



Avaya Solution & Interoperability Test Lab

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

Abstract

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

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

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

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1 Introduction

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

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

2 General Test Approach and Test Results

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

2.1 Interoperability Compliance Testing

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

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

2.2 Test Results

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

The following observations were made:

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

2.3 Support

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

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

3 Reference Configuration

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

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

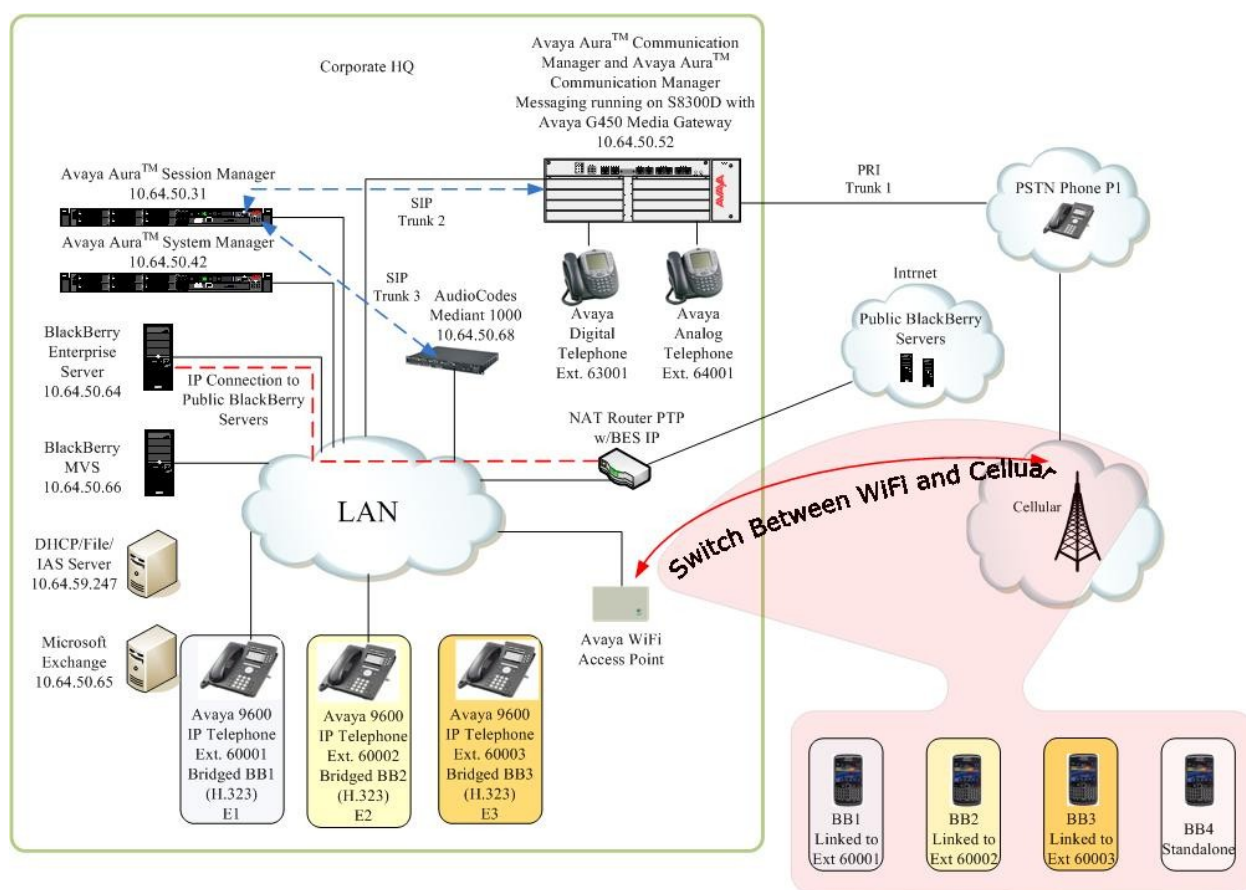


Figure 1: Avaya/RIM MVS Solution Diagram

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

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

Inbound call (Device-initiated calling)

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

Inbound call (PBX-initiated calling)

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

Outbound call (Device-initiated calling)

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

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

Outbound call (PBX-initiated calling)

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

4 Equipment and Software Validated

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

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

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

5 Configure Avaya Aura® Communication Manager

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

5.1 System Parameters Customer Options

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

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

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

display system-parameters customer-options		Page	1 of 11
OPTIONAL FEATURES			
G3 Version: V16	Software Package: Enterprise		
Location: 2	System ID (SID): 1		
Platform: 28	Module ID (MID): 1		
		USED	
	Platform Maximum Ports: 6400	54	
	Maximum Stations: 2400	9	
	Maximum XMOBILE Stations: 2400	0	
	Maximum Off-PBX Telephones – EC500: 9600	5	
	Maximum Off-PBX Telephones – OPS: 9600	5	
	Maximum Off-PBX Telephones – PBFMC: 9600	0	
	Maximum Off-PBX Telephones – PVFMC: 9600	0	
	Maximum Off-PBX Telephones – SCCAN: 0	0	
	Maximum Survivable Processors: 313	0	

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

display system-parameters customer-options		Page	3 of	11
OPTIONAL FEATURES				
Abbreviated Dialing Enhanced List?	y	Audible Message Waiting?	y	
Access Security Gateway (ASG)?	n	Authorization Codes?	y	
Analog Trunk Incoming Call ID?	y	CAS Branch?	n	
A/D Grp/Sys List Dialing Start at 01?	y	CAS Main?	n	
Answer Supervision by Call Classifier?	y	Change COR by FAC?	n	
ARS?	y	Computer Telephony Adjunct Links?	y	
ARS/AAR Partitioning?	y	Cvg Of Calls Redirected Off-net?	y	
ARS/AAR Dialing without FAC?	y	DCS (Basic)?	y	
ASAI Link Core Capabilities?	n	DCS Call Coverage?	y	
ASAI Link Plus Capabilities?	n	DCS with Rerouting?	y	
Async. Transfer Mode (ATM) PNC?	n	Digital Loss Plan Modification?	y	
Async. Transfer Mode (ATM) Trunking?	n	DS1 MSP?	y	
ATM WAN Spare Processor?	n	DS1 Echo Cancellation?	y	
ATMS?	y			
Attendant Vectoring?	y			

5.2 Dial Plan and Access Codes

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

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

display dialplan analysis			DIAL PLAN ANALYSIS TABLE						Page	1 of	12
			Location: all						Percent Full: 2		
Dialed String	Total Length	Call Type	Dialed String	Total Length	Call Type	Dialed String	Total Length	Call Type			
6	5	ext									
8	1	fac									
9	1	fac									
*	4	dac									

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

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

5.3 Configure IP Node Names

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

change node-names ip		Page 1 of 2
IP NODE NAMES		
Name	IP Address	
default	0.0.0.0	
msgserver	10.64.50.52	
procr	10.64.50.52	
procr6	::	
sm5031	10.64.50.31	
(5 of 5 administered node-names were displayed)		
Use 'list node-names' command to see all the administered node-names		
Use 'change node-names ip xxx' to change a node-name 'xxx' or add a node-name		

5.4 Configure IP Network Region

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

change ip-network-region 1

Page 1 of 20

IP NETWORK REGION

Region: 1

Location: 1 **Authoritative Domain: avaya.com**

Name:

MEDIA PARAMETERS

Codec Set: 1

Intra-region IP-IP Direct Audio: yes

Inter-region IP-IP Direct Audio: yes

UDP Port Min: 49152

IP Audio Hairpinning? y

UDP Port Max: 65535

DIFFSERV/TOS PARAMETERS

Call Control PHB Value: 46

Audio PHB Value: 46

Video PHB Value: 26

802.1P/Q PARAMETERS

Call Control 802.1p Priority: 6

Audio 802.1p Priority: 6

Video 802.1p Priority: 5

AUDIO RESOURCE RESERVATION PARAMETERS

H.323 IP ENDPOINTS

RSVP Enabled? n

H.323 Link Bounce Recovery? y

Idle Traffic Interval (sec): 20

Keep-Alive Interval (sec): 5

Keep-Alive Count: 5

5.5 Configure IP Codec Set

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

change ip-codec-set 1 Page 1 of 2

IP Codec Set

Codec Set: 1

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

Media Encryption

1: none

2:

3:

5.6 Configure SIP Trunk to Avaya Aura® Session Manager

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

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

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

add signaling-group 2		Page 1 of 1
SIGNALING GROUP		
Group Number: 10	Group Type: sip	
IMS Enabled? n	Transport Method: tcp	
Q-SIP? n		SIP Enabled LSP? n
IP Video? n		Enforce SIPS URI for SRTP? y
Peer Detection Enabled? y	Peer Server: Others	
Near-end Node Name: procr	Far-end Node Name: sm5031	
Near-end Listen Port: 5060	Far-end Listen Port: 5060	
	Far-end Network Region: 1	
Far-end Domain: avaya.com		
Incoming Dialog Loopbacks: eliminate	Bypass If IP Threshold Exceeded? n	
DTMF over IP: rtp-payload	RFC 3389 Comfort Noise? n	
Session Establishment Timer(min): 3	Direct IP-IP Audio Connections? y	
Enable Layer 3 Test? y	IP Audio Hairpinning? n	
H.323 Station Outgoing Direct Media? n	Initial IP-IP Direct Media? n	
	Alternate Route Timer(sec): 6	

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

add trunk-group 2		Page 1 of 21
TRUNK GROUP		
Group Number: 2	Group Type: sip	CDR Reports: y
Group Name: To sm5032	COR: 1	TN: 1 TAC: *002
Direction: two-way	Outgoing Display? n	
Dial Access? n		Night Service:
Queue Length: 0		
Service Type: tie	Auth Code? n	
	Member Assignment Method: auto	
	Signaling Group: 2	
	Number of Members: 10	

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

add trunk-group 2		Page 3 of 21
TRUNK FEATURES		
ACA Assignment? n	Measured: none	Maintenance Tests? y
Numbering Format: private		
	UI Treatment: service-provider	
	Replace Restricted Numbers? n	
	Replace Unavailable Numbers? n	
Modify Tandem Calling Number: no		
Show ANSWERED BY on Display? y		

5.7 Configure Route Pattern

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

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

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

change route-pattern 3													Page		1 of 3				
Pattern Number: 3													Pattern Name: To sm5031						
SCCAN? n													Secure SIP? n						
Grp	FRL	NPA	Pfx	Hop	Toll	No.	Inserted						DCS/	IXC					
No			Mrk	Lmt	List	Del	Digits						QSIG						
							Dgts						Intw						
1:	2	0						0						n	user				
2:													n	user					
3:													n	user					
4:													n	user					
5:													n	user					
6:													n	user					
BCC VALUE													TSC	CA-TSC	ITC BCIE Service/Feature PARM		No.	Numbering	LAR
0 1 2 M 4 W														Request			Dgts	Format	
															Subaddress				
1:	y	y	y	y	y	n	n	rest					pub-unk	none					
2:	y	y	y	y	y	n	n	rest						none					
3:	y	y	y	y	y	n	n	rest						none					
4:	y	y	y	y	y	n	n	rest						none					
5:	y	y	y	y	y	n	n	rest						none					
6:	y	y	y	y	y	n	n	rest						none					

5.8 Configure Automatic Alternate Routing

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

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

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

change aar analysis 6							
AAR DIGIT ANALYSIS TABLE							
Location: all							
Percent Full: 2							
	Dialed String	Total Min	Total Max	Route Pattern	Call Type	Node Num	ANI Reqd
	6	5	5	3	aar		n

change aar analysis 130							
AAR DIGIT ANALYSIS TABLE							
Location: all							
Percent Full: 2							
	Dialed String	Total Min	Total Max	Route Pattern	Call Type	Node Num	ANI Reqd
	130	11	11	3	aar		n

5.9 Incoming Call Treatment for the PSTN Trunk

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

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

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

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

5.10 Stations and Off-PBX Station Mapping For Mobile Devices

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

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

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

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

add station 60001		Page	1 of	5
STATION				
Extension: 60001	Lock Messages? n	BCC: 0		
Type: 9620	Security Code: 123456	TN: 1		
Port: IP	Coverage Path 1: 99	COR: 1		
Name: Station 60009	Coverage Path 2:	COS: 1		
	Hunt-to Station:			
STATION OPTIONS				
	Time of Day Lock Table:			
Loss Group: 19	Personalized Ringing Pattern: 1			
	Message Lamp Ext: 60001			
Speakerphone: 2-way	Mute Button Enabled? y			
Display Language: english				
Survivable GK Node Name:				
Survivable COR: internal	Media Complex Ext:			
Survivable Trunk Dest? y	IP SoftPhone? n			
	IP Video? n			
Short/Prefixed Registration Allowed: default				
Customizable Labels? y				

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

add station 60001		Page 4 of 5
STATION		
SITE DATA		
Room:		Headset? n
Jack:		Speaker? n
Cable:		Mounting: d
Floor:		Cord Length: 0
Building:		Set Color:
ABBREVIATED DIALING		
List1:	List2:	List3:
BUTTON ASSIGNMENTS		
1: call-appr	4: ec500	Timer? n
2: call-appr	5:	
3: call-appr	6:	
voice-mail		

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

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

add off-pbx-telephone station-mapping							Page 1 of 3
STATIONS WITH OFF-PBX TELEPHONE INTEGRATION							
Station Extension	Application	Dial Prefix	CC	Phone Number	Trunk Selection	Config Set	Dual Mode
60001	EC500	-		60001	aar	5	
-							

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

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

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

change off-pbx-telephone configuration-set 5	Page 1 of 1
 CONFIGURATION SET: 5 Configuration Set Description: RIM Calling Number Style: network CDR for Origination: phone-number CDR for Calls to EC500 Destination? y Fast Connect on Origination? n Post Connect Dialing Options: dtmf Cellular Voice Mail Detection: none Barge-in Tone? n Calling Number Verification? n Call Appearance Selection for Origination: primary-first Confirmed Answer? n Use Shared Voice Connections for Second Call Answered? n Use Shared Voice Connections for Second Call Initiated? n	

6 Configure Avaya Aura® Session Manager

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

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

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

6.1 Specify SIP Domain

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

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

Click **Commit**.

The screenshot shows the Avaya Aura® System Manager 6.1 web interface. The top navigation bar includes the Avaya logo, the title 'Avaya Aura® System Manager 6.1', and links for 'Help', 'About', 'Change Password', and 'Log off admin'. Below the navigation bar, there are tabs for 'Routing' and 'Home'. The left sidebar contains a tree view with the following items: Routing, Domains (selected), Locations, Adaptations, SIP Entities, Entity Links, Time Ranges, Routing Policies, Dial Patterns, Regular Expressions, and Defaults. The main content area is titled 'Domain Management' and shows a table with one item. The table has columns for Name, Type, Default, and Notes. The 'Name' column contains 'avaya.com', the 'Type' column contains 'sip', and the 'Default' column contains a checkbox. There are 'Commit' and 'Cancel' buttons at the bottom right of the table. A message '* Input Required' is displayed below the table.

Name	Type	Default	Notes
* avaya.com	sip	<input type="checkbox"/>	

6.2 Add Locations

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

Under *General*:

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

Under *Location Pattern*:

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

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

AVAYA Avaya Aura® System Manager 6.1 [Help](#) | [About](#) | [Change Password](#) | [Log off admin](#)

[Routing](#) [Home](#)

Home / Elements / Routing / Locations - Location Details

Location Details [Help ?](#) [Commit](#) [Cancel](#)

Call Admission Control has been set to ignore SDP. All calls will be counted using the Default Audio Bandwidth.
See Session Manager -> Session Manager Administration -> Global Setting

General

* **Name:**

Notes:

Overall Managed Bandwidth

Managed Bandwidth Units:

Total Bandwidth:

Per-Call Bandwidth Parameters

* **Default Audio Bandwidth:**

Location Pattern

[Add](#) [Remove](#)

1 Item [Refresh](#) [Filter: Enable](#)

<input type="checkbox"/>	IP Address Pattern	Notes
<input type="checkbox"/>	* 10.64.50.*	<input type="text"/>

Select : All, None

* Input Required [Commit](#) [Cancel](#)

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

AVAYA Avaya Aura® System Manager 6.1 [Help](#) | [About](#) | [Change Password](#) | [Log off admin](#)

[Routing](#) × [Home](#)

Home / Elements / Routing / Locations - Location Details [Help ?](#)

Location Details [Commit](#) [Cancel](#)

Call Admission Control has been set to ignore SDP. All calls will be counted using the Default Audio Bandwidth.
See Session Manager -> Session Manager Administration -> Global Setting

General

* **Name:**

Notes:

Overall Managed Bandwidth

Managed Bandwidth Units:

Total Bandwidth:

Per-Call Bandwidth Parameters

* **Default Audio Bandwidth:**

Location Pattern

[Add](#) [Remove](#)

1 Item [Refresh](#) [Filter: Enable](#)

<input type="checkbox"/>	IP Address Pattern	Notes
<input type="checkbox"/>	* <input type="text" value="10.64.50.68"/>	<input type="text"/>

Select : All, None

* **Input Required** [Commit](#) [Cancel](#)

6.3 Add SIP Entities

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

6.3.1 Avaya Aura® Session Manager

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

Under *General*:

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

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

The screenshot displays the Avaya Aura® System Manager 6.1 web interface. The top navigation bar includes the Avaya logo, the title "Avaya Aura® System Manager 6.1", and links for "Help", "About", "Change Password", and "Log off admin". A breadcrumb trail shows "Home / Elements / Routing / SIP Entities - SIP Entity Details". On the left, a sidebar menu lists various configuration options, with "SIP Entities" highlighted. The main content area is titled "SIP Entity Details" and features a "General" tab. The form contains the following fields: "Name" (value: sm5031), "FQDN or IP Address" (value: 10.64.50.31), "Type" (dropdown menu showing "Session Manager"), "Notes" (empty text area), "Location" (dropdown menu), "Outbound Proxy" (dropdown menu), "Time Zone" (dropdown menu showing "America/Denver"), and "Credential name" (empty text area). At the bottom, there is a "SIP Link Monitoring" section with a dropdown menu set to "Use Session Manager Configuration". "Commit" and "Cancel" buttons are located in the top right corner of the form area.

6.3.2 Avaya Aura® Communication Manager

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

Under *General*:

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

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

The screenshot displays the Avaya Aura® System Manager 6.1 web interface. The top navigation bar includes the Avaya logo, the title "Avaya Aura® System Manager 6.1", and links for "Help", "About", "Change Password", and "Log off admin". A breadcrumb trail shows "Home / Elements / Routing / SIP Entities - SIP Entity Details". On the left, a sidebar menu lists various configuration categories, with "SIP Entities" highlighted. The main content area is titled "SIP Entity Details" and features a "General" tab. The form contains several fields: "Name" (cm5052), "FQDN or IP Address" (10.64.50.52), "Type" (CM), "Notes", "Adaptation", "Location" (Avaya CO), "Time Zone" (America/Denver), "Override Port & Transport with DNS SRV" (unchecked), "SIP Timer B/F (in seconds)" (4), "Credential name", "Call Detail Recording" (none), and "SIP Link Monitoring" (Use Session Manager Configuration). "Commit" and "Cancel" buttons are located at the top right of the form area.

6.3.3 AudioCodes Mediant 1000

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

Under *General*:

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

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

AVAYA Avaya Aura® System Manager 6.1 [Help](#) | [About](#) | [Change Password](#) | [Log off admin](#)

[Routing](#) [Home](#)

[Home / Elements / Routing / SIP Entities - SIP Entity Details](#)

SIP Entity Details [Help ?](#) [Commit](#) [Cancel](#)

General

* **Name:** Audio_Codes_Med_1000

* **FQDN or IP Address:** 10.64.50.68

Type: Other

Notes:

Adaptation:

Location: RIM

Time Zone: America/Denver

Override Port & Transport with DNS SRV: ☐

* **SIP Timer B/F (in seconds):** 4

Credential name:

Call Detail Recording: none

SIP Link Monitoring

SIP Link Monitoring: Link Monitoring Enabled

* **Proactive Monitoring Interval (in seconds):** 900

* **Reactive Monitoring Interval (in seconds):** 120

* **Number of Retries:** 1

6.4 Add Entity Links

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

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

Repeat configuration for Communication Manager and the AudioCodes Mediant 1000.

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

Session Manager ↔ Communication Manager

The screenshot shows the Avaya Aura System Manager 6.1 interface. The left sidebar contains a navigation menu with options: Routing, Domains, Locations, Adaptations, SIP Entities, Entity Links (selected), Time Ranges, Routing Policies, Dial Patterns, Regular Expressions, and Defaults. The main content area is titled 'Entity Links' and shows a table with one configuration row. The table has columns: Name, SIP Entity 1, Protocol, Port, SIP Entity 2, Port, Connection Policy, and Notes. The configuration row shows: Name: cm5052, SIP Entity 1: sm5031, Protocol: TCP, Port: 5060, SIP Entity 2: cm5052, Port: 5060, Connection Policy: Trusted, and Notes: (empty). There are 'Commit' and 'Cancel' buttons at the top right and bottom right of the table. A 'Filter: Enable' link is also present. A message '* Input Required' is displayed at the bottom left of the table area.

Name	SIP Entity 1	Protocol	Port	SIP Entity 2	Port	Connection Policy	Notes
* cm5052	* sm5031	TCP	* 5060	* cm5052	* 5060	Trusted	

Session Manager ↔ AudioCodes Mediant 1000

Avaya Aura® System Manager 6.1

Help | About | Change Password | Log off admin

Routing x Home

Home / Elements / Routing / Entity Links - Entity Links

Entity Links

1 Item Refresh Filter: Enable

Name	SIP Entity 1	Protocol	Port	SIP Entity 2	Port	Connection Policy	Notes
* Audio_Codes_Med_1	* sm5031	TCP	* 5060	* Audio_Codes_Med_1000	* 5060	Trusted	

* Input Required

Commit Cancel

6.5 Add Routing Policies

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

Under *General*:

Enter a descriptive name in **Name**.

Under *SIP Entity as Destination*:

Click **Select**, and then select the appropriate SIP entity to which this routing policy applies.

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

Avaya Aura® System Manager 6.1

Help | About | Change Password | Log off admin

Routing x Home

Home / Elements / Routing / Routing Policies - Routing Policy Details

Routing Policy Details

General

* Name: To cm50502

Disabled: ☐

Notes:

SIP Entity as Destination

Select

Name	FQDN or IP Address	Type	Notes
cm5052	10.64.50.52	CM	

Commit Cancel

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

The screenshot shows the Avaya Aura System Manager 6.1 web interface. The left sidebar contains a navigation menu with options: Routing, Domains, Locations, Adaptations, SIP Entities, Entity Links, Time Ranges, Routing Policies (highlighted), Dial Patterns, Regular Expressions, and Defaults. The main content area is titled 'Routing Policy Details' and shows the 'General' tab. The 'Name' field is set to 'Audio_Codes_Med_1000'. There are checkboxes for 'Disabled' and a 'Notes' field. Below this, the 'SIP Entity as Destination' section has a 'Select' button. At the bottom, a table lists the configuration details:

Name	FQDN or IP Address	Type	Notes
Audio_Codes_Med_1000	10.64.50.68	Other	

6.6 Add Dial Patterns

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

Under *General*:

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

Under *Originating Locations and Routing Policies*:

Click **Add**, and then select the appropriate location and routing policy from the list.

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



Routing

Domains

Locations

Adaptations

SIP Entities

Entity Links

Time Ranges

Routing Policies

Dial Patterns

Regular Expressions

Defaults

Home / Elements / Routing / Dial Patterns - Dial Pattern Details

Dial Pattern Details

[Help ?](#)

Commit

Cancel

General

* Pattern: 6

* Min: 5

* Max: 5

Emergency Call: ☐

SIP Domain: -ALL-

Notes:

Originating Locations and Routing Policies

Add

Remove

1 Item | Refresh

Filter: Enable

<input type="checkbox"/>	Originating Location Name 1 ▲	Originating Location Notes	Routing Policy Name	Rank 2 ▲	Routing Policy Disabled	Routing Policy Destination	Routing Policy Notes
<input type="checkbox"/>	-ALL-	Any Locations	cm5052	0	<input type="checkbox"/>	cm5052	


Select : All, None

Denied Originating Locations

Add

Remove

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


Avaya Aura® System Manager 6.1
[Help](#) | [About](#) | [Change Password](#) | [Log off admin](#)

[Routing](#) * [Home](#)

Routing

- Domains
- Locations
- Adaptations
- SIP Entities
- Entity Links
- Time Ranges
- Routing Policies
- Dial Patterns**
- Regular Expressions
- Defaults

[Home](#) / [Elements](#) / [Routing](#) / [Dial Patterns - Dial Pattern Details](#)

Dial Pattern Details
[Help ?](#)
[Commit](#) [Cancel](#)

General

* **Pattern:**

* **Min:**

* **Max:**

Emergency Call: ☐

SIP Domain:

Notes:

Originating Locations and Routing Policies

[Add](#) [Remove](#)

1 Item [Refresh](#) [Filter: Enable](#)

<input type="checkbox"/>	Originating Location Name ¹	Originating Location Notes	Routing Policy Name	Rank ²	Routing Policy Disabled	Routing Policy Destination	Routing Policy Notes
<input type="checkbox"/>	-ALL-	Any Locations	To cm50502	0	<input type="checkbox"/>	cm5052	

Select : All, None

Denied Originating Locations

[Add](#) [Remove](#)

0 Items [Refresh](#) [Filter: Enable](#)

<input type="checkbox"/>	Originating Location	Notes
--------------------------	----------------------	-------

* **Input Required** [Commit](#) [Cancel](#)

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

AVAYA Avaya Aura® System Manager 6.1 [Help](#) | [About](#) | [Change Password](#) | [Log off admin](#)

[Routing](#) * [Home](#)

Home / Elements / Routing / Dial Patterns - Dial Pattern Details

Dial Pattern Details [Help ?](#) [Commit](#) [Cancel](#)

General

* **Pattern:**

* **Min:**

* **Max:**

Emergency Call: ☐

SIP Domain:

Notes:

Originating Locations and Routing Policies

[Add](#) [Remove](#) [Refresh](#) [Filter: Enable](#)

<input type="checkbox"/>	Originating Location Name ¹	Originating Location Notes	Routing Policy Name	Rank ²	Routing Policy Disabled	Routing Policy Destination	Routing Policy Notes
<input type="checkbox"/>	Avaya CO		Audio_Codes_Med_1000	0	<input type="checkbox"/>	Audio_Codes_Med_1000	

Select : All, None

Denied Originating Locations

[Add](#) [Remove](#) [Refresh](#) [Filter: Enable](#)

<input type="checkbox"/>	Originating Location	Notes
<input type="checkbox"/>		

* Input Required [Commit](#) [Cancel](#)

Example:

DNIS call-back number is 13035383606 and the table entry “130” is used for routing.

7 Configure AudioCodes Mediant 1000 VoIP Media Gateway

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

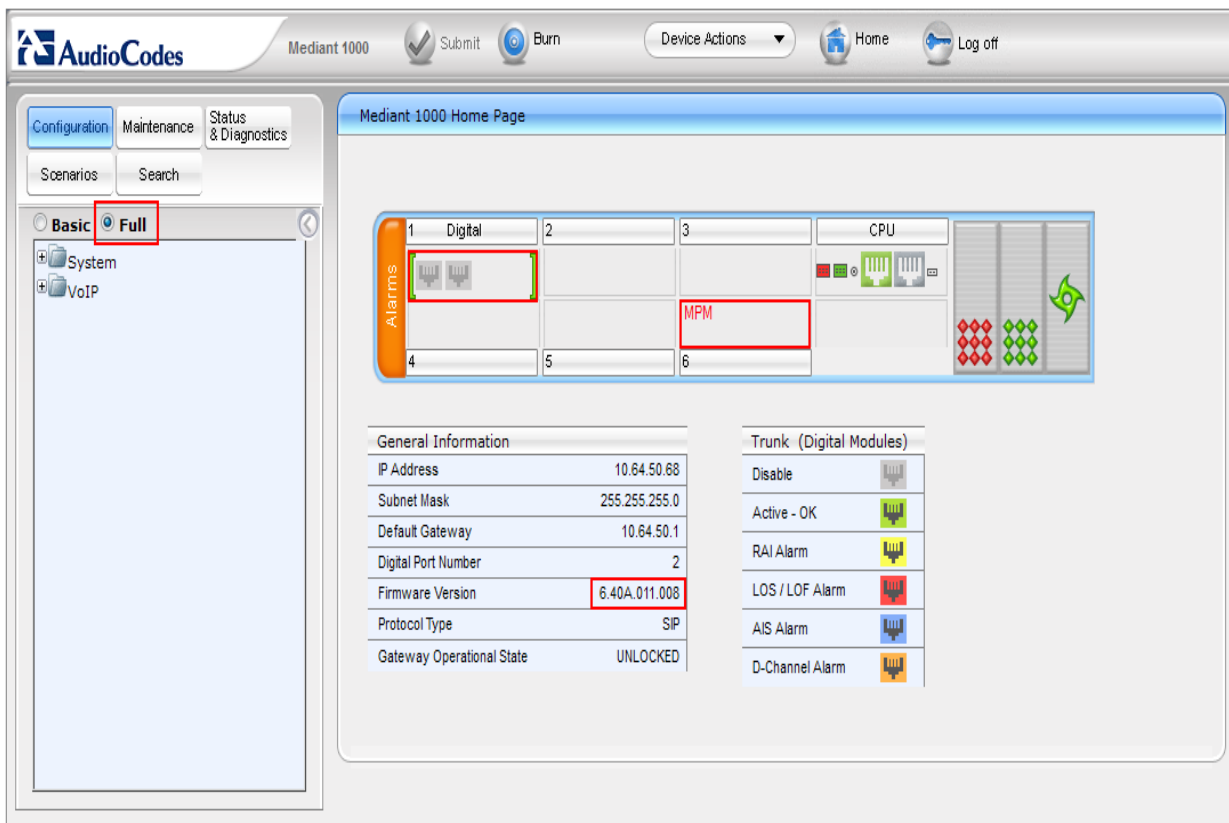
The Mediant 1000 configuration procedures include the following areas:

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

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

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

Note: The AudioCodes GW may display “Trunk”, “MPM” or both.



7.1 System Settings

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

7.1.1 Application Settings

7.1.1.1 Disable DHCP Client

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

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

The screenshot shows the AudioCodes Mediant 1000 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 contains tabs for 'Configuration', 'Maintenance', and 'Status & Diagnostics', with sub-tabs for 'Scenarios' and 'Search'. Below these are radio buttons for 'Basic' and 'Full' configuration. The 'System' tree is expanded, showing 'Application Settings', 'Syslog Settings', 'Regional Settings', 'Certificates', 'Management', and 'VoIP'. The 'Application Settings' page is displayed, featuring several sections: 'NTP Settings' with fields for NTP Server IP Address (0.0.0.0), NTP UTC Offset (Hours: 0, Minutes: 0), and NTP Updated Interval (Hours: 24, Minutes: 0); 'Day Light Saving Time' with a 'Day Light Saving Time' dropdown (Disable), 'Start Time' (Jan 01 0:00), 'End Time' (Jan 01 0:00), and 'Offset [min]' (60); 'STUN Settings' with 'Enable STUN' (Disable), 'STUN Server Primary IP' (0.0.0.0), and 'STUN Server Secondary IP' (0.0.0.0); 'NFS Settings' with an 'NFS Table' button; and 'DHCP Settings' with 'Enable DHCP' (Disable). The 'Enable DHCP' dropdown is highlighted with a red box. A 'Submit' button is located at the bottom right of the main content area.

7.1.2 Syslog Settings

7.1.2.1 Enable Syslog

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

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

Note: Syslog server is required.

The screenshot displays the 'Syslog Settings' page in the AudioCodes Mediant 1000 web interface. The left sidebar shows a tree view with 'System' expanded, containing 'Application Settings', 'Syslog Settings', 'Regional Settings', 'Certificates', 'Management', and 'VoIP'. The main content area is titled 'Syslog Settings' and contains two sections:

- Syslog Settings:** A table with the following fields:

Field	Value
Enable Syslog	Enable
Syslog Server IP Address	10.64.59.204
Syslog Server Port	514
Debug Level	7
Trunks Filter	-1
- Activity Types to Report via 'Activity Log' Messages:** A list of activity types with checkboxes for reporting:

Activity Type	Report
Parameters Value Change	<input checked="" type="checkbox"/>
Auxiliary Files Loading	<input checked="" type="checkbox"/>
Device Reset	<input checked="" type="checkbox"/>
Flash Memory Burning	<input checked="" type="checkbox"/>
Device Software Update	<input checked="" type="checkbox"/>
Access to Restricted Domains	<input checked="" type="checkbox"/>
Non-Authorized Access	<input checked="" type="checkbox"/>
Sensitive Parameters Value Change	<input checked="" type="checkbox"/>
Login and Logout	<input checked="" type="checkbox"/>

A 'Submit' button is located at the bottom right of the page.

7.2 VoIP Settings

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

7.2.1 TDM

7.2.2 TDM Bus Settings

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

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

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

The screenshot shows the AudioCodes Mediant 1000 configuration interface. The top navigation bar includes 'Submit' and 'Burn' buttons, along with 'Device Actions', 'Home', 'Help', and 'Log off' links. The left sidebar contains a tree view with categories like 'System', 'VoIP', 'Network', 'TDM', 'Security', 'PSTN', 'Signaling', 'Media', 'Services', 'Applications Enabling', 'Control Network', 'SIP Definitions', and 'Coders And Profiles'. The 'TDM' category is expanded, and 'TDM Bus Settings' is selected. The main content area is titled 'TDM Bus Settings' and contains a table of parameters. The 'TDM Bus Clock Source' parameter is highlighted with a red box, showing its value as 'Internal'. Other parameters include 'PCM Law Select' (MuLaw), 'TDM Bus PSTN Auto FallBack Clock' (Disable), 'TDM Bus PSTN Auto Clock Reverting' (Disable), 'Idle PCM Pattern' (255), 'Idle ABCD Pattern' (0x0F), 'TDM Bus Local Reference' (1), and 'TDM Bus Type' (Framers). A 'Submit' button is located at the bottom right of the configuration area.

Parameter	Value
PCM Law Select	MuLaw
TDM Bus Clock Source	Internal
TDM Bus PSTN Auto FallBack Clock	Disable
TDM Bus PSTN Auto Clock Reverting	Disable
Idle PCM Pattern	255
Idle ABCD Pattern	0x0F
TDM Bus Local Reference	1
TDM Bus Type	Framers

7.3 SIP Definitions

7.3.1 SIP General Parameters

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

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

Default values may be retained for all other fields.

The screenshot shows the AudioCodes Mediant 1000 configuration interface. The left sidebar contains a navigation tree with the following items: System, VoIP, Network, PSTN, Media, Services, Applications Enabling, Control Network, SIP Definitions (expanded), General Parameters (selected), Advanced Parameters, Account Table, Proxy & Registration, Coders And Profiles, and GW and IP to IP. The main area is titled 'SIP General Parameters' and contains a table of parameters. The parameters and their values are as follows:

Parameter	Value
NAT IP Address	0.0.0.0
PRACK Mode	Disable
Channel Select Mode	Ascending
Enable Early Media	Enable
Session-Expires Time	0
Minimum Session-Expires	90
Session Expires Method	Re-INVITE
Asserted Identity Mode	Disabled
Fax Signaling Method	T.38 Relay
SIP Transport Type	UDP
SIP UDP Local Port	5060
SIP TCP Local Port	5060
SIP TLS Local Port	5061
Enable SIPS	Disable
SIP Destination Port	5060
Enable Remote Party ID	Enable
Enable History-Info Header	Disable
Play Ringback Tone to IP	Don't Play
Play Ringback Tone to Tel	Prefer IP
Enable Reason Header	Enable

Below the table is a section for 'Retransmission Parameters'. A 'Submit' button is located at the bottom right of the interface.

7.3.2 SIP Advanced Parameters

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

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

Default values may be retained for all other fields.

The screenshot shows the AudioCodes Mediant 1000 configuration interface. The left sidebar contains a tree view with the following structure:

- Basic
- Full
 - System
 - VoIP
 - Network
 - TDM
 - Security
 - PSTN
 - Signaling
 - Media
 - Services
 - Applications Enabling
 - Control Network
 - SIP Definitions
 - General Parameters
 - Advanced Parameters (selected)
 - Account Table
 - Proxy & Registration
 - RADIUS Accounting Settings
 - Coders And Profiles
 - GW and IP to IP
 - IP Media

The main configuration area is titled 'Advanced Parameters' and contains the following sections:

- Basic Parameter List**
 - Enable Digit Delivery to IP: Disable
 - PSTN Alert Timeout: 180
 - QoS Statistics in SIP Release Call: Disable
- Disconnect and Answer Supervision**
 - Disconnect on Broken Connection: No
 - Broken Connection Timeout [100 msec]: 100
 - Disconnect Call on Silence Detection: No
 - Silence Detection Period [sec]: 120
 - Silence Detection Method: Voice/Energy Detectors
 - Enable Fax Re-Routing: Disable
- CDR and Debug**
 - CDR Server IP Address: 10.64.59.204
 - CDR Report Level: Start & End & Connect Call
- Misc. Parameters**
 - Progress Indicator to IP: No PI
 - Enable X-Channel Header: Disable
 - Enable Early 183: Disable
 - Enable Busy Out: Disable
 - Graceful Busy Out Timeout [sec]: 0
 - Default Release Cause: 3

A 'Submit' button is located at the bottom right of the configuration area.

7.4 Network

7.4.1 IP Settings

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

The screenshot shows a web-based configuration interface for network settings. On the left is a navigation tree with categories like System, VoIP, Network, DNS, TDM, Security, PSTN, Signaling, Media, and Services. The 'Network' category is expanded, showing 'IP Settings' as the selected option. The main area is titled 'Multiple Interface Table' and contains a table with the following data:

Index	Application Type	IP Address	Prefix Length	Gateway	VLAN ID	Interface Name	Primary DNS Server IP Address	Secondary DNS Server IP Address
0	OAMP + Media + Control	10.64.50.68	24	10.64.50.1	1	O+M+C	0.0.0.0	0.0.0.0

Below the table, there are configuration options: 'VLAN Mode' set to 'Disable', 'Native VLAN ID' set to '1', and an 'IP Interface Status Table' button. At the bottom right is a 'Submit' button with a checkmark icon.

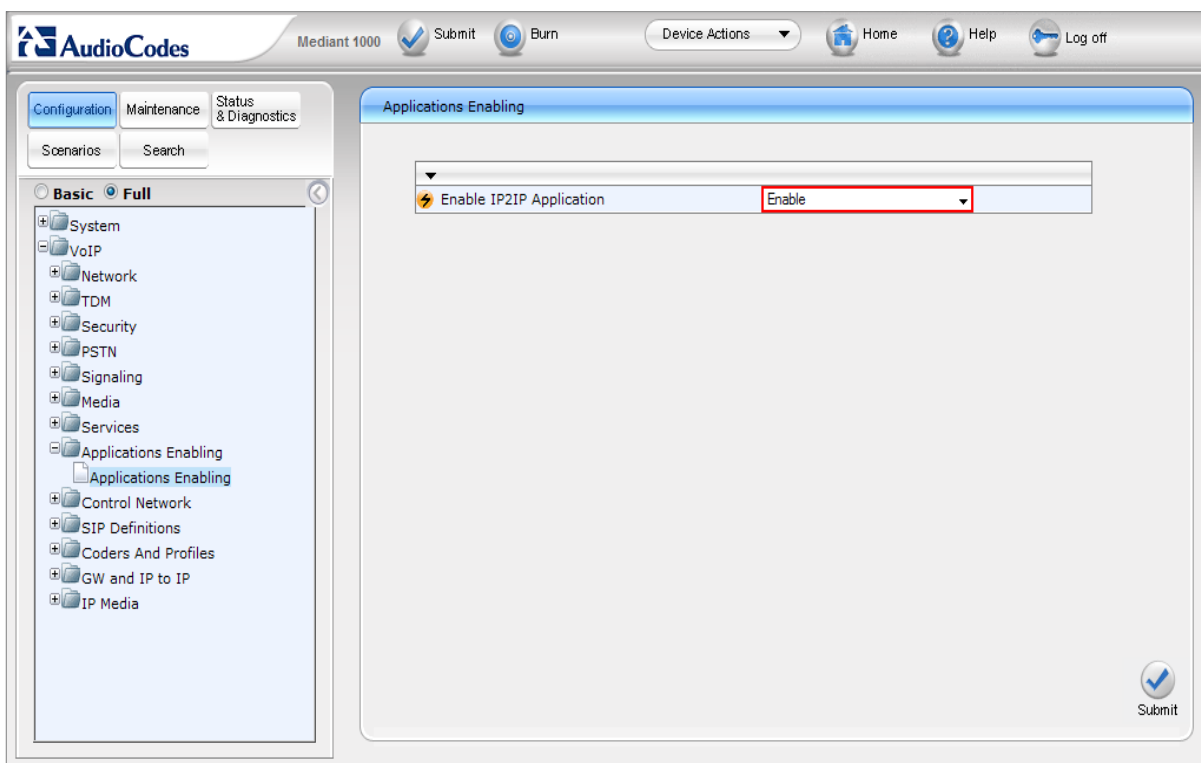
7.5 Applications Enabling

7.5.1.1 Applications Enabling

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

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

Note: Requires system reset



7.6 Media

7.6.1 Voice Settings

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

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

Default values may be retained for all other fields.

The screenshot shows the AudioCodes Mediant 1000 configuration interface. The left sidebar contains a tree view with the following structure:

- Basic (selected)
- Full
- System
- VoIP
 - Network
 - TDM
 - Security
 - PSTN
 - Signaling
 - Media
 - Voice Settings (selected)
 - Fax/Modem/CID Settings
 - RTP/RTCP Settings
 - IPMedia Settings
 - General Media Settings
 - Media Realm Configuration
 - Media Security
- Services
- Applications Enabling
- Control Network
- SIP Definitions
- Coders And Profiles
- GW and IP to IP

The main area displays the 'Voice Settings' page with a 'Basic Parameter List' table:

Parameter	Value
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

A red box highlights the 'DTMF Transport Type' and 'DTMF Volume' fields. The 'Submit' button is located at the bottom right of the configuration area.

IP Media Settings

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

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

Default values may be retained for all other fields.

Note: Four channels are required for each MVS call.

The screenshot shows the AudioCodes Mediant 1000 configuration interface. The left sidebar shows a tree view with 'Media' expanded and 'IPMedia Settings' selected. The main panel displays the 'IPMedia Settings' configuration page. The 'Number of Media Channels' field is highlighted with a red box and contains the value 12. Other fields include 'Answer Machine Detector Sensitivity' (checked), 'Answer Machine Detector Beep Detection Timeout' (200), 'Answer Machine Detector Beep Detection Sensitivity' (0), 'Enable AGC' (Disable), 'AGC Slope' (3), 'AGC Redirection' (0), 'AGC Target Energy' (19), 'Enable Energy Detector' (Disable), 'Energy Detector Quality Factor' (4), 'Energy Detector Threshold' (3), 'Enable Pattern Detector' (Disable), 'Active Speakers Min Interval' (20), 'Configure Audio Playback' (PCMA), 'Configure Audio Recording' (PCMA), and 'End Of Record Time' (60). A 'Submit' button is at the bottom right.

Parameter	Value
Answer Machine Detector Sensitivity	<input checked="" type="checkbox"/>
Answer Machine Detector Beep Detection Timeout	200
Answer Machine Detector Beep Detection Sensitivity	0
Enable AGC	Disable
AGC Slope	3
AGC Redirection	0
AGC Target Energy	19
Enable Energy Detector	Disable
Energy Detector Quality Factor	4
Energy Detector Threshold	3
Enable Pattern Detector	Disable
Active Speakers Min Interval	20
Number of Media Channels	12
Configure Audio Playback	PCMA
Configure Audio Recording	PCMA
End Of Record Time	60
Record Audio Format	PCMA

7.7 Control Network

7.7.1 Proxy Sets Table

7.7.1.1 Proxy Sets Table BlackBerry® MVS

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

Configure the parameters as described below.

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

Default values may be retained for all other fields.

The screenshot displays the AudioCodes Mediant 1000 web interface. The left sidebar shows a tree view with 'Control Network' expanded, and 'Proxy Sets Table' selected. The main content area is titled 'Proxy Sets Table'. At the top, there is a dropdown for 'Proxy Set ID' with '1' selected. Below this is a table with 5 rows. The first row is highlighted, showing '10.64.50.66' in the 'Proxy Address' column and 'UDP' in the 'Transport Type' column. Below the table, there is a section for 'Enable Proxy Keep Alive' with a dropdown set to 'Disable'. Other fields include 'Proxy Keep Alive Time' (60), 'Proxy Load Balancing Method' (Disable), 'Is Proxy Hot Swap' (No), 'Proxy Redundancy Mode' (Not Configured), 'SRD Index' (0), and 'Classification Input' (IP only). A 'Submit' button is located at the bottom right of the configuration area.

	Proxy Address	Transport Type
1	10.64.50.66	UDP
2		
3		
4		
5		

Enable Proxy Keep Alive	Disable
Proxy Keep Alive Time	60
Proxy Load Balancing Method	Disable
Is Proxy Hot Swap	No
Proxy Redundancy Mode	Not Configured
SRD Index	0
Classification Input	IP only

7.7.1.2 Proxy Sets Table Avaya Aura® Session Manager

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

Configure the parameters as described below.

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

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

The screenshot shows the AudioCodes Mediant 1000 configuration interface. The left sidebar contains a tree view with categories like System, VoIP, Network, TDM, Security, PSTN, Signaling, Media, Services, Applications Enabling, and Control Network. Under Control Network, the 'Proxy Sets Table' is selected. The main area displays the 'Proxy Sets Table' configuration. At the top, 'Proxy Set ID' is set to 2. Below this is a table with 5 rows for Proxy Address and Transport Type. The first row is filled with '10.64.50.31' and 'TCP'. Below the table, there are several configuration fields: 'Enable Proxy Keep Alive' (set to Disable), 'Proxy Keep Alive Time' (60), 'Proxy Load Balancing Method' (Disable), 'Is Proxy Hot Swap' (No), 'Proxy Redundancy Mode' (Not Configured), 'SRD Index' (0), and 'Classification Input' (IP only). A 'Submit' button is at the bottom right.

	Proxy Address	Transport Type
1	10.64.50.31	TCP
2		
3		
4		
5		

Enable Proxy Keep Alive	Disable
Proxy Keep Alive Time	60
Proxy Load Balancing Method	Disable
Is Proxy Hot Swap	No
Proxy Redundancy Mode	Not Configured
SRD Index	0
Classification Input	IP only

7.7.2 IP Group Table

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

Configure the following parameters.

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

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

The screenshot shows the AudioCodes Mediant 1000 configuration interface. The left sidebar contains a tree view with categories like System, VoIP, Network, TDM, Security, PSTN, Signaling, Media, Services, Applications Enabling, Control Network, SIP Interface Table, IP Group Table, Proxy Sets Table, NAT Translation Table, SIP Definitions, Coders And Profiles, GW and IP to IP, and IP Media. The 'IP Group Table' is selected under the 'Control Network' category. The main panel displays the 'IP Group Table' configuration form. It includes a 'Basic Parameter List' section with fields for Index (1), Type (SERVER), Description (MVS 10.64.50.66), Proxy Set ID (1), SIP Group Name, Contact User, SRD (0), Media Realm, and IP Profile ID (1). Below this is a 'Gateway Parameters' section with fields for Always Use Route Table (No), Routing Mode (Not Configured), SIP Re-Routing Mode (Standard), Enable Survivability (Disable), and Serving IP Group ID (2). A 'Submit' button is located at the bottom right of the form.

Repeat the configuration for the Avaya Aura® Session Manager.

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

Mediant 1000
Submit
Burn
Device Actions
Home
Help
Log off

Configuration

Maintenance

Status & Diagnostics

Scenarios

Search

Basic

Full

- System
- VoIP
 - Network
 - TDM
 - Security
 - PSTN
 - Signaling
 - Media
 - Services
 - Applications Enabling
 - Control Network
 - SRD Table
 - SIP Interface Table
 - IP Group Table
 - Proxy Sets Table
 - NAT Translation Table
 - SIP Definitions
 - Coders And Profiles
 - GW and IP to IP
 - IP Media

IP Group Table

Basic Parameter List ▲

Index

2

Common Parameters

Type	SERVER
Description	Avaya SM 10.64.50.31
Proxy Set ID	2
SIP Group Name	
Contact User	
SRD	0
Media Realm	
IP Profile ID	2

Gateway Parameters

Always Use Route Table	No
Routing Mode	Not Configured
SIP Re-Routing Mode	Standard
Enable Survivability	Disable
Serving IP Group ID	1

Submit

7.8 Coders and Profiles

7.8.1 Coders

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

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

The screenshot shows the AudioCodes Mediant 1000 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 contains a tree view with categories like 'System', 'VoIP', 'Network', 'TDM', 'Security', 'PSTN', 'Signaling', 'Media', 'Services', 'Applications Enabling', 'Control Network', 'SIP Definitions', 'Coders And Profiles', 'GW and IP to IP', and 'IP Media'. Under 'Coders And Profiles', the 'Coders' option is selected. The main area displays the 'Coders Table' with the following data:

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

A 'Submit' button is located at the bottom right of the table area.

7.8.2 IP Profiles

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

Default values were retained for all other fields.

The screenshot shows the AudioCodes Mediant 100 web interface. The top navigation bar includes the AudioCodes logo, 'Mediant 100', and buttons for 'Submit', 'Burn', 'Device Actions', 'Home', 'Help', and 'Log off'. The left sidebar contains a tree view with categories: Configuration, Maintenance, and Status & Diagnostics. Under Configuration, there are sub-items: Scenarios, Search, Basic, and Full. The 'Full' category is expanded, showing a list of settings including System, VoIP, Network, TDM, Security, PSTN, Signaling, Media, Services, Applications Enabling, Control Network, SIP Definitions, Coders And Profiles, Coders, Coders Group Settings, Tel Profile Settings, IP Profile Settings (highlighted), GW and IP to IP, and IP Media. The main content area is titled 'IP Profile Settings'. It features a 'Basic Parameter List' section with a dropdown for 'Profile ID' (set to 1) and a text field for 'Profile Name' (set to 10.64.50.66). Below this is a 'Common Parameters' section with a table of settings: RTP IP DiffServ (46), Signaling DiffServ (40), Disconnect on Broken Connection (No), Dynamic Jitter Buffer Minimum Delay [msec](*) (10), Dynamic Jitter Buffer Optimization Factor(*) (10), RTP Redundancy Depth(*) (0), Echo Canceled(*) (Enable), Input Gain (-32 to 31 dB)(*) (0), and Voice Volume (-32 to 31 dB)(*) (0). A 'Gateway Parameters' section is partially visible at the bottom, showing 'Fax Signaling Method' set to 'T.38 Relay'. A 'Submit' button is located at the bottom right of the main content area.

Basic Parameter List	
Profile ID	1
Profile Name	10.64.50.66

Common Parameters	
RTP IP DiffServ	46
Signaling DiffServ	40
Disconnect on Broken Connection	No
Dynamic Jitter Buffer Minimum Delay [msec](*)	10
Dynamic Jitter Buffer Optimization Factor(*)	10
RTP Redundancy Depth(*)	0
Echo Canceled(*)	Enable
Input Gain (-32 to 31 dB)(*)	0
Voice Volume (-32 to 31 dB)(*)	0

Gateway Parameters	
Fax Signaling Method	T.38 Relay

AudioCodes Mediant 1000

Configuration Maintenance Status & Diagnostics

Scenarios Search

Basic Full

- System
- VoIP
 - Network
 - TDM
 - Security
 - PSTN
 - Signaling
 - Media
 - Services
 - Applications Enabling
 - Control Network
 - SIP Definitions
 - Coders And Profiles
 - Coders
 - Coders Group Settings
 - Tel Profile Settings
 - IP Profile Settings
 - GW and IP to IP
 - IP Media

IP Profile Settings

Basic Parameter List ▲

Media Security Behavior	Disable
CNG Detector Mode	Events Only
Modems Transport Type	Enable Bypass
NSE Mode	Disable
Number of Calls Limit	-1
Progress Indicator to IP	Not Configured
Profile Preference	1
Coder Group	Default Coder Group
Remote RTP Base UDP Port	0
First Tx DTMF Option	INFO(Cisco)
Second Tx DTMF Option	INFO(Cisco)
Declare RFC 2833 in SDP	Yes
Add IE In SETUP	
AMD Sensitivity Parameter Suit	0
AMD Sensitivity Level	8
AMD Max Greeting Time	300
AMD Max Post Silence Greeting Time	400
Enable Hold	Enable

Submit

Repeat the configuration for the Avaya Aura® Session Manager.

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

Default values were retained for all other fields.

AudioCodes Mediant 1000 Submit Burn Device Actions Home Help Log off

Configuration Maintenance Status & Diagnostics

Scenarios Search

Basic **Full**

- System
 - VoIP
 - Network
 - TDM
 - Security
 - PSTN
 - Signaling
 - Media
 - Services
 - Applications Enabling
 - Control Network
 - SIP Definitions
 - Coders And Profiles
 - Coders
 - Coders Group Settings
 - Tel Profile Settings
 - IP Profile Settings**
 - GW and IP to IP
 - IP Media

IP Profile Settings

Basic Parameter List ▲

Profile ID: 2
Profile Name: 10.64.50.31

Common Parameters

RTP IP DiffServ	46
Signaling DiffServ	40
Disconnect on Broken Connection	No
Dynamic Jitter Buffer Minimum Delay [msec](*)	10
Dynamic Jitter Buffer Optimization Factor(*)	10
RTP Redundancy Depth(*)	0
Echo Canceled(*)	Enable
Input Gain (-32 to 31 dB)(*)	0
Voice Volume (-32 to 31 dB)(*)	0

Gateway Parameters

Fax Signaling Method	T.38 Relay
----------------------	------------

Submit

AudioCodes Mediant 1000 Submit Burn Device Actions Home Help Log off

Configuration Maintenance Status & Diagnostics

Scenarios Search

Basic **Full**

- System
 - VoIP
 - Network
 - TDM
 - Security
 - PSTN
 - Signaling
 - Media
 - Services
 - Applications Enabling
 - Control Network
 - SIP Definitions
 - Coders And Profiles
 - Coders
 - Coders Group Settings
 - Tel Profile Settings
 - IP Profile Settings**
 - GW and IP to IP
 - IP Media

IP Profile Settings

Basic Parameter List ▲

Media Security Behavior	Disable
CNG Detector Mode	Events Only
Modems Transport Type	Enable Bypass
NSE Mode	Disable
Number of Calls Limit	-1
Progress Indicator to IP	Not Configured
Profile Preference	1
Coder Group	Default Coder Group
Remote RTP Base UDP Port	0
First Tx DTMF Option	RFC 2833
Second Tx DTMF Option	RFC 2833
Declare RFC 2833 in SDP	Yes
Add IE In SETUP	
AMD Sensitivity Parameter Suit	0
AMD Sensitivity Level	8
AMD Max Greeting Time	300
AMD Max Post Silence Greeting Time	400
Enable Hold	Enable

Submit

7.9 GW and IP to IP

7.9.1 Manipulation Tables

7.9.1.1 Dest Number IP to Tel

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

The screenshot shows the AudioCodes Mediant 1000 web interface. The left sidebar contains a tree view with categories like System, VoIP, Network, TDM, Security, PSTN, Signaling, Media, Services, Applications Enabling, Control Network, SIP Definitions, Coders And Profiles, GW and IP to IP, Trunk Group, and Manipulations. Under Manipulations, 'Dest Number IP->Tel' is selected. The main area displays a table titled 'Destination Phone Number Manipulation Table for IP -> Tel Calls'. The table has columns: Index, Destination Prefix, Source Prefix, Source IP Address, Stripped Digits From Left, Stripped Digits From Right, Prefix to Add, Suffix to Add, and Nt. The table contains five rows of data. The first row has a red border around the 'Destination Prefix' cell 'XXXX#'. The second row has a red border around the 'Destination Prefix' cell '13035383606'. The third row has a red border around the 'Destination Prefix' cell '13035383592'. The fourth row has a red border around the 'Destination Prefix' cell '1*'. The fifth row has a red border around the 'Destination Prefix' cell 'XXXXXXXXXX'. The 'Stripped Digits From Left' and 'Stripped Digits From Right' columns are all 0. The 'Prefix to Add' column is empty for the first four rows and contains '9' for the fifth row. The 'Suffix to Add' column is empty for all rows. The 'Nt' column contains the value 255 for all rows.

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

Scroll to the right to see the remaining fields.

7.9.1.2 Dest Number Tel to IP

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

The screenshot shows the AudioCodes Mediant 1000 configuration interface. The left sidebar contains a tree view with categories like Configuration, Maintenance, and Status & Diagnostics. Under Configuration, the 'Full' tab is selected, and the 'GW and IP to IP' section is expanded, showing 'Dest Number Tel->IP' as the active configuration.

The main area displays the 'Destination Phone Number Manipulation Table for Tel -> IP Calls'. It includes an 'Add' button and a table with the following data:

Index	Source Trunk Group	Source IP Group	Destination Prefix	Source Prefix	Stripped Digits From Left	Stripped Digits From Right	Prefix to Add	Suffix to Add	Number of D Leave
1	-1	-1	7*	*	1	0			255
3	-1	-1	9123	*	4	0			255

7.9.1.3 Source Number IP to Tel

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

The screenshot shows the AudioCodes Mediant 1000 configuration interface. The left sidebar contains a tree view with categories like Security, PSTN, Signaling, Media, Services, Applications Enabling, Control Network, SIP Definitions, Coders And Profiles, GW and IP to IP, Trunk Group, and Manipulations. The 'Manipulations' category is expanded, showing 'General Settings', 'Dest Number IP->Tel', 'Dest Number Tel->IP', 'Calling Name IP->Tel', 'Calling Name Tel->IP', 'Source Number IP->Tel', and 'Source Number Tel->IP'. The 'Source Number IP->Tel' option is selected. The main panel displays the 'Source Phone Number Manipulation Table for IP -> Tel Calls'. It includes an 'Add' button and a table with the following columns: Index, Destination Prefix, Source Prefix, Source IP Address, Stripped Digits From Left, Stripped Digits From Right, Prefix to Add, Suffix to Add, and Number of Digits to Leave. A single row is visible with Index 1, Destination Prefix '+', Source Prefix '-', Source IP Address '-', Stripped Digits From Left 1, Stripped Digits From Right 0, Prefix to Add, Suffix to Add, and Number of Digits to Leave 255.

Index	Destination Prefix	Source Prefix	Source IP Address	Stripped Digits From Left	Stripped Digits From Right	Prefix to Add	Suffix to Add	Number of Digits to Leave
1	+	-	-	1	0			255

Scroll to the right to see the remaining fields.

7.9.1.4 Source Number Tel to IP

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

The screenshot shows the AudioCodes Mediant 1000 configuration interface. The left sidebar is the same as in the previous screenshot, but the 'Source Number Tel->IP' option under the 'Manipulations' category is selected. The main panel displays the 'Source Phone Number Manipulation Table for Tel -> IP Calls'. It includes an 'Add' button and a table with the following columns: Index, Source Trunk Group, Source IP Group, Destination Prefix, Source Prefix, Stripped Digits From Left, Stripped Digits From Right, Prefix to Add, Suffix to Add, and Number of Digits to Leave. A single row is visible with Index 1, Source Trunk Group -1, Source IP Group -1, Destination Prefix '+', Source Prefix '-', Stripped Digits From Left 1, Stripped Digits From Right 0, Prefix to Add, Suffix to Add, and Number of Digits to Leave 255.

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

7.9.2 Routing

7.9.2.1 Tel to IP Routing

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

The screenshot shows the 'Tel to IP Routing' configuration page in the AudioCodes Mediant 1000 interface. The left sidebar shows the navigation tree with 'Routing' expanded. The main area displays a table with the following data:

	Src. IPGroupID	Src. Trunk Group ID	Dest. Phone Prefix	Source Phone Prefix	->	Dest. IP Address	Port	Transport Type	Dest. IPGroup ID	Dest. SRD	IP Profile ID
1	2	*	*	*		10.64.50.66		Not Configured	1	-1	1
2	1	*	*	*		10.64.50.31		Not Configured	2	-1	2
3	-1							Not Configured	-1		
4	-1							Not Configured	-1		
5	-1							Not Configured	-1		
6	-1							Not Configured	-1		
7	-1							Not Configured	-1		

Scroll to the right to see the remaining fields.

7.9.2.2 IP to Trunk Group Routing

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

The screenshot shows the 'IP To Trunk Group Routing Table' configuration page in the AudioCodes Mediant 1000 interface. The left sidebar shows the navigation tree with 'Routing' expanded. The main area displays a table with the following data:

	Dest. Host Prefix	Source Host Prefix	Dest. Phone Prefix	Source Phone Prefix	Source IP Address	->	Trunk Group ID	IP Profile ID	Source IPGroup ID
1			*	*	10.64.50.31		-1	2	2
2			*	*	10.64.50.66		-1	1	1
3									
4									
5									
6									
7									

Scroll to the right to see the remaining fields.

7.9.3 DTMF and Supplementary

7.9.3.1 DTMF and Dialing

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

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

Default values may be retained for all other fields.

The screenshot shows the AudioCodes Mediant 1000 configuration web interface. The left sidebar contains a tree view with categories like System, VoIP, Network, PSTN, Media, Services, Applications Enabling, Control Network, SIP Definitions, Coders And Profiles, GW and IP to IP, Trunk Group, Manipulations, Routing, DTMF and Supplementary, DTMF & Dialing, Digital Gateway, and Advanced Applications. The 'DTMF & Dialing' option is selected. The main panel displays the 'DTMF & Dialing' configuration page with an 'Advanced Parameter List' dropdown. The parameters are as follows:

Parameter	Value
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
Default Destination Number	1000

A red box highlights the 'Declare RFC 2833 in SDP', '1st Tx DTMF Option', and '2nd Tx DTMF Option' fields. The 'Submit' button is located at the bottom right of the configuration area.

7.10 Additional settings for the ini file

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

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

The screenshot shows the 'Admin Page' of an AudioCodes gateway. The browser address bar displays '10.64.50.68/AdminPage'. On the left sidebar, the 'ini Parameters' link is highlighted. The main area shows the configuration for the parameter 'PLAYDTMFDURINGHOLD'. The 'Parameter Name' field contains 'PLAYDTMFDURINGHOLD', and the 'Enter Value' field contains '1'. An 'Apply New Value' button is visible. Below this, an 'Output Window' displays the following text: 'Parameter Name: PLAYDTMFDURINGHOLD', 'Parameter New Value:1', and 'Parameter Description:Enable/Disable Playing DTMF to TEL during hold'.

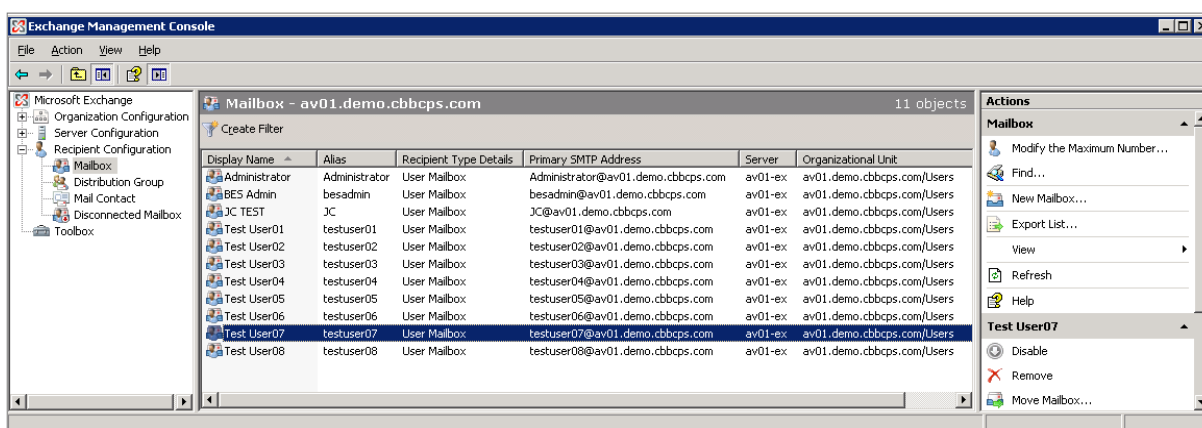
The screenshot shows the 'Admin Page' of an AudioCodes gateway. The browser address bar displays '10.64.50.68/AdminPage'. On the left sidebar, the 'ini Parameters' link is highlighted. The main area shows the configuration for the parameter 'IP2IPTRANSFERMODE'. The 'Parameter Name' field contains 'IP2IPTRANSFERMODE', and the 'Enter Value' field contains '1'. An 'Apply New Value' button is visible. Below this, an 'Output Window' displays the following text: 'Parameter Name: IP2IPTRANSFERMODE', 'Parameter New Value:1', and 'Parameter Description:IP2IP Transfer Mode'.

8 Research in Motion Mobile Voice System Configuration

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

8.1 Microsoft Exchange Server

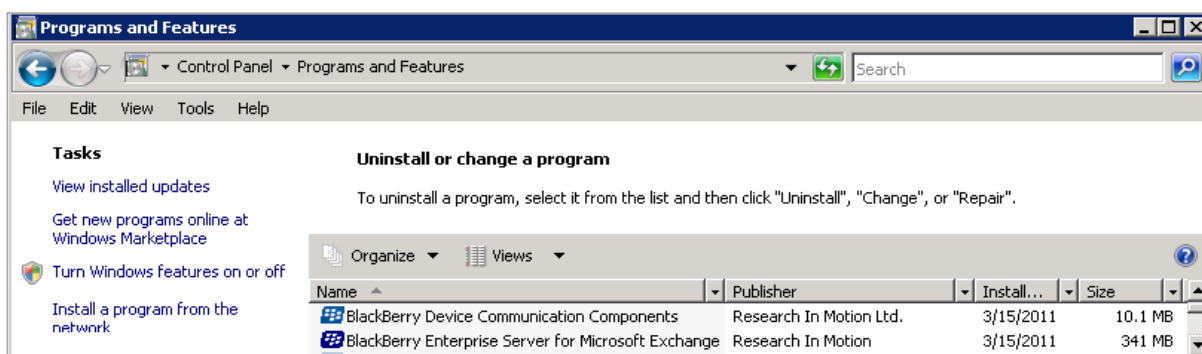
Verify that each user that will be configured on the MVS is also configured as a user within Microsoft Exchange. Log in to the Exchange Management Console. Navigate to **Microsoft Exchange → Recipient Configuration → Mailbox** to view the existing users.



8.2 BlackBerry® Enterprise Server Configuration

8.2.1 Verify Software Version

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



8.2.2 Create Users

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

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

The screenshot shows the BlackBerry Administration Service interface. The top header includes the BlackBerry logo, the text 'BlackBerry® Administration Service', and user information: 'System Administrator', 'Log out | Home | Help', and the date 'Friday, January 20, 2012'. The left sidebar contains a 'Quick user search' box with a 'Name:' field and a search icon. Below this are three main sections: 'BlackBerry solution management' with a tree view showing 'User' (selected), 'Group', 'Role', 'Software', 'Policy', and 'Administrator user'; 'Devices' with 'Attached devices', 'Deployment jobs', and 'Wireless activations'; and 'Servers and components' with 'BlackBerry Solution topology'. The 'Preferences' section at the bottom shows 'My setup'. The main content area is titled 'User > Create a user' and 'Create a BlackBerry enabled user'. It includes a sub-header 'Search messaging users' and a form with 'Email user criteria' (fields for 'Messaging server display name' and 'Email address') and 'Sort criteria' (a 'Sort by:' dropdown set to 'Display name' and radio buttons for 'A to Z' and 'Z to A'). At the bottom right of the form is a red-bordered box containing a magnifying glass icon, the text 'Search', and a 'Clear' button. Below the form are links for 'Cancel', 'Import new users', and 'Add user from company directory'. The footer contains copyright information: 'Copyright © 1997 - 2012 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 8.1** have already been added, thus only the users named that have not been associated are shown as available users to be added. Select a user to add by clicking the box next to the user name. Click **Continue**.

BlackBerry Administration Service

System Administrator
Log out | Home | Help

Friday, January 20, 2012

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.

Quick user search

Name:

BlackBerry solution management

- User
 - Create a user
 - Manage users
- Group
- Role
- Software
- Policy
- Administrator user

Devices

- Attached devices
- Deployment jobs
- Wireless activations

Servers and components

- BlackBerry Solution topology

Preferences

- My setup

Search messaging users

Email user criteria

Messaging server display name: Email address:

Sort criteria

Sort by: Display name

☒ A to Z ☐ Z to A

Showing 1 - 5 of 5

Messaging server display name	Email address
<input type="checkbox"/> Administrator	Administrator@av01.demo.cbbcps.com
<input type="checkbox"/> BES Admin	besadmin@av01.demo.cbbcps.com
<input type="checkbox"/> JC TEST	JC@av01.demo.cbbcps.com
<input type="checkbox"/> Test User05	testuser05@av01.demo.cbbcps.com
<input checked="" type="checkbox"/> Test User08	testuser08@av01.demo.cbbcps.com

Showing 1 - 5 of 5

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Version: 5.0.3.31

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.

BlackBerry® Administration Service

System Administrator
[Log out](#) | [Home](#) | [Help](#)
 Friday, January 20, 2012

Quick user search
 Name:

BlackBerry solution management
 User
 Create a user
 Manage users
 Group
 Role
 Software
 Policy
 Administrator user

Devices
 Attached devices
 Deployment jobs
 Wireless activations

Servers and components
 BlackBerry Solution topology

Preferences
 My setup

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
Test User08	testuser08@av01.demo.cbbcps.com

Available BlackBerry Enterprise Server instances

BlackBerry Enterprise Server:	
AV01-BES	

Available groups

Administrators	<input type="button" value="Add"/> <input type="button" value="Add all"/> <input type="button" value="Remove"/> <input type="button" value="Remove all"/>
BlackBerry Web Desktop Manager users	
Help desk representatives	

Current groups

--

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 Version: 5.0.3.31

8.2.3 Manage Users

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

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

The screenshot shows the BlackBerry Administration Service interface. The top navigation bar includes the BlackBerry logo, 'BlackBerry® Administration Service', and user information: 'System Administrator', 'Log out | Home | Help', and the date 'Friday, January 20, 2012'. The main content area is titled 'User > Manage users > View user (Test User01)'. On the left is a sidebar with navigation links for 'User', 'Group', 'Role', 'Software', 'Policy', and 'Administrator user'. The main area displays user details for 'Test User01'. A red box highlights the 'Associated device properties' section, which includes fields for PIN, Home Carrier, Phone number, Associated BlackBerry Enterprise Server, Device IT policy, Queued IT policy status, Last contact date, Result of last transaction to the device, Device model, Current Carrier, Software version, Device IT policy time, and Last message sent. Below this are sections for 'Messaging configuration', 'BlackBerry Enterprise Server status', 'Device activation', and 'Device deployment'. The footer contains copyright information: 'Copyright © 1997 - 2012 Research In Motion Limited. All rights reserved. Version: 5.0.3.31'.

Quick user search
Name:

BlackBerry solution management
User
Create a user
Manage users
Group
Role
Software
Policy
Administrator user

Role
Software
Policy
Administrator user

Role
Software
Policy
Administrator user

Policy
Administrator user

Devices
Attached devices
Deployment jobs
Wireless activations

Servers and components
BlackBerry Solution topology

Preferences
My setup

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: 26

Authentication type User name Password
Active Directory 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	25FC3A18	Device model	9810
Home Carrier	Research In Motion	Current Carrier	AT&T
Phone number	6505043557	Software version	7.0.0.261 (Platform 5.0.0.469)
Associated BlackBerry Enterprise Server	AV01-BES		
Device IT policy	Default	Device IT policy time	11/23/11 3:54:40 PM
Queued IT policy status	Applied successfully		
Last contact date	12/14/11 8:46:38 AM	Last message sent	12/14/11 8:46:37 AM
Result of last transaction to the device	Received from device		

Messaging configuration Description
Default configuration 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


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Version: 5.0.3.31



System Administrator
[Log out](#) | [Home](#) | [Help](#)
Friday, January 20, 2012

Quick user search
Name: 

BlackBerry solution management

-  User
 -  Create a user
 -  Manage users
-  Group
-  Role
-  Software
 -  Create a user
 -  Manage users
-  Group
-  Role
-  Software
 -  Create a user
 -  Manage users
-  Group
-  Role
-  Software
 -  Create a user
 -  Manage users
-  Group
-  Role
-  Software
 -  Create a user
 -  Manage users
-  Group
-  Role
-  Software
 -  Create a user
 -  Manage users
-  Group
-  Role
-  Software
 -  Create a user
 -  Manage users
-  Group
-  Role
-  Software
 -  Create a user
 -  Manage users
-  Group
-  Role
-  Software
 -  Create a user
 -  Manage users
-  Policy
-  Administrator user

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:	25FC3A18	User-device configuration:	Default configuration		
Available memory (KB):	228427	Battery level (%):	70		
Uptime:	1 day(s), 18 hours, 1 minutes, 14 seconds				
Properties status					
Last updated:	12/14/11 8:46:37 AM				
Hardware					
BlackBerry device model:	9810	Network type:	3G		
Memory (MB):	512	Serial number (IMEI):	004401.13.744150.3		
Frequencies:	GSM 850, GSM 900, GSM 1800, GSM 1900	Secured boot ROM:	No		
Home carrier:	Research In Motion	Display screen height:	480		
Display screen width:	640				
Software					
Platform version:	5.0.0.469	BlackBerry Device Software version:	7.0.0.261		
Phone number:	6505043557	Security password:	No		
Current carrier:	AT&T	Direct connect ID:			
Last reported IT policy on device					
IT policy name:	Default	IT policy time:	11/23/11 3:55:32 PM		
Policies update status					
Last updated:	12/12/11 2:38:29 PM				
Queued IT policy					
IT policy name:	Default	IT policy status:	Applied successfully		
IT policy sent:	11/23/11 3:54:37 PM	IT policy received:	11/23/11 3:54:40 PM		
BlackBerry Enterprise Server information					
Associated BlackBerry Enterprise Server:	AV01-BES				

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

Devices

-  Attached devices
-  Deployment jobs
-  Wireless activations


Servers and components

-  BlackBerry Solution topology

Preferences

-  My setup

Device activation

-  Specify new device password and lock device
-  Delete all device data and remove device
-  Delete only the organization data and remove device

Device data

-

8.3 BlackBerry® Mobile Voice System Server Configuration

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

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

The screenshot shows the BlackBerry Mobile Voice System (MVS) console interface. The top header includes the BlackBerry logo, the text "BlackBerry® Mobile Voice System", and user information "User: sysadmin" with links for "About", "Change Password", and "Log Off". The "About" link is highlighted with a red box. The left sidebar contains a menu with the following sections:

- Home
 - Dashboard
- User Management
 - Users
 - Templates
 - Class of Service
- System Configuration
 - Telephony Connectors
 - Locations
 - Voice Mail Connectors
 - Licensing
- Administrator Management
 - Administrative Roles
 - Administrators
- Servers and Components
 - MVS Topology
 - Server View
 - Component View
 - High Availability
- Reporting
 - Call Detail Record Report
 - Disconnected Call Report

The main content area displays "About the BlackBerry MVS" information:

Version:	5.1.1 (Bundle 21)
Serial Number:	MA3370912

Database Information:

Host or IP:	10.64.50.64
Port:	1433
Database Name:	BESMgmt
Authentication Type:	Microsoft® Windows

You can use the configuration tool of the BlackBerry MVS to change the values of the database configuration settings. The tool is located on the computer that hosts this instance of the BlackBerry MVS Console.

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8.3.1 Create a Mobile Voice System Server

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

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



BlackBerry® Mobile Voice System

User: sysadmin | [About](#) | [Change Password](#) | [Log Off](#)

Friday, January 20, 2012

Home
Dashboard

User Management
Users
Templates
 Add
 Manage
Class of Service
 Add
 Manage

System Configuration
Telephony Connectors
Locations
Voice Mail Connectors
Licensing

Administrator Management
Administrative Roles
Administrators

Servers and Components
MVS Topology
 Server View
 Component View
High Availability

Reporting
Call Detail Record Report
Disconnected Call Report

Server View
The following table lists all the BlackBerry MVS Servers and indicates which BlackBerry MVS components are configured on the BlackBerry MVS Servers.

Instance Name	Configured					Delete
	FMC Phone	Event Manager	BES Connector	Data Manager	Witness Server	
AV01-MVS	Yes	Yes	Yes	Yes	Yes	Delete

[Add MVS Server](#)

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

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

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

BlackBerry Mobile Voice System

User: sysadmin | [About](#) | [Change Password](#) | [Log Off](#)

Friday, January 20, 2012

Home

- Dashboard

User Management

- Users
- Templates
 - Add
 - Manage
- Class of Service
 - Add
 - Manage

System Configuration

- Telephony Connectors
 - Add
 - Manage
- Locations
- Voice Mail Connectors
 - Licensing

Administrator Management

- Administrative Roles
- Administrators

Servers and Components

- MVS Topology
 - Server View
 - Component View
- High Availability

Reporting

- Call Detail Record Report
- Disconnected Call Report

Edit MVS Server - AV01-MVS

Instance Name: AV01-MVS *

IP Address: 10.64.50.66 *

Session Manager

FMC Phone

You cannot disable this FMC Phone because it is part of a high availability installation.

Configured: Yes

SIP IP Address: 10.64.50.66 *

Line Port: 5060 *

Trunk Port: 6060 *

Telephony Connector	DID/DDI Number
Audio codes 1K	+13035383606
Telephony Connector	No Data Coverage Number
Audio codes 1K	+13035551111

Event Manager

Configured: Yes

Listen Port: 17635 *

☐ Enable Event Email Notifications

SMTP Server:

SMTP Port:

"To" Email Address:

"From" Email Address:

☐ Does the BlackBerry MVS require SMTP authentication to send email notifications?

BlackBerry Enterprise Server Connector

You cannot enable this BlackBerry Enterprise Server Connector because it is part of a high availability installation.

Configured: Yes

Listen Port: 17633 *

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

Data Manager

Configured: Yes

Listen Port: 17631 *

Witness Server

Configured: Yes

Listen Port: 17634 *


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8.3.2 Create a Telephony Connector

Create a telephony connector to use as a SIP gateway to communicate with the Communication Manager. In the case of the compliance test, the telephony connector was the AudioCodes Mediant 1000. To add a telephony connector, navigate to **System Configuration → Telephony Connectors → Add** from the left-hand navigation menu described at the top of **Section 8.3**.

Configure the parameters as described below. After creation, if the Telephone Connector needs to be modified, it can be edited by navigating to **System Configuration → Telephony Connectors → Manage**.

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



BlackBerry® Mobile Voice System

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Edit Telephony Connector - Audio codes 1K

Display Name

Audio codes 1K

Type

SIP Gateway

IP Address

10.64.50.68

Host Name

10.64.50.68

Trunk Port



5060

IP-IP Mode

☒

PBX Initiated Calling

Caller Identification Number *

Number	Action
	
3035383592	

Phone Number Translation

Location

United States of America

International Direct Dialing

011

National Direct Dialing

1

Home Country Code

1

Save

Cancel

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8.3.3 Create User Account Template

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

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

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

The screenshot shows the BlackBerry Mobile Voice System (MVS) configuration interface. The top navigation bar includes the BlackBerry logo, the text "BlackBerry® Mobile Voice System", and user information: "User: sysadmin | About | Change Password | Log Off". The date "Friday, January 20, 2012" is displayed on the right. The left-hand navigation menu is expanded, showing "User Management" with sub-items: "Users" (Add, Manage, Import), "Templates" (Add, Manage), "Class of Service" (Add, Manage), "System Configuration" (Telephony Connectors, Locations, Voice Mail Connectors, Licensing), "Administrator Management" (Administrative Roles, Administrators), "Servers and Components" (MVS Topology, Server View, Component View, High Availability), and "Reporting" (Call Detail Record Report, Disconnected Call Report). The main content area is titled "Edit Template - default" and contains several sections: "BlackBerry MVS Line Configuration" with fields for "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), and "Network Handoff" (Automatic handoff with user prompt); "Caller Restrictions" with "Caller Restriction" (All Callers Except Blocked) and a checked "Allow calls from unknown numbers"; "Blocked Callers (max limit = 20)" and "Allowed Callers (max limit = 20)" tables, both currently empty; and "Call Scheduling (in office time zone)" with "Office Time Zone" set to "(GMT-05:00) Eastern Time (US & Canada)" and a table for "Day of Week" scheduling. The "Day of Week" table has columns for "Day of Week", "Allow Calls", "Start of Day (HHMM)", and "End of Day (HHMM)". The rows are: Weekly (Always, Start of Day, End of Day), Sunday (Always), Monday (Always), Tuesday (Always), Wednesday (Always), Thursday (Always), Friday (Always), and Saturday (Always). At the bottom of the form are "Save" and "Cancel" buttons. The footer text reads "Copyright © 2011 Research In Motion Limited".

8.3.4 Create Class of Service

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

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

The example below shows the DNIS pool class of service.

The example below shows the ANI pool class of service.

BlackBerry Mobile Voice System

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Edit Class of Service - ANI Pool

Class of Service Name: ANI Pool *

BlackBerry MVS Call Features

- ☒ User can transfer a call
- ☒ User can move a call to the desk phone
- ☒ User can move a call from the desk phone to the BlackBerry device
- ☒ User can move a call to the mobile phone number
- ☒ User can move a call to a one time number
- ☒ User can move a call from Voice over Wi-Fi to Voice over Mobile
- ☒ User can move a call from Voice over Mobile to Voice over Wi-Fi
- ☒ User can have a schedule for incoming calls
- ☒ User can add participants to an active call
- ☒ Device can initiate automatic handoffs between Voice over Wi-Fi and Voice over Mobile
- ☒ Device can set the mobile phone number

BlackBerry MVS Only Calling

☐ Restrict calls to use only the BlackBerry MVS Line

Call Direction

Incoming Calls: ☒ PBX → BlackBerry Smartphone
☐ BlackBerry Smartphone → PBX

Outgoing Calls: ☒ PBX → BlackBerry Smartphone
☐ BlackBerry Smartphone → PBX

Wi-Fi Profile Control

☒ Allow Voice over Wi-Fi calling on user created Wi-Fi profiles

Device Settings

RTP Start Port: 51100 *

Preferred Order of Device Codecs: G.711 u-law, G.711 A-law, G.729a, G.729b

User Edit Control

- ☒ User may change the mobile phone number
- ☒ User may change the call move to desk phone number
- ☒ User may change the BlackBerry MVS line label
- ☒ User may change the default line for outgoing calls
- ☒ User may change the outgoing call setup sound
- ☒ User may change the Voice Mail access number
- ☒ User may change caller restrictions
- ☒ User may change the default network for BlackBerry MVS calls
- ☒ User may select the automatic handoff method
- ☒ User may enable and disable automatic handoff

Scheduling Control

Select All - Select None

- ☒ User may set Monday scheduling
- ☒ User may set Tuesday scheduling
- ☒ User may set Wednesday scheduling
- ☒ User may set Thursday scheduling
- ☒ User may set Friday scheduling
- ☒ User may set Saturday scheduling
- ☒ User may set Sunday scheduling


Save Cancel

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8.3.5 Add Mobile Voice System Users

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

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



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Add BlackBerry MVS Users

?

User Name

BlackBerry Enterprise Server

AV01-BES

Search

Clear

Template

default

?

Class of Service

DNIS Pool

*

MVS Server

AV01-MVS

Telephony Connector

Audio codes 1K

Voice Mail Connector

None

First

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Page 1

of 1

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Last

1 selected user

	Phone Numbers				
<input checked="" type="checkbox"/>	User Name	Mobile	Extension	Desk	Direct Dial
<input checked="" type="checkbox"/>	Test User04	6505548868	1234	1234	1234

First

< Previous

Page 1

of 1

Next >

Last

Add MVS User(s)

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8.3.6 Manage Mobile Voice System Users

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

BlackBerry Mobile Voice System

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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
Extension
BlackBerry Enterprise Server
MVS Server
Telephony Connector
Voice Mail Connector
Class of Service

Search Clear


Search Results

1 selected user

User Name	Extension	Enabled	Class of Service	Telephony Connector	Voice Mail Connector	BlackBerry Enterprise Server	MVS Server
<input checked="" type="checkbox"/> Test User06	1111	No	ANI Pool	Audio codes 1K	None	AV01-BES	AV01-MVS
<input type="checkbox"/> Test User07	60002	Yes	ANI Pool	Audio codes 1K	None	AV01-BES	AV01-MVS
<input type="checkbox"/> Test User01	3333	No	ANI Pool	Audio codes 1K	None	AV01-BES	AV01-MVS
<input type="checkbox"/> Test User02	61001	Yes	ANI Pool	Audio codes 1K	None	AV01-BES	AV01-MVS
<input type="checkbox"/> Test User03	60001	Yes	DNIS Pool	Audio codes 1K	None	AV01-BES	AV01-MVS
<input type="checkbox"/> Test User04	4444	No	DNIS Pool	Audio codes 1K	None	AV01-BES	AV01-MVS

Remove MVS Users Change MVS User Settings

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


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 BlackBerry® Mobile Voice System

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Change Users

General Attributes

MVS Service

Select To Change

Device Number Prefix or Replacement

Extension

Add Prefix

Change Prefix From

To

Call Move To Desk Number

Add Prefix

Change Prefix From

To

MVS Server

MVS Server

Select To Change

Telephony Connector

Telephony Connector

Select To Change

Voice Mail Connector

Voice Mail Connector

Select To Change

BlackBerry MVS Line Configuration

BlackBerry MVS Line Label

Default line for outgoing calls

Default network for BlackBerry MVS calls

Outgoing call setup sound

Voice Mail Access Number

MWI Notifications

Network Handoff

Caller Restrictions

Caller Restriction

Allow calls from unknown numbers

Numbers to Add to User Call Lists

Blocked Callers (max limit = 20)

Number	Name	Action

Allowed Callers (max limit = 20)

Number	Name	Action

Numbers to Remove from User Call Lists

Blocked Callers (max limit = 20)

Number	Action

Allowed Callers (max limit = 20)

Number	Action

Call Scheduling (in office time zone)

Office Time Zone

Day of Week	Allow Calls	Start of Day (HHMM)	End of Day (HHMM)
Weekly	Always		
Sunday	Select To Change		
Monday	Select To Change		
Tuesday	Select To Change		
Wednesday	Select To Change		
Thursday	Select To Change		
Friday	Select To Change		
Saturday	Select To Change		

Class of Service

Class of Service

Select To Change

Review Changes

Reset

Cancel

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9 Verification Steps

The following steps may be used to verify the configuration:

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

```
status signaling-group 2
                        STATUS SIGNALING GROUP

      Group ID: 2
      Group Type: sip

      Group State: in-service
```

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

```
status trunk 2
                        TRUNK GROUP STATUS

Member   Port      Service State      Mtce Connected Ports
                        Busy

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

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

The screenshot displays the Avaya Aura® System Manager 6.1 web interface. The top navigation bar includes the Avaya logo, the product name, and links for Help, About, Change Password, and Log off admin. A breadcrumb trail shows the path: Home / Elements / Session Manager / System Status / SIP Entity Monitoring - SIP Entity Monitoring. The left sidebar contains a tree view with categories like Session Manager, Network Configuration, Device and Location Configuration, Application Configuration, System Status, and System Tools. The main content area is titled 'SIP Entity Link Monitoring Status Summary' and includes a 'Run Monitor' button. Below this is a table with 1 item, showing details for 'sm5031'. A second section, 'All Monitored SIP Entities', also has a 'Run Monitor' button and a table with 2 items: 'Audio Codes Med 1000' and 'cm5052'.

Avaya Aura® System Manager 6.1

Help | About | Change Password | Log off admin

Session Manager * Home

Home / Elements / Session Manager / System Status / SIP Entity Monitoring - SIP Entity Monitoring

SIP Entity Link Monitoring Status Summary

This page provides a summary of Session Manager SIP entity link monitoring status.

Entity Link Status for All Session Manager Instances

Run Monitor

Session Manager Name	Entity Links Down/Total	Entity Links Partially Down	SIP Entities - Monitoring Not Started	SIP Entities - Not Monitored
sm5031	0/2	0	0	0

Select : All, None

All Monitored SIP Entities

Run Monitor

2 Items Refresh Show ALL Filter: Enable

SIP Entity Name
Audio Codes Med 1000
cm5052

Select : All, None

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

10 Conclusion

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

11 Additional References

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

- [1] *Administering Avaya Aura® Communication Manager*, June 2010, Release 6.0, Issue 6.0, Document Number 03-300509, available at <http://support.avaya.com>.
- [2] *Administering Avaya Aura® Session Manager*, October 2010, Issue 1.1, Release 6.1, Document Number 03-603324, available at <http://support.avaya.com>.
- [3] *Avaya one-X Deskphone Edition for 9600 Series IP Telephones Administrator Guide Release 3.1*, November 2009, Document Number 16-300698.
- [4] *Implementing Avaya Aura® Communication Manager Messaging*, May 2011, Document Number 18-603644.

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

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

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

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

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