



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.

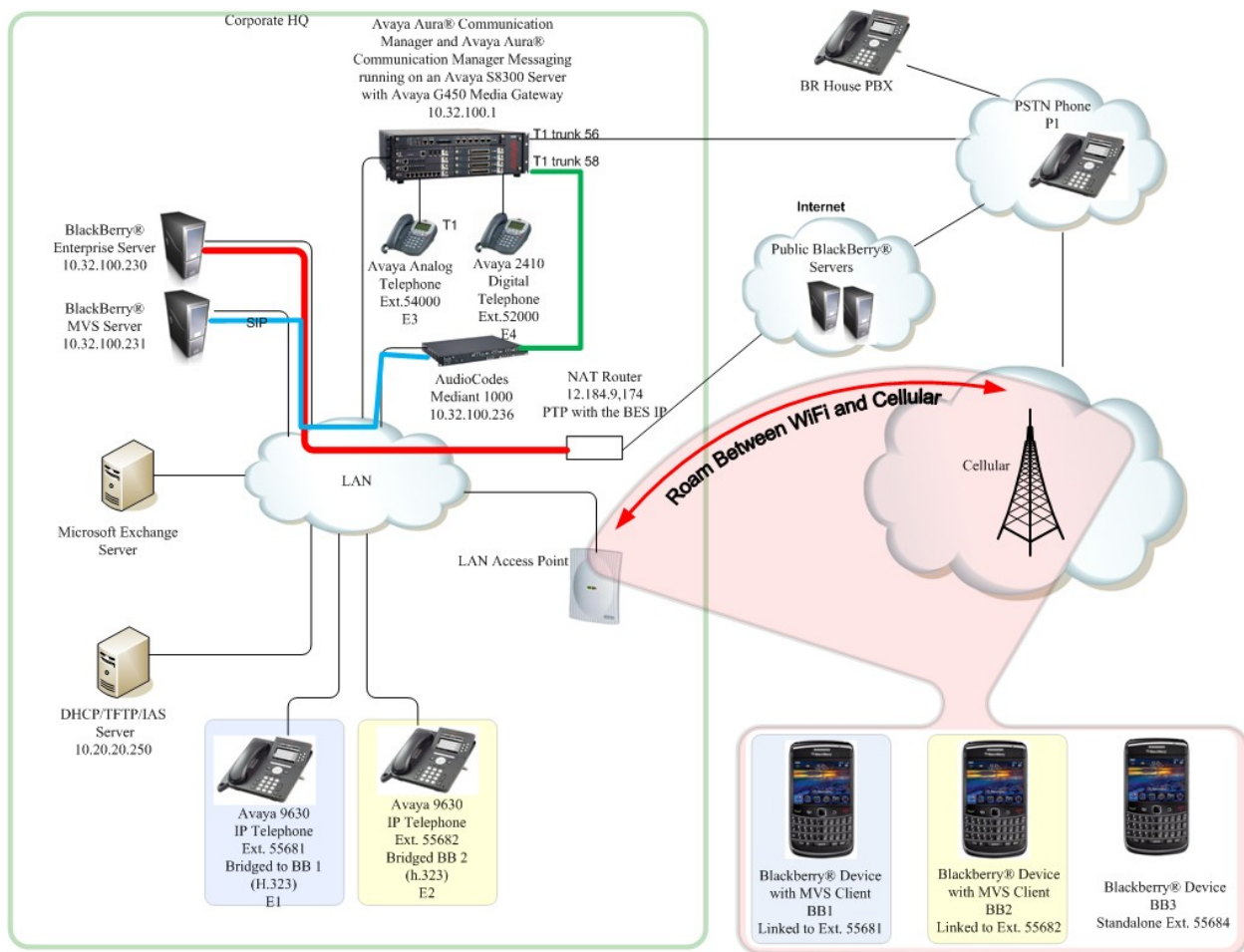


Figure 1: Avaya/RIM MVS Solution Diagram

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
<i>Avaya PBX Products</i>	
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
<i>Avaya Messaging (Voice Mail) Products</i>	
Avaya Aura® Communication Manager Messaging (CMM)	6.0.1
<i>Avaya Telephony Sets</i>	
Avaya 96xx Series IP Telephones	(H.323 3.1.1)
Avaya 2410 Digital Telephone	5.0
<i>RIM Products</i>	
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
<i>AudioCodes Products</i>	
AudioCodes Mediant 1000	6.00A.037
<i>Microsoft products</i>	
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.	<p>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.</p> <div><pre>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</pre></div>

Step	Description
2.	<p>Automatic Route Selection (ARS) will be used to route calls to the PSTN trunk. On Page 3, verify that ARS is set to y.</p> <div> 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 Digital Loss Plan Modification? y Async. Transfer Mode (ATM) Trunking? n DS1 MSP? y ATM WAN Spare Processor? n ATMS? y DS1 Echo Cancellation? y Attendant Vectoring? y </div>
3.	<p>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.</p> <div> display system-parameters customer-options Page 4 of 11 OPTIONAL FEATURES Emergency Access to Attendant? y IP Stations? y Enable 'dadmin' Login? y ISDN Feature Plus? n Enhanced Conferencing? y ISDN/SIP Network Call Redirection? y Enhanced EC500? y ISDN-BRI Trunks? y Enterprise Survivable Server? n ISDN-PRI? y Enterprise Wide Licensing? n 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 Multifrequency Signaling? y Forced Entry of Account Codes? y Multimedia Call Handling (Basic)? y Global Call Classification? y Multimedia Call Handling (Enhanced)? y Hospitality (G3V3 Enhancements)? y Multimedia IP SIP Trunking? y IP Trunks? y IP Attendant Consoles? y </div>

Step	Description
4.	<p>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.</p> <div data-bbox="264 378 1438 934" style="border: 1px solid black; padding: 10px;"> <pre> display system-parameters customer-options OPTIONAL FEATURES Multinational Locations? n Multiple Level Precedence & Preemption? y Multiple Locations? n Personal Station Access (PSA)? y PNC Duplication? n Port Network Support? n Posted Messages? y Private Networking? y Processor and System MSP? y Processor Ethernet? y Remote Office? y Restrict Call Forward Off Net? y Secondary Data Module? y Station and Trunk MSP? y Station as Virtual Extension? y System Management Data Transfer? n Tenant Partitioning? y Terminal Trans. Init. (TTI)? y Time of Day Routing? y TN2501 VAL Maximum Capacity? y Uniform Dialing Plan? y Usage Allocation Enhancements? y Wideband Switching? y Wireless? n </pre> </div>

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.

Step	Description
1.	<p>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.</p> <pre> display dialplan analysis 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>
2.	<p>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.</p> <pre> change feature-access-codes 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>

5.3 System Parameters Features

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

Step	Description
1.	<p>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.</p> <ul style="list-style-type: none"> • 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. <div style="border: 1px solid black; padding: 10px; margin-top: 10px;"> <pre> change system-parameters features Page 8 of 19 FEATURE-RELATED SYSTEM PARAMETERS ISDN PARAMETERS Send Non-ISDN Trunk Group Name as Connected Name? n Display Connected Name/Number for ISDN DCS Calls? y Send ISDN Trunk Group Name on Tandem Calls? n Send Custom Messages Through QSIG? n 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> </div>

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**.

Step	Description
3.	<p>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.</p> <div><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: V9: gateway-announcements ANN VMM Max Survivable IP Ext: 8</pre></div>

Step	Description
4.	<p>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:</p> <ul style="list-style-type: none"> • 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. <div data-bbox="264 762 1438 1325"> <pre> add ds1 001v5 DS1 CIRCUIT PACK Page 1 of 2 Location: 001V5 Bit Rate: 1.544 Line Coding: b8zs Framing Mode: esf Signaling Mode: isdn-pri Connect: pbx Interface: peer-master Peer Protocol: Q-SIG Side: a TN-C7 Long Timers? n Interworking Message: PROgress Interface Companding: mulaw 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 </pre> </div>

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.	<p>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:</p> <ul style="list-style-type: none"> • 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. <div> <pre> 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 </pre> </div>
2.	<p>On Page 2, set the Supplementary Service Protocol to b.</p> <div> <pre> 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 </pre> </div>

Step	Description
3.	<p>On Page 3, set the following:</p> <ul style="list-style-type: none"> 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. <div data-bbox="264 495 1438 1085"> <pre> add trunk-group 58 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 UII IE Treatment: shared Maximum Size of UII IE Contents: 128 Replace Restricted Numbers? n Replace Unavailable Numbers? n Send Connected Number: y Hold/Unhold Notifications? y Send UII IE? y Modify Tandem Calling Number: no Send UCID? n BSR Reply-best DISC Cause Value: 31 Send Codeset 6/7 LAI IE? y Dsl Echo Cancellation? n Apply Local Ringback? n Show ANSWERED BY on Display? y </pre> </div>
4.	<p>On Page 4 of the trunk group form, verify that Path Replacement is set to y.</p> <div data-bbox="264 1197 1424 1598"> <pre> add trunk-group 58 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 </pre> </div>

Step	Description
5.	<p>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:</p> <ul style="list-style-type: none"> • 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. <div data-bbox="264 835 1453 1123"> <pre> 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 </pre> </div>

Step	Description																																																																																																
6.	<p>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:</p> <ul style="list-style-type: none">In the Port column, enter a value xxxxzz, where xxxx 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.																																																																																																
	<div><div>change trunk-group 58</div><div>Page 5 of 21</div><div>TRUNK GROUP</div><div>Administered Members (min/max): 1/23</div><div>GROUP MEMBER ASSIGNMENTS</div><div>Total Administered Members: 23</div><table><tr><th>Port</th><th>Code</th><th>Sfx</th><th>Name</th><th>Night</th><th>Sig Grp</th></tr><tr><td>1: 001V501</td><td>MM710</td><td>b</td><td></td><td></td><td>58</td></tr><tr><td>2: 001V502</td><td>MM710</td><td>b</td><td></td><td></td><td>58</td></tr><tr><td>3: 001V503</td><td>MM710</td><td>b</td><td></td><td></td><td>58</td></tr><tr><td>4: 001V504</td><td>MM710</td><td>b</td><td></td><td></td><td>58</td></tr><tr><td>5: 001V505</td><td>MM710</td><td>b</td><td></td><td></td><td>58</td></tr><tr><td>6: 001V506</td><td>MM710</td><td>b</td><td></td><td></td><td>58</td></tr><tr><td>7: 001V507</td><td>MM710</td><td>b</td><td></td><td></td><td>58</td></tr><tr><td>8: 001V508</td><td>MM710</td><td>b</td><td></td><td></td><td>58</td></tr><tr><td>9: 001V509</td><td>MM710</td><td>b</td><td></td><td></td><td>58</td></tr><tr><td>10: 001V510</td><td>MM710</td><td>b</td><td></td><td></td><td>58</td></tr><tr><td>11: 001V511</td><td>MM710</td><td>b</td><td></td><td></td><td>58</td></tr><tr><td>12: 001V512</td><td>MM710</td><td>b</td><td></td><td></td><td>58</td></tr><tr><td>13: 001V513</td><td>MM710</td><td>b</td><td></td><td></td><td>58</td></tr><tr><td>14: 001V514</td><td>MM710</td><td>b</td><td></td><td></td><td>58</td></tr><tr><td>15:</td><td></td><td></td><td></td><td></td><td></td></tr></table></div>	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:					
Port	Code	Sfx	Name	Night	Sig Grp																																																																																												
1: 001V501	MM710	b			58																																																																																												
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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																																																																																												
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12: 001V512	MM710	b			58																																																																																												
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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.	<p>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.</p> <ul style="list-style-type: none"> 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. <pre> change route-pattern 58 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 Intw 1: 58 0 2: 3: 4: 5: 6: 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 </pre>

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.

Step	Description
1.	<p>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.</p> <ul style="list-style-type: none">• 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.

change aar analysis 5

Page 1 of 2

AAR DIGIT ANALYSIS TABLE
Location: allPercent Full: 3

Dialed String	Total Min	Total Max	Route Pattern	Call Type	Node Num	ANI Req'd
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

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.	<p>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:</p> <ul style="list-style-type: none">• 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. <div><div>change inc-call-handling-trmt trunk-group 56<div>Page1 of 3</div></div><div>INCOMING CALL HANDLING TREATMENT</div><table><tr><th>Service/ Feature</th><th>Number Len</th><th>Number Digits</th><th>Del</th><th>Insert</th><th>Per Call CPN/BN</th><th>Night Serv</th></tr><tr><td>tie</td><td>10</td><td>9088485681</td><td>10</td><td>55681</td><td></td><td></td></tr><tr><td>tie</td><td>10</td><td>9088485682</td><td>10</td><td>55682</td><td></td><td></td></tr><tr><td>tie</td><td>10</td><td>9088485683</td><td></td><td>3</td><td></td><td></td></tr><tr><td>tie</td><td>10</td><td>9088485684</td><td>10</td><td>55684</td><td></td><td></td></tr><tr><td>tie</td><td></td><td></td><td></td><td></td><td></td><td></td></tr></table></div>	Service/ Feature	Number Len	Number Digits	Del	Insert	Per Call CPN/BN	Night Serv	tie	10	9088485681	10	55681			tie	10	9088485682	10	55682			tie	10	9088485683		3			tie	10	9088485684	10	55684			tie						
Service/ Feature	Number Len	Number Digits	Del	Insert	Per Call CPN/BN	Night 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.

Step	Description
1.	<p>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.</p> <p>Use the add station 55684 command to create the station for this user.</p> <div><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: Personalized Ringing Pattern: 1 Message Lamp Ext: 55684 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></div>

Step	Description																																								
2.	<p>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.</p> <div><div>add station 55684Page 4 of 5</div><div><div>STATION</div><div><div>SITE DATA</div><div><div>Room:Headset? n</div><div>Jack:Speaker? n</div><div>Cable:Mounting: d</div><div>Floor:Cord Length: 0</div><div>Building:Set Color:</div></div></div><div><div>ABBREVIATED DIALING</div><div><div>List1:List2:List3:</div></div></div><div><div>BUTTON ASSIGNMENTS</div><div><div>1: call-appr5:</div><div>2: call-appr6:</div><div>3: call-appr7:</div><div>4: ec500Timer? n8:</div><div>voice-mail</div></div></div></div></div>																																								
3.	<p>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.</p> <ul style="list-style-type: none">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. <div><div>add off-pbx-telephone station-mapping 55681Page 1 of 3</div><div><div>STATIONS WITH OFF-PBX TELEPHONE INTEGRATION</div><table><tr><th>Station Extension</th><th>Application</th><th>Dial Prefix</th><th>CC</th><th>Phone Number</th><th>Trunk Selection</th><th>Config Set</th><th>Dual Mode</th></tr><tr><td>55681</td><td>EC500</td><td>-</td><td></td><td>55681</td><td>aar</td><td>5</td><td></td></tr><tr><td>55682</td><td>EC500</td><td></td><td></td><td>55682</td><td>aar</td><td>5</td><td></td></tr><tr><td>55684</td><td>EC500</td><td></td><td></td><td>55684</td><td>aar</td><td>5</td><td></td></tr><tr><td></td><td></td><td>-</td><td></td><td></td><td></td><td></td><td></td></tr></table></div></div>	Station Extension	Application	Dial Prefix	CC	Phone Number	Trunk Selection	Config Set	Dual Mode	55681	EC500	-		55681	aar	5		55682	EC500			55682	aar	5		55684	EC500			55684	aar	5				-					
Station Extension	Application	Dial Prefix	CC	Phone Number	Trunk Selection	Config Set	Dual Mode																																		
55681	EC500	-		55681	aar	5																																			
55682	EC500			55682	aar	5																																			
55684	EC500			55684	aar	5																																			
		-																																							

Step	Description
4.	<p>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.</p> <ul style="list-style-type: none"> • Configuration Set Description – Enter a meaningful name/description. • Calling Number Verification? – Set to n. <div data-bbox="264 562 1438 1150" style="border: 1px solid black; padding: 10px; margin-top: 20px;"> <div style="display: flex; justify-content: space-between;"> change off-pbx-telephone configuration-set 5 Page 1 of 1 </div> <div style="text-align: center; margin-top: 20px;"> <p>CONFIGURATION SET: 5</p> <p>Configuration Set Description: for RIM</p> <p>Calling Number Style: network</p> <p>CDR for Origination: phone-number</p> <p>CDR for Calls to EC500 Destination? y</p> <p>Fast Connect on Origination? n</p> <p>Post Connect Dialing Options: dtmf</p> <p>Cellular Voice Mail Detection: none</p> <p>Barge-in Tone? n</p> <p>Calling Number Verification? n</p> <p>Call Appearance Selection for Origination: primary-first</p> <p>Confirmed Answer? n</p> <p>Use Shared Voice Connections for Second Call Answered? n</p> <p>Use Shared Voice Connections for Second Call Initiated? n</p> </div> </div>

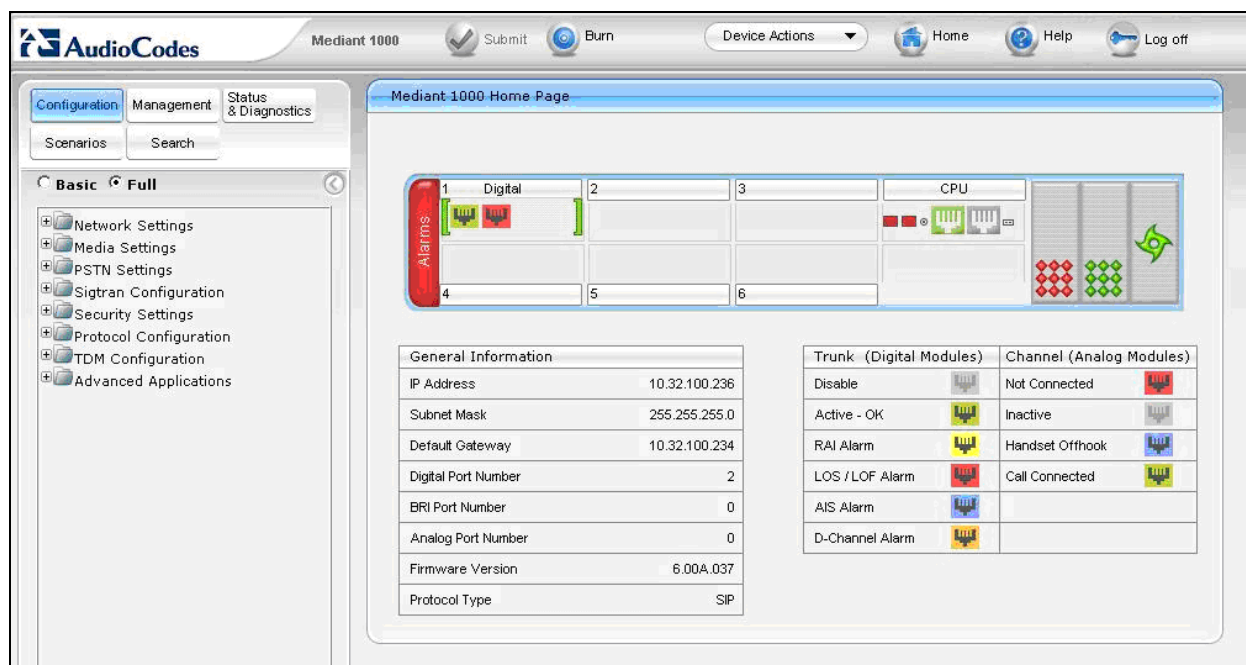
6 Configure AudioCodes Mediant 1000 VoIP Media Gateway

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**.

The screenshot shows the AudioCodes Mediant 1000 web interface. The left sidebar contains a tree view with 'Network Settings' expanded, showing 'IP Settings' selected. The main content area is titled 'Multiple Interface Table' and includes a 'Note: Select row index to modify the relevant row.' Below this is a table with the following data:

Index	Application Type	IP Address	Prefix Length	Gateway	VLAN ID	Interface Name
0	DAMP + Media + Control	10.32.100.236	24	10.32.100.234	0	0+M+C

Below the table, there is a 'VLAN Mode' dropdown set to 'Disable' and a 'Native VLAN ID' input field set to '1'. A 'Done' button is located at the top right of the configuration area.

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.

The screenshot shows the AudioCodes Mediant 1000 web interface with 'Voice Settings' selected in the left sidebar. The main content area displays a 'Basic Parameter List' with the following settings:

Voice Volume (-32 to 31 dB)	0
Input Gain (-32 to 31 dB)	0
Silence Suppression	Disable
DTMF Transport Type	RFC2833 Relay DTMF
DTMF Volume (-31 to 0 dB)	0
NTE Max Duration	-1
CAS Transport Type	CASEventsOnly
DTMF Generation Twist	0
Echo Canceller	Enable

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.

The screenshot shows the AudioCodes Mediant 1000 web interface. The left sidebar contains a tree view with categories like Network Settings, Media Settings, and PSTN Settings. The 'PSTN Settings' category is expanded, and 'Trunk Settings' is selected. The main content area is titled 'Trunk Settings' and contains three sections: General Settings, Trunk Configuration, and ISDN Configuration. The General Settings section includes Module ID (1), Trunk ID (1), Trunk Configuration State (Active), and Protocol Type (T1 QSIG). The Trunk Configuration section includes Clock Master (Recovered), Auto Clock Trunk Priority (0), Line Code (B8ZS), Line Build Out Loss (0 dB), Trace Level (Full ISDN Trace), Line Build Out Overwrite (OFF), and Framing Method (T1 FRAMING ESF CRC6). The ISDN Configuration section includes ISDN Termination Side (User side), Q931 Layer Response Behavior (0x40000000), Outgoing Calls Behavior (0x400), Incoming Calls Behavior (0x0), General Call Control Behavior (0x0), NFAS Group Number (0), IUA Interface ID (1), NFAS Interface ID (255), and D-channel Configuration (PRIMARY). At the bottom of the page, there are 'Submit' and 'Deactivate' buttons.

General Settings	
Module ID	1
Trunk ID	1
Trunk Configuration State	Active
Protocol Type	T1 QSIG

Trunk Configuration	
Clock Master	Recovered
Auto Clock Trunk Priority	0
Line Code	B8ZS
Line Build Out Loss	0 dB
Trace Level	Full ISDN Trace
Line Build Out Overwrite	OFF
Framing Method	T1 FRAMING ESF CRC6

ISDN Configuration	
ISDN Termination Side	User side
Q931 Layer Response Behavior	0x40000000
Outgoing Calls Behavior	0x400
Incoming Calls Behavior	0x0
General Call Control Behavior	0x0
NFAS Group Number	0
IUA Interface ID	1
NFAS Interface ID	255
D-channel Configuration	PRIMARY

Continued from previous page.

The screenshot displays the AudioCodes Mediant 1000 configuration web interface. The top navigation bar includes the AudioCodes logo, the device name 'Mediant 1000', and buttons for 'Submit', 'Burn', 'Device Actions', 'Home', 'Help', and 'Log off'. The left sidebar contains a tree view of configuration categories: Configuration (selected), Management, and Status & Diagnostics. Under Configuration, there are sub-categories like Scenarios and Search, and a list of settings including Network Settings, IP Settings, Application Settings, IP Routing Table, QoS Settings, SCTP Settings, Media Settings, Voice Settings, Fax/Modem/CID Settings, RTP/RTCP Settings, IP Media Settings, General Media Settings, Media Security, PSTN Settings (selected), CAS State Machines, Trunk Settings (selected), Sigtran Configuration, Security Settings, Protocol Configuration, TDM Configuration, and Advanced Applications.

The main content area is titled 'Trunk Settings'. It features a 'Framing Method' dropdown set to 'T1 FRAMING ESF CRC6'. Below this is an 'ISDN Configuration' section with the following parameters:

Parameter	Value	Action
ISDN Termination Side	User side	
Q931 Layer Response Behavior	0x40000000	Reset
Outgoing Calls Behavior	0x400	Reset
Incoming Calls Behavior	0x0	Reset
General Call Control Behavior	0x0	Reset
NFAS Group Number	0	
IUA Interface ID	-1	
NFAS Interface ID	255	
D-channel Configuration	PRIMARY	

Below the ISDN Configuration is a 'PSTN Configuration' section with the following parameters:

Parameter	Value
PSTN Alert Timeout	-1
Transfer Mode	Path Replacement Transfer
Local ISDN Ringback Tone Source	PBX
Set PI in Rx Disconnect Message	Not Configured
ISDN Transfer Capabilities	Not Configured
Progress Indicator to ISDN	Not Configured
Select Receiving of Overlap Dialing	None
B-channel Negotiation	Not Configured
Out-Of-Service Behavior	Default
Remove Calling Name	Use Global Parameter
Play Ringback Tone to Trunk	Not Configured

At the bottom of the interface, there are 'Submit' and 'Deactivate' buttons, and a 'Stop Trunk' button with a red stop icon.

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.

The screenshot displays the AudioCodes Mediant 1000 configuration web interface. The left sidebar shows a tree view of configuration categories, with 'Trunk Group' selected under 'Protocol Configuration'. The main area is titled 'Trunk Group Table' and contains a table for configuring call routing settings. Above the table are two dropdown menus: 'Add Phone Context As Prefix' (set to 'Disable') and 'Trunk Group Index' (set to '1-10').

Group Index	Module	From Trunk	To Trunk	Channels	Phone Number	Trunk Group ID	Tel Profile ID
1	Module 1 PRI	1	1	1-23		1	0
2							
3							
4							
5							
6							
7							
8							
9							
10							

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.

The screenshot displays the 'Trunk Group Settings' page in the AudioCodes Mediant 1000 web interface. The left sidebar shows the navigation tree with 'Trunk Group Settings' selected under 'Protocol Configuration'. The main area contains a table with 10 rows for configuring trunk groups. The first row is populated with the following values:

Index	Trunk Group ID	Channel Select Mode	Registration Mode	Serving IP Group ID	Gateway Name	Contact User
1	1	Ascending				
2						
3						
4						
5						
6						
7						
8						
9						
10						

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.

The screenshot displays the AudioCodes Mediant 1000 web interface. The top navigation bar includes 'Submit', 'Burn', 'Device Actions', 'Home', 'Help', and 'Log off'. The left sidebar shows a tree view of configuration options, with 'SIP General Parameters' selected under 'Protocol Definition'. The main content area is titled 'SIP General Parameters' and contains a table of configuration fields. A 'Basic Parameter List' link is visible in the top right of the main area.

SIP General	
NAT IP Address	0.0.0.0
PRACK Mode	Disable
Channel Select Mode	Cyclic Ascending
Enable Early Media	Enable
183 Message Behavior	Progress
Session-Expires Time	300
Minimum Session-Expires	90
Session Expires Method	ReINVITE
Asserted Identity Mode	Disabled
Fax Signaling Method	No Fax
Detect Fax on Answer Tone	Initiate T.38 on Preamble
SIP Transport Type	UDP
SIP UDP Local Port	5060
SIP TCP Local Port	5060
SIP TLS Local Port	5061
Enable SIPs	Disable
Enable TCP Connection Reuse	Enable
TCP Timeout	0
SIP Destination Port	5060
Use user=phone in SIP URL	No
Use user=phone in From Header	No
Use Tel URI for Asserted Identity	Disable
Tel to IP No Answer Timeout	180
Enable Remote Party ID	Enable
Add Number Plan and Type to RPI Header	Yes
Enable History-Info Header	Disable
Use Source Number as Display Name	No
Use Display Name as Source Number	No
Enable Contact Restriction	Disable
Play Ringback Tone to IP	Don't Play

Submit

Continued from previous page.

The screenshot displays the AudioCodes Mediant 1000 configuration web interface. The top navigation bar includes the AudioCodes logo, the device name 'Mediant 1000', and buttons for 'Submit', 'Burn', 'Device Actions', 'Home', 'Help', and 'Log off'. The left sidebar contains a tree view with categories: Configuration, Management, and Status & Diagnostics. Under Configuration, there are sub-sections for Scenarios and Search. The main content area is titled 'SIP General Parameters' and features a 'Basic Parameter List' on the right. The parameters are organized into two sections: a main list of SIP parameters and a 'Retransmission Parameters' section at the bottom. Each parameter has a corresponding dropdown menu or text input field.

SIP General Parameters	
Enable Remote Party ID	Enable
Add Number Plan and Type to RPI Header	Yes
Enable History-Info Header	Disable
Use Source Number as Display Name	No
Use Display Name as Source Number	No
Enable Contact Restriction	Disable
Play Ringback Tone to IP	Don't Play
Play Ringback Tone to Tel	Prefer IP
Use Tgrp information	Disable
Enable GRUU	Disable
User-Agent Information	
SDP Session Owner	AudiocodesGW/
Play Busy Tone to Tel	Don't Play
Subject	
Multiple Packetization Time Format	None
Enable Semi-Attended Transfer	Enable
3xx Behavior	Forward
Enable P-Charging Vector	Disable
Enable VoiceMail URI	Disable
Retry-After Time	0
Enable P-Associated-URI Header	Disable
Source Number Preference	
Forking Handling Mode	Parallel handling
Enable Comfort Tone	Disable
Add Trunk Group ID as Prefix to Source	No
Enable Reason Header	Enable

Retransmission Parameters	
SIP T1 Retransmission Timer [msec]	500
SIP T2 Retransmission Timer [msec]	4000
SIP Maximum RTX	7

Submit

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.

The screenshot shows the AudioCodes Mediant 1000 configuration interface. The left sidebar contains a tree view with categories: Configuration, Management, and Status & Diagnostics. Under Configuration, there are sub-categories: Scenarios, Search, Basic, and Full. The 'Basic' category is selected, and the 'DTMF & Dialing' option is highlighted. The main panel displays the 'DTMF & Dialing' configuration page with a 'Basic Parameter List' table.

Basic Parameter List	
Max Digits In Phone Num	30
Inter Digit Timeout for Overlap Dialing [sec]	4
Declare RFC 2833 in SDP	Yes
1st Tx DTMF Option	INFO(Cisco)
2nd Tx DTMF Option	INFO(Cisco)
RFC 2833 Payload Type	101
Digit Mapping Rules	
Min Routing Overlap Digits	1
ISDN Overlap IP to Tel Dialing	Disable
Min Routing Overlap Digits	1
ISDN Overlap IP to Tel Dialing	Disable
Default Destination Number	1000
Special Digit Representation	Special

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.

The screenshot displays the AudioCodes Mediant 1000 configuration web interface. The top navigation bar includes the AudioCodes logo, 'Mediant 1000', and buttons for 'Submit', 'Burn', 'Device Actions', 'Home', 'Help', and 'Log off'. The left sidebar shows a tree view of configuration categories: Configuration (selected), Management, and Status & Diagnostics. Under Configuration, there are sub-categories: Scenarios and Search. The 'Basic' tab is active, showing a list of configuration items: Network Settings, Media Settings, PSTN Settings, Sigtran Configuration, Security Settings, Protocol Configuration (expanded), Media Realm Configuration, Trunk Group, Protocol Definition, Application Network Setting, Proxies, Registration, IP Groups, Coders And Profile Definitions (expanded), Coders (selected), Coders Group Settings, and Tel Profile Settings. The main area is titled 'Coders Table' and contains a table with five columns: 'Coder Name', 'Packetization Time', 'Rate', 'Payload Type', and 'Silence Suppression'. The table has 10 rows. The first row is pre-filled with 'G.711U-law', '20', '64', '0', and 'Disabled'. The remaining 9 rows have empty dropdown menus for each column.

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

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.

Index	Destination Prefix	Source Prefix	Source IP Address	Stripped Digits From Left	Stripped Digits From Right	Prefix to Add	Suffix to Add
0	xxxxx#	*	*	0	0		
1	xxxxxxxxxxxx#	*	*	0	0	91	
2	1xxxxxxxxx#	*	*	0	0	9	
3	+1xxxxxxxxx#	*	*	1	0	9	
4	+	*	*	1	0	9	

Scroll to the right to see the remaining fields.

IP Address	Stripped Digits From Left	Stripped Digits From Right	Prefix to Add	Suffix to Add	Number of Digits to Leave	NPI	TON
0	0	0			255	Not Configured	Not Configured
0	0	0	91		255	Not Configured	Not Configured
0	0	0	9		255	Not Configured	Not Configured
1	0	0	9		255	Not Configured	Not Configured
1	0	0	9		255	Not Configured	Not Configured

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.)

AudioCodes Mediant 1000

Configuration Management Status & Diagnostics

Scenarios Search

Basic Full

Network Settings
Media Settings
PSTN Settings
Sigtran Configuration
Security Settings
Protocol Configuration
Media Realm Configuration
Trunk Group
Protocol Definition
Application Network Setting
Proxies, Registration, IP Groups
Coders And Profile Definitions
SIP Advanced Parameters
Manipulation Tables
General Settings
Dest Number IP->Tel
Dest Number Tel->IP

Destination Phone Number Manipulation Table for Tel-> IP Calls

Note: Select row index to modify the relevant row.

Basic Parameter List

Add

Index	Source Trunk Group	Destination Prefix	Source Prefix	Stripped Digits From Left	Stripped Digits From Right	Prefix to Add	Suffix to Add	Number of Digits to Leave
0	1	5683		0	0	1908848		255
1								

Scroll to the right to see the remaining fields.

AudioCodes Mediant 1000

Configuration Management Status & Diagnostics

Scenarios Search

Basic Full

Network Settings
Media Settings
PSTN Settings
Sigtran Configuration
Security Settings
Protocol Configuration
Media Realm Configuration
Trunk Group
Protocol Definition
Application Network Setting
Proxies, Registration, IP Groups
Coders And Profile Definitions
SIP Advanced Parameters
Manipulation Tables
General Settings
Dest Number IP->Tel
Dest Number Tel->IP

Destination Phone Number Manipulation Table for Tel-> IP Calls

Note: Select row index to modify the relevant row.

Basic Parameter List

Add

Index	Source Trunk Group	Destination Prefix	Source Prefix	Stripped Digits From Left	Stripped Digits From Right	Prefix to Add	Suffix to Add	Number of Digits to Leave
0	1	5683		0	0	1908848		255
1								

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 route phone calls to the BlackBerry® MVS Server.

	Src. Trunk Group ID	Dest. Phone Prefix	Source Phone Prefix	Dest. IP Address	Port	Transport Type	Dest. IPGroup ID
1	1	*	*	10.32.100.231		Not Configured	0
2						Not Configured	
3						Not Configured	
4						Not Configured	
5						Not Configured	
6						Not Configured	
7						Not Configured	
8						Not Configured	
9						Not Configured	
10						Not Configured	

Scroll to the right to see the remaining fields.

Source Phone Prefix	Dest. IP Address	Port	Transport Type	Dest. IPGroup ID	IP Profile ID	Status
	10.32.100.231		Not Configured	0		n/a
			Not Configured			
			Not Configured			
			Not Configured			
			Not Configured			
			Not Configured			
			Not Configured			
			Not Configured			
			Not Configured			
			Not Configured			

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.

AudioCodes Mediant 1000

Configuration Management Status & Diagnostics

Scenarios Search

Basic Full

- Network Settings
- Media Settings
- PSTN Settings
- Sigtran Configuration
- Security Settings
- Protocol Configuration
 - Media Realm Configuration
 - Trunk Group
 - Protocol Definition
 - Application Network Setting
 - Proxies, Registration, IP Groups
 - Coders And Profile Definitions
 - SIP Advanced Parameters
 - Manipulation Tables
 - Routing Tables
 - Alternative Routing
 - Routing General Parameters
 - Tel to IP Routing
 - IP to Trunk Group Routing**
 - Internal DNS Table

IP To Trunk Group Routing Table

Basic Parameter List

Routing Index: 1-10

IP To Tel Routing Mode: Route calls after manipulation

	Dest. Host Prefix	Source Host Prefix	Dest. Phone Prefix	Source Phone Prefix	Source IP Address	Trunk Group ID	IP
1			*	*	10.32.100.231	1	0
2							
3							
4							
5							
6							
7							
8							
9							
10							

Scroll to the right to see the remaining fields.

AudioCodes Mediant 1000

Configuration Management Status & Diagnostics

Scenarios Search

Basic Full

- Network Settings
- Media Settings
- PSTN Settings
- Sigtran Configuration
- Security Settings
- Protocol Configuration
 - Media Realm Configuration
 - Trunk Group
 - Protocol Definition
 - Application Network Setting
 - Proxies, Registration, IP Groups
 - Coders And Profile Definitions
 - SIP Advanced Parameters
 - Manipulation Tables
 - Routing Tables
 - Alternative Routing
 - Routing General Parameters
 - Tel to IP Routing
 - IP to Trunk Group Routing**
 - Internal DNS Table

IP To Trunk Group Routing Table

Basic Parameter List

Routing Index: 1-10

IP To Tel Routing Mode: Route calls after manipulation

Prefix	Source Host Prefix	Dest. Phone Prefix	Source Phone Prefix	Source IP Address	Trunk Group ID	IP Profile ID	Source IP Group ID
		*	*	10.32.100.231	1	0	-1

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.

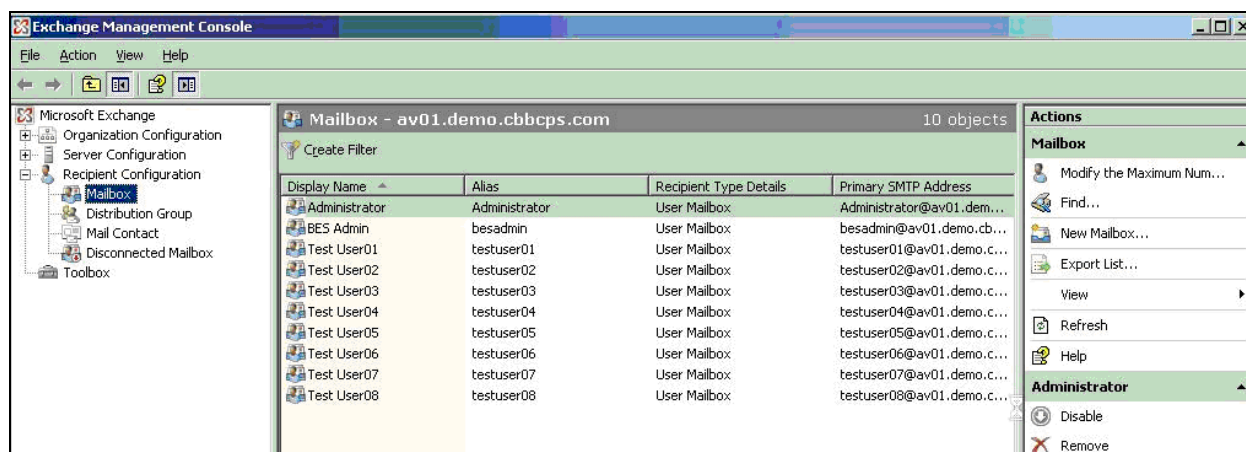
The **Burn and Restart** command from the main screen needs to be executed to apply the above parameter changes.

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

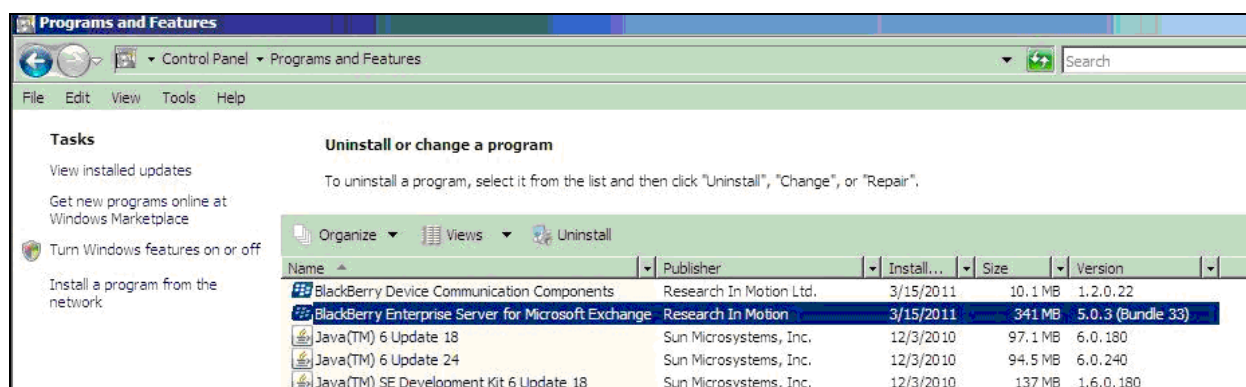
Verify that each user that will be configured on the MVS is also configured as a user within Microsoft Exchange. Log in to the Exchange Management Console. Navigate to **Microsoft Exchange → Recipient Configuration → Mailbox** to view the existing users.



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.

The screenshot shows the BlackBerry Administration Service interface. The left sidebar contains the following sections:

- Quick user search**: A search bar with the label "Name:".
- BlackBerry solution management**: A list of links including "User", "Create a user", "Manage users", "Group", "Role", "Software", "Policy", and "Administrator user".
- Devices**: A list of links including "Attached devices", "Deployment jobs", and "Wireless activations".
- Servers and components**: A link for "BlackBerry Solution topology".
- Preferences**: A link for "My setup".

The main content area is titled "Create a BlackBerry enabled user" and includes the following elements:

- A breadcrumb trail: "User > Create a user".
- A heading: "Create a BlackBerry enabled user".
- A sub-heading: "Search messaging users".
- Form fields for "Email user criteria": "Messaging server display name:" and "Email address:".
- Form fields for "Sort criteria": "Sort by:" with a dropdown menu set to "Display name" and radio buttons for "A to Z" and "Z to A".
- Buttons for "Search" and "Clear".
- Buttons for "Cancel", "Import new users", and "Add user from company directory" (highlighted with a red rectangle).

At the bottom of the page, there is a copyright notice: "Copyright © 1997 - 2011 Research In Motion Limited. All rights reserved. Version: 5.0.3.31".

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**.

User > Create a user

Create a BlackBerry enabled user

You can create a user so that you can assign and activate a BlackBerry® device to the user. The user must exist on your organization's messaging server.



Search messaging users

Email user criteria


Messaging server display name: Email address:

Sort criteria



Sort by: Display name ▾
☒ A to Z ☐ Z to A


 Search  Clear


Showing 1 - 2 of 2


	Messaging server display name	Email address
<input type="checkbox"/>	Administrator	Administrator@av01.demo.cbbcps.com
<input checked="" type="checkbox"/>	BES Admin	besadmin@av01.demo.cbbcps.com

Showing 1 - 2 of 2

 Continue  Cancel

 Import new users

 Refresh available user list from company directory

 Add user from company directory

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.

User > Create a user

Create a BlackBerry enabled user

You can create a user so that you can assign and activate a BlackBerry® device to the user. The user must exist on your organization's messaging server.

Messaging server display name	Email address
BES Admin	besadmin@av01.demo.cbbcps.com

Available BlackBerry Enterprise Server instances

BlackBerry Enterprise Server: AV01-BES

Available groups		Current groups
Administrators BlackBerry Web Desktop Manager users Help desk representatives	Add Add all Remove Remove all	

[Create a user with activation password](#)
[Create a user with generated activation password](#)
[Create a user without activation password](#)
[Cancel](#)

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.

User > Manage users > View user (Test User01)

Manage users

You must search for a user to manage. You can update user information, add or change the groups and roles that a user is assigned to, and delete users.

User Information	Groups	Roles	Software tokens	Component information	Access control rules	Software configuration	Policies
Wi-Fi profiles	VPN profiles						

User information

Display name:	Test User01	User ID:	7
---------------	-------------	----------	---

Authentication type

Active Directory	User name	Password
	The entered data retrieved a user identification from the Active Directory system. The authentication will use the associated Active Directory credentials.	

Associated device properties

PIN	2205EA6E	Device model	9800
Home Carrier	Research In Motion	Current Carrier	AT&T
Phone number	6508923725	Software version	6.0.0.246 (Platform 6.4.0.105)
Associated BlackBerry Enterprise Server	AV01-BES		
Device IT policy	Default	Device IT policy time	4/13/11 7:44:19 AM
Queued IT policy status	Applied successfully		
Last contact date	5/3/11 6:28:24 PM	Last message sent	5/3/11 6:28:23 PM
Result of last transaction to the device	Delivered to device		

Messaging configuration

Default configuration	Description
	The default configuration is created automatically when the BlackBerry Enterprise Server is installed.

Edit user
 Send message to user
 Back to search
 Back to previous search results

BlackBerry Enterprise Server status

- Switch BlackBerry user to different BlackBerry Enterprise Server
- Disable as BlackBerry user

Status

- Delete user
- Reload user

Device activation

- Specify an activation password
- Generate an activation email
- Clear activation password
- Specify new device password and lock device

Device deployment

- Resend service books to a device
- Resend IT policy to a device
- View tasks

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

User > Manage users > View user (Test User01) > View associated device

Devices

You can view and edit information on a BlackBerry® device.

Device information	Messaging	Capabilities	Service books	Applications	Modules
--------------------	-----------	--------------	---------------	--------------	---------

Properties

PIN:	2205EA6E	User-device configuration:	Default configuration
Available memory (KB):	300670	Battery level (%):	20
Uptime:	1 day(s), 0 hours, 1 minutes, 4 seconds		

Properties status

Last updated:	5/3/11 4:40:52 PM
---------------	-------------------

Hardware

BlackBerry device model:	9600	Network type:	3G
Memory (MB):	500	Serial number (IMEI):	004401.13.600464.1
Frequencies:	GSM 850, GSM 900, GSM 1800, GSM 1900	Secured boot ROM:	Yes
Home carrier:	Research In Motion	Display screen height:	360
Display screen width:	480		

Software

Platform version:	6.4.0.105	BlackBerry Device Software version:	6.0.0.246
Phone number:	6508923725	Security password:	No
Current carrier:	AT&T	Direct connect ID:	

Last reported IT policy on device

IT policy name:	Default	IT policy time:	4/13/11 10:44:23 AM
-----------------	---------	-----------------	---------------------

Policies update status

Last updated:	5/3/11 5:43:47 AM
---------------	-------------------

Queued IT policy

IT policy name:	Default	IT policy status:	Applied successfully
IT policy sent:	4/13/11 7:44:12 AM	IT policy received:	4/13/11 7:44:19 AM

BlackBerry Enterprise Server information

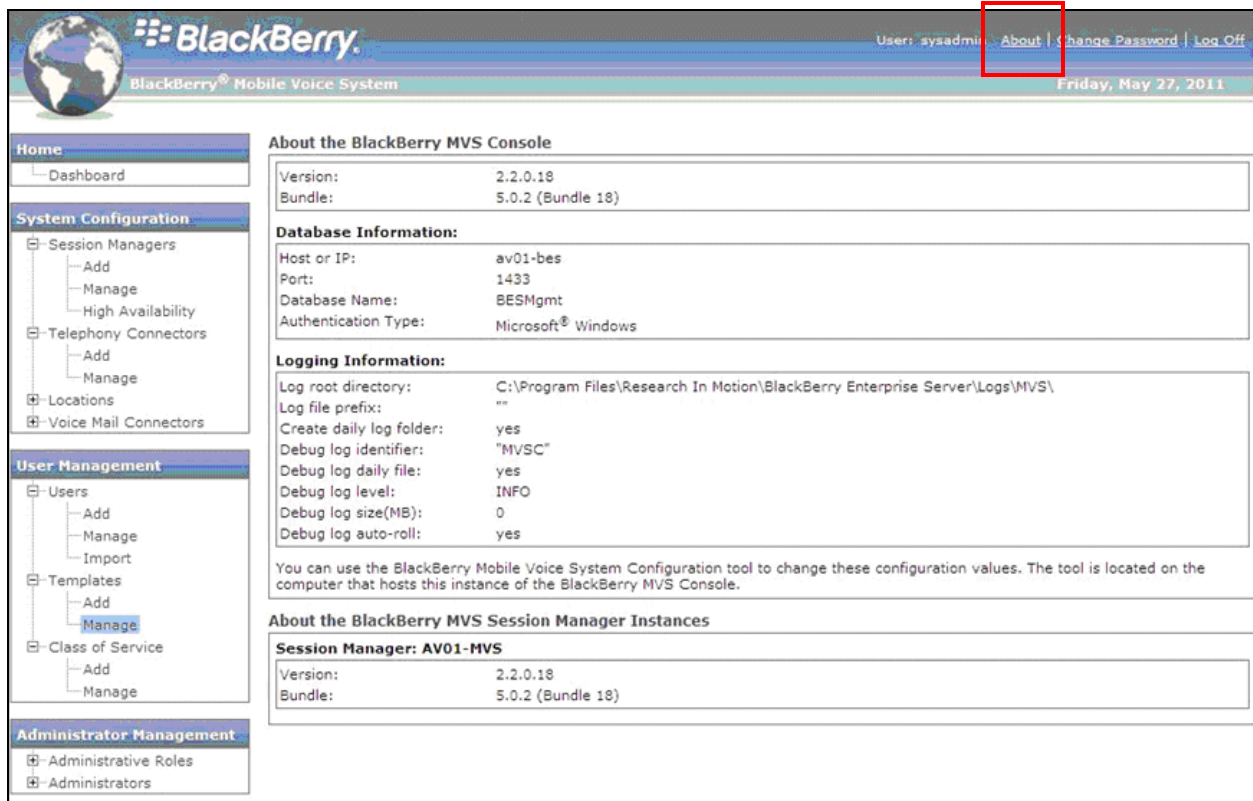
Associated BlackBerry Enterprise Server:	AV01-BES
--	----------

[View user information](#)
[Back to search](#)
[Back to previous search results](#)

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.



The screenshot displays the BlackBerry Mobile Voice System (MVS) console interface. The top navigation bar includes the BlackBerry logo, the text "BlackBerry® Mobile Voice System", and user information "User: sysadmin". The "About" link is highlighted with a red box. The left-hand navigation menu is organized into sections: Home (Dashboard), System Configuration (Session Managers, Telephony Connectors, Locations, Voice Mail Connectors), User Management (Users, Templates, Class of Service), and Administrator Management (Administrative Roles, Administrators). The main content area shows the "About the BlackBerry MVS Console" page, which includes the following information:

About the BlackBerry MVS Console	
Version:	2.2.0.18
Bundle:	5.0.2 (Bundle 18)

Database Information:

Host or IP:	av01-bes
Port:	1433
Database Name:	BESMgmt
Authentication Type:	Microsoft® Windows

Logging Information:

Log root directory:	C:\Program Files\Research In Motion\BlackBerry Enterprise Server\Logs\MVS\
Log file prefix:	""
Create daily log folder:	yes
Debug log identifier:	"MVSC"
Debug log daily file:	yes
Debug log level:	INFO
Debug log size(MB):	0
Debug log auto-roll:	yes

You can use the BlackBerry Mobile Voice System Configuration tool to change these configuration values. The tool is located on the computer that hosts this instance of the BlackBerry MVS Console.

About the BlackBerry MVS Session Manager Instances

Session Manager: AV01-MVS	
Version:	2.2.0.18
Bundle:	5.0.2 (Bundle 18)

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. 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**.

Add Session Manager

Enable High Availability ☐

Instance Name: AV01-MVS *

SIP IP Address: 10.32.100.231 *

Line Port: 5060 *

Trunk Port: 6060 *

DID/DDI number for BlackBerry device-initiated calling:
Audio Codes: +19088485683

BlackBerry Enterprise Servers *

Associated	Instance Name	Secure Connection Password
<input checked="" type="checkbox"/>	AV01-BES	The BlackBerry MVS will use the default secure connection password. <input type="checkbox"/> Change the secure connection password that the BlackBerry MVS uses.

Save Cancel

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**.

Add Telephony Connector

Display Name: Audio Codes *

Type: SIP Gateway *

IP Address: 10.32.100.236 *

Host Name: 10.32.100.236 *

Trunk Port: 5060 *

PBX Initiated Calling

Caller Identification Number *

Number	Action
	+
9088485471	✖

Phone Number Translation

Location: United States of America

International Direct Dialing: 011

National Direct Dialing: 1

Home Country Code: 1

Save Cancel

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).

Add Template

Template Name: Default *

BlackBerry MVS Line Configuration

BlackBerry MVS Line Label: Work *

Default line for outgoing calls: BlackBerry MVS Line

When the BlackBerry MVS Line is unavailable use the Mobile Line: Ask

Default network for BlackBerry MVS calls: Wi-Fi

When Wi-Fi is unavailable, use Mobile: Yes

Outgoing call setup sound: Tones

Voice Mail Access Number:

MWI Notifications: Enabled

Caller Restrictions

Caller Restriction: All Callers Except Blocked

☒ Allow calls from unknown numbers


Blocked Callers (max limit = 20)

Number	Name	Action
		+


Allowed Callers (max limit = 20)

Number	Name	Action
		+

Scroll down to view the remaining fields.

Call Scheduling (in office time zone) 

Office Time Zone (GMT-08:00) Pacific Time (US & Canada) ▼

Day of Week	Allow Calls	Start of Day (HHMM)	End of Day (HHMM)
Weekly	Always ▼	<input type="text"/>	<input type="text"/> 
Sunday	Always ▼		
Monday	Always ▼		
Tuesday	Always ▼		
Wednesday	Always ▼		
Thursday	Always ▼		
Friday	Always ▼		
Saturday	Always ▼		

Save Cancel

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.

The screenshot displays the 'Add Class of Service' configuration interface. The 'Class of Service Name' field is set to 'DNIS Pool'. The 'BlackBerry MVS Call Features' section contains several checked options: 'User can transfer a call', 'User can move a call to the desk phone number', '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', and 'Device can set the mobile phone number'. The 'Call Direction' section shows 'Incoming Calls' and 'Outgoing Calls' both set to 'BlackBerry Smartphone → PBX'. The 'Wi-Fi Profile Control' section has 'Allow Voice over Wi-Fi calling on user created Wi-Fi profiles' checked. The 'Device Settings' section shows 'RTP Local Port' as 51100 and 'Preferred Order of Device Codecs' as G.711 μ-law, G.711 A-law, G.729a, and G.729b.

Add Class of Service	
Class of Service Name	DNIS Pool *
BlackBerry MVS Call Features	
<input checked="" type="checkbox"/> User can transfer a call	
<input checked="" type="checkbox"/> User can move a call to the desk phone number	
<input checked="" type="checkbox"/> User can move a call to the mobile phone number	
<input checked="" type="checkbox"/> User can move a call to a one time number	
<input checked="" type="checkbox"/> User can move a call from Voice over Wi-Fi to Voice over Mobile	
<input checked="" type="checkbox"/> User can move a call from Voice over Mobile to Voice over Wi-Fi	
<input checked="" type="checkbox"/> User can have a schedule for incoming calls	
<input checked="" type="checkbox"/> Device can set the mobile phone number	
Call Direction	
Incoming Calls	<input type="radio"/> PBX → BlackBerry Smartphone
	<input checked="" type="radio"/> BlackBerry Smartphone → PBX
Outgoing Calls	<input type="radio"/> PBX → BlackBerry Smartphone
	<input checked="" type="radio"/> BlackBerry Smartphone → PBX
Wi-Fi Profile Control	
<input checked="" type="checkbox"/> Allow Voice over Wi-Fi calling on user created Wi-Fi profiles	
Device Settings	
RTP Local Port	51100 *
Preferred Order of Device Codecs	G.711 μ-law G.711 A-law G.729a G.729b

Scroll down to see the remaining values.

The screenshot shows a 'User Edit Control' dialog box with two main sections: 'User Edit Control' and 'Scheduling Control'. The 'User Edit Control' section contains eight checked checkboxes for various user permissions. The 'Scheduling Control' section contains a 'Select All' and 'Select None' link, followed by seven checked checkboxes for daily scheduling. At the bottom are 'Save' and 'Cancel' buttons.

Section	Item	Status
User Edit Control	User may change the mobile phone number	Checked
	User may change the call move to desk phone number	Checked
	User may change the BlackBerry MVS line label	Checked
	User may change the default line for outgoing calls	Checked
	User may change the outgoing call setup sound	Checked
	User may change the Voice Mail access number	Checked
	User may change caller restrictions	Checked
	User may change the default network for BlackBerry MVS calls	Checked
Scheduling Control	Select All - Select None	Link
	User may set Monday scheduling	Checked
	User may set Tuesday scheduling	Checked
	User may set Wednesday scheduling	Checked
	User may set Thursday scheduling	Checked
	User may set Friday scheduling	Checked
	User may set Saturday scheduling	Checked
	User may set Sunday scheduling	Checked

The example below shows the ANI pool class of service.

The screenshot shows an 'Add Class of Service' dialog box. The 'Class of Service Name' field is highlighted with a red box and contains the text 'ANI Pool'. Below this are several sections: 'BlackBerry MVS Call Features' with eight checked checkboxes; 'Call Direction' with two sections (Incoming and Outgoing Calls) each having two radio button options, with the first option in each section selected; 'Wi-Fi Profile Control' with one checked checkbox; and 'Device Settings' with an 'RTP Local Port' field set to '51100' and a 'Preferred Order of Device Codecs' list containing 'G.711 µ-law', 'G.711 A-law', 'G.729a', and 'G.729b'.

Section	Item	Status
BlackBerry MVS Call Features	User can transfer a call	Checked
	User can move a call to the desk phone number	Checked
	User can move a call to the mobile phone number	Checked
	User can move a call to a one time number	Checked
	User can move a call from Voice over Wi-Fi to Voice over Mobile	Checked
	User can move a call from Voice over Mobile to Voice over Wi-Fi	Checked
	User can have a schedule for incoming calls	Checked
	Device can set the mobile phone number	Checked
Call Direction	Incoming Calls: PBX → BlackBerry Smartphone	Selected
	Incoming Calls: BlackBerry Smartphone → PBX	Not Selected
	Outgoing Calls: PBX → BlackBerry Smartphone	Selected
	Outgoing Calls: BlackBerry Smartphone → PBX	Not Selected
Wi-Fi Profile Control	Allow Voice over Wi-Fi calling on user created Wi-Fi profiles	Checked
Device Settings	RTP Local Port	51100
	Preferred Order of Device Codecs	G.711 µ-law, G.711 A-law, G.729a, G.729b

Scroll down to see the remaining values.

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

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

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)**.

Add BlackBerry MVS Users

Use this page to add BlackBerry MVS users. All users that you want to add to the BlackBerry MVS must have BlackBerry Enterprise Server user accounts.

User Name

BlackBerry Enterprise Server AV01-BES

Search Clear Cancel

Template

Class of Service

Telephony Connector

Voice Mail Connector

Default

DNIS Pool *

Audio Codes *

None

(1 selected user)

	User Name	Mobile	Extension	Desk	Direct Dial
<input checked="" type="checkbox"/>	Test User02	55682	55682 *	55682 *	9088485682

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**.

Manage BlackBerry MVS Users

Use this page to manage BlackBerry MVS users that are already configured to use the BlackBerry MVS. Use the search criteria to search for existing BlackBerry MVS users.

Search Criteria

User Name Telephony Connector

Extension Voice Mail Connector

BlackBerry Enterprise Server Class of Service

Search Results

(No selected users)

<input type="checkbox"/> User Name	Extension	Class of Service	MVS Service	Telephony Connector
<input type="checkbox"/> Test User02	55684	DNIS pool	Enabled	Audio Codes
<input type="checkbox"/> Test User03	55685	ANI Pool	Enabled	Audio Codes
<input type="checkbox"/> Test User04	55681	DNIS pool	Enabled	Audio Codes
<input type="checkbox"/> Test User08	55682	ANI Pool	Enabled	Audio Codes

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.

Edit User - Test User04

General Attributes

MVS Service

Extension

Direct Dial Number

Mobile Phone Number

Desk Phone Number

Telephony Connector

Telephony Connector

Voice Mail Connector

Voice Mail Connector

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

Edit User - Test User04

General Settings Class of Service

BlackBerry MVS Line Configuration

BlackBerry MVS Line Label: *

Default line for outgoing calls:

When the BlackBerry MVS Line is unavailable use the Mobile Line:

Default network for BlackBerry MVS calls:

When Wi-Fi is unavailable, use Mobile:

Outgoing call setup sound:

Voice Mail Access Number:

MWI Notifications:

Caller Restrictions

Caller Restriction:

☒ Allow calls from unknown numbers

Blocked Callers (max limit = 20)

Number	Name	Action
<input type="text"/>	<input type="text"/>	<input type="button" value="Add"/>

Allowed Callers (max limit = 20)

Number	Name	Action
<input type="text"/>	<input type="text"/>	<input type="button" value="Add"/>

Call Scheduling (in office time zone)

Office Time Zone:

Day of Week	Allow Calls	Start of Day (HHMM)	End of Day (HHMM)
Weekly	<input type="text" value="Always"/>	<input type="text"/>	<input type="text"/> <input type="button" value="f"/>
Sunday	<input type="text" value="Always"/>		
Monday	<input type="text" value="Always"/>		
Tuesday	<input type="text" value="Always"/>		
Wednesday	<input type="text" value="Always"/>		
Thursday	<input type="text" value="Always"/>		
Friday	<input type="text" value="Always"/>		
Saturday	<input type="text" value="Always"/>		

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

Edit User - Test User04

General
Settings
Class of Service

Class of Service Name

Class of Service Name
DNIS pool

BlackBerry MVS Call Features

☒ User can transfer a call
☐ User can move a call to the desk phone number
☒ 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
☒ Device can set the mobile phone number

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 Local Port
51100
Preferred Order of Device Codes

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

Scheduling Control

☒ 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

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, Document 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>.

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