Classification? y Multimedia Call Handling (Basic)? y
	             Hospitality (Basic)? y Multimedia Call Handling (Enhanced)? y
	 Hospitality (G3V3 Enhancements)? y Multimedia IP SIP Trunking? y
	                       IP Trunks? y
	
	           IP Attendant Consoles? y</pre>

Step	Description
4.	Automatic Alternate Routing will be used to route calls to the trunk connected to the Mediant 1000 and ultimately to the RIM MVS. On **Page 5**, verify that **Private Networking** is set to **y**.

```
display system-parameters customer-options                    Page   5 of  11
                                OPTIONAL FEATURES

                Multinational Locations? n           Station and Trunk MSP? y
 Multiple Level Precedence & Preemption? y     Station as Virtual Extension? y
                     Multiple Locations? n
                                             System Management Data Transfer? n
          Personal Station Access (PSA)? y             Tenant Partitioning? y
                        PNC Duplication? n       Terminal Trans. Init. (TTI)? y
                   Port Network Support? n             Time of Day Routing? y
                        Posted Messages? y    TN2501 VAL Maximum Capacity? y
                                                       Uniform Dialing Plan? y
                     Private Networking? y     Usage Allocation Enhancements? y
               Processor and System MSP? y
                     Processor Ethernet? y               Wideband Switching? y
                                                                   Wireless? n
                          Remote Office? y
          Restrict Call Forward Off Net? y
                  Secondary Data Module? 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) and dial access codes (dac) can be used to invoke specific PBX features.

<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>1.</td><td>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.
<pre>
display dialplan analysis   Page   1 of 12
                             DIAL PLAN ANALYSIS TABLE
                                   Location: all   Percent Full: 3

    Dialed   Total Call   Dialed   Total Call   Dialed   Total Call
    String   Length Type   String   Length Type   String   Length Type
   0   3   fac
   1   3   fac
   2   5   ext
   3   1   dac
   4   5   aar
   5   5   ext
   6   5   ext
   7   1   fac
   8   1   fac
   9   1   fac
</pre></td></tr>
<tr><td>2.</td><td>Assign an access code for AAR and ARS (if not already assigned) that is consistent with the dial plan shown in Step 1. A fac or dac can be used for this purpose.
<pre>
change feature-access-codes   Page   1 of 11
                               FEATURE ACCESS CODE (FAC)
         Abbreviated Dialing List1 Access Code: *600
         Abbreviated Dialing List2 Access Code: *601
         Abbreviated Dialing List3 Access Code: *602
Abbreviated Dial - Prgm Group List Access Code:
                      Announcement Access Code: *604
                       Answer Back Access Code: *650
                         Attendant Access Code:
      Auto Alternate Routing (AAR) Access Code: 3
    Auto Route Selection (ARS) - Access Code 1: 9   Access Code 2:
                 Automatic Callback Activation: *605   Deactivation: *606
Call Forwarding Activation Busy/DA: *607   All: *608   Deactivation: *609
</pre></td></tr>
</table>

5.3 System Parameters Features

Verify that the necessary system level features are enabled to support the configuration.

<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>1.</td><td>To use QSIG on the ISDN-PRI trunk, the following parameters need to be configured. Issue the change system-parameters features command, navigate to Page 8 and perform the following.

Set the QSIG/ETSI TSC Extension to any unused valid extension
Set Path Replacement with Measurements to y.
Set QSIG Path Replacement Extension to any unused valid extension.
Set the Send QSIG Path Replacement Conf. Event to ASAI field is set to y.

<pre>
change system-parameters features   Page   8 of 19
                        FEATURE-RELATED SYSTEM PARAMETERS

ISDN PARAMETERS
                                                           PARAMETERS FOR CREATING
 Send Non-ISDN Trunk Group Name as Connected Name? n   QSIG SELECTION NUMBERS
 Display Connected Name/Number for ISDN DCS Calls? y   Network Level:
       Send ISDN Trunk Group Name on Tandem Calls? n   Level 2 Code:
                Send Custom Messages Through QSIG? n   Level 1 Code:
                          QSIG/ETSI TSC Extension: 59998
 MWI - Number of Digits Per Voice Mail Subscriber: 5

                              National CPN Prefix:
                         International CPN Prefix:
                                Pass Prefixed CPN: ASAI? n   VDN/Vector? n
      Delay for USNI Calling Name for Analog Caller ID Phones (seconds): 0
    Unknown Numbers Considered Internal for AUDIX? y   Maximum Length: 5
             USNI Calling Name for Outgoing Calls? n
               Path Replacement with Measurements? y
                  QSIG Path Replacement Extension: 59997
   Send QSIG Path Replacement Conf. Event to ASAI? y
            Path Replace While in Queue/Vectoring? N
</pre></td></tr>
</table>

5.4 DS1 Media Module Configuration

This section describes the steps for configuring the DS1 media module to support the ISDN-PRI QSIG trunk between the Communication Manager and the Mediant 1000 in the sample configuration shown in **Figure 1**.

<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>3.</td><td>Issue the command display media-gateway 1 to display the Media Gateway information. On Page 2, verify there is a DS1 media module (DS1 MM) available. If not, install the media module.

<pre>
display media-gateway 1   Page   2 of   2
                             MEDIA GATEWAY 1

                                 Type: g450

Slot   Module Type   Name   DSP Type FW/HW version
 V1:   S8300   ICC MM   MP80   45   3
 V2:   MM711   ANA MM
 V3:   MM712   DCP MM
 V4:   MM710   DS1 MM
 V5:   MM710   DS1 MM
 V6:
 V7:
 V8:   Max Survivable IP Ext: 8
 V9:   gateway-announcements ANN VMM
</pre>
</td></tr>
</table>

Step	Description
4.	Enter the **add ds1 xxxxx** command, where **xxxxx** is the carrier/slot location of the DS1 connected to the Mediant 1000 as shown in **Step 1**. For this configuration, location **001v5** was used. On **Page 1** of the **ds1** form, configure the following: • **Name** – Enter a meaningful description. • **Bit Rate** – Set to **1.544.** • **Line Coding** – Set to **b8zs.** • **Framing Mode** – Set to **esf.** • **Signaling Mode** – Set to **isdn-pri.** • **Connect** – Set to **pbx.** • **Interface:** – Set to **peer-master.** • **Peer Protocol:** – Set to **Q-SIG.** • **Side –** Set to **a.**

```
add ds1 001v5                                                  Page   1 of   2
                                DS1 CIRCUIT PACK

            Location: 001V5                         Name: ToAC
            Bit Rate: 1.544                  Line Coding: b8zs
   Line Compensation: 1                     Framing Mode: esf
      Signaling Mode: isdn-pri
             Connect: pbx                      Interface: peer-master
   TN-C7 Long Timers? n                    Peer Protocol: Q-SIG
Interworking Message: PROGress                      Side: a
Interface Companding: mulaw                         CRC? n
           Idle Code: 11111111
                              DCP/Analog Bearer Capability: 3.1kHz

                                           T303 Timer(sec): 4
                                          Disable Restarts? y

      Slip Detection? n                 Near-end CSU Type: other

   Echo Cancellation? N
```

5.5 ISDN-PRI QSIG Trunk and Signaling Group

This section creates the QSIG trunk and the signaling group associated with the trunk.

Step	Description
1.	Enter the **add trunk-group x** command, where **x** is an available trunk group number. Trunk group 58 is chosen for the QSIG trunk to the Mediant 1000. On **Page 1** of the trunk group form, configure the following: • **Group Type** – Set to **isdn.** • **Group Name** – Enter a meaningful name/description. • **TAC** – Enter a Trunk Access Code that is valid under the provisioned dial plan. • **Carrier Medium** – Set to **PRI/BRI.** • **Service Type** – Set to **tie.**
2.	On **Page 2**, set the **Supplementary Service Protocol** to **b**.

```
add trunk-group 58                                           Page   1 of  21
                                TRUNK GROUP

Group Number: 58                   Group Type: isdn          CDR Reports: y
  Group Name: ToAC                        COR: 1        TN: 1        TAC: *058
   Direction: two-way        Outgoing Display? n        Carrier Medium: PRI/BRI
 Dial Access? y              Busy Threshold: 255  Night Service:
Queue Length: 0
Service Type: tie                   Auth Code? n            TestCall ITC: rest
                         Far End Test Line No:
TestCall BCC: 4
```

```
add trunk-group 58                                             Page   2 of  21
      Group Type: isdn

TRUNK PARAMETERS
         Codeset to Send Display: 6     Codeset to Send National IEs: 6
        Max Message Size to Send: 260   Charge Advice: none
  Supplementary Service Protocol: b     Digit Handling (in/out): enbloc/enbloc

            Trunk Hunt: cyclical
                                                   Digital Loss Group: 13
Incoming Calling Number - Delete:     Insert:                 Format:
              Bit Rate: 1200        Synchronization: async    Duplex: full
 Disconnect Supervision - In? y  Out? y
 Answer Supervision Timeout: 0
          Administer Timers? n        CONNECT Reliable When Call Leaves ISDN? n
             XOIP Treatment: auto    Delay Call Setup When Accessed Via IGAR? n
```

Step	Description
3.	On **Page 3**, set the following: • Set the **NCA-TSC Trunk Member** to one of the trunk members/channels configured in **Step 6**. • Set **Send Name** to **y**. • Set **Send Calling Number** to **y**. • Set **Send Connected Number** to **y**.
4.	On **Page 4** of the trunk group form, verify that **Path Replacement** is set to **y**.

Step 3 screen:

```
add trunk-group 58                                            Page   3 of  21
TRUNK FEATURES
          ACA Assignment? n            Measured: none      Wideband Support? n
                                 Internal Alert? n       Maintenance Tests? y
                               Data Restriction? n     NCA-TSC Trunk Member: 3
                                      Send Name: y      Send Calling Number: y
            Used for DCS? n             Hop Dgt? n     Send EMU Visitor CPN? n
   Suppress # Outpulsing? n    Format: public
 Outgoing Channel ID Encoding: preferred     UUI IE Treatment: shared
                                         Maximum Size of UUI IE Contents: 128
                                               Replace Restricted Numbers? n
                                              Replace Unavailable Numbers? n
                                                    Send Connected Number: y
                                                Hold/Unhold Notifications? y
             Send UUI IE? y     Modify Tandem Calling Number: no
               Send UCID? n              BSR Reply-best DISC Cause Value: 31
 Send Codeset 6/7 LAI IE? y                        Ds1 Echo Cancellation? n

    Apply Local Ringback? n
 Show ANSWERED BY on Display? y
```

Step 4 screen:

```
add trunk-group 58                                            Page   4 of  21
                       QSIG TRUNK GROUP OPTIONS



   TSC Method for Auto Callback: drop-if-possible
           Diversion by Reroute? y
               Path Replacement? y
Path Replacement with Retention? n
        Path Replacement Method: better-route
                           SBS? n
  Display Forwarding Party Name? y
    Character Set for QSIG Name: eurofont
               QSIG Value-Added? n
```

Step	Description
5.	Enter the **add signaling-group x** command, where **x** is an available signaling group number. On **Page 1** of the signaling group form, configure the following: • **Group Type** – Set to **isdn-pri**. • **Associated Signaling** – Set to **y**. • **Primary D-Channel** – Enter **xxxxx24**, where **xxxxx** is the location of the DS1 media module configured in **Section 5.4, Step 4** and connected to the Mediant 1000 (24 is the D-Channel in a T1 ISDN-PRI). • **Trunk Group for Channel Selection** – Enter the number of the trunk group configured in **Step 1**. • **TSC Supplementary Service Protocol:** – **b** • **Max number of NCA TSC: -- 12** • **Max number of CA TSC: -- 12** • **Trunk Group for NCA TSC: --** Enter the number of the trunk group configured in **Step 1.**

```
add signaling-group 58                                          Page   1 of   1
                                SIGNALING GROUP

 Group Number: 58               Group Type: isdn-pri
                     Associated Signaling? y          Max number of NCA TSC: 12
                        Primary D-Channel: 001V524     Max number of CA TSC: 12
                                                     Trunk Group for NCA TSC: 58
       Trunk Group for Channel Selection: 58      X-Mobility/Wireless Type: NONE
      TSC Supplementary Service Protocol: b          Network Call Transfer? N
```

Step	Description
6.	After the signaling group has been created, return to the trunk group form using the **change trunk-group** command used in **Step 1**. On **Page 5** of the trunk group form, add 10 or more trunk members by entering the following: • In the **Port** column, enter a value **xxxxxzz**, where **xxxxx** is the location of the DS1 media module configured in **Section 5.4, Step 4** and **zz** is a channel in the T1 ISDN-PRI trunk. • In the **Sig Grp** column, enter the signaling group configured in **Step 3**.

```
change trunk-group 58                                           Page   5 of  21
                                    TRUNK GROUP
                                         Administered Members (min/max):   1/23
GROUP MEMBER ASSIGNMENTS                     Total Administered Members:  23

       Port    Code Sfx Name         Night            Sig Grp
  1: 001V501  MM710     b                               58
  2: 001V502  MM710     b                               58
  3: 001V503  MM710     b                               58
  4: 001V504  MM710     b                               58
  5: 001V505  MM710     b                               58
  6: 001V506  MM710     b                               58
  7: 001V507  MM710     b                               58
  8: 001V508  MM710     b                               58
  9: 001V509  MM710     b                               58
 10: 001V510  MM710     b                               58
 11: 001V511  MM710     b                               58
 12: 001V512  MM710     b                               58
 13: 001V513  MM710     b                               58
 14: 001V514  MM710     b                               58
 15:
```

5.6 Configure Route Pattern

A route pattern is configured to use the trunk defined in **Section 5.5, Step 1**. 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 to the route pattern defined below.

Step	Description
1.	To configure a route pattern, use the **change route-pattern x** command, where **x** is an available route pattern number. For the compliance test, route pattern 58 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 5.5**, **Step 1**. • 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. • Set **TSC** to **y**. • Default values may be used for all other fields.

```
change route-pattern 58                                         Page   1 of   3
                    Pattern Number: 58  Pattern Name: toAC
                             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: 58   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  y  none      rest                                           none
 2: y y y y y n  n            rest                                           none
 3: y y y y y n  n            rest                                           none
 4: y y y y y n  n            rest                                           none
 5: y y y y y n  n            rest                                           none
 6: y y y y y n  n            rest                                           none
```

5.7 Configure Automatic Alternate Routing

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

<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>1.</td><td>To create 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 for each mobile user extension supported by the configuration in Figure 1. This includes mobile extension 55684 even though this device is not associated with an enterprise desktop extension. In addition, a DNIS call-back number must be assign 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 Mediant 1000. In the example below, this number is 9088485683. 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 5.6 that directs calls to the trunk connected to the Mediant 1000.
Set the Call Type to aar.

<pre>
change aar analysis 5   Page   1 of   2
                             AAR DIGIT ANALYSIS TABLE
                                   Location: all   Percent Full: 3

          Dialed   Total   Route   Call   Node ANI
          String   Min Max Pattern   Type   Num   Reqd
    5   5   5   1   aar   n
    55681   5   5   58   aar   n
    55682   5   5   58   aar   n
    55684   5   5   58   aar   n
    59999   5   5   99   aar   n
    6   7   7   254   aar   n
    7   7   7   254   aar   n
    8   7   7   254   aar   n
    9088485683   10   10   58   aar   n
</pre>
</td></tr>
</table>

5.8 Incoming Call Treatment for the PSTN Trunk

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

Step	Description
1.	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 56** command to create the entries in the example below. Trunk group 56 is used because this is the trunk group connected to the PSTN as shown in **Figure 1**. The DID numbers 9088485681, 9088485682 and 9088485684 are associated with the internal extensions 55681, 55682 and 55684 respectively. By deleting all 10 digits of these numbers and inserting the internal extension, the inbound DID is converted to an internal extension. The DID number 9088485683 is the DNIS call-back number so it is prepended with a 3 (the AAR prefix) to route this call to AAR for further processing. The parameters in the table are defined as follows: • Set the **Service/Feature** to **tie**. • 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 56                  Page   1 of   3
                       INCOMING CALL HANDLING TREATMENT
 Service/      Number   Number      Del Insert          Per Call Night
 Feature        Len      Digits                         CPN/BN   Serv
 tie            10  9088485681      10   55681
 tie            10  9088485682      10   55682
 tie            10  9088485683           3
 tie            10  9088485684      10   55684
 tie
```

5.9 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 55681, 55682 and 55684 were configured on Communication Manager but extensions 55681 and 55682 had physical phones logged in to them and extension 55684 did not. (See **Figure 1)**.

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

<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>1.</td><td>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 (such as extension x55684 in Figure 1), then a station must be added.

Use the add station 55684 command to create the station for this user.

<pre>
add station 55684   Page   1 of   5
                                     STATION

Extension: 55684   Lock Messages? n   BCC: 0
     Type: 9640   Security Code: 123456   TN: 1
     Port: IP   Coverage Path 1: 99   COR: 1
     Name: RIM3 Test   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: 55684
            Speakerphone: 2-way   Mute Button Enabled? y
        Display Language: english   Button Modules: 0
 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
</pre></td></tr>
</table>

Step	Description

2. 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 55684                                                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                           5:
 2: call-appr                           6:
 3: call-appr                           7:
 4: ec500      Timer? n                 8:

    voice-mail
```

3. 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.
- 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 55681                    Page   1 of   3
                  STATIONS WITH OFF-PBX TELEPHONE INTEGRATION

 Station       Application Dial   CC  Phone Number    Trunk       Config  Dual
 Extension                 Prefix                     Selection   Set     Mode
 55681            EC500         -     55681            aar        5
 55682            EC500               55682            aar        5
 55684            EC500               55684            aar        5
                                -
```

Step	Description
4.	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: for 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
```

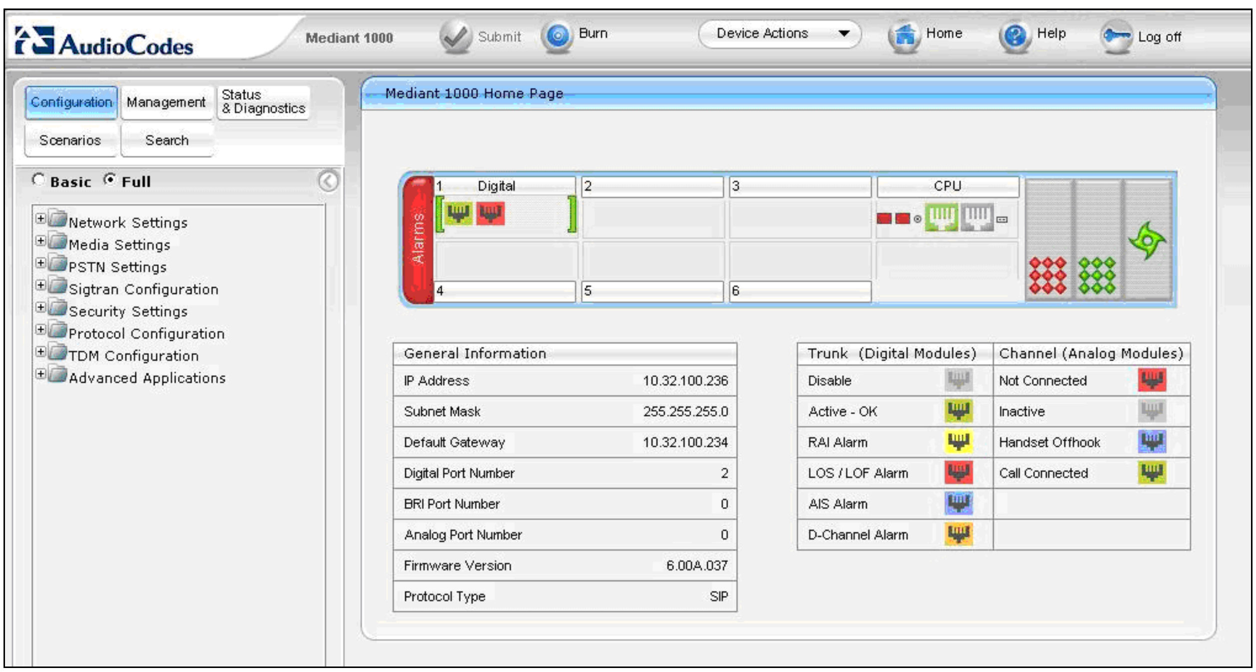
6 Configure AudioCodes Mediant 1000 VoIP Media Gateway

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:

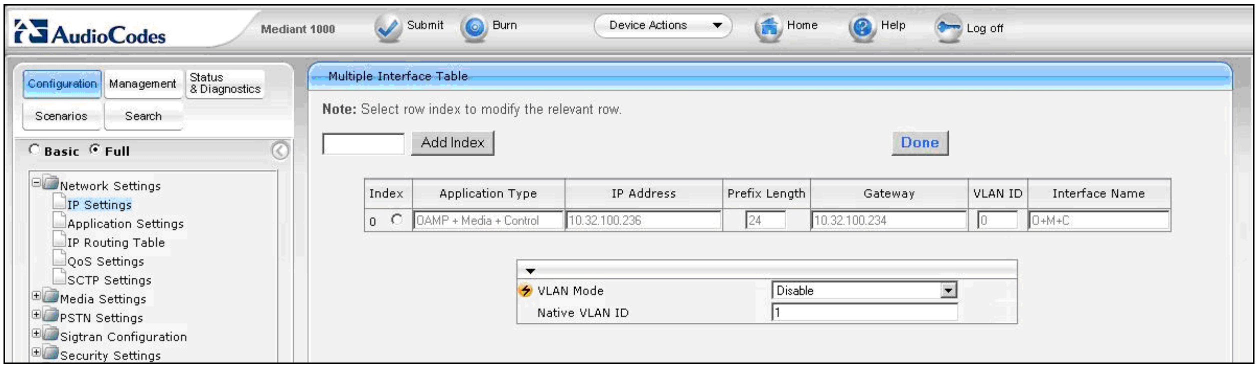
- Network IP settings
- Media Settings
- PSTN trunk settings
- Protocol Configuration
- Protocol Definition
- Coders and Profile
- Manipulation Table
- Routing Table

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.



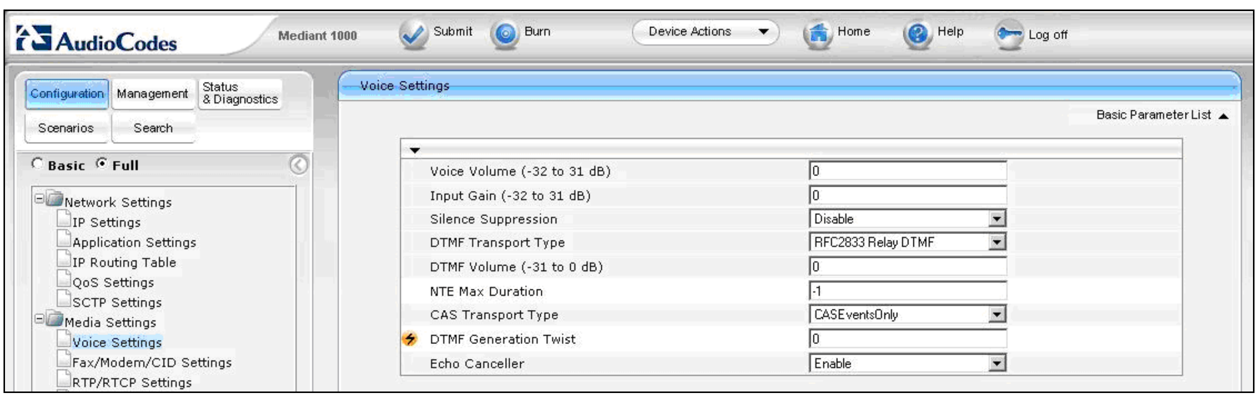
6.1 Network IP settings

The network settings that were configured during installation can be viewed by navigating to **Network Settings → 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 (not shown). 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**.



6.2 Media Settings

Navigate to **Media Settings → Voice Settings**. For **DTMF Transport Type**, select ***RFC2833 Relay DTMF***, set the **DTMF Volume** setting to ***0***. Default values may be retained for all other fields.

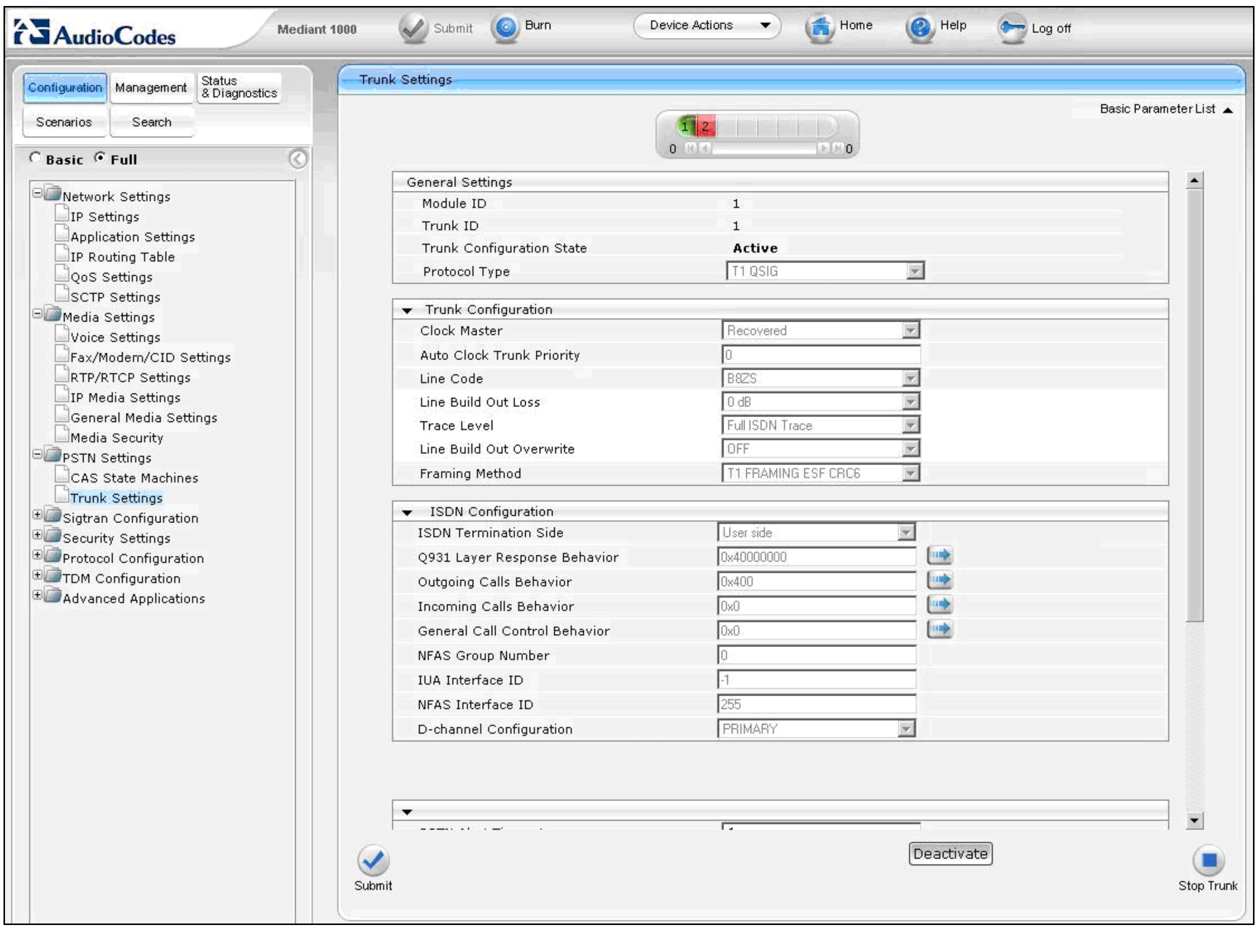


6.3 PSTN Settings

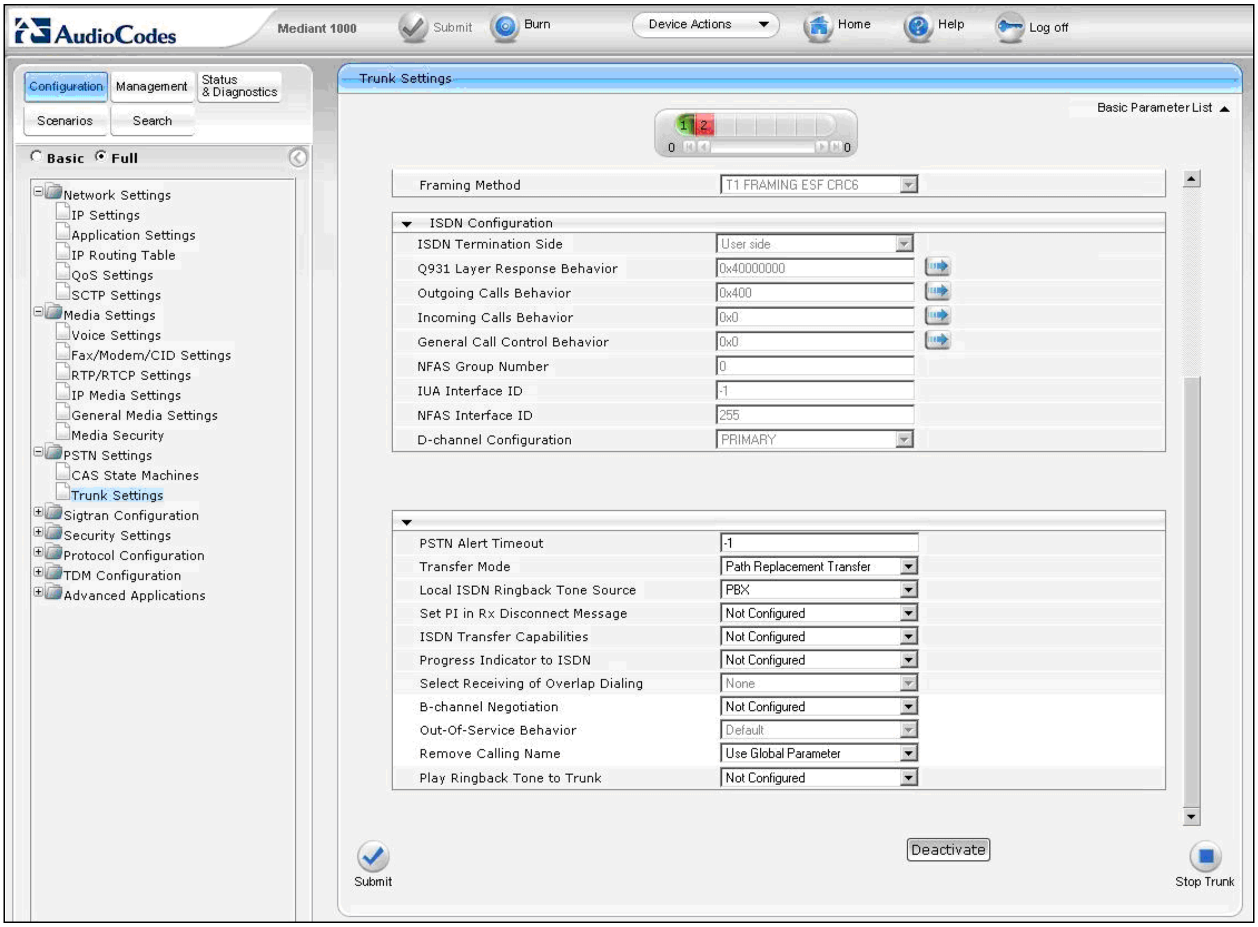
Navigate to **PSTN Settings → Trunk Settings** to configure the line side T1 interface to Communication Manager. These settings must be consistent with the DS1 settings on Communication Manager (**Section 5.5**). Configure the following parameters.

- Set **Protocol Type** to ***T1 QSIG***.
- Set **Line Code** to ***B8ZS***.
- Set **Framing Method** to ***T1 FRAMING ESF CRC6***.

Default values may be retained for all other fields. Click the **Apply Trunk Settings** icon button at the bottom of the screen (not shown). Use the scroll bar to continue.



Continued from previous page.

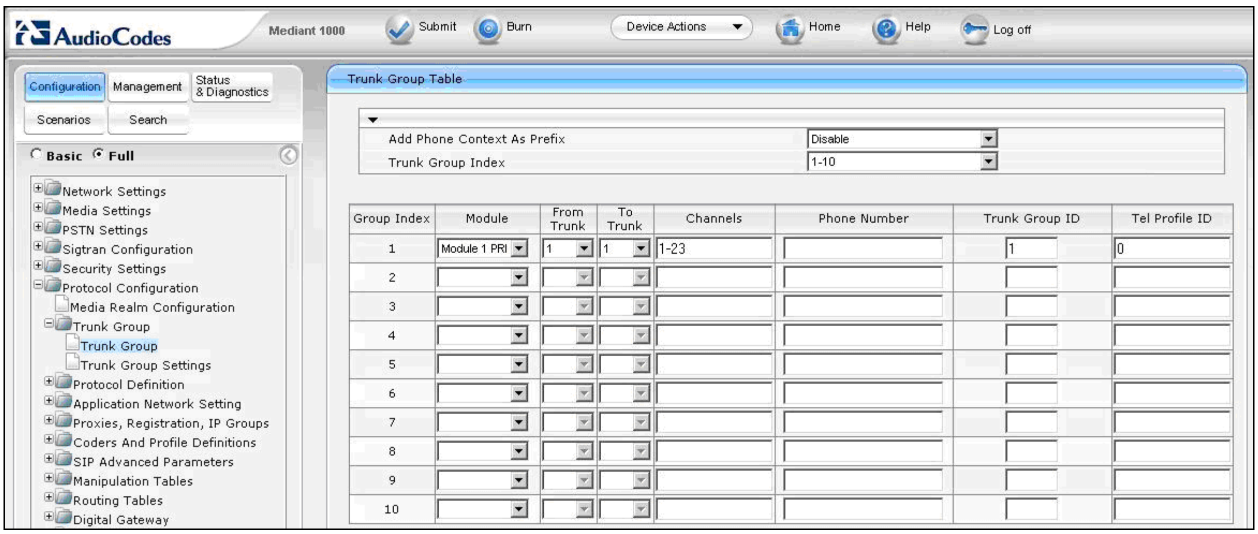


6.4 Protocol Configuration

6.4.1 Trunk Group

Navigate to **Protocol Configuration → Trunk Group → Trunk Group.** The Trunk Group Table is used to configure the call routing settings for the trunk that is configured to the BlackBerry® MVS Server.

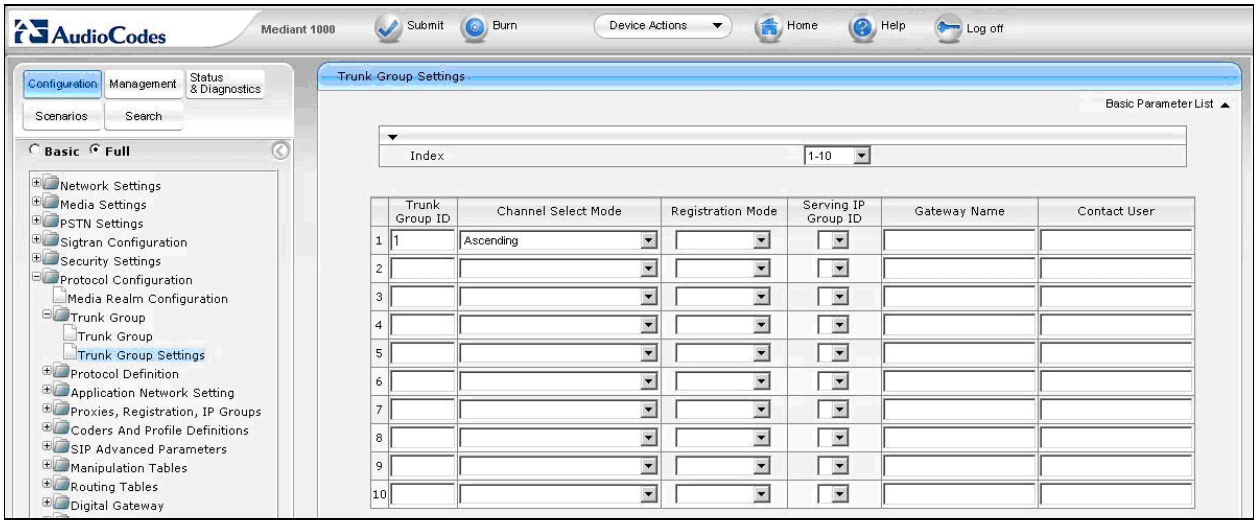
In the **From Trunk** and **To Trunk** columns, enter the starting and ending trunks to be assigned. In the **Channel(s)** column, enter the range of channels on those trunks to be assigned. The setting ***1-23*** means 23 channels are assigned to each trunk as defined in the **From Trunk** and **To Trunk** columns. The default value may be used for all other settings.



6.4.2 Trunk Group Settings

Navigate to **Protocol Configuration → Trunk Group → Trunk Group Settings**. Configure the parameters as described below.

- For **Trunk Group ID**, enter ***1*** as configured for Trunk Group.
- Select the **Channel Select Mode** to use a **hunt order that is opposite the hunt order used by the PBX to avoid glare conditions**. The channels in this trunk group are treated as a pool and can be selected to use an ascending or descending order. The screenshot below shows that for the compliance test, the **Channel Select Mode** was selected as **Ascending.** However, as a general recommendation is to best to use **Descending** order since the PBX will generally use an ascending order by default.



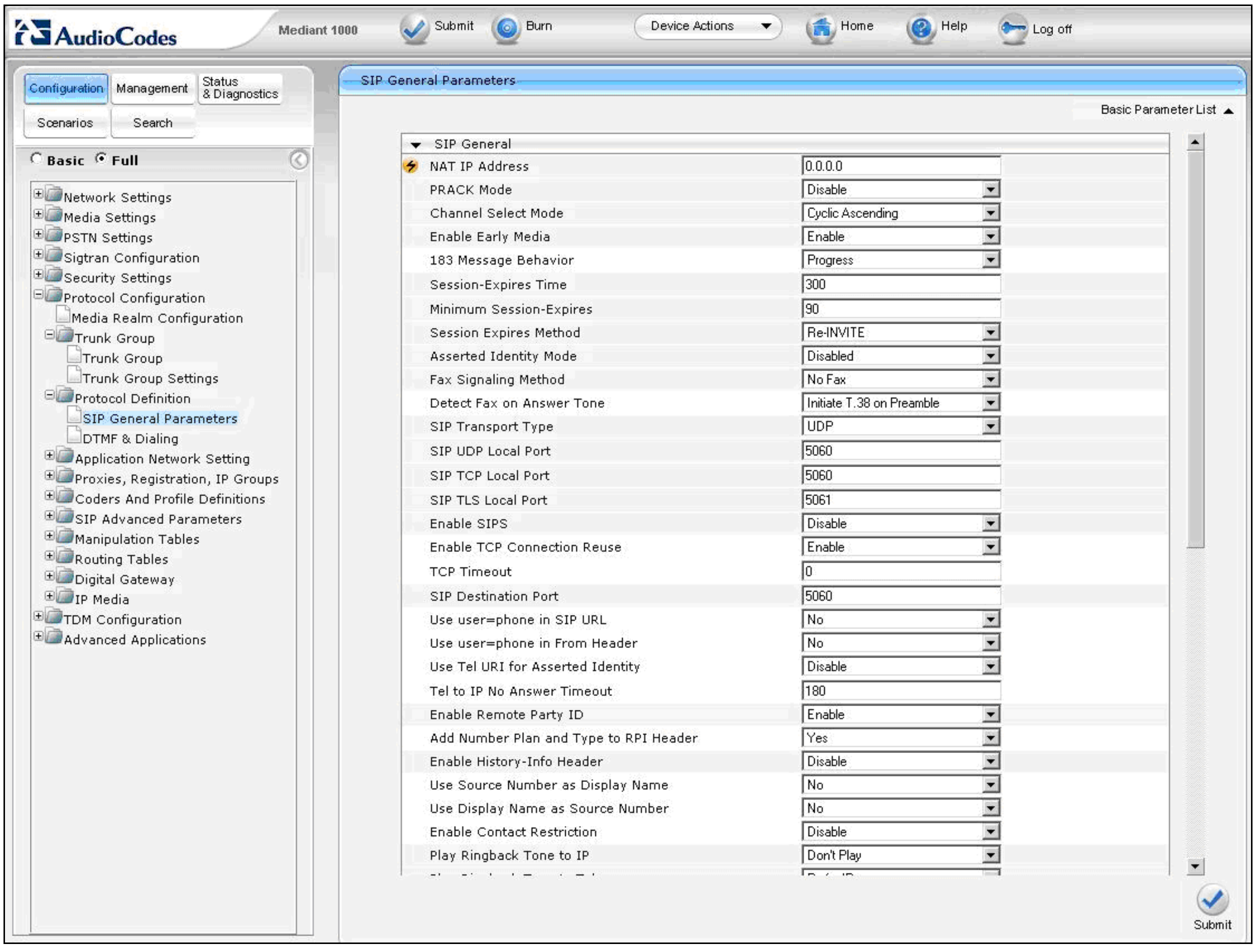
6.5 Protocol Definition

6.5.1 SIP General Parameters

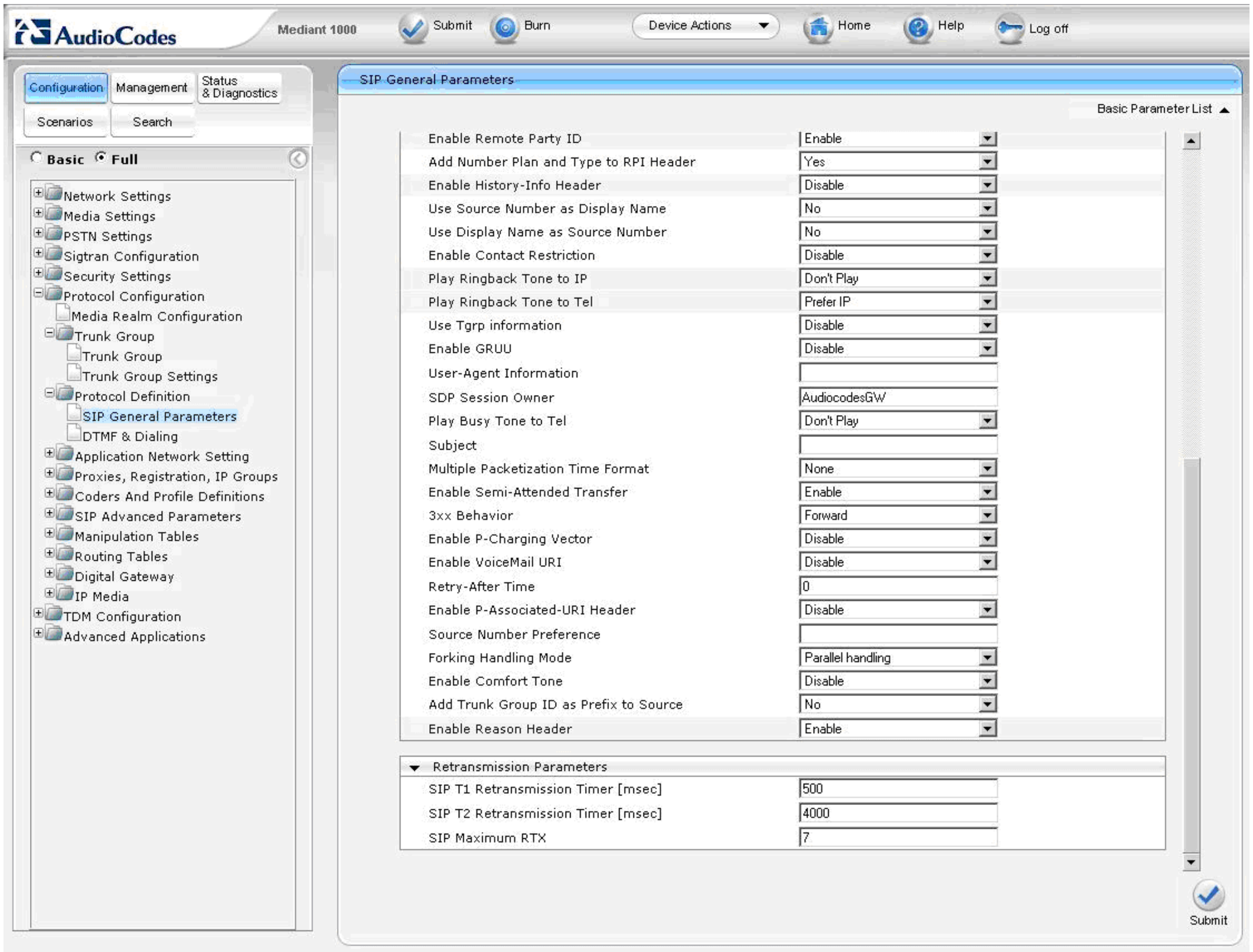
Navigate to **Protocol Configuration → Protocol Definition → SIP General Parameters**. Configure the parameters as described below.

- 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. Use the scroll bar to continue.



Continued from previous page.

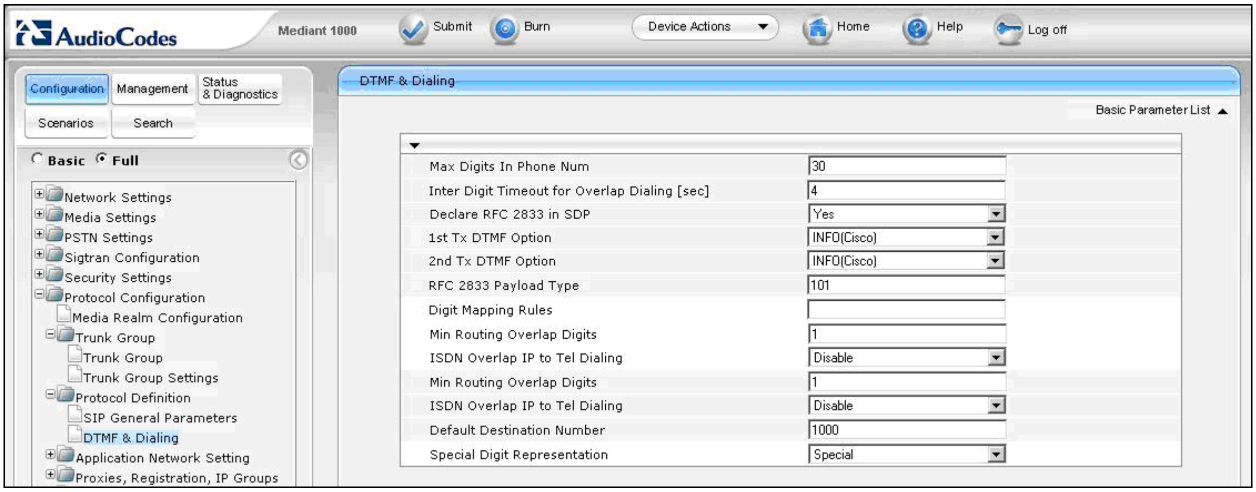


6.5.2 DTMF and Dialing

Navigate to **Protocol Configuration → Protocol Definition → 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.

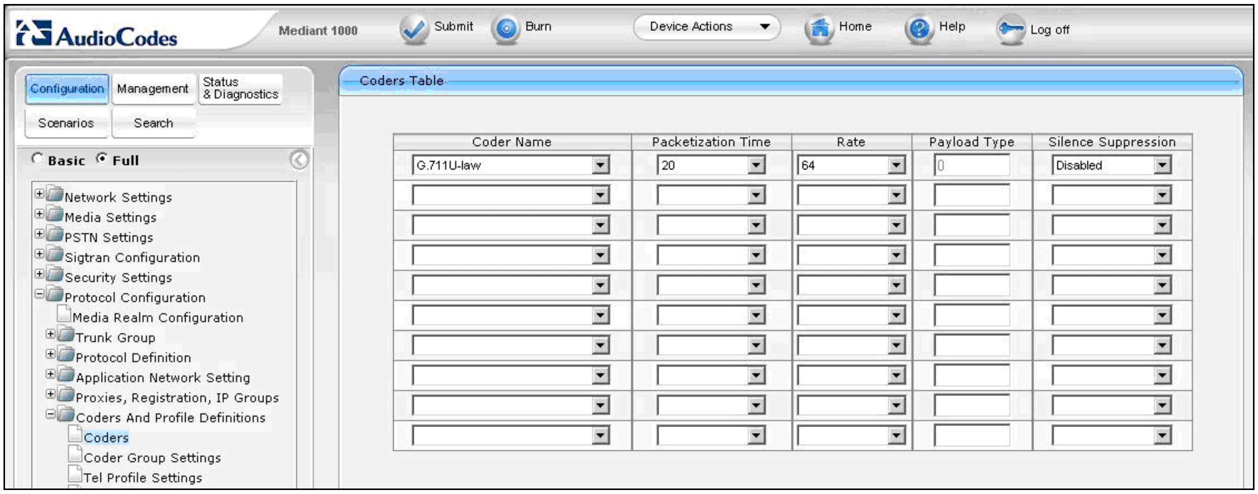


6.6 Coders and Profile Definitions

6.6.1 Coders

Navigate to **Protocol Configuration → Coders and Profile Definitions → 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.

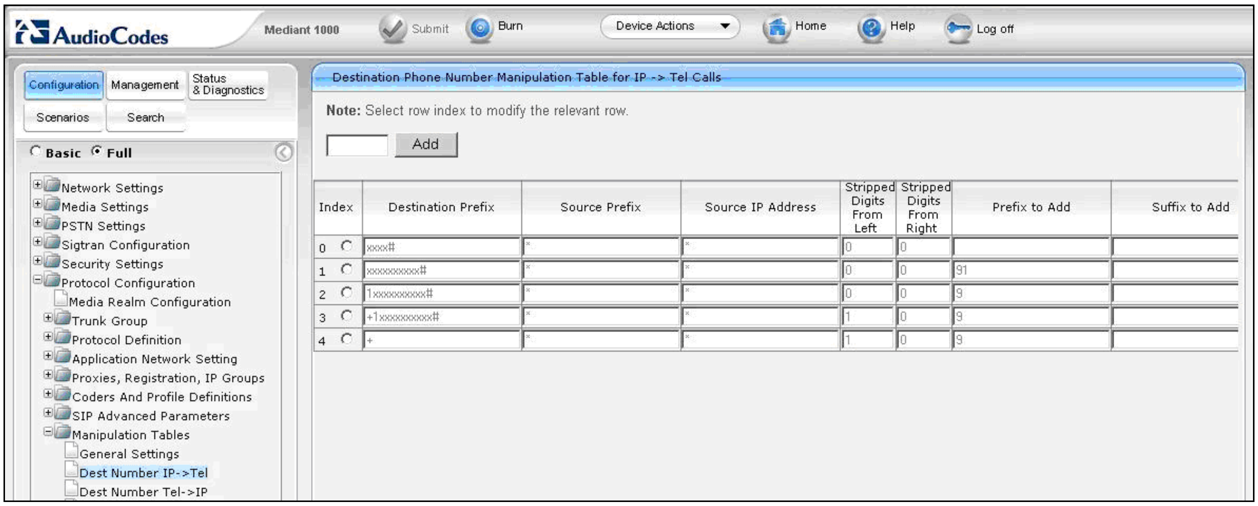
The codec list used during compliance testing is shown in the example below. ***G.711U-law*** was selected as the most preferred codec. Default values were retained for all other fields.



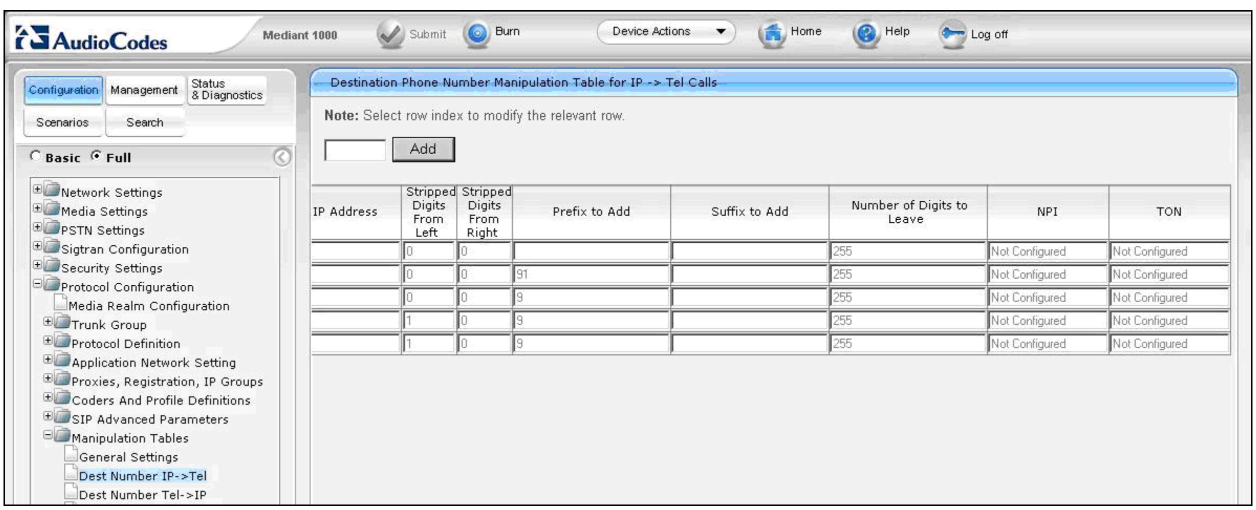
6.7 Manipulation Tables

6.7.1 Dest Number IP to Tel

Navigate to **Protocol Configuration → Manipulation Tables → 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**.**

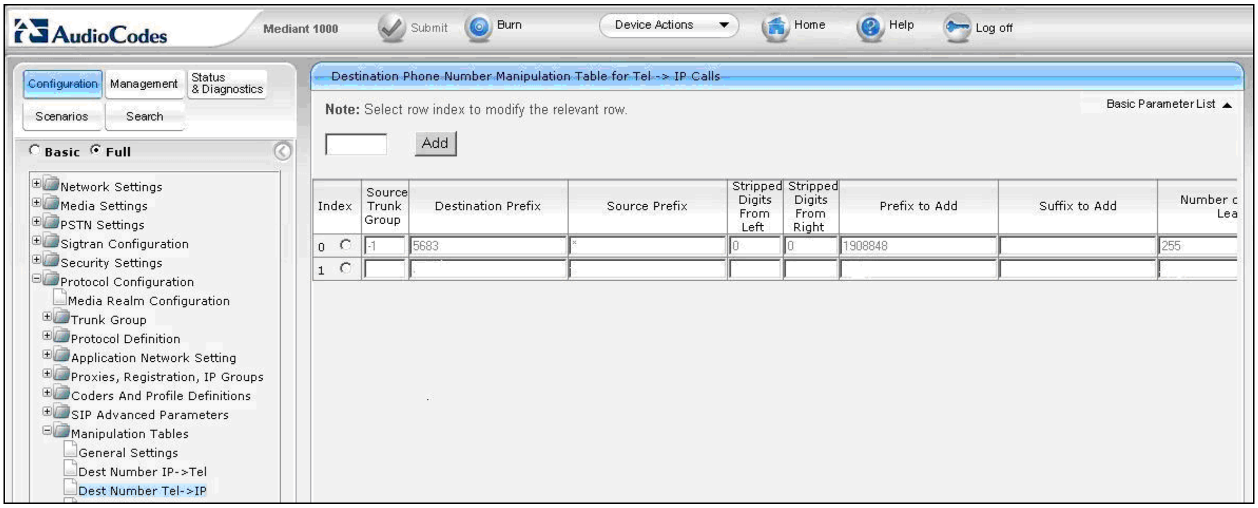


Scroll to the right to see the remaining fields.

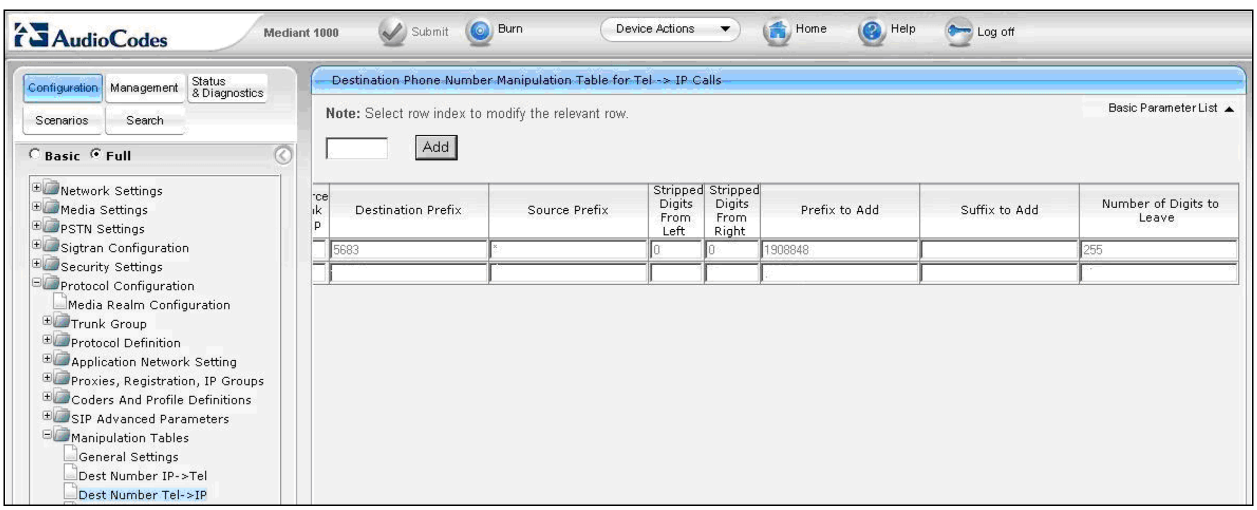


6.7.2 Dest Number Tel to IP

Navigate to **Protocol Configuration → Manipulation Tables → 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.)



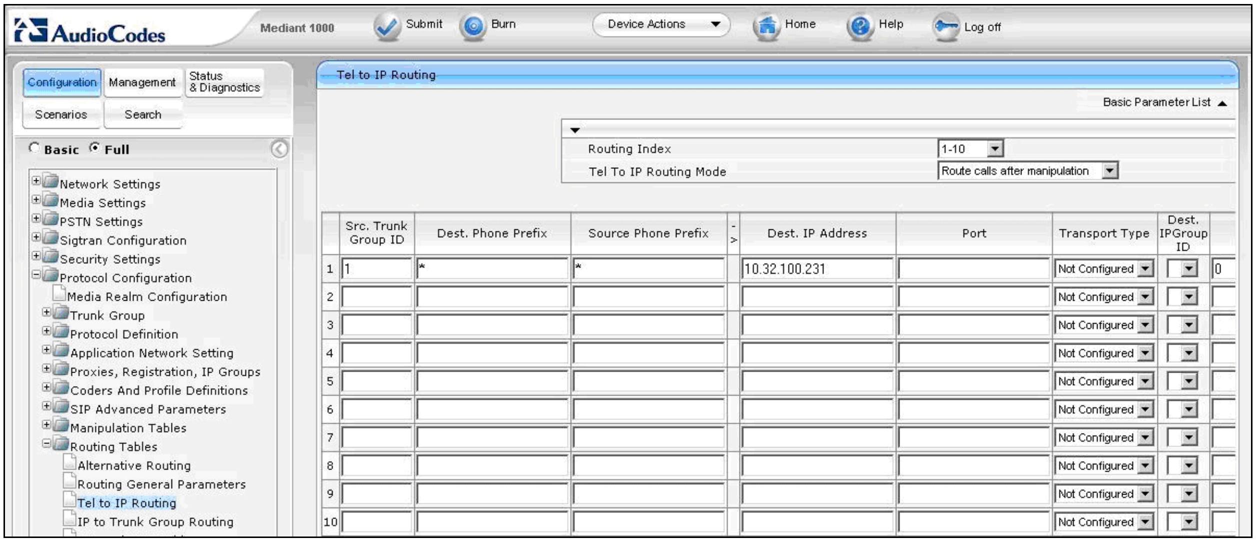
Scroll to the right to see the remaining fields.



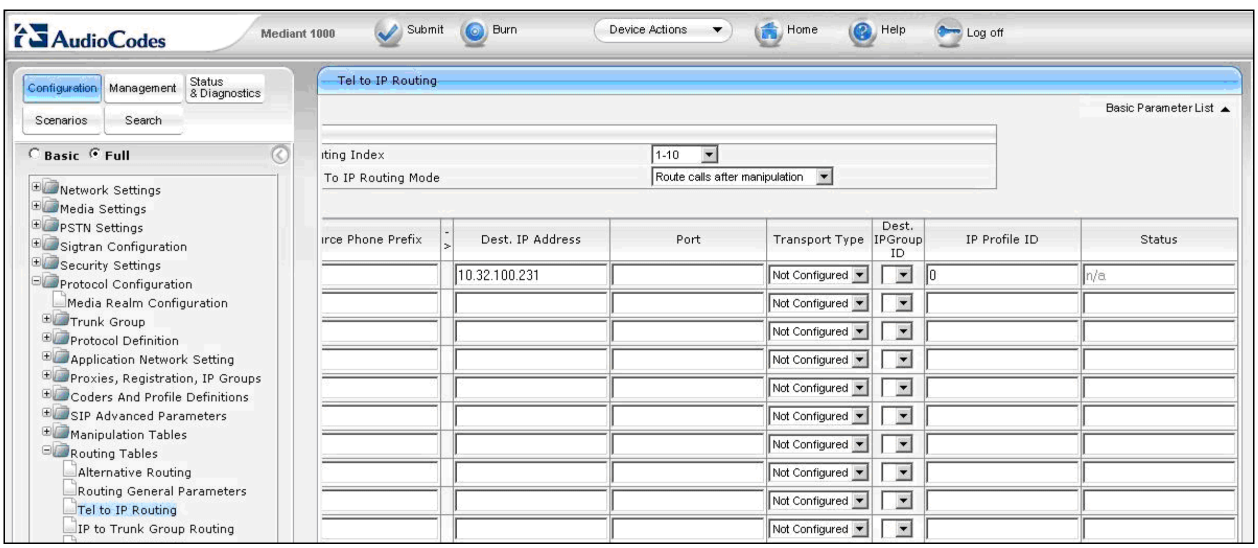
6.8 Routing Table

6.8.1 Tel to IP Routing

Navigate to **Protocol Configuration → Routing Tables → Tel to IP Routing**. Use these settings to routes phone calls to the BlackBerry® MVS Server.

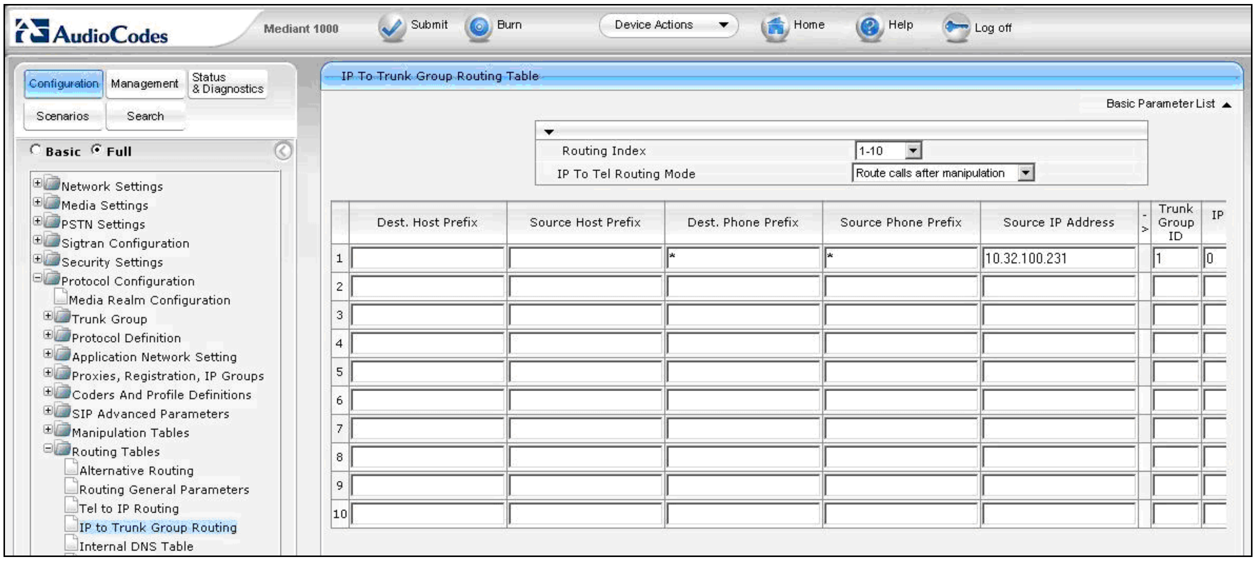


Scroll to the right to see the remaining fields.

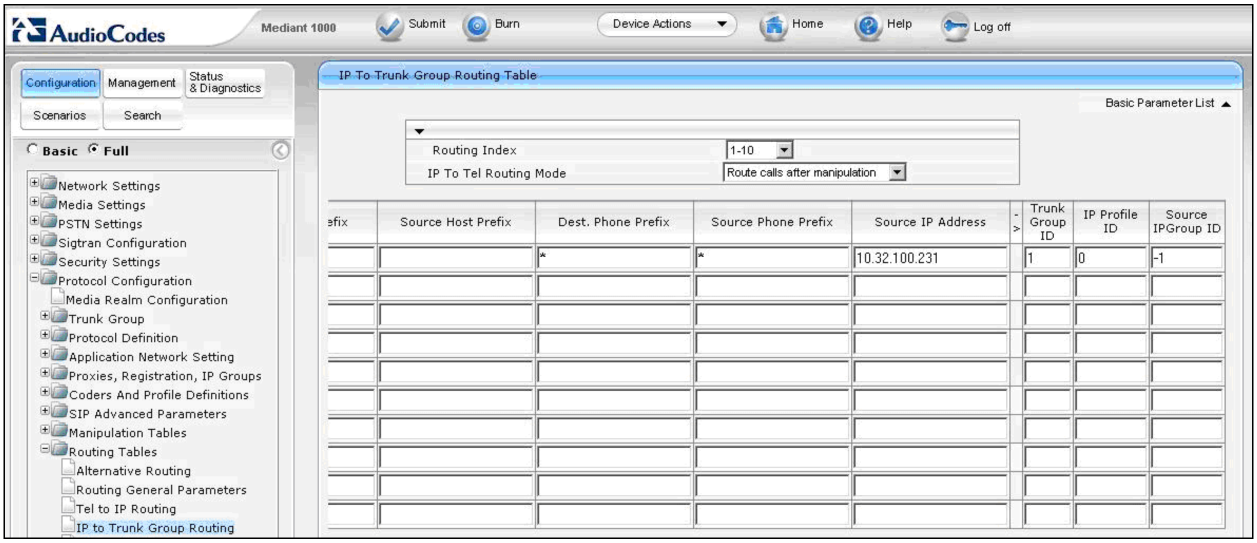


6.8.2 IP to Trunk Group Routing

Navigate to **Protocol Configuration → Routing Tables → 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.



6.9 Additional settings for the ini file

In the AudioCodes UI (<IP address of AudioCodes gateway/AdminPage>), add the following entry. These settings allow the user to hear DTMF tones during hold. (The Parameter Name is not case sensitive).

- **PlayDTMFduringHold 1**
- **TrunkTransferMode 2**
- **IsdnIBehaviour 1073741824**
- **EnableEarlyMedia 1**
- **ProgressIndicator2Ip 0**

The **IsdnIBehaviour** parameter setting is required for call transfer interoperability.

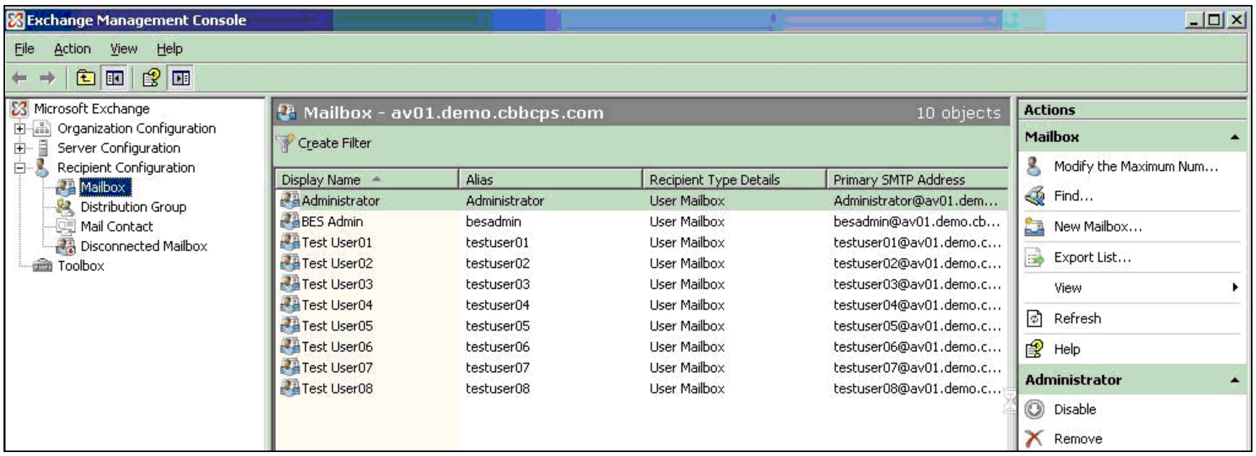
The **Burn and Restart** command from the main screen needs to be executed to apply the above parameter changes.

7 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.

7.1 Microsoft Exchange Server

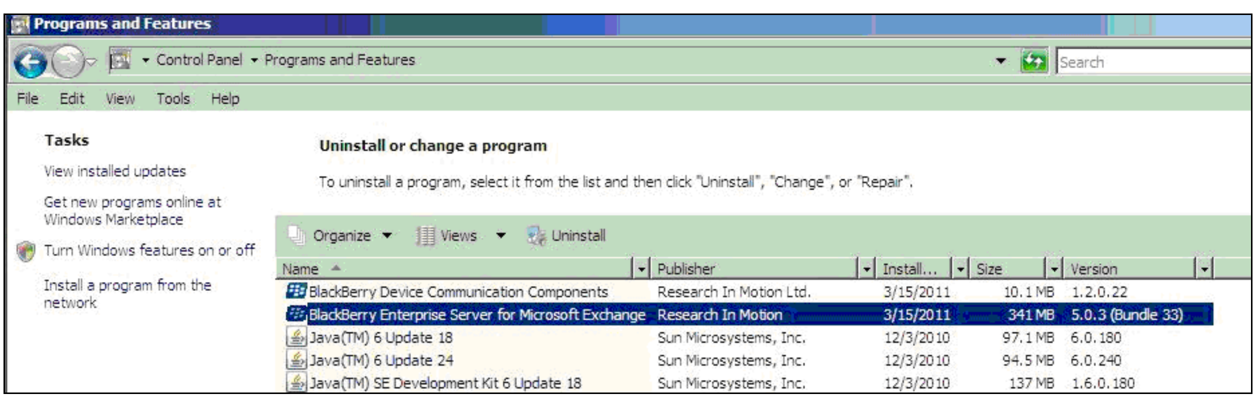
Verify that each user that will be configured on the MVS is also configured as a user within Microsoft Exchange. Log in to the Exchange Management Console. Navigate to **Microsoft Exchange → Recipient Configuration → Mailbox** to view the existing users.



7.2 BlackBerry® Enterprise Server Configuration

7.2.1 Verify Software Version

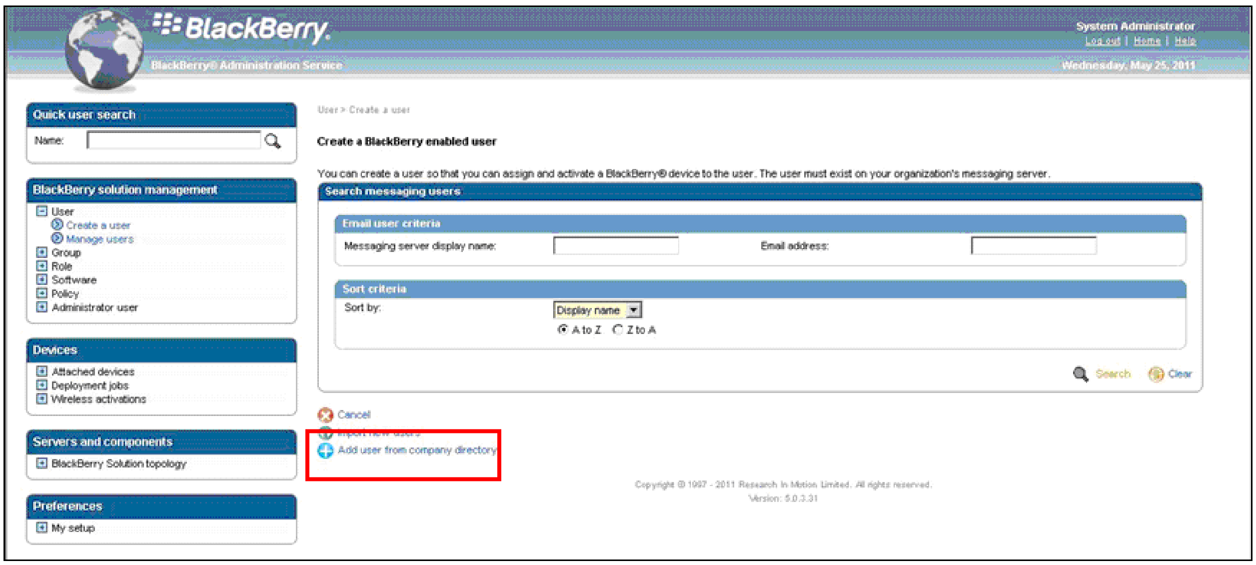
Log in to the Blackberry® Enterprise Server running on Microsoft Windows 2008 Server. From the Windows menus, navigate to **Control Panel → Programs and Features**. A list of installed programs will be displayed. Verify that ***Blackberry Enterprise Server for Microsoft Exchange*** is listed along with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



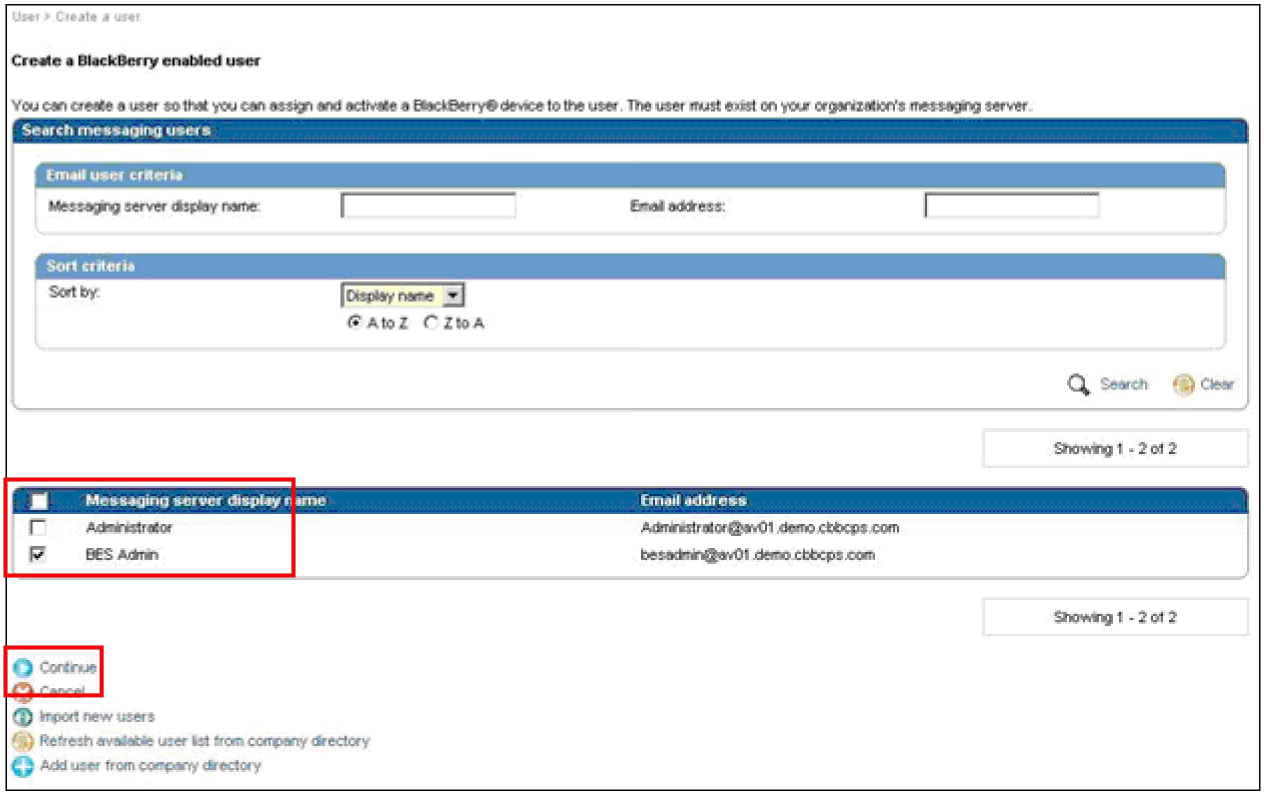
7.2.2 Create Users

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.

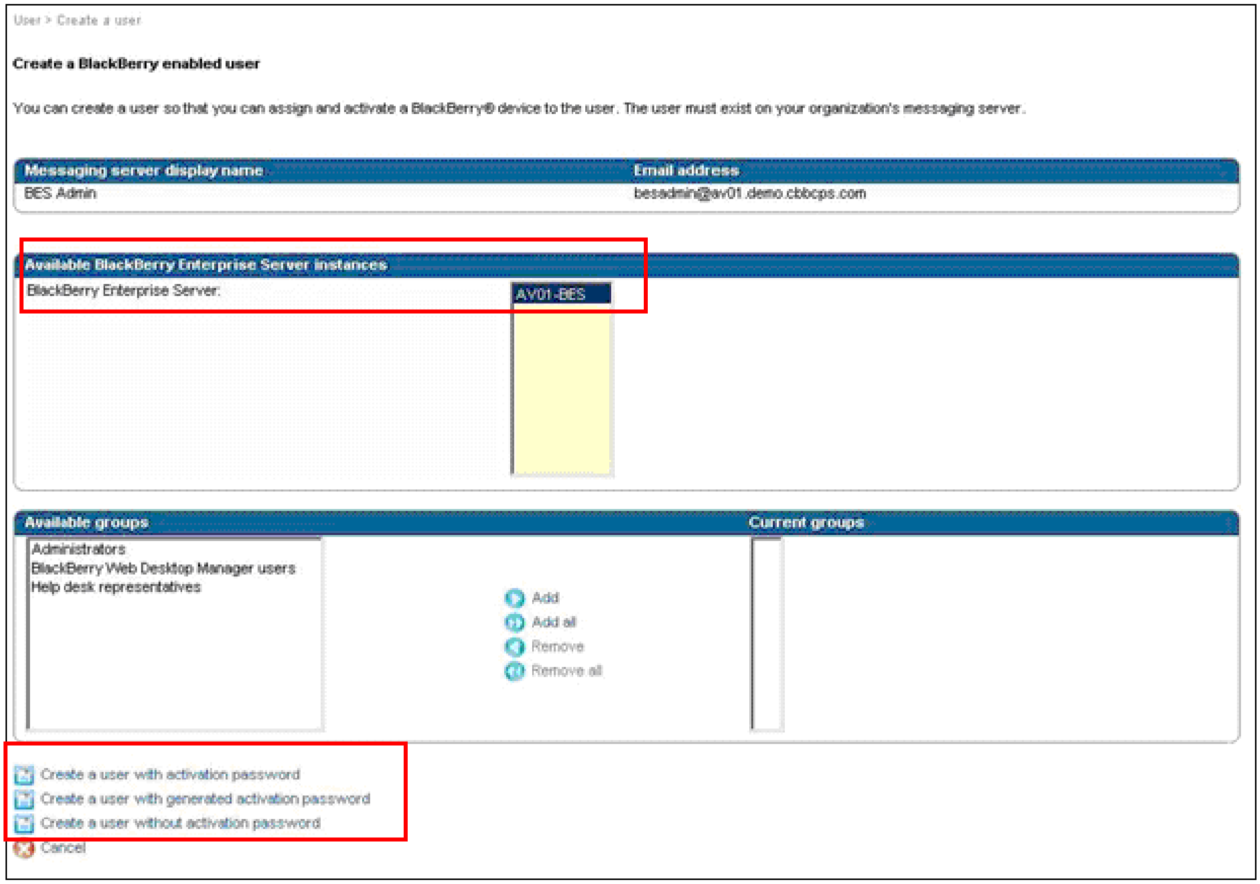
To create a user account, navigate to **Blackberry solution management → Users → Create a user**. The **Create a Blackberry enabled user** screen appears as shown below. The easiest way to add a new user is to add a user from the corporate directory. To do this, click the **Add user from company directory** link 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 7.1** have already been added, thus only the users named **Administrator** and **BES Admin** are shown as available users to be added. Select a user to add by clicking the box next to the user name. Click **Continue**.



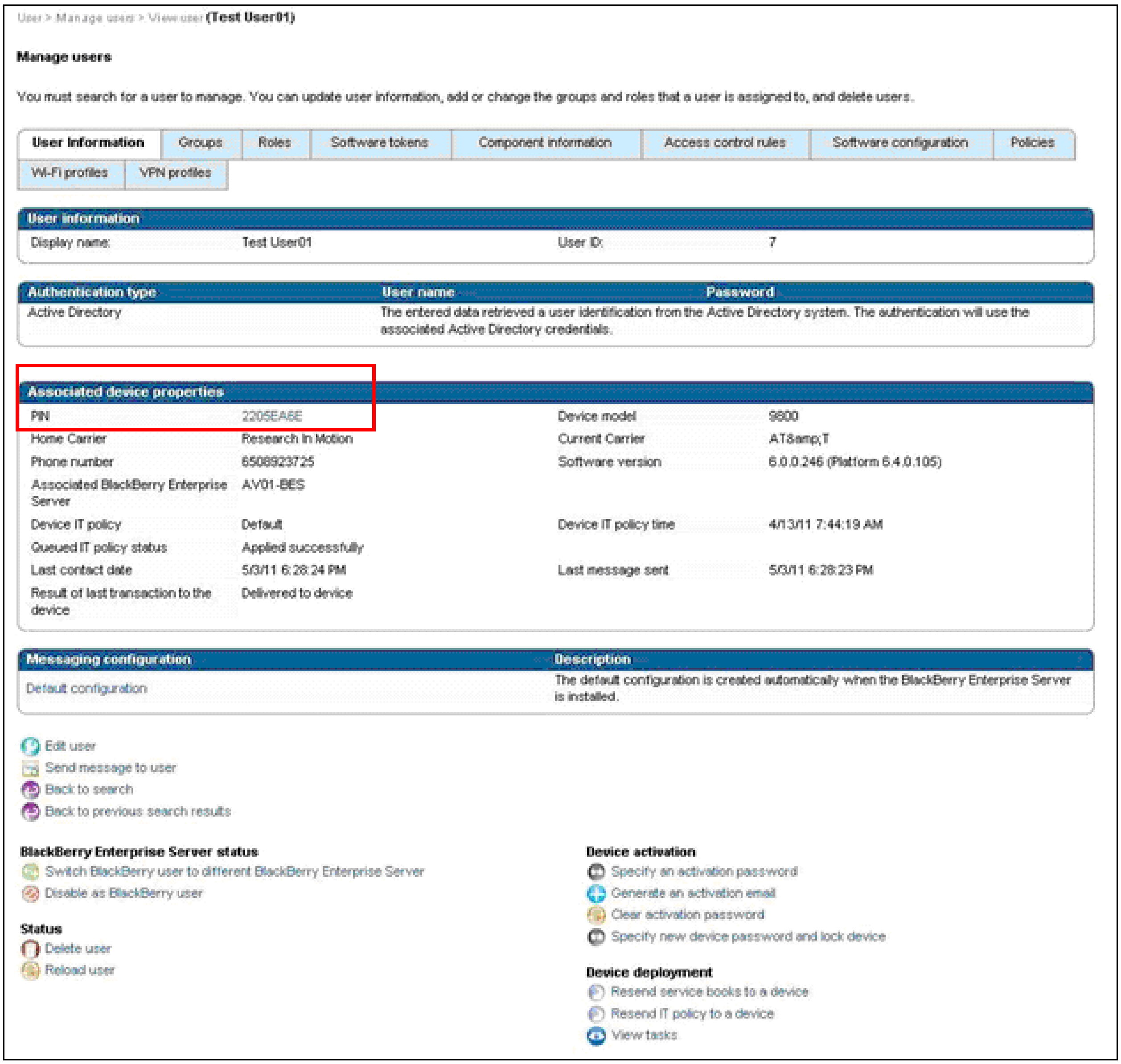
In the next screen, select an available **Blackberry Enterprise Server** to associate with this user from those listed in the middle of the screen. In the case of the compliance test, there was only one server **AV01-BES** to select. Click one of the create options listed at the bottom of the page to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



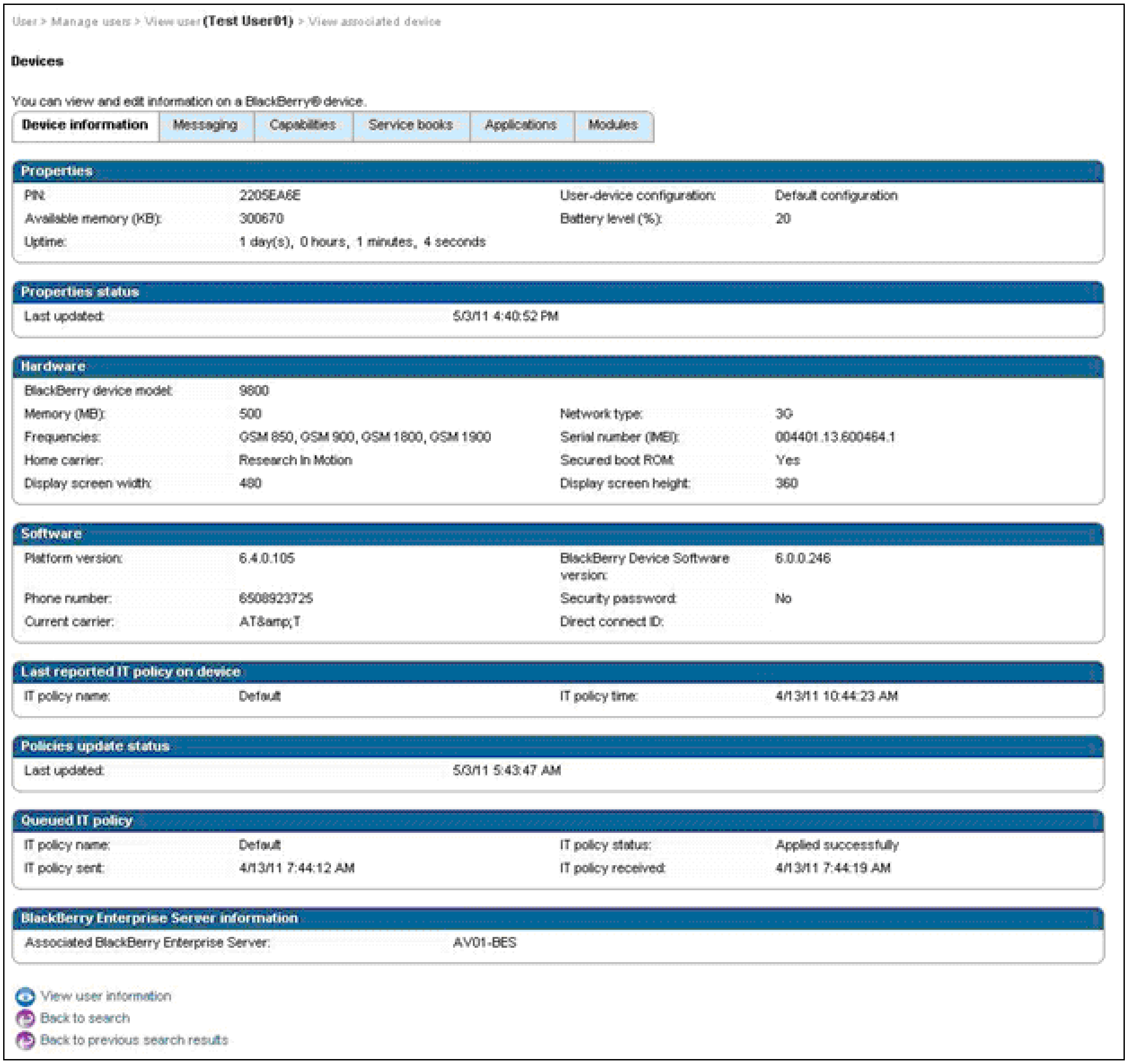
7.2.3 Manage Users

After a user is created, a user account may be modified by navigating to **Blackberry solution management → Users → Manage a user**. 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.



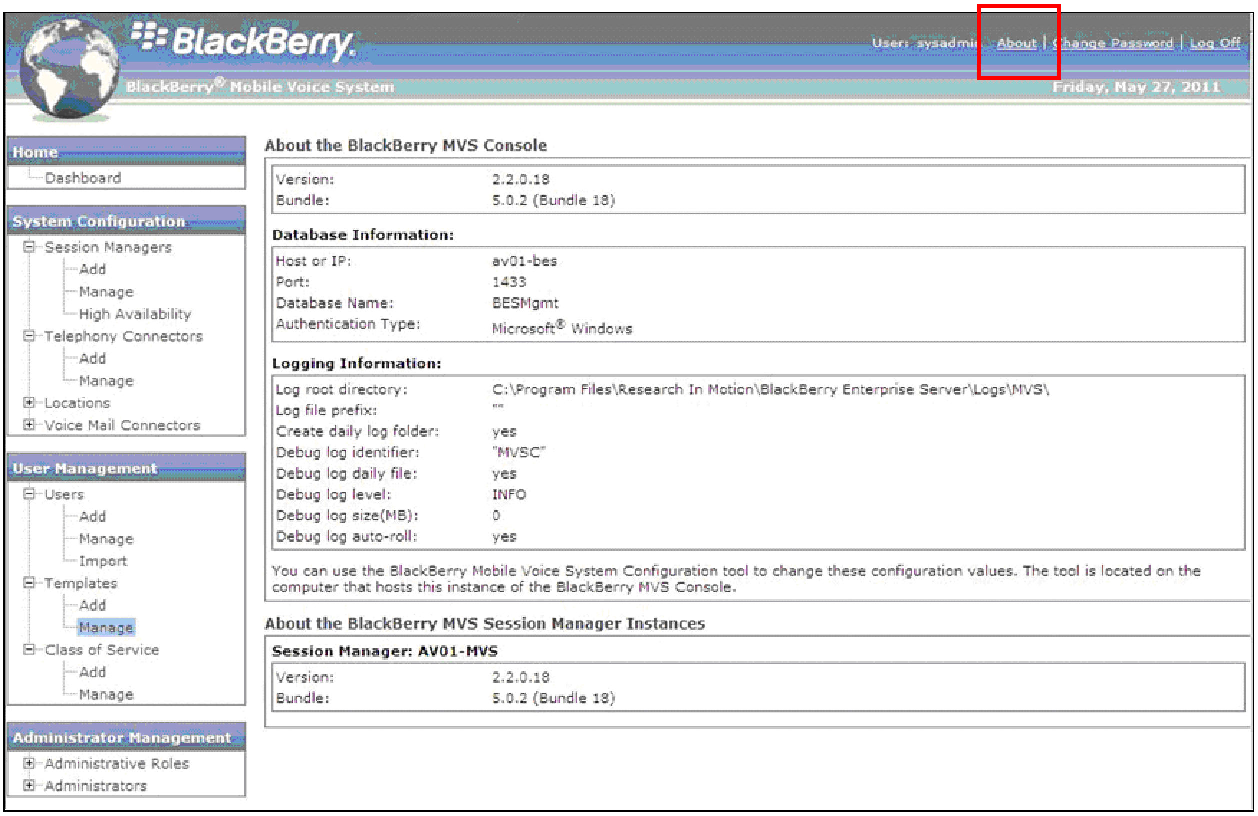
After clicking the **PIN** number, the device details are displayed below.



7.3 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.

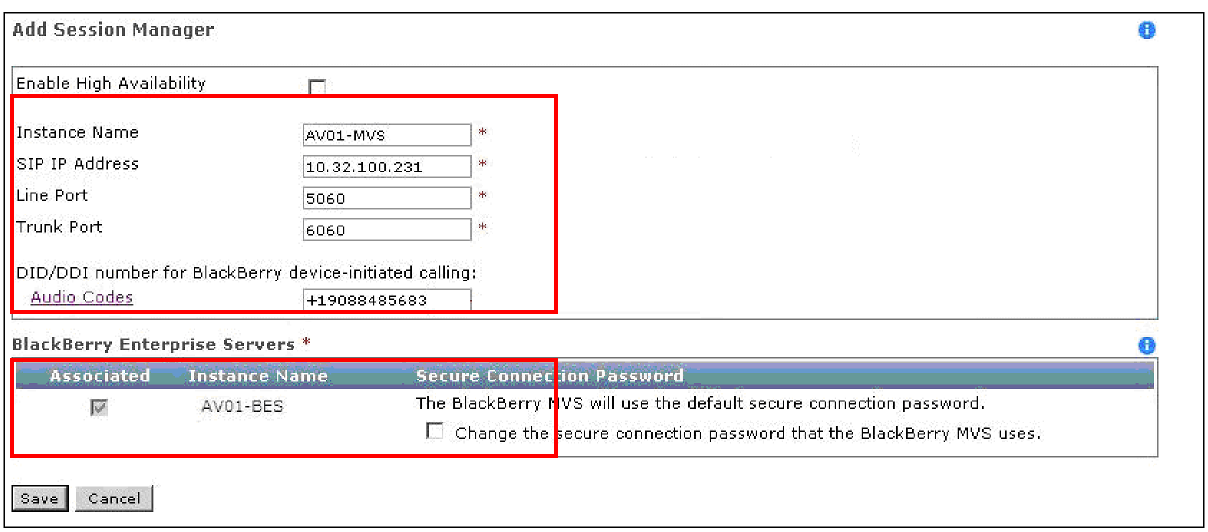


7.3.1 Create a Mobile Voice System Session Manager

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 **System Configuration → Session Managers → Add** from the left-hand navigation menu described at the top of **Section 7.3**. Configure the parameters as described below. After creation, if the Session Manager needs to be modified, it can be edited by navigating to **System Configuration → Session Managers → Manage**.

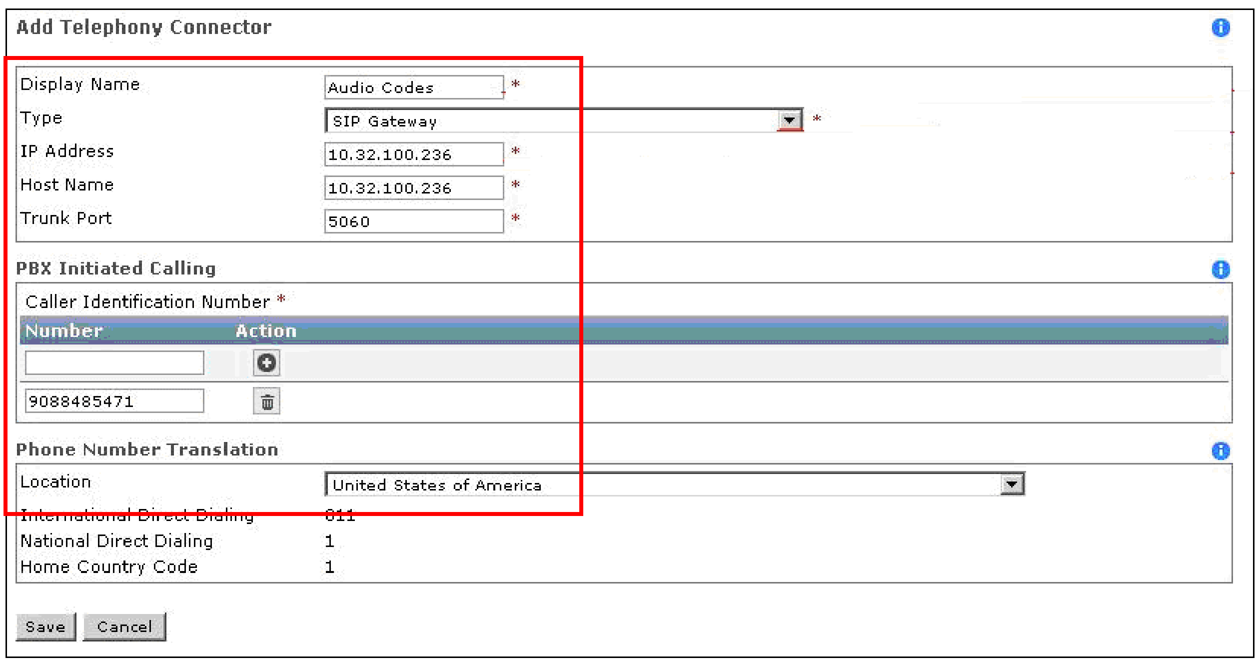
- In the **Instance Name** field, type the instance name that was specified when the MVS Session Manager was installed.
- In the **SIP IP Address** field, type the IP address that was specified when the MVS Session Manager was installed.
- In the **Line Port** field, type the UDP port number that the 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 that you want to associate with the MVS Session Manager.
- Click **Save**.



7.3.2 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 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 7.3**. 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 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. When you click a 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**.

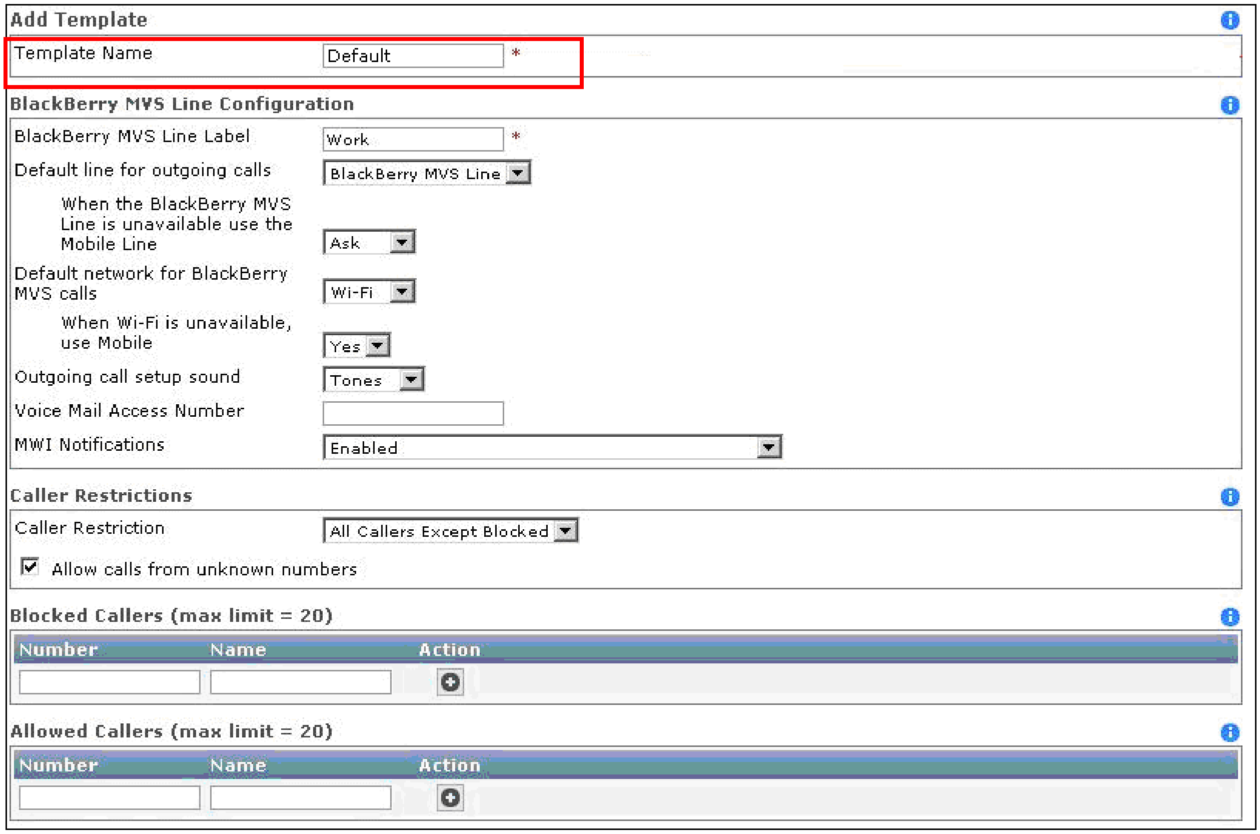


7.3.3 Create User Account Template

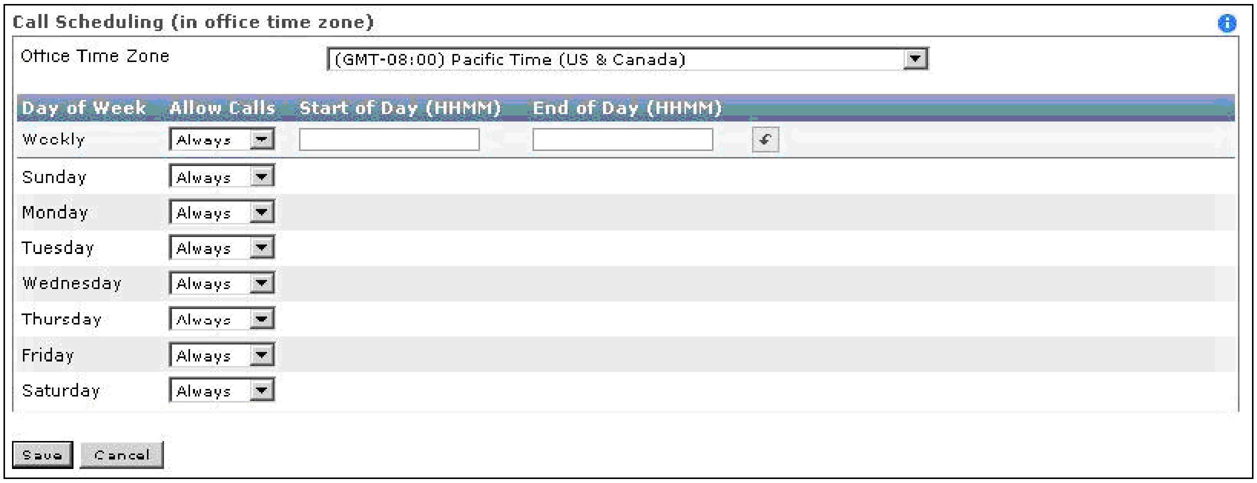
Templates must be carefully planned. The BlackBerry® MVS does not apply future template changes to user accounts that already have the template applied to them.

To add a template, navigate to **System Configuration → Templates → Add** from the left-hand navigation menu described at the top of **Section 7.3**. Configure the parameters as described below. After creation, if the Template needs to be modified, it can be edited by navigating to **System Configuration → 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).



Scroll down to view the remaining fields.

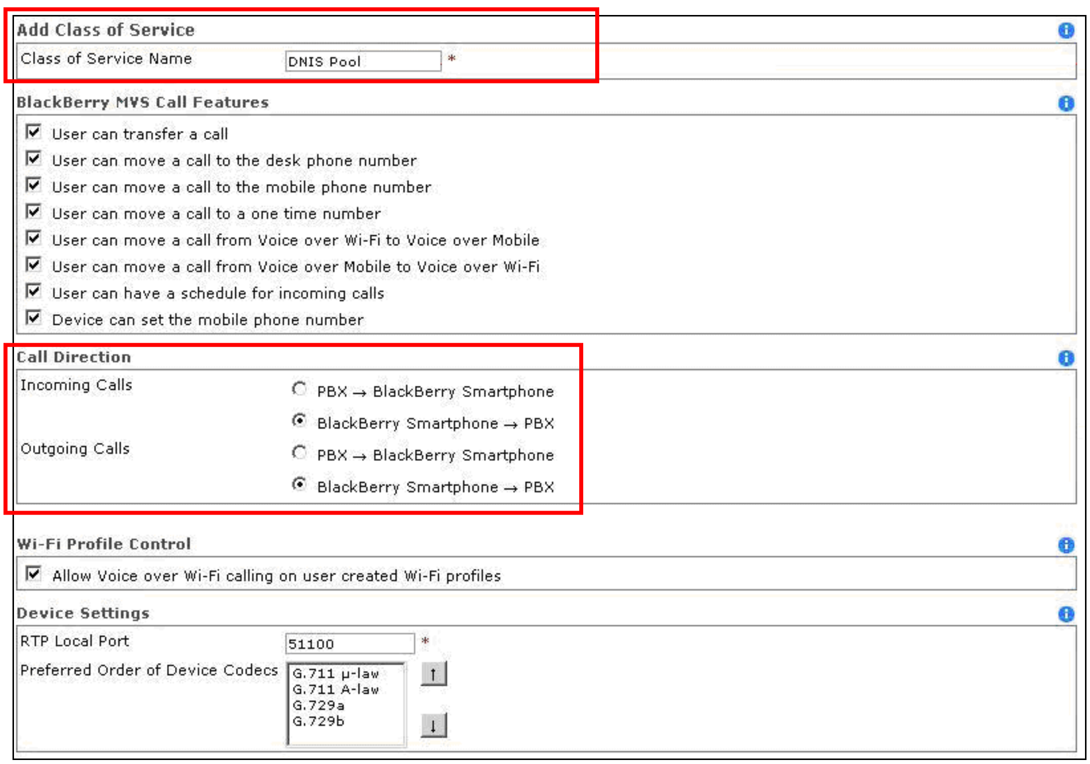


7.3.4 Create Class of Service

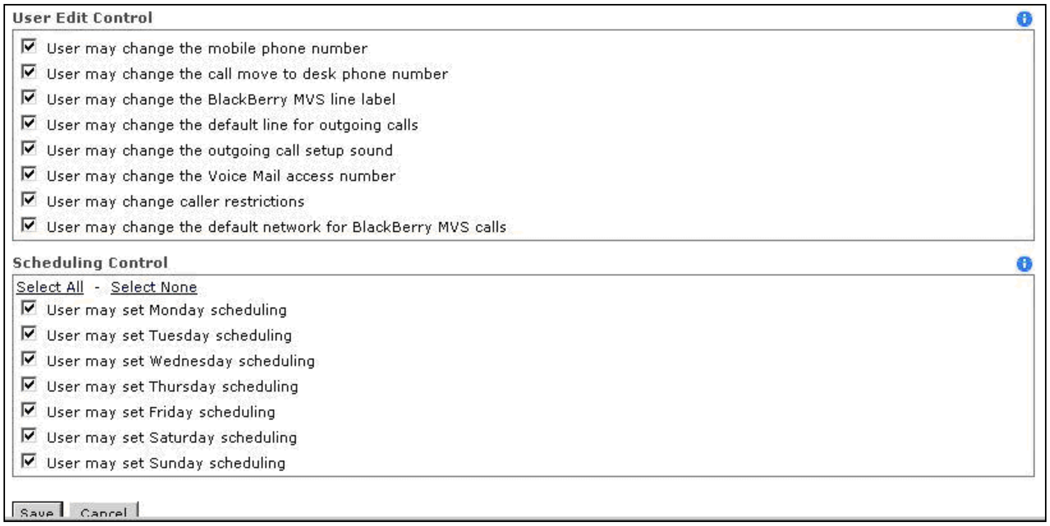
To add a template, navigate to **System Configuration → Class of Service → Add** from the left-hand navigation menu described at the top of **Section 7.3**. Configure the parameters as described below. After creation, if the Class of Service needs to be modified, it can be edited by navigating to **System Configuration → 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).

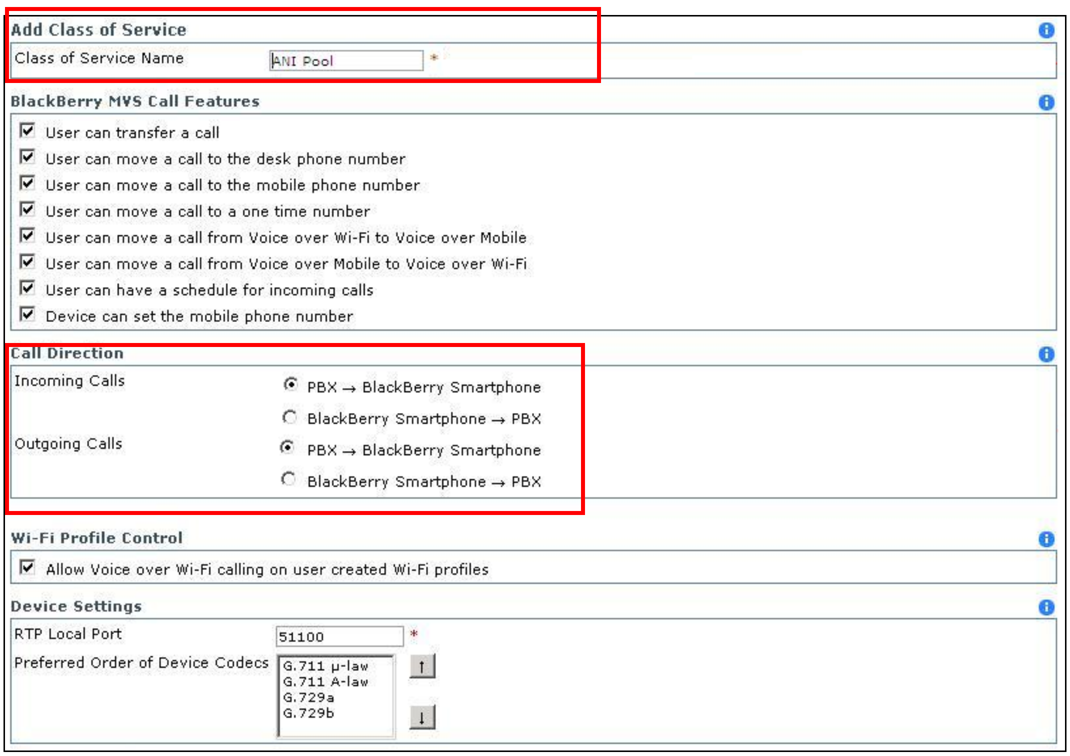
The example below shows the DNIS pool class of service.



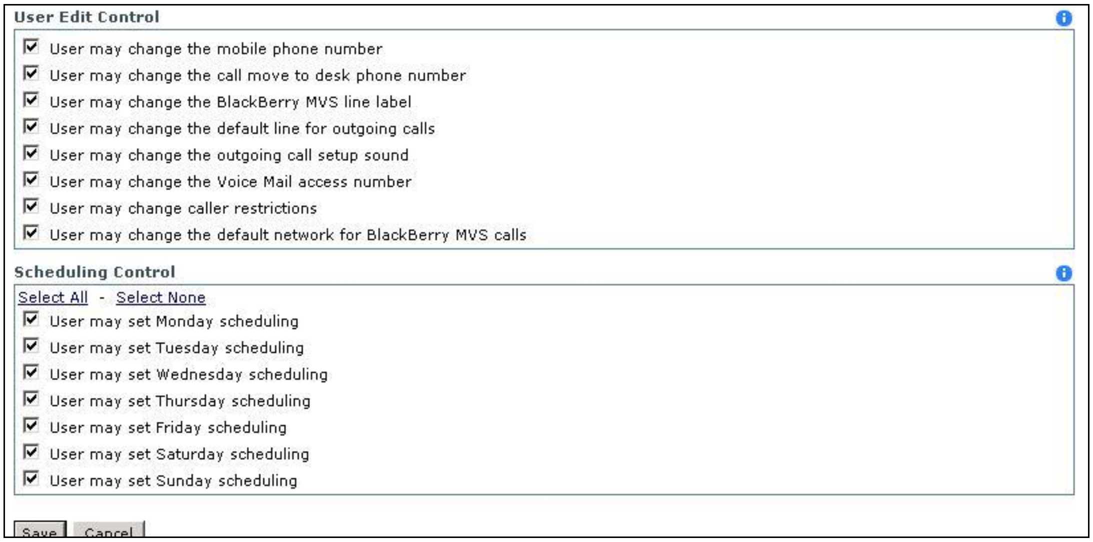
Scroll down to see the remaining values.



The example below shows the ANI pool class of service.



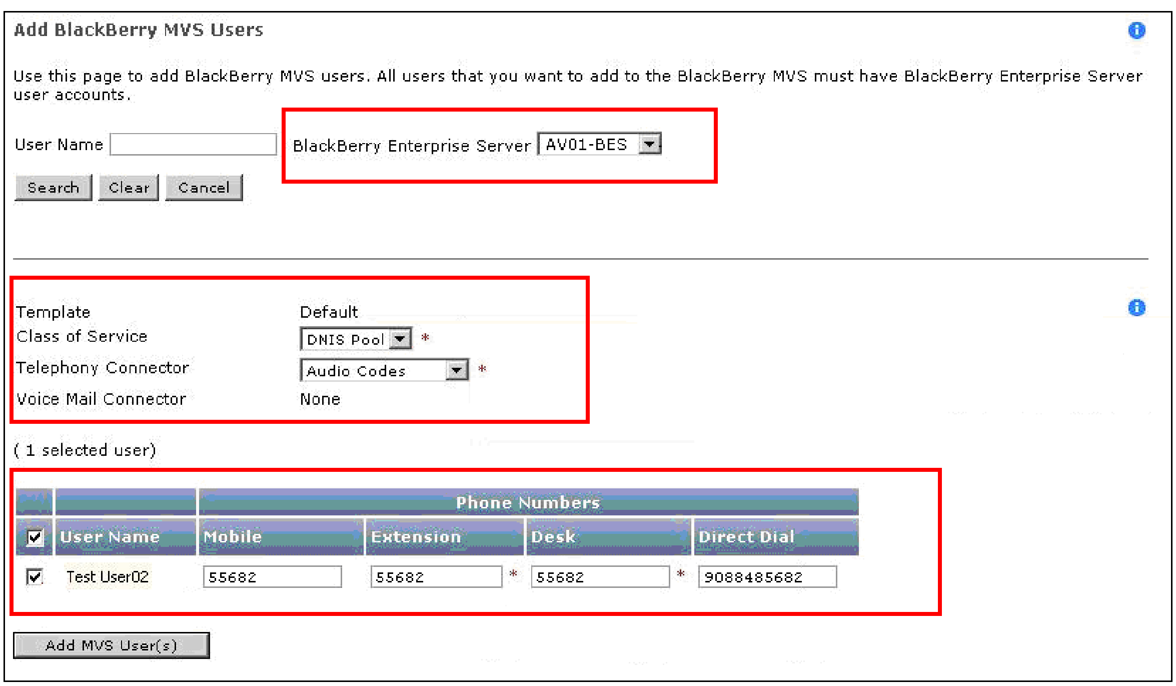
Scroll down to see the remaining values.



7.3.5 Add Mobile Voice System Users

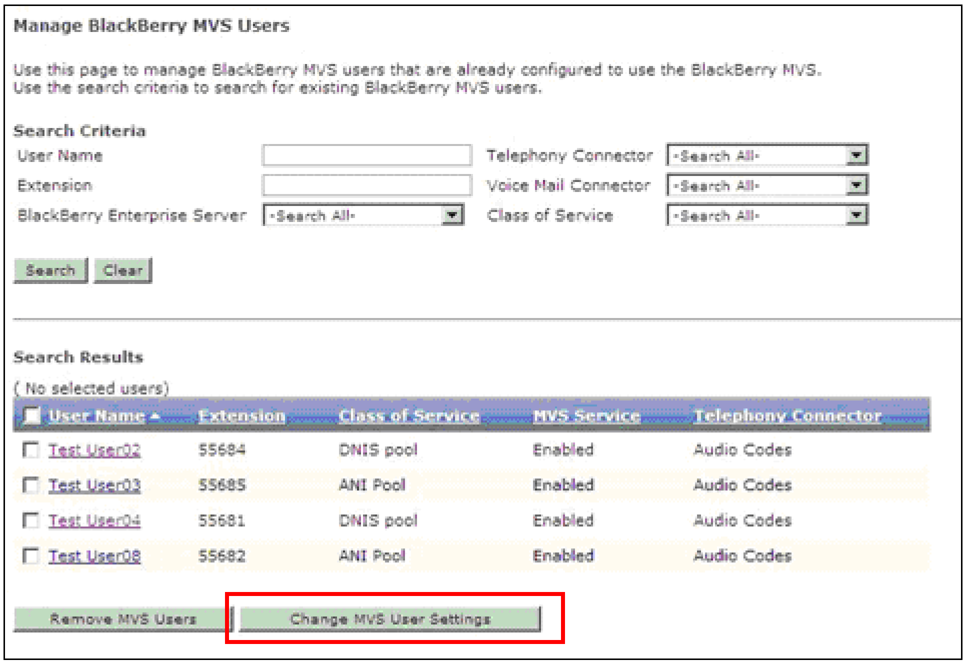
To add a MVS user, navigate to **System Configuration → Users → Add** from the left-hand navigation menu described at the top of **Section 7.3**. Configure the parameters as described below.

- 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 7.3.3** 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 7.3.4**. 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 7.3.2**.
- Select the check box beside each BlackBerry® Enterprise Server user account that you want to add.
- If necessary, configure the settings for the BlackBerry® Enterprise Server user accounts that you selected. Fields that are marked with an asterisk (*) are required.
- Click **Add MVS User(s)**.

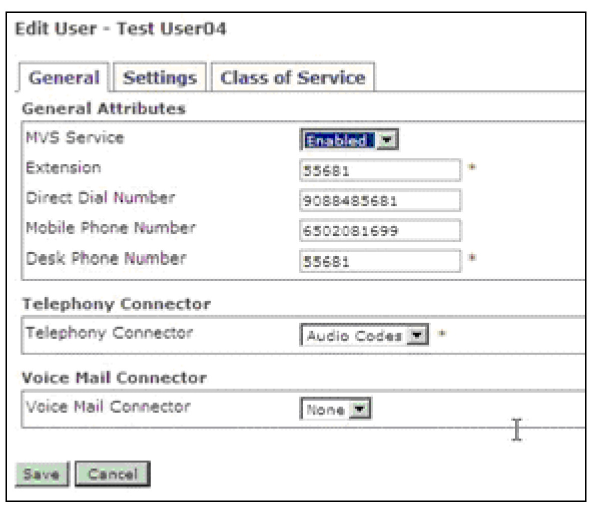


7.3.6 Manage Mobile Voice System Users

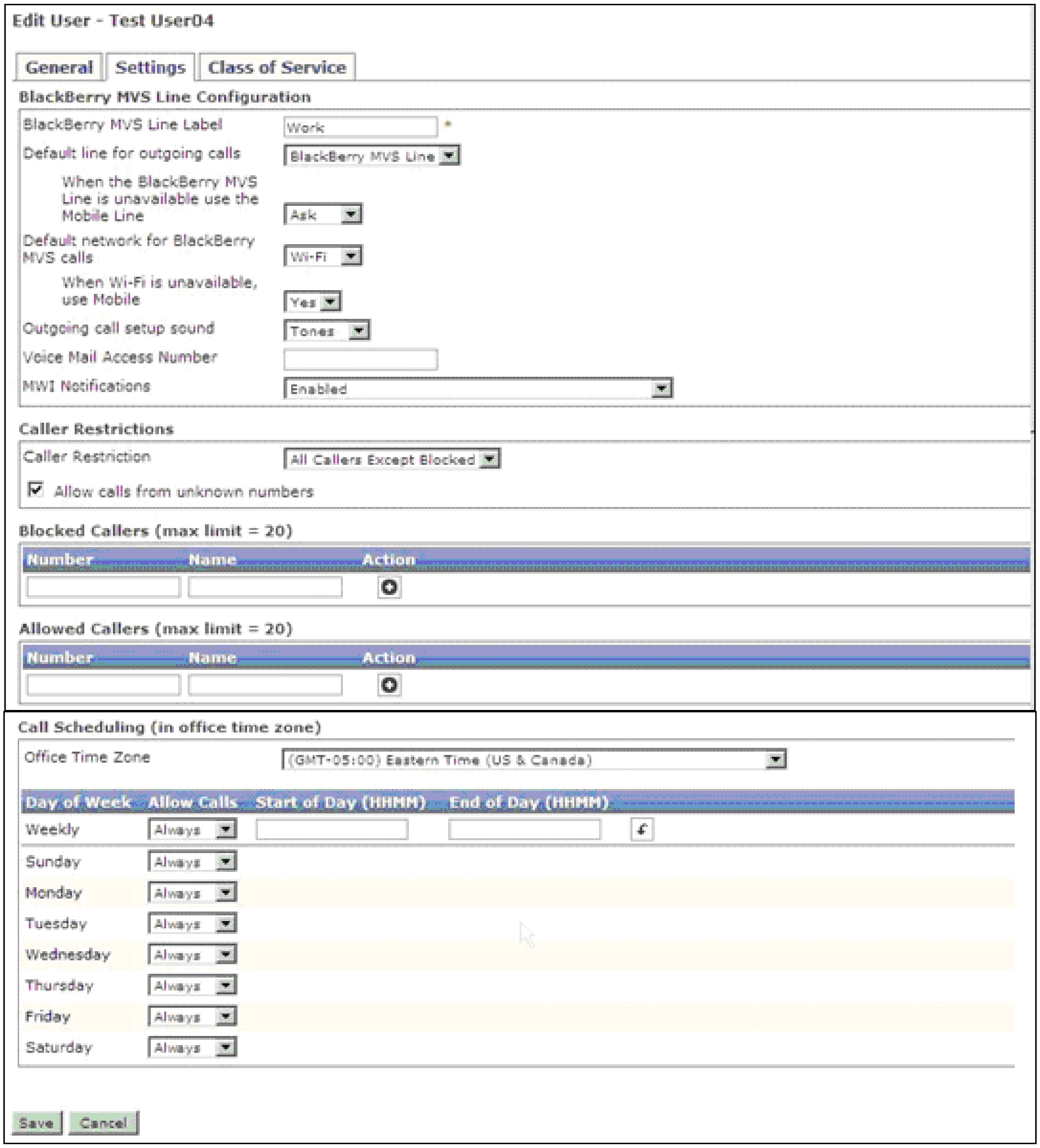
After creation, if the user needs to be modified, it can be edited by navigating to **System Configuration → User → Manage**. Select the check box next to the **User Name** in the search results that is to be modified. Click **Change MVS User Settings**.



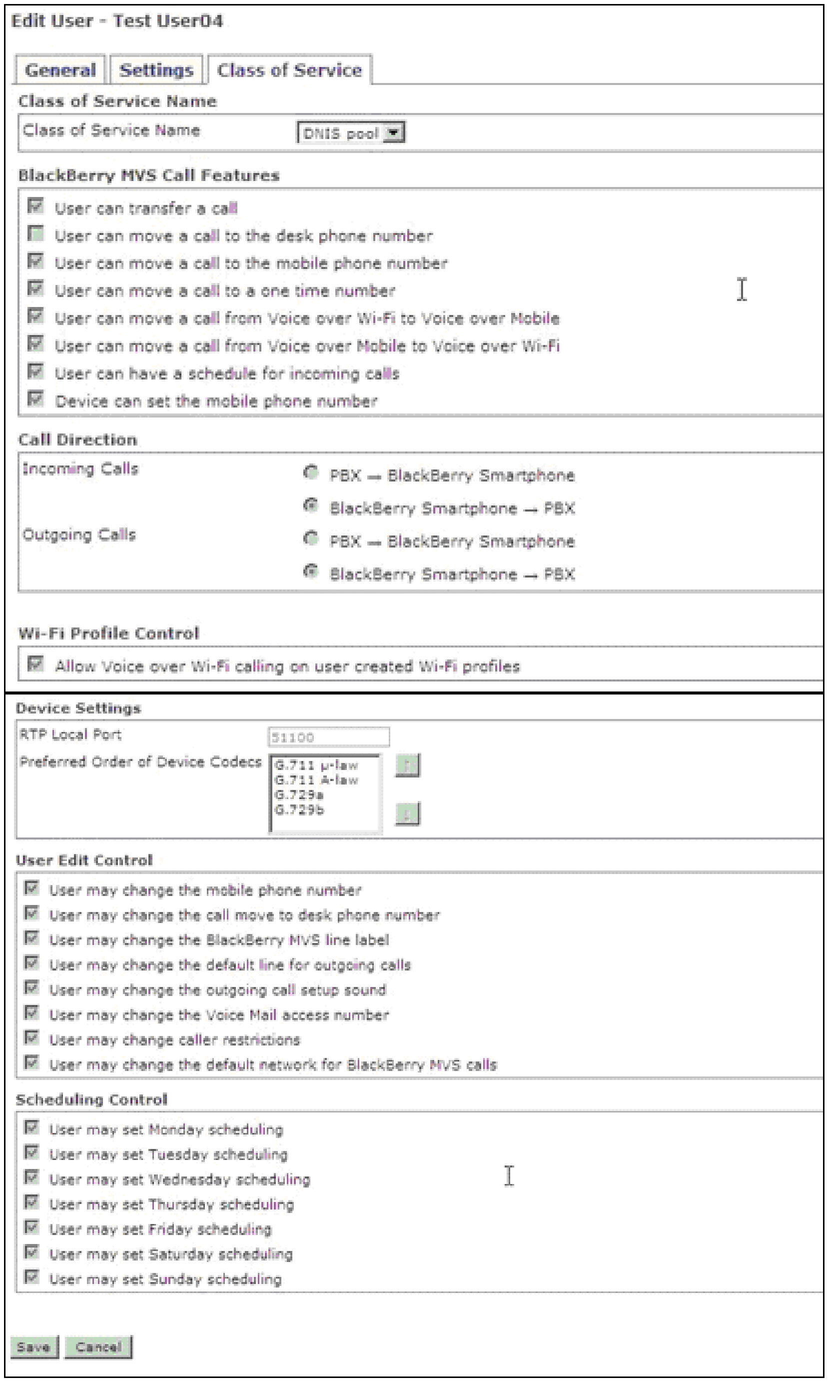
The following screens show all the user parameters that are available using the example of **Test User04**. The screen below shows the parameters on the **General** tab.



The following screen shows the parameters on the **Settings** tab.



The following screen shows the parameters on the **Class of Service** tab.



8 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 group configured in **Section 5.5**, **Step 5** is in-service.
- From the Avaya Communication Manager SAT, use the **status trunk-group** command to verify that the trunk group configured in **Section 5.4**, **Step 1** is in-service.
- 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.

9 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 Communication 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.

10 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*, August 2010, Document Number 03-300509.

[2] *Avaya Aura® Communication Manager Feature Description and Implementation,* August 2010, *D*ocument Number 555-245-205.

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

[4] *Avaya one-X Deskphone SIP for 9600 Series IP Telephones Administrator Guide, Release 2.0,* Document Number 16-601944.

[5] *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:

http://docs.blackberry.com/en/admin/subcategories/?userType=2&category=BlackBerry+MVS&subCategory=&url=%2Fadmin%2Fsubcategories%2F&versionId=779

Product documentation for the AudioCodes Mediant 1000 VoIP Media Gateway can be obtained from AudioCodes at the following links: http://www.audiocodes.com/products/mediant-1000 and http://audiocodes.com/support.

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.