



Avaya Solution & Interoperability Test Lab

Configuring Avaya Aura® Messaging 6.1 as a Voice Messaging Solution for Avaya Business Communication Manager 450 Release 6.0 with SIP trunking through Avaya Aura® Session Manager 6.1 – Issue 1.0

Abstract

These Application Notes describe a sample configuration of Avaya Aura® Messaging 6.1 as a voice mail solution for Avaya Business Communication Manager 450 6.0. In this configuration Avaya Aura® Messaging and Business Communication Manager 450 are connected to Avaya Aura® Session Manager R6.1 over SIP trunks. Avaya Aura® Session Manager provides SIP proxy/routing functionality, routing SIP sessions across a TCP/IP network with centralized routing policies. Avaya Aura® Messaging supports Business Communication Manager 450 endpoints for voice messaging features such as greeting menu, user mailbox services and transfer functionalities.

These Application Notes provide information for the setup, configuration, and verification of the call flows tested for this solution.

1. Introduction

These Application Notes describe a sample configuration of Avaya Aura® Messaging 6.1 as a voice mail solution for Avaya Business Communication Manager (BCM) 450 6.0. In this configuration Avaya Aura® Messaging and Avaya BCM are connected to Avaya Aura® Session Manager over SIP trunks. Avaya Aura® Session Manager provides SIP proxy/routing functionality, routing SIP sessions across a TCP/IP network with centralized routing policies. Avaya Aura® Messaging provides unified communications features such as greeting menu, user mailbox services and transfer functionalities. Avaya Aura® Communication Manager is setup as an emulated PSTN connected to Avaya BCM through T1 connection.

2. Interoperability Testing and Test Result

Interoperability was tested between Avaya Business Communication Manager and Avaya Aura® Messaging with SIP trunking through Avaya Aura® Session Manager.

2.1. Interoperability Compliance Testing

Interoperability testing was executed between a variety of Avaya telephones such as Digital, UNISim registered to Business Communication Manager, Avaya SIP phones registered to Session Manager and Avaya H323, Digital phones registered to Communication Manager. The focus was to verify call and messaging functionality between Communication Manager, Business Communication Manager and Avaya Aura® Messaging in a SIP network with trunking through Session Manager.

The following Avaya Aura® Messaging capabilities were covered:

- No Answer
- Personal Greetings
- Bypass Greetings
- Message Waiting Indication
- Reply
- Call Forwarding
- Multiple Call Forwarding
- Call Transfer
- Simultaneous Calls
- Personal Operator
- Personal Operator – No Answer
- Auto Attendant
- Auto Attendant – No Answer
- Call to Forward All (forward to messaging access number) endpoint.
- Call to Busy endpoint (messaging access number is set if this endpoint busy) All the call is forwarded to pilot number.

The following Avaya Aura® Messaging capabilities were not in scope for this testing:

- Call Sender
- Reach Me
- Notify Me

2.2. Test Results and Observations

Interoperability testing of Avaya Aura® Messaging 6.1 Single Server as a voice mail solution for Avaya Business Communication Manager with SIP Trunking through Avaya Aura® Session Manager R6.1 was successful.

3. Reference Configuration

Figure 1 below illustrates the configuration used in these Application Notes. The sample configuration shows an enterprise with Avaya BCM communicating with the Avaya Aura® Messaging via a SIP trunk. The Avaya BCM has an analog, a digital and an IP Telephone connected as endpoints.

For security purposes public IP addresses have been masked out or altered in this document.

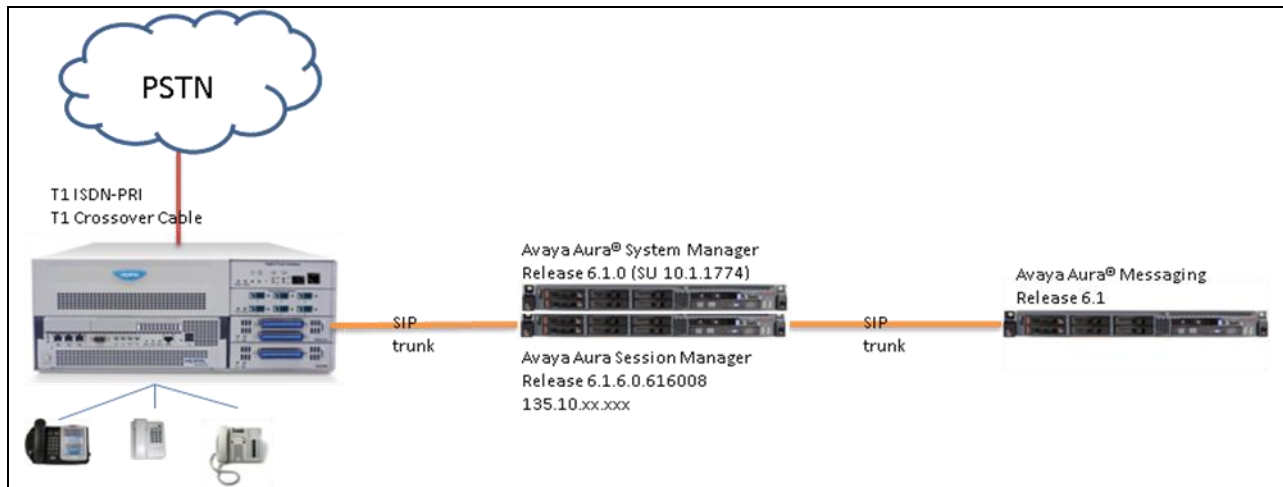


Figure 1: Network Configuration

4. Equipment and Software Validated

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

Equipment	Software
S8800 Server	Avaya Aura® Messaging Release 6.1
S8800 Server	Avaya Aura® Session Manager Release 6.1
S8800 Server	Avaya Aura® System Manager Release 6.1
G450 with S8300D, emulated PSTN	Avaya Aura® Communication 6.0 PRI, Digital Trunk Interface Module
Business Communication Manager 450	Avaya BCM450 R6 SU 011-201205
2 – Avaya 1140E IP Telephone (SIP)	Firmware Version: 0625C8J
2- Digital Phones T7316	n/a
2 – Analog Phones	n/a

5. Configure Avaya Business Communication Manager with SIP Trunking to Avaya Aura® Session Manager

This section describes the procedure for setting up Avaya BCM. The following administration activities will be described:

- Configure Proxy for Private SIP Trunking
- Configure the Global setting for SIP Trunking
- Configure general info for IP Trunks
- Configure the routing
- Configure the destination code
- Configure the private network for dialing plan
- Configuring the Public network for the dialing plan
- Configure Target Lines
- Assign a target line to a selected set
- Configure Active Sets

The highlights in the following screens indicate the values used during the compliance test. Default values may be used for all other fields. Please keep in mind that the values used in this guide may be unique to the example shown. User will have to use values unique to their site, where this solution is being deployed (e.g., site's IP address, extension numbers, etc).

Avaya BCM configurations can be performed through Business Element Management only.

5.1. SIP Trunking Configuration

This section explains the steps to configure a SIP trunk routing entry that will access the Messaging via Session Manager from the Avaya BCM.

5.1.1. Configure Proxy for Private SIP Trunking

After logging into the BCM element manager, configure a private proxy for the SIP trunking by selecting, *Configuration* → *Resources* → *IP Trunks* → *SIP Trunking*

Select the tab *Private* → *Proxy* to add a proxy as shown in figure below:

- **Domain:** the defined domain that the Avaya Aura® Messaging and Avaya Aura® Session Manager system is assigned to. During compliance test bwdev.com domain is used.
- **IP Address:** is Session Manager's IP.

The screenshot displays the Avaya BCM Element Manager configuration interface for SIP Trunking. On the left is the Task Navigation Panel with a tree view showing the navigation path: Configuration → Resources → IP Trunks → SIP Trunking. The main window is titled 'SIP Trunking' and has tabs for 'Public', 'Private', 'Global Settings', and 'Media Parameters'. The 'Private' tab is selected, and within it, the 'Proxy' sub-tab is active. The 'SIP Proxy' configuration section includes a 'Domain' field with the value 'bwdev.com', a 'Route all calls using proxy' checkbox (unchecked), and an 'MCDN Protocol' dropdown menu set to 'None'. To the right, there is an 'Optional IP Address for legacy routing' section with an 'IP Address' field containing '135.10.198' and a 'Port' field containing '5060'. Below this is the 'Outbound Proxy Table' which contains one entry:

Domain	IP Address	Port	Load-balancing Weight	Keep alive
bwdev.com	135.10.198	5060		0/None

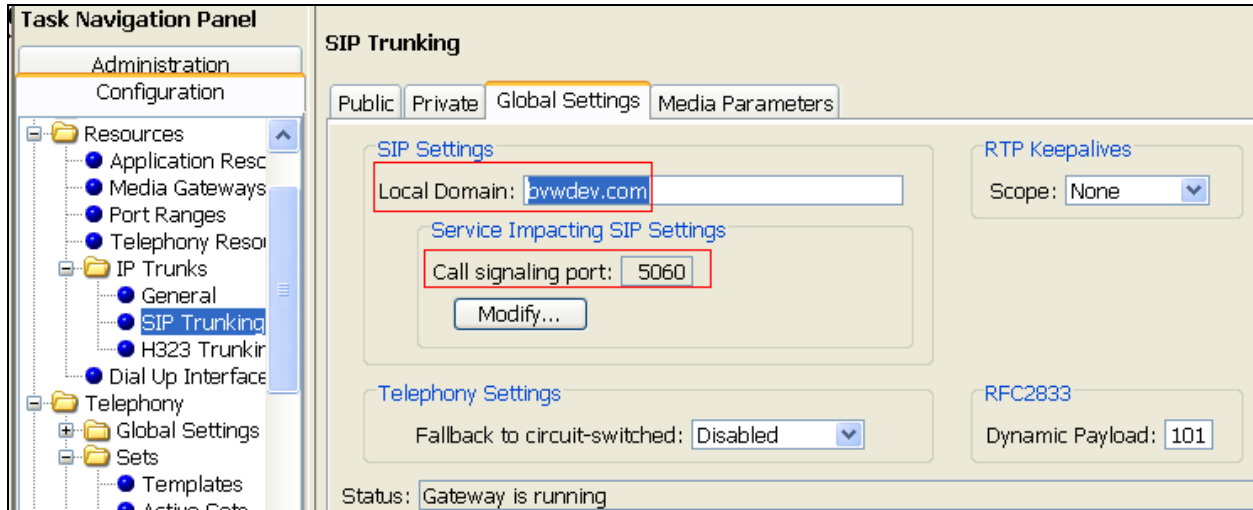
At the bottom of the configuration area are 'Add...' and 'Delete' buttons.

5.1.2. Configure the Global settings for SIP Trunking:

Navigate to *Configuration* → *Resources* → *IP Trunks* → *SIP Trunking*

Select the tab *Global Settings* as shown in figure below:

- **Local Domain:** is the defined domain that the AAM and ASM system is assigned to.
- **Call Signaling port:** 5060



5.2. IP Trunks Configuration

This section describes how to configure the general settings for IP trunk.

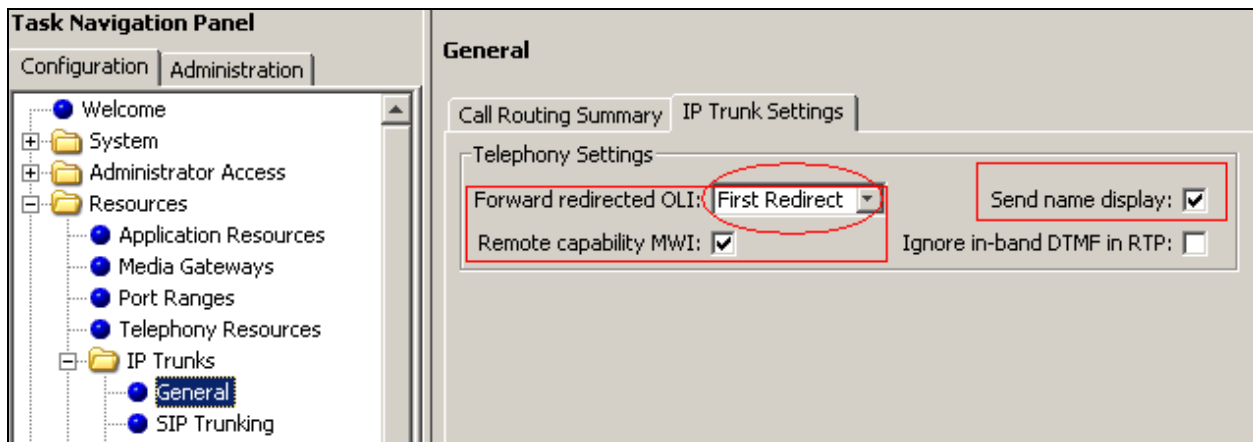
5.2.1. General IP trunk settings:

Navigate to *Configuration* → *Resources* → *IP Trunks* → *General* → *IP Trunk Settings*

Setup the general information for the IP trunk as below:

- **Forward redirected OLI:** First Redirect.
- **Remote capability MWI:** checked
- **Send name display:** checked.

Note: for detail of these setting please refer to Avaya BCM documentation listed in reference **Section 12**



5.3. Dialing Plan Configuration

This section describes how to configure the dialing plan, routes and pool that will be used by the Avaya BCM to communicate with the Avaya Aura Messaging.

5.3.1. Configure the routing:

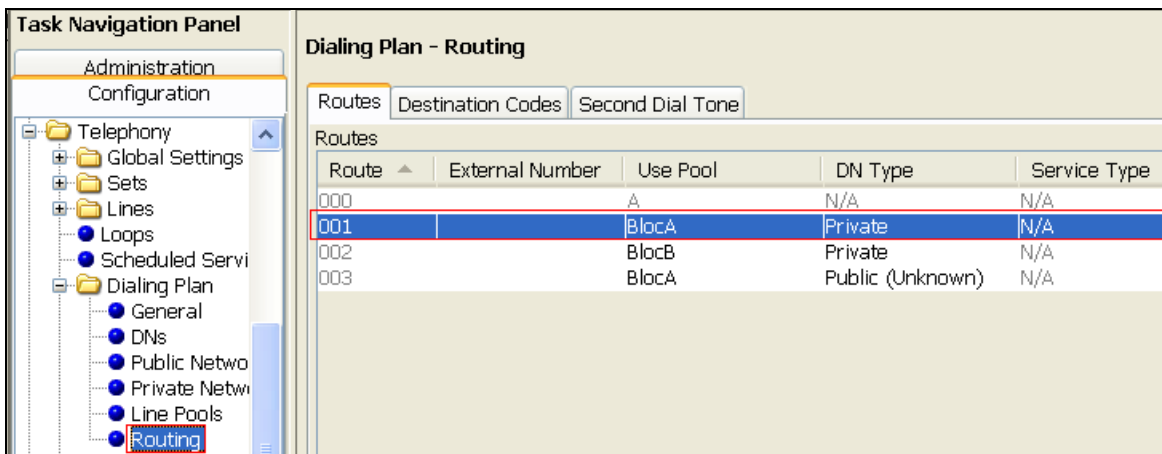
Navigate to *Configuration* → *Telephony* → *Dialing Plan* → *Routing*

In *Routes* tab to add a new route by click on the *Add* button. Enter the route number 001 and click *OK* when Done.

Double click on new created Route and assign value to the route as below:

- **Use Pool:** BlocA.
- **DN Type:** Private

The rest of the values leave them as default.

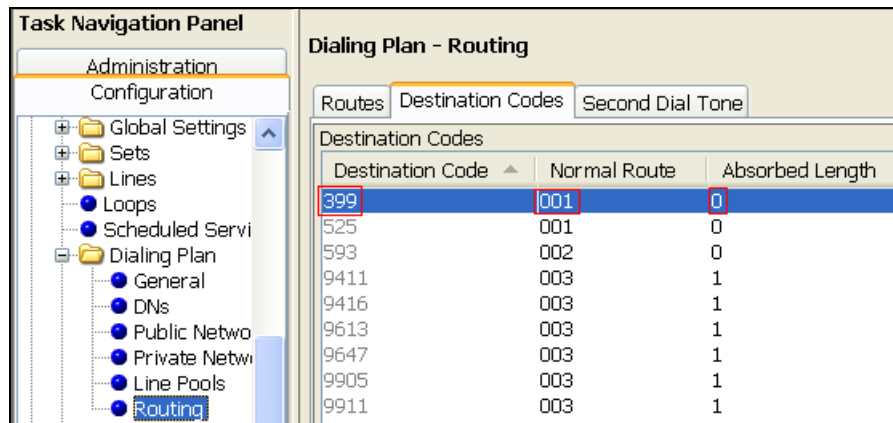


5.3.2. Configure the destination code:

Navigate to *Configuration* → *Telephony* → *Dialing Plan* → *Routing*

In *Destination Codes* tab add a destination code as shown in figure below:

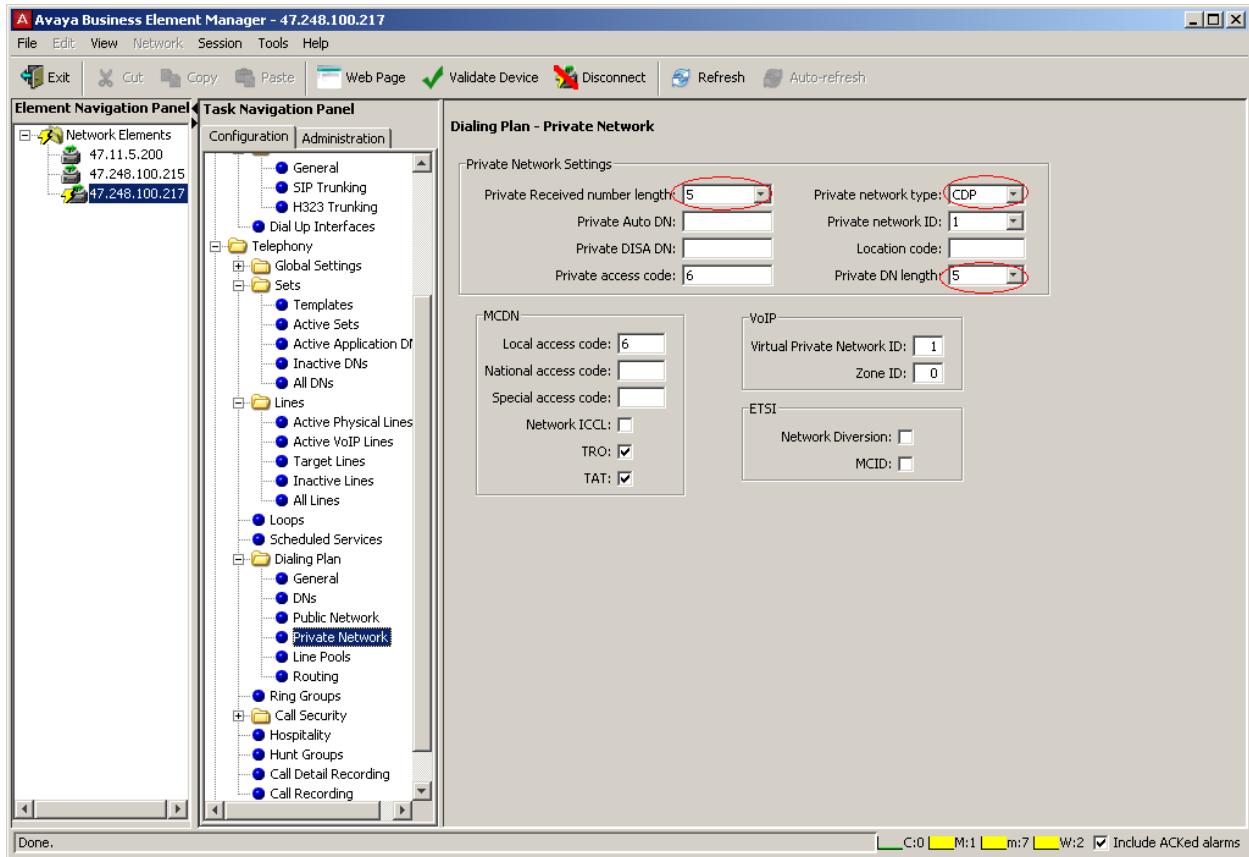
- **Destination Code:** 399. The destination code 399 is chosen because the AAM pilot number used in the example is 39990.
- **Normal Route:** 001
- **Absorbed Length:** 0



5.3.3. Configure the dialing plan private network: *Configuration* → *Telephony* → *Dialing Plan* → *Private Network*

Configure the private network as shown in figure below:

- **Private Received number length:** 5.
- **Private network type:** CDP.
- **Private DN length:** 5



5.4. Target Lines Configuration

This section describes how to configure target lines which will be assigned to telephones that will be used as endpoints connected to the BCM.

5.4.1. Configure a target line to a selected set:

Navigate to *Configuration* → *Telephony* → *Lines* → *Target Lines*

In the Target Lines screen, select a **Line** and enter **DN** to selected **Line**

For example in the figure shown below:

- **Line:** 413 has been selected.
- **DN:** 22235 has been assigned by clicking on the **Add** button under the **Assigned DNs** tab.

Note: Add unique set DN to one Target line. Require one line assignment for every telephone device in the system.

The screenshot shows the 'Target Lines' configuration window. On the left is a 'Task Navigation Panel' with a tree view containing 'Administration', 'Configuration', 'System', 'Administrator Access', 'Resources', 'Telephony', 'Global Settings', 'Sets', 'Lines', 'Active Physical Lines', 'Active VoIP Lines', 'Target Lines', 'Inactive Lines', 'All Lines', 'Loops', 'Scheduled Services', 'Dialing Plan', 'Ring Groups', 'Call Security', 'Hospitality', 'Hunt Groups', 'Call Detail Recording', 'Call Recording', 'Data Services', and 'Applications'. The 'Target Lines' table is as follows:

Line	Trunk ...	Name	Control Set	Line Type	Prime Set	Pub. Received #	Priv. Received #	Distinct Ring
365	Target line	Line365	22231	Public	22231	22232	22232	None
413	Target line	Line413	22231	Public	22231	22235	22235	Pattern 2
999	Target line	Line999	22263	Public	22263	9134400150	22263	None
896	Target line	Line896	20224	Public	22231		22264	None
998	Target line	Line998	22263	Public	22263	9134400061	22268	None
994	Target line	Line994	20224	Public	22231		22349	None
997	Target line	Line997	22263	Public	22263		22524	None
426	Target line	Line426	22231	Public	22231	22525	22525	None
995	Target line	Line995	22263	Public	22263	22624	22624	None
996	Target line	Line996	22263	Public	22263	9134404664		None

Below the table are buttons for 'Copy', 'Paste...', and 'Renumber'. The 'Details for Line: 413' section shows the 'Assigned DNs' tab with the following table:

DN	Appearance Type	Appearances	Caller ID Set	Vmsg Set
22235	Appr&Ring	1	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>

Buttons for 'Add...' and 'Delete' are also present.

Also for the assigned set to generate busy tone while it is busy.

Select **Preferences** tab:

- **If Busy** field: Busy tone has to be selected as shown in figure below.

The screenshot shows the 'Target Lines' configuration window with line 365 selected. The 'Target Lines' table is as follows:

Line	Trunk Type	Name	Control Set	Line Type	Prime Set	Pub. Received #	Priv. Received #
361	Target line	Line361	22231	Public	22231	22441	
362	Target line	Line362	22231	Public	22231	22234	22234
363	Target line	Line363	22231	Public	22231	22235	22235
364	Target line	Line364	22231	Public	22231	22221	22221
365	Target line	Line365	22231	Public	22231	22232	22232
366	Target line	Line366	22234	Public	22234		22334
367	Target line	Line367	22231	Public	22231		
368	Target line	Line368	22231	Public	22231		
369	Target line	Line369	22231	Public	22231		
370	Target line	Line370	22231	Public	22231		
371	Target line	Line371	22231	Public	22231		
372	Target line	Line372	22231	Public	22231		
373	Target line	Line373	22231	Public	22231		

The 'Details for Line: 365' section shows the 'Preferences' tab with the following settings:

- Aux. ringer:
- Distinct rings in use: Pattern 2
- If Busy: **Busy tone** (selected in a dropdown menu)
- Voice message center: 1
- Redirect to: (empty dropdown)

5.5. Active Sets Configuration

This section describes the steps to configure the sets that have been assigned to a line as explained in **Section 5.4**

5.5.1. Configure the Active sets:

Select **Configuration** → **Telephony** → **Sets** → **Active Sets**

Example configuration for **Line Access** tab of selected active set:

- Select **DN**: 22235
- **Fwd No Answer**: 39990
- **Fwd Busy**: 39990

Note: **39990** is the pilot number of Avaya Messaging.

Example configuration for **Line Assignment** tab of selected active set:

- **Vmsg Set**: checked so that voice mail can be accessed by the DN 22235
- **Priv. Received#**: 22235
- **Pub. Received #**: 22235.

Task Navigation Panel

- Administration
 - Configuration
- Welcome
- System
- Administrator Acce
- Resources
- Telephony
 - Global Settings
 - Sets
 - Templates
 - Active Sets
 - Active Applic.
 - Inactive DNs
 - All DNs
 - Lines
 - Active Physic
 - Active VoIP L
 - Target Lines
 - Inactive Lines
 - All Lines
 - Loops
 - Scheduled Servi
- Dialing Plan
- Ring Groups

Active Sets

Line Access | Capabilities and Preferences | Restrictions

DN	Model	Name	Port	Pub. OLI	Priv. OLI	Fwd No Answer	Fwd Delay	Fwd Busy
22229	Analog	22229	1006				N/A	
22230	Analog	22230	1007				N/A	
22231	Analog	22231	1001			22301	3	
22232	1140E/2004/200...	22232	0248	22232	22232	39990	4	39990
22233	1140E/2004/200...	22233	0233	22233	22233	39990	4	39990
22234	T7316/M7310	22234	2001	22234	22234	39990	2	39990
22235	T7316/M7310	22235	2002	22235	22235	39990	2	39990
22236	Analog	22236	4001	22236	22236		N/A	
22237	Analog	22237	4002	22237	22237		N/A	
22254	1140E/2004/200...	22254	0243				N/A	

Copy | Paste... | Renumber...

Details for DN: 22235

Line Assignment | Line Pool Access | Answer DNs | MeetMe Conferencing

Assigned Lines

Line	Appearance Type	Appearances	Caller ID Set	Vmsg Set	Priv. Received #	Pub. Received #
413	Appr&Ring	1	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	22235	22235

Figures below show the additional configurations to be done to the selected DN which has to be member of the BlocA pool found in the **Line Pool Access** tab.

Task Navigation Panel

Administration
Configuration

Welcome
System
Administrator Access
Resources
Application Resources
Media Gateways
Port Ranges
Telephony Resources
IP Trunks
Dial Up Interfaces
Telephony
Global Settings
Sets
Templates
Active Sets
Active Application D...
Inactive DNs
All DNs
Lines
Loops
Scheduled Services
Dialing Plan
General
DNs
Public Network
Private Network
Line Pools
Routing
Ring Groups
Call Security
Hospitality

Active Sets

Line Access Capabilities and Preferences Restrictions

DN	Model	Name	Port	Pub. OLI	Priv. OLI	Fwd No Answer	Fwd Delay	Fwd Busy
22225	Analog	22225	1002				N/A	
22226	Analog	22226	1003				N/A	
22227	Analog	22227	1004				N/A	
22228	Analog	22228	1005			22301	4	22301
22229	Analog	22229	1006				N/A	
22230	Analog	22230	1007				N/A	
22231	Analog	22231	1001			22301	3	
22232	1140E/2004/2007/2050...	22232	0248	22232	22232	39990	4	39990
22233	1140E/2004/2007/2050...	22233	0233	22233	22233	39990	4	39990
22234	T7316/M7310	22234	2001	22234	22234	39990	2	39990
22235	T7316/M7310	22235	2002	22235	22235	39990	2	39990
22236	Analog	22236	4001	22236	22236	39990	2	39990

Copy Paste... Renumber...

Details for DN: 22235

Line Assignment Line Pool Access Answer DNs MeetMe Conferencing

Line Pools

Line Pool

A
BlocA

Add... Delete

In the *Capabilities and Preferences* tab make sure the following options are selected:

- **DND on Busy:** checked.
- **Allow redirect:** checked.

Task Navigation Panel

Administration
Configuration

Welcome
System
Administrator Access
Resources
Application Resources
Media Gateways
Port Ranges
Telephony Resources
IP Trunks
Dial Up Interfaces
Telephony
Global Settings
Sets
Templates
Active Sets
Active Application D...
Inactive DNs
All DNs
Lines
Loops
Scheduled Services

Active Sets

Line Access Capabilities and Preferences Restrictions

DN	Model	Name	Prime Line	Intercom K...	Control ...	First Dis...	Auto Called...
22234	T7316/M7310	22234	I/C	2	22231	Name	<input checked="" type="checkbox"/>
22235	T7316/M7310	22235	I/C	2	22231	Name	<input checked="" type="checkbox"/>
22236	Analog	22236	I/C	N/A	22231	Name	<input type="checkbox"/>

Copy Paste...

Details for DN: 22235

Capabilities SWCA Call Group Preferences Button Programming Table Button Programming User Speed Dial

Handsfree: Auto
Pickup group:
Page zone: 1
Direct dial: 1
Intrusion protection level: None

HF answerback:
DND on Busy:
Paging:
Auto hold for incoming page:
Priority call:
Auto hold:

Allow redirect:
Redirect ring:
Receive short tones:
Silent monitor supervisor:

6. Configure Avaya Business Manager 450 with PRI trunk to PSTN

6.1. Administer Resources

This section describes how to configure a PRI Trunk on BCM to PSTN.

6.1.1. Administer Application Resource for PRI Trunks

These Application Notes assume that the basic configuration has already been administered.

This section describes steps for configuring Application Resource for PRI Trunks on BCM to work with Service Provider system.

For further information on Avaya Business Communication Manager 450 configuration, please consult references in **Section 12**.

Enable the PRI device on Avaya BCM by select **Resources** → **Telephony Resources**.

Under **Configured Device** column, select DTM + PRI and then click **Enable** button if it is not already enable as show in figure below.

Configure PRI trunk parameter as highlighted in red box. Others are left as default:

- **Trunk type:** PRI
- **Protocol:** NI-2
- **NSF extension:** None.
- **B channel selection:** Descending Sequential.
- **Clock source:** Internal.
- **CO fail:** TIA-547A.
- **Interface levels:** ISDN.
- **Framing:** ESF.
- **Line config:** B8ZS.

The screenshot shows the Avaya Business Manager 450 configuration interface. On the left is a Task Navigation Panel with a tree view containing categories like Administration, Configuration, System, Identification, Date and Time, Keycodes, IP Subsystem, Telephony Regions, Administrator Access, Resources, Application Resources, Media Gateways, Port Ranges, Telephony Resources, IP Trunks, Dial Up Interfaces, Telephony, Global Settings, Sets, Lines, Loops, Scheduled Services, Dialing Plan, Ring Groups, Call Security, Hospitality, Hunt Groups, Call Detail Recording, Call Recording, and Data Services. The main area is titled 'Telephony Resources' and contains a table of modules. The row for 'Main MBM 3 DTM-PRI' is selected and highlighted in blue. Below the table are buttons for 'Disable', 'Enable', 'Deconfigure...', and 'Configure...'. Below these buttons is a section for 'Details for Module: Main MBM 3 DTM-PRI' with tabs for 'Trunk Module Parameters', 'Call-by-Call Service Selection', 'Trunk Port Details', and 'Provision Lines'. The 'Trunk Module Parameters' tab is active, showing various configuration options. The 'Trunk type' is set to 'PRI', 'Protocol' to 'NI-2', 'NSF extension' to 'None', 'B channel selection' to 'Descending Sequential', and 'Clock source' to 'Internal'. The 'T1 Parameters' section shows 'CO fail' set to 'TIA-547A', 'Interface levels' to 'ISDN', 'Framing' to 'ESF', 'Line coding' to 'B8ZS', 'Internal CSU' checked, and 'CSU line build' to '0 dB'. Red boxes highlight the 'Trunk type', 'Protocol', 'NSF extension', 'B channel selection', 'Clock source', 'CO fail', 'Interface levels', 'Framing', 'Line coding', and 'CSU line build' fields.

Location	Configured Device	Dip Switch	Bus	State	Low	High	Active	Busy
Internal	IP Trunks	N/A	N/A	Enabled	001	008	8	0
Internal	IP Sets	N/A	N/A	Enabled	22221	22627	18	0
Internal	Applications	N/A	N/A	Enabled	22300	22399	62	N/A
Main MBM 1	ASM/ASM+ MBM	All On	10.1	Enabled	20224	22231	7	0
Main MBM 2	DSM16/DSM16+ MBM	All On	20.1	Enabled	22234	22639	2	0
Main MBM 3	DTM-PRI	All On	30.1	Enabled	009	031	23	0
Main MBM 4	ASM/ASM+ MBM	All On	40.1	Enabled	22236	22531	8	0
Expansion 1	None	N/A	N/A	N/A	N/A	N/A	N/A	N/A

6.1.2. Routing Settings

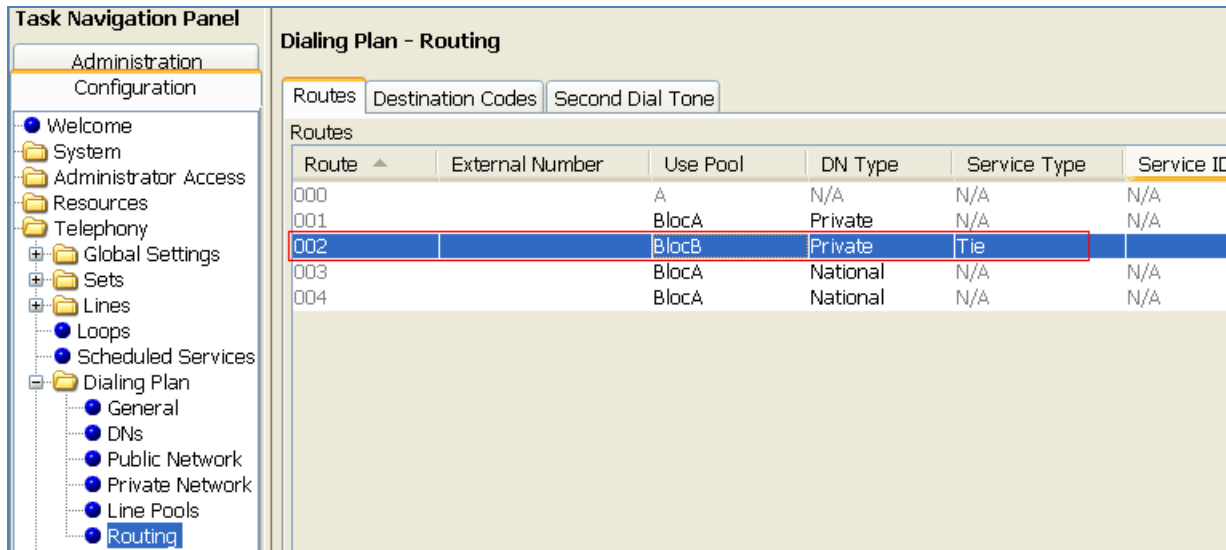
This section describes how to configure the dialing plan, routes and pool that will be used by the Avaya BCM to connect to PSTN.

Navigate to **Telephony → Dialing Plan → Routing**.

In **Routes** tab to add a new route to PSTN by click on the **Add** button. Enter the route number 002 and click OK when Done.

Double click on new created Route and assign value to the route as below, other leave as default:

- **Use Pool:** BlocB.
- **DN Type:** Private.
- **Service Type:** Tie.



Route	External Number	Use Pool	DN Type	Service Type	Service ID
000		A	N/A	N/A	N/A
001		BlocA	Private	N/A	N/A
002		BlocB	Private	Tie	
003		BlocA	National	N/A	N/A
004		BlocA	National	N/A	N/A

6.1.3. Administer Destination Codes

To assign Destination Codes to dial to PSTN via PRI. Perform similar step as shown in section 5.3.2 for with the following information

- **Destination Code:** 4521. Extension on Avaya Communication manager is 52xxx. Digit 4 to let Avaya BCM route the call through route 002
- **Normal Route:** 002.
- **Absorbed Length:** 1. Drop the first digit 4.

6.1.4. Administer Telephony Lines

Assign the pool to telephone line by navigate to **Telephony → Lines → Active Physical Lines**. Double click on a selected line under the **Line Type**, choose **Pool: BlocB** in this example as shown below.

Line	Trunk Type	Name	Control Set	Line Type	Prime Set	Pub. Received #	Priv. Received #	Distinct Ring
009	PR1	Line009	22231	Pool:BlcB	22231	N/A	N/A	None
010	PR1	Line010	22231	Pool:BlcB	22231	N/A	N/A	None
011	PR1	Line011	22231	Pool:BlcB	22231	N/A	N/A	None
012	PR1	Line012	22231	Pool:BlcB	22231	N/A	N/A	None
013	PR1	Line013	22231	Pool:BlcB	22231	N/A	N/A	None
014	PR1	Line014	22231	Pool:BlcB	22231	N/A	N/A	None
015	PR1	Line015	22231	Pool:BlcB	22231	N/A	N/A	None
016	PR1	Line016	22231	Pool:BlcB	22231	N/A	N/A	None
017	PR1	Line017	22231	Pool:BlcB	22231	N/A	N/A	None
018	PR1	Line018	22231	Pool:BlcB	22231	N/A	N/A	None
019	PR1	Line019	22231	Pool:BlcB	22231	N/A	N/A	None
020	PR1	Line020	22231	Pool:BlcB	22231	N/A	N/A	None
021	PR1	Line021	22231	Pool:BlcB	22231	N/A	N/A	None

Copy Paste... Renumber

Details for Line: 009

Preferences Restrictions

Distinct rings in use: Pattern 2

6.1.5. Administer Telephony Target Lines

Assign a DN: 22234 to an available target line **Line: 362**. See **Section 5.4** for detail procedure.

6.1.6. User/Telephone Sets Configuration for Incoming/Outgoing Call

This section show how to configure telephone sets to specific physical line for incoming/outgoing calls to/from a digital set.

Select **Telephony** → **Sets** → **Active Sets**. In the **Line Access**, select the available digital set which has the **Model** is T7316/M7310.

Assign **Priv. OLI: 22234**. This will allow the delivery of the **Calling Line Identification Display**.

For incoming call: Assign the line that configured in Section **6.1.5** o this phone by click on the **Add** button in **Line Assignment** tab. Enter line number, in this example 362 and click OK.

Modify the detail information of the line parameters as values highlighted in red boxes in the below figure:

- **Caller ID Set:** checked.
- **Vmsg Set:** checked.
- **Priv. Received #:** 22234.
- **Pub. Received #:** 22234.

Task Navigation Panel

- Administration
 - Configuration
- Welcome
- System
- Administrator Access
- Resources
- Telephony
 - Global Settings
 - Sets
 - Templates
 - Active Sets
 - Active Applicator
 - Inactive DNs
 - All DNs
 - Lines
 - Active Physical Li
 - Active VoIP Lines
 - Target Lines
 - Inactive Lines
 - All Lines
 - Loops
 - Scheduled Services
- Dialing Plan
 - General
 - DNs
 - Public Network
 - Private Network
 - Line Pools
 - Routing

Active Sets

Line Access | Capabilities and Preferences | Restrictions

DN	Model	Name	Port	Pub. OLI	Priv. OLI	Fwd No Answer	Fwd Delay	Fwd Busy
22232	1140E/2004/2007/2050...	22232	0248	22232	22232	39990	2	39990
22233	1140E/2004/2007/2050...	22233	0233	22233	22233	39990	2	39990
22234	T7316/M7310	22234	2001	22234	22234	39990	2	39990
22235	T7316/M7310	22235	2002	22235	22235	39990	2	39990
22236	Analog	22236	4001	22236	22236	39990	2	39990
22237	Analog	22237	4002	22237	22237	39990	2	39990
22441	1120E/2002	22441	0235	22441			N/A	
22268	1140E/2004/2007/2050...	22268	0240	9134400061	22268		N/A	
22263	1120E/2002	22263	0238	9134400150	22263	96139675279	4	
22221	1140E/2004/2007/2050...	22221	0242			22301	4	22301
22222	1140E/2004/2007/2050...	22222	0249				N/A	

Copy Paste... Renumber...

Details for DN: 22234

Line Assignment | Line Pool Access | Answer DNs | MeetMe Conferencing

Assigned Lines

Line	Appearance Type	Appearances	Caller ID Set	Vmsg Set	Priv. Received #	Pub. Received #
362	Appr&Ring	1	✓	✓	22234	22234

For Outgoing Call: Select tab **Line Pool Access** tab, click **Add** button to add **BlocB**. Click **OK** from the **Add Line Pool** pop up to complete as shown below:

Task Navigation Panel

- Administration
 - Configuration
- Welcome
- System
- Administrator Access
- Resources
- Telephony
 - Global Settings
 - Sets
 - Templates
 - Active Sets
 - Active Applicator
 - Inactive DNs
 - All DNs
 - Lines
 - Active Physical Li
 - Active VoIP Lines
 - Target Lines
 - Inactive Lines
 - All Lines
 - Loops
 - Scheduled Services
- Dialing Plan
 - General
 - DNs
 - Public Network
 - Private Network
 - Line Pools
 - Routing
 - Ring Groups
 - Call Security
 - Hospitality

Active Sets

Line Access | Capabilities and Preferences | Restrictions

DN	Model	Name	Port	Pub. OLI	Priv. OLI	Fwd No Answer	Fwd Delay	Fwd Busy
22232	1140E/2004/2007/2050...	22232	0248	22232	22232	39990	2	39990
22233	1140E/2004/2007/2050...	22233	0233	22233	22233	39990	2	39990
22234	T7316/M7310	22234	2001	22234	22234	39990	2	39990
22235	T7316/M7310	22235	2002	22235	22235	39990	2	39990
22236	Analog	22236	4001	22236	22236	39990	2	39990
22237	Analog	22237	4002	22237	22237	39990	2	39990
22441	1120E/2002	22441	0235	22441			N/A	
22268	1140E/2004/2007/2050...	22268	0240	9134400061	22268		N/A	
22263	1120E/2002	22263	0238	9134400150	22263	96139675279	4	
22221	1140E/2004/2007/2050...	22221	0242			22301	4	22301
22222	1140E/2004/2007/2050...	22222	0249				N/A	

Copy Paste... Renumber...

Details for DN: 22234

Line Assignment | Line Pool Access | Answer DNs | MeetMe Conferencing

Line Pools

Line Pool

BlocA

BlocB

Add... Delete

7. Configure Avaya Aura® Communication Manager as Emulated PSTN – PRI Trunk Configuration

This section focuses on configuring the T1 trunks on Avaya Communication Manager to serve as service provider to Avaya Business Communication Manager, and provides a sample routing using Automatic Alternate Routing (AAR). The configuration procedures include the following areas:

- Administer DS1 circuit pack
- Administer trunk group
- Administer signaling group
- Administer trunk group members
- Administer route pattern
- Administer public unknown numbering
- Administer uniform dial plan
- Administer AAR analysis

7.1. Administer DS1 circuit pack

Log into the System Access Terminal (SAT), and administer a DS1 circuit pack to be used for Connectivity to BCM. Use the **add ds1 001v6** command. Note that the actual slot number may vary. In this case “001v6” is used as the slot number. Enter the following values for the specified fields, and retain the default values for the remaining fields. Submit these changes.

Note: The **Interface** field must be complementary on both switches. For the sample configuration, Avaya Communication Manager is administered as the *network/master* (“peer-master”), and Avaya BCM is administered as the “user/slave”.

```
add ds1 001v6                                     Page 1 of 2
                                                DS1 CIRCUIT PACK
Location: 001V6
Bit Rate: 1.544
Line Compensation: 1
Connect: pbx
TN-C7 Long Timers? n
Interworking Message: PROGRESS
Interface Companding: mulaw
Idle Code: 11111111
DCP/Analog Bearer Capability: 3.1kHz
T303 Timer(sec): 4
Disable Restarts? n
Slip Detection? n
Near-end CSU Type: other
Echo Cancellation? n
Name: To BCM
Line Coding: b8zs
Framing Mode: esf
Signaling Mode: isdn-pri
Interface: peer-master
Peer Protocol: O-SIG
Side: b
CRC? n
```

7.2. Administer Trunk Group

Administer an ISDN trunk group to interface with Avaya BCM. Use the **add trunk-group n** command; where **n** is an available trunk group number. Enter the following values for the specified fields, and retain the default values for the remaining fields.

```
add trunk-group 1                               Page 1 of 21
                                                TRUNK GROUP
Group Number: 1
Group Name: Tie Route to BCM
Direction: two-way
Dial Access? n
Queue Length: 0
Service Type: tie
TestCall BCC: 4
Group Type: isdn
COR: 1
Outgoing Display? n
Busy Threshold: 255
Auth Code? n
Far End Test Line No:
CDR Reports: y
TN: 1
TAC: 100
Carrier Medium: PRI/BRI
Night Service:
```

Navigate to **Page 2**. For the **Supplementary Service Protocol** field, enter “b” for Q-SIG. For the **Format** field, enter “unk-unk”. Retain the default values for the remaining fields.

```
add trunk-group 1                                     Page 2 of 21
  Group Type: isdn

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

  Trunk Hunt: cyclical

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

Navigate to **Page 3**. Enable the **Send Name, Send Calling Number, and Send Connected Number** fields. For the **Format** field, enter “unknown”. Submit these changes.

```

add trunk-group 1                                     Page 3 of 21
TRUNK FEATURES
  ACA Assignment? n                               Measured: none       Wideband Support? n
                                           Internal Alert? n     Maintenance Tests? y
                                           Data Restriction? n   NCA-TSC Trunk Member: 23
                                           Send Name: y         Send Calling Number: y
  Used for DCS? n                               Hop Dgt? n           Send EMU Visitor CPN? n
  Suppress # Outpulsing? n                     Format: unknown
Outgoing Channel ID Encoding: preferred          UII IE Treatment: service-provider
                                           Replace Restricted Numbers? n
                                           Replace Unavailable Numbers? n
                                           Send Connected Number: y
                                           Hold/Unhold Notifications? y
  Send UII IE? y                               Modify Tandem Calling Number: no
  Send UCID? n
  Send Codeset 6/7 LAI IE? y                   Dsl Echo Cancellation? n
  Apply Local Ringback? n
  Show ANSWERED BY on Display? y
                                           Network (Japan) Needs Connect Before Disconnect? n

```

7.3. Administer Signaling Group

Administer an ISDN signaling group for the new trunk group to use for signaling. Use the **add signaling-group n** command, where **n** is an available signaling group number. For the **Primary D-Channel** field, enter the slot number for the DS1 circuit pack from **Section 7.1**.

For the **Trunk Group for NCA TSC** and **Trunk Group for Channel Selection** fields, enter the ISDN trunk group number. For the **Supplementary Service Protocol** field, enter “b” for QSIG. Maintain the default values for the remaining fields, and submit these changes.

```

add signaling-group 1
                                SIGNALING GROUP
Group Number: 1                 Group Type: isdn-pri
Associated Signaling? y         Max number of NCA TSC: 10
Primary D-Channel: 001V624    Max number of CA TSC: 10
                                Trunk Group for NCA TSC: 1
Trunk Group for Channel Selection: 1 X-Mobility/Wireless Type: NONE
TSC Supplementary Service Protocol: b Network Call Transfer? n

```

7.4. Administer Trunk Group Members

Use the **change trunk-group n** command, where **n** is the trunk group number added in **Section 7.2**. Navigate to **Page 3**. For the **NCA-TSA Trunk Member** field, enter the highest trunk group member number to use for routing of tandem QSIG call independent signaling connections.

```
change trunk-group 1                                     Page 3 of 21
21
TRUNK FEATURES
  ACA Assignment? n                                     Measured: none      Wideband Support? n
                                                         Internal Alert? n    Maintenance Tests? y
                                                         Data Restriction? n  NCA-TSC Trunk Member: 23
                                                         Send Name: y        Send Calling Number: y
  Used for DCS? n                                       Hop Dgt? n          Send EMU Visitor CPN? n
  Suppress # Outpulsing? n                               Format: unknown
  Outgoing Channel ID Encoding: preferred              UUI IE Treatment: service-
provider
                                                         Replace Restricted Numbers? n
                                                         Replace Unavailable Numbers? n
                                                         Send Connected Number: y
                                                         Hold/Unhold Notifications? y
  Send UUI IE? y                                         Modify Tandem Calling Number: no
  Send UCID? n
  Send Codeset 6/7 LAI IE? y                             Dsl Echo Cancellation? n
  Apply Local Ringback? n
  Show ANSWERED BY on Display? y
                                                         Network (Japan) Needs Connect Before Disconnect? n
```

Navigate to **Page 4**. Shown below are default values that were used during testing.

```
change trunk-group 1                                     Page 4 of 21
QSIG TRUNK GROUP OPTIONS
TSC Method for Auto Callback: drop-if-possible
  Diversion by Reroute? y
  Path Replacement? y
Path Replacement with Retention? n
  Path Replacement Method: better-route
  SBS? n
Display Forwarding Party Name? y
  Character Set for QSIG Name: eurofont
  QSIG Value-Added? n
```

Navigate to **Page 5** and **6**. Enter all 23 ports of the DS1 circuit pack into the **Port** fields, and the corresponding **Code** and **Sfx** fields will be populated automatically. Enter the ISDN signaling group number into the **Sig Grp** fields as shown below. Submit these changes.

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

Port	Code	Sfx	Name	Night	Sig Grp
1: 001V601	MM710	B			1
2: 001V602	MM710	B			1
3: 001V603	MM710	B			1
4: 001V604	MM710	B			1
5: 001V605	MM710	B			1
6: 001V606	MM710	B			1
7: 001V607	MM710	B			1

7.5. Administer Route Pattern

Create a route pattern for the new ISDN trunk group to use for routing. Use the **change route pattern n** command, where **n** is an available route pattern. Enter the following values for the specified fields, and retain the default values for the remaining fields. Submit these changes.

```
add route-pattern 1                                     Page 1 of 3
                                     Pattern Number: 5 Pattern Name: BCM-Qsig-Route
                                     SCCAN? n Secure SIP? n
Grp FRL NPA Pfx Hop Toll No. Inserted DCS/ IXC
No      Mrk Lmt List Del Digits  QSIG
                                     Dgts Intw
1: 1 0                                     n user
2:                                     n user
3:                                     n user
4:                                     n user
5:                                     n user
6:                                     n user

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

7.6. Administer Public Unknown Numbering

Use the **change public-unknown-numbering 0** command, to define the calling party number to be sent to Avaya Business Communication Manager. Add an entry for the trunk group defined in **Section 7.2**. In the example shown below, all calls originating from a 6-digit extension beginning with 7 and routed to trunk group 1 will result in the 5-digit calling number to be sent. Submit these changes.

```
change public-unknown-numbering 0                               Page 1 of 2
                                NUMBERING - PUBLIC/UNKNOWN FORMAT
```

Ext Len	Ext Code	Trk Grp(s)	CPN Prefix	Total CPN Len
6	7222	1		6

Total Administered: 1
Maximum Entries: 9999

7.7. Administer Uniform Dial Plan

This section provides a sample AAR routing used for routing calls with dialed digits 7xxxxx to Avaya BCM. Use the **change uniform-dialplan 0** command, and add an entry to specify use of AAR for routing of digits 7xxxxx. Enter the following values for the specified fields and retain the default values for the remaining fields. Submit these changes.

```
change uniform-dialplan 0                                     Page 1 of 2
                                UNIFORM DIAL PLAN TABLE
                                Percent Full: 0
```

Matching Pattern	Len	Del	Insert Digits	Net Conv	Node Num
7222	6	1		aar	n

7.8. Administer AAR Analysis

Use the **change aar analysis 0** command, and add an entry to specify how to route the calls to Avaya BCM. Enter the following values for the specified fields and retain the default values for the remaining fields. Submit these changes.

```
change aar analysis 0                                       Page 1 of 2
                                AAR DIGIT ANALYSIS TABLE
                                Location: all                 Percent Full: 1
```

Dialed String	Total Min	Total Max	Route Pattern	Call Type	Node Num	ANI Req'd
7222	6	6	1	aar		n

8. Configure Avaya Aura® Session Manager

This section provides the procedures for configuring Session Manager as provisioned in the reference configuration. Session Manager is comprised of two functional components: the Session Manager server and the System Manager server. All SIP call provisioning for Session Manager is performed through the System Manager Web interface and is then downloaded into Session Manager.

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

In this section, the following topics are discussed:

- SIP Domains
- Locations
- SIP Entities
- Entity Links
- Time Ranges
- Routing Policy
- Dial Patterns
- Synchronization

8.1. Configure SIP Domain

Launch a web browser, enter <http://<IP address of System Manager>/SMGR> in the URL, and log in with the appropriate credentials.

Navigate to **Routing** → **Domains**, and click on the **New** button (not shown) to create a new SIP Domain. Enter the following values and use default values for remaining fields:

- **Name** – Enter the Authoritative Domain Name, e.g “bvwdev.com”.
- **Type** – Select SIP

Click **Commit** to save. The following screen shows the Domains page used during the compliance test.

The screenshot displays the Avaya Aura System Manager web interface. The left sidebar shows a navigation menu with 'Domains' highlighted. The main content area is titled 'Domain Management' and contains a table with the following data:

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

At the bottom of the page, there are 'Commit' and 'Cancel' buttons, with the 'Commit' button highlighted in red. A red asterisk and the text '* Input Required' are visible at the bottom left of the form area.

8.2. Configure Locations

Locations are used to identify logical and/or physical locations where SIP Entities reside, for purposes of bandwidth management or location-based routing.

Navigate to **Routing** → **Locations**, and click on the **New** button (not shown) to create a new SIP endpoint location.

General section

Enter the following values and use default values for remaining fields.

- Enter a descriptive Location name in the Name field.
- Enter a description in the **Notes** field if desired.

Location Pattern section

Click **Add** and enter the following values:

- Enter the IP address information for the IP address Pattern (e.g. “10.1.2.*”)
- Enter a description in the **Notes** field if desired.

Repeat steps in the Location Pattern section if the Location has multiple IP segments.

Modify the remaining values on the form, if necessary; otherwise, retain the default values.

Click on the **Commit** button. Repeat all the steps for each new Location. The following screen shows the Locations page used during the compliance test.

Routing > Home / Elements / Routing / Locations - Location Details

Location Details Commit

Call Admission Control has been set to ignore SDR. 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

2 Items | Refresh Filter:

<input type="checkbox"/>	IP Address Pattern	Notes
<input type="checkbox"/>	* 10.1.2.*	<input type="text"/>
<input type="checkbox"/>	* 10.1.1.*	<input type="text"/>

8.3. Configure SIP Entities

A SIP Entity must be added for Session Manager and for each network component that has a SIP trunk provisioned to Session Manager. During the compliance test, the following SIP Entities were configured:

- Session Manager itself.
- Avaya Aura Messaging
- BCM

Navigate to **Routing** → **SIP Entities**, and click on the **New** button (not shown) to create a new SIP entity. Provide the following information:

General section

Enter the following values and use default values for remaining fields.

- Enter a descriptive Location name in the Name field.
- Enter IP address for signaling interface on each BCM, Avaya Aura Messaging.
- From the **Type** drop down menu select a type that best matches the SIP Entity.
 - For Session Manager, select Session Manager
 - For Messaging, select Modular Messaging
 - For BCM, select Others
- Enter a description in the **Notes** field if desired.
- Select the appropriate time zone.
- Accept the other default values.

Click on the **Commit** button to save each SIP entity. The following screens show the SIP Entity page used during the compliance test.

The screenshot shows the 'SIP Entity Details' configuration page. The left sidebar contains a navigation menu with 'SIP Entities' selected. The main content area is titled 'SIP Entity Details' and 'General'. The form fields are: Name: BCM450_34; FQDN or IP Address: 135.10.; Type: Other; Notes: (empty); Adaptation: (empty); Location: Belleville; Time Zone: America/New_York; Override Port & Transport with DNS SRV: (unchecked); SIP Timer B/F (in seconds): 4; Credential name: (empty); Call Detail Recording: none; SIP Link Monitoring: Use Session Manager Configuration. There are 'Commit' and 'Cancel' buttons at the top right.

Repeat all the steps for each new entity

8.4. Configure Entity Links

Entity Links define the connections between the SIP Entities and Session Manager. In the compliance test, the following entity links are defined from Session Manager.

- Session Manager ⇔ BCM
- Session Manager ⇔ Avaya Aura Messaging

Navigate to **Routing** → **Entity Links**, and click on the **New** button (not shown) to create a new entity link. Provide the following information:

- Enter a descriptive name in the **Name** field.
- In the **SIP Entity 1** drop down menu, select the Session Manager SIP Entity created in **Section 0**.
- In the **Protocol** drop down menu, select the protocol to be used (e.g. “UDP” or “TCP”).
- In the **Port** field, enter the port to be used (e.g. “5060”).
- In the **SIP Entity 2** drop down menu, select an entity.
- In the **Port** field, enter the port to be used (e.g. “5060”).
- Check the **Trusted** box.
- Enter a description in the **Notes** field if desired.

Click on the **Commit** button to save each Entity Link definition. The following screen shows an Entity Links page (between Session Manager and Messaging) used during the compliance test.

Name	SIP Entity 1	Protocol	Port	SIP Entity 2	Port	Trusted	Notes
* DevASM_DevAAM_S	* DevASM	TCP	* 5060	* DevAAM_SM	* 5060	<input checked="" type="checkbox"/>	

Repeat the steps to define Entity Links between Session Manager and Avaya BCM.

8.5. Time Ranges

The Time Ranges form allows admission control criteria to be specified for Routing Policies. In the reference configuration, no restrictions were used.

To add a Time Range, navigate to **Routing** → **Time Ranges**, and click on the **New** button (not shown). Provide the following information:

- Enter a descriptive Location name in the **Name** field (e.g. “24/7”).
- Check each day of the week.
- In the **Start Time** field, enter “00:00”.
- In the **End Time** field, enter “23:59”.
- Enter a description in the **Notes** field if desired.

Click the **Commit** button. The following screen shows the Time Range page used during the compliance test.

Time Ranges

1 Item | Refresh Filter: Enable

<input type="checkbox"/>	Name	Mo	Tu	We	Th	Fr	Sa	Su	Start Time	End Time	Notes
<input type="checkbox"/>	24/7	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	00:00	23:59	Time Range 24/7

Select : All, None

8.6. Configure Routing Policy

Routing Policies associates destination SIP Entities with Time of Day admission control parameters and Dial Patterns. In the reference configuration, Routing Policies are defined for: Business Communication Manager.

To add a Routing Policy, navigate to **Routing** → **Routing Policy**, and click on the **New** button (not shown) on the right. Provide the following information:

General section

- Enter a descriptive name in the **Name** field.
- Enter a description in the **Notes** field if desired.

SIP Entity as Destination section

- Click the **Select** button.
- Select the SIP Entity that will be the destination for this call (not shown).
- Click the **Select** button and return to the Routing Policy Details form.

Time of Day section

- Leave default values.

Click **Commit** to save Routing Policy definition. The following screen shows the Routing Policy used for the compliance test.

The screenshot displays the 'Routing Policy Details' configuration page. The left sidebar shows a navigation menu with 'Routing Policies' selected. The main content area is divided into sections: 'General' and 'SIP Entity as Destination'. In the 'General' section, the 'Name' field is set to 'RouteToBCM450', the 'Disabled' checkbox is unchecked, and the 'Notes' field contains 'RouteToBCM450'. In the 'SIP Entity as Destination' section, there is a 'Select' button. Below this, a table lists available SIP entities:

Name	FQDN or IP Address	Type	Notes
BCM450_34	135.10. . .	Other	

Repeat the steps to define routing policies to others Entities.

8.7. Dial Patterns

Dial Patterns define digit strings to be matched for inbound and outbound calls. In addition, the domain in the request URI is also examined. In the compliance test, the following dial patterns are defined from Session Manager.

- 222xx – SIP endpoints in BCM
- 39990 – Avaya Aura Messaging Pilot Number.

To add a Dial Pattern, select **Routing → Dial Patterns**, and click on the **New** button (not shown) on the right. During the compliance test, 5 digit dial plan was utilized. Provide the following information:

General section

- Enter a unique pattern in the **Pattern** field (e.g. “399”).
- In the **Min** field enter the minimum number of digits (e.g. “5”).
- In the **Max** field enter the maximum number of digits (e.g. “5”).
- In the **SIP Domain** field drop down menu select the domain that will be contained in the Request URI *received* by Session Manager from BCM and AAM.
- Enter a description in the **Notes** field if desired.

Originating Locations and Routing Policies section

- Click on the **Add** button and a window will open (not shown).
- Click on the boxes for the appropriate Originating Locations and Routing Policies that pertain to this Dial Pattern.
 - Location All.
 - Routing Policies **SM_to_AAM**.
 - Click on the **Select** button and return to the Dial Pattern window.

Click the **Commit** button to save the new definition. The following screen shows the dial pattern used for Messaging during the compliance test. Repeat the same for Avaya BCM with Pattern: 222.

The screenshot displays the 'Dial Pattern Details' configuration page. The 'General' section includes the following fields:

- Pattern:** 3999
- Min:** 5
- Max:** 5
- Emergency Call:**
- SIP Domain:** bvwddev.com
- Notes:** Dial pattern to call AAM

The 'Originating Locations and Routing Policies' section shows a table with one item:

<input type="checkbox"/>	Originating Location Name	Originating Location Notes	Routing Policy Name	Rank	Routing Policy Disabled	Routing Policy Destination	Routing Policy Notes
<input type="checkbox"/>	Belleville,Ont,Ca	Belleville DevConnect lab	SM_TO_AAM	0	<input type="checkbox"/>	DevAAM_SM	Route from SM to AAM

9. Configure Avaya Aura® Messaging

Messaging was configured for SIP communication with Session Manager and also to add Avaya BCM, Communication Manager Subscribers. The procedures include the following areas:

- Administer Sites
- Administer Telephony Integration
- Administer Dial Rules
- Administer Class of Service to enable Message Waiting
- Administer Subscribers

See references in **Section 12** for standard installation and configuration information. General knowledge of the configuration tools and interfaces is assumed.

9.1. Administer Sites

A Messaging access number and a Messaging Auto Attendant number needs to be defined. Log into the Avaya Aura Messaging System Management Interface (SMI) and go to **Administration** → **Messaging** → **Messaging System (Storage)** → **Sites**. In the right panel fill in the following:

Under **Main Properties**:

- **Messaging access number (internal)** Enter a Messaging Pilot number

The screenshot shows the Avaya Aura Messaging System Management Interface (SMI) with the following configuration details:

- Navigation:** Administration / Messaging / Messaging System (Storage) / Sites
- Sites Section:** Site: Default (dropdown), Add New... (button), Delete (button)
- Main Properties Section:**
 - Name: Default
 - ID: 1
 - Messaging access number (external): 39991
 - Messaging access number (internal): 39990

Scroll down to the **Site Internal Dial Plan** section.

Under **Site Internal Dial Plan**:

- **Short Extension Length** Enter the number of digits in extensions
- **Short Mailbox Length** Enter the number of digits in mailbox numbers

The screenshot shows the Avaya Administration web interface. At the top left is the Avaya logo. Below it is a navigation bar with 'Help' and 'Log Off' on the left, and 'Administration' in the center. Underneath is a breadcrumb trail 'Administration / Messaging'. A left-hand navigation menu lists various system management options. The main content area is titled 'Administration / Messaging' and contains several configuration fields. The 'Site Internal Dial Plan' section is highlighted with a red box. This section includes a description: 'Describe the internal dial plan applicable to this site.' Below this are two input fields: 'Short extension length:' with a value of '5' and 'Short mailbox length:' with a value of '5'. Other fields include 'Subscriber number length (within this site's national destination code):', 'Outside line prefix:', 'Extension style for telephony integration:' (set to 'Short'), 'Site prefix:', and 'National mailbox number convention:' (set to 'Choose One').

Scroll down to the **Auto Attendant** section.

Under **Auto Attendant**:

- **Auto Attendant** Select “Enabled”
- **Auto Attendant pilot number** Enter an Auto Attendant number
- **Keypad entry** Select “ENHANCED”
- **Speech recognition** Select “Enabled”

Click **Save** to save changes.

The screenshot shows the Avaya Administration console interface. The top navigation bar includes the Avaya logo, 'Help', 'Log Off', and 'Administration'. Below this is a sub-header 'Administration / Messaging'. The left sidebar contains a tree view of system components, including 'Messaging System (Storage)', 'Reports (Storage)', and 'Server Information'. The main content area displays configuration options for the 'Auto Attendant' section, which is highlighted with a red box. The settings are as follows:

- Operator (live attendant) extension:** [Empty text box]
- General mailbox:** [Empty text box]
- Auto Attendant:** enabled disabled
- Auto Attendant pilot number:** [39995]
- Additional sites included in the directory:** [None]
- Keypad entry:** [ENHANCED]
- Speech recognition:** enabled disabled

At the bottom right of the configuration area, there are two buttons: 'Save' (highlighted with a red box) and 'Cancel'.

9.2. Administer Telephony Integration

A SIP trunk needs to be configured from Messaging to Session Manager. Log into the Messaging System Management Interface (SMI) and go to **Administration** → **Messaging** → **Telephony Settings (Application)** → **Telephony Integration**. In the right panel fill in the following:

Under **Basic Configuration**:

- **Extension Length:** Enter the length of extensions
- **Switch Integration Type:** SIP

Under **SIP Specific Configuration**:

- **Transport Method:** “TCP”
- **Connection 1:** Enter the Session Manager signaling IP address and TCP port number
- **Messaging Address:** Enter the Messaging IP address and TCP port number
- **SIP Domain:** Enter the Messaging and Session Manager domain names

Click **Save** to save changes.

Telephony Integration

The Telephony Integration page is used for administration of the switch link parameters of the messaging system.

BASIC CONFIGURATION

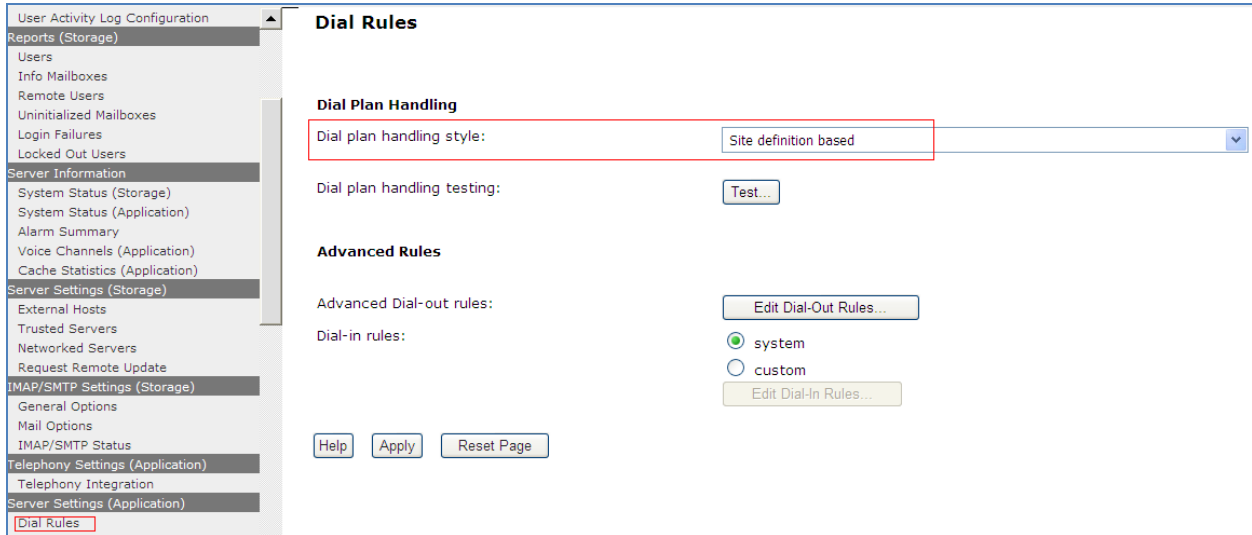
Switch Number	1
Extension Length	5
Switch Integration Type	SIP
IP Address Version	IPv4

SIP SPECIFIC CONFIGURATION

Transport Method	TCP
Far-end Connections	1
Connection 1	IP 135.10. . Port 5060
Messaging Address	IP 10.32. . Port 5060
SIP Domain	Messaging: bvwdev.com Switch: bvwdev.com
Messaging Ports	Call Answer Ports: 100 Maximum: 100 Transfer Ports: 20
Switch Trunks	Total: 120 Maximum: 120

9.3. Configure Dial Rules

Navigate to Administration *Messaging* → *Server Settings (Application)* → *Dial Rules* to configure the dial rules. Set the **Dial plan handling style**: Site definition based, as shown below.



Next select the **Edit Dial-Out Rules** button to verify the appropriate parameters for outbound dialing from Avaya Aura® Messaging were set above. These dial rules help Avaya Aura® Messaging send the correct number and combination of digits when originating a call to Communication Manager, whether the call is destined for another extension or ultimately expected to be routed to the PSTN.

For the sample configuration, 7-digit extensions were used on Avaya Communication Manager so any time Avaya Aura Messaging originates a call to an extension it should send the 7-digit number and not attempt to insert or delete any digits.

Scroll down to the section titled **Dial-out Test Numbers**. Enter in a number in the appropriate section and select the **Test** button to see how Avaya Aura® Messaging would dial that number. As shown below the number **7785002** is treated as an internal number and is dialed intact, whereas the test number **408-555-7086** is treated as a long-distance national number which requires a **9** prefixed as an access code.

Dial-Out Test Numbers

```
# Examples below.  
# Add more phone numbers to test for your specific configuration.  
  
# Extension (example):  
2001  
7785002  
(212) 555-7086  
  
# Local number (example):  
555-7086  
333-3030  
  
# Long-distance number (example):  
(408) 555-7086
```

Test

Save

Dial-Out Test Results

Input Phone Number	→	Call Type	Output Phone Number
2001	→	INTERNAL	2001
7785002	→	INTERNAL	7785002
555-7086	→	INTERNAL	5557086
333-3030	→	INTERNAL	3333030
(408) 555-7086	→	LONGDISTANCE	914085557086

9.4. Configure Class of Service

Verify Messaging Waiting is enabled for all subscribers.

Use *Administration* → *Messaging* → *Messaging System (Storage)* → *Class of Service*.

Select **Standard** from the **Class of Service** drop-down menu.

Under **General** section, enter the following value and use default values for remaining fields.

- **Set Message Waiting Indicator (MWI) on user's desk phone:** checked.
- **Dial-out privilege:** Local.

Under **Greetings** section, enter for **Two Greetings (different greetings for busy and noanswer)** field to allow subscribers to record different personal greetings for busy and no-answer scenarios.

Click **Save** (not shown) to save changes.

The following screen shows the settings defined for the “Standard” **Class of Service** in the test configuration.

Class of Service

Class of Service:

General

Name:

ID:

Required seat license:

Telephone User Interface:

User can send to system distribution lists (ELAs)

Fax support:

Dial-out privilege:

User can use Reach Me

Allow voice recognition for addressing (user can select recipients by saying their name)

IMAP4/POP3 access: (for Avaya Message Store users)

Set Message Waiting Indicator (MWI) on user's desk phone

Enable password aging

User can send system broadcast messages

9.5. Administer Subscribers

Log into the Messaging System Management Interface (SMI) and go to *Administration* → *Messaging*. In the left panel, under *Messaging System (Storage)* select *User Management*. In the right panel fill in the following:

- **First Name** Enter first name
- **Last Name** Enter last name
- **Display Name** Enter display name
- **ASCII name** Enter the ASCII name
- **Site** Enter site defined in **Section 9.1**
- **Mailbox Number** Enter desired mailbox number i.e. “22235”
- **Internal identifier** Enter the name for internal use
- **Numeric address** Enter the mailbox number
- **Extension** Enter desired extension number i.e. “22235”

The screenshot shows the 'User Management > Properties for BCM 22235' configuration page. The left sidebar contains a navigation menu with categories like 'Messaging System (Storage)', 'Reports (Storage)', 'Server Information', 'Server Settings (Storage)', 'IMAP/SMTP Settings (Storage)', and 'Telephony Settings (Application)'. The 'User Management' option is selected. The main content area is titled 'User Management > Properties for BCM 22235' and contains the following fields:

- User Properties:** First name: BCM; Last name: 22235; Display name: BCM 22235; ASCII name: BCM 22235.
- Site:** Default (dropdown).
- Mailbox number:** 22235; Internal identifier: BCM.22235 (with @sp-aamess1.avaya.com); Numeric address: 22235.
- Extension:** 22235; Include in Auto Attendant directory.
- Class of Service:** Standard (dropdown); Pronounceable name: BCM 22235.
- MWI enabled:** Yes (dropdown).

Scroll down on the page to Class of Service.

- **Class of Service** Select a Class of Service
- **Pronounceable Name** Enter a pronounceable name to be used when dialing the extension using voice commands
- **MWI Enabled** Select “Yes” to enable the MWI light on phones
- **New Password/Confirm Password** Enter desired extension password
- **Next logon password change** Select the **Checkbox**

Click **Save** to save changes.

AVAYA

Help Log Off Administration

Administration / Messaging

Messaging System (Storage)

- User Management
- Class of Service
- Sites
- Topology
- Storage Destinations
- System Policies
- Enhanced List Management
- System Mailboxes
- System Ports and Access
- User Activity Log Configuration

Reports (Storage)

- Users
- Info Mailboxes
- Remote Users
- Uninitialized Mailboxes
- Login Failures
- Locked Out Users

Server Information

- System Status (Storage)
- System Status (Application)
- Alarm Summary
- Voice Channels (Application)
- Cache Statistics (Application)

Server Settings (Storage)

- External Hosts
- Trusted Servers
- Networked Servers

Class of Service: Standard

Pronounceable name: BCM 22235

MWI enabled: Yes

Miscellaneous 1:

Miscellaneous 2:

New password: ●●●●●●

Confirm password: ●●●●●●

User must change voice messaging password at next logon

Voice messaging password expired

Locked out from voice messaging

Save Delete


10. Verification Steps

10.1. Verify Avaya Aura® Session Manager Operational Status

10.1.1. Verify Avaya Aura® Session Manager is Operational

Navigate to *Elements* → *Session Manager* → *Dashboard* (not shown) to verify the overall system status for Session Manager.

Specifically, verify the status of the following fields as shown below:

- **Tests Pass:** 
- **Security Module:** **Up** 
- **Service State:** **Accept New Service** 

Home / Elements / Session Manager - Session Manager Help ?

Session Manager Dashboard

This page provides the overall status and health summary of each administered Session Manager.

Session Manager Instances

Service State Shutdown System As of 1:51 PM

1 Item | Refresh | Show ALL Filter: Enable

<input type="checkbox"/>	Session Manager	Type	Alarms	Tests Pass	Security Module	Service State	Entity Monitoring	Active Call Count	Registrations	Version
<input type="checkbox"/>	DevASM	Core	24069/2177/2118	✓	Up	Accept New Service	12/44	0	9	6.1.1.0.611023

Session Manager Instances status.


Navigate to *Elements* → *Session Manager* → *System Status* → *Security Module Status* (not shown) to view more detailed status information on the status of Security Module for the specific Session Manager. Verify the **Status** column displays “Up” as shown below.

Security Module Status

This page allows you to view the status of each Session Manager's Security Module and to perform certain actions.

Reset Synchronize Certificate Management Connection Status

1 Item | Refresh | Show ALL Filter: Enable

	Details	Session Manager	Type	Status	Connections	IP Address	VLAN	Default Gateway	NIC Bonding	Entity Links (expected / actual)	Certificate Used
	Show	DevASM	SM	Up	46	135.10. . . /26	---	135.10. . .	Disabled	57/57	SIP CA

10.1.2. Verify SIP Entity Link Status

Navigate to *Elements* → *Session Manager* → *System Status* → *SIP Entity Monitoring* (not shown) to view more detailed status information for one of the SIP Entity Links.

Select the SIP Entity for BCM from the **All Monitored SIP Entities** table (not shown) to open the *SIP Entity, Entity Link Connection Status* page.

In the All Entity Links to SIP Entity: BCM450_34table, verify the **Conn. Status** for the link is “Up” as shown below.

SIP Entity, Entity Link Connection Status

This page displays detailed connection status for all entity links from all Session Manager instances to a single SIP entity.

All Entity Links to SIP Entity: BCM450_34

Summary View

1 Item | Refresh Filter: Enable

Details	Session Manager Name	SIP Entity Resolved IP	Port	Proto.	Conn. Status	Reason Code	Link Status
▶ Show	DevASM	135.10. .	5060	UDP	Up	200 OK	Up

As described above the Entity link connection status can also be viewed for the AAM. Verify that the **Conn. Status** is also **Up** (not shown).

10.2. Verify Avaya Aura® Messaging

10.2.1. Verify no answer call

Make a call from a Avaya BCM endpoint to another Avaya BCM endpoint and verify that the call covers to Messaging upon no answer. Leave a voice message. Verify that the MWI light of the called phone turns on. From the receiving Avaya BCM endpoint, dial the Messaging access number to retrieve the message. Verify that the Messaging system identifies the Avaya BCM endpoint and that the voice message can be retrieved. Verify that the MWI light turns off.

Log into the Messaging System Management Interface (SMI) and go to **Administration** → **Messaging**. In the left panel, under **Logs** select **User Activity**. In the right panel fill in the following:

Under **User Activity Log**:

- **Mailbox Number** Enter the BCM extension that received the voicemail.
- **Start Date** Enter an appropriate start date and time.
- **End Date** Enter an appropriate end date and time.

Click **Display** button and verify that a listing of the detailed activities is displayed into the bottom portion of the right hand pane. Verify that there is an entry showing the message left by a subscriber (in this case 22234). Also verify that there is an entry showing the message being retrieved.

User Activity Log

The User Activity Log page allows the selection of the Mailbox and Start and End times to display the activity log.

Mailbox Number	<input type="text" value="22234"/>		
Start Date	April <input type="text" value="30"/> , 2012	Time	09 : 13
End Date	May <input type="text" value="1"/> , 2012	Time	09 : 13

Name: Bon, Ba **Mailbox Number:** 22234

<u>Date</u>	<u>Time</u>	<u>Activity</u>	<u>Description</u>
05/01/2012	8:50	received	VM message from 22232 new=1(v=1 f=0 e=0 dsn=0) un=0 o=0 d=0 x=0
05/01/2012	8:50	inbox-sel	id=6f778 port=55143 IP=127.0.0.1 msgs=1
05/01/2012	8:50	inbox-dsel	id=6f778 port=55143 IP=127.0.0.1 msgs=1
05/01/2012	8:50	inbox-sel	id=6f778 port=55143 IP=127.0.0.1 msgs=1
05/01/2012	8:50	inbox-dsel	id=6f778 port=55143 IP=127.0.0.1 msgs=1
05/01/2012	8:50	status	changed from new to old for message received 5/01/12 at 8:50

10.2.2. Test calls can be made from Avaya Aura Messaging to phones that are configured with mailboxes.

To perform this test, select *Administration* → *Messaging*. In the left panel, under *Diagnostics* select *Diagnostics (Application)*. In the right panel fill in the following:

- **Select the test(s) to run:** Select **Call-out** from the drop down menu.
- **Telephone number:** Enter the number to call.

Click on **Run Tests** to start the test. The phone will ring and when answered a test message is played. The **Results** section of the page will update indicating that the call was ok as shown below.

Diagnostics (Application)

Selection & Configuration

Select the test(s) to run:

This calls out to the specified extension. When the phone is picked up, a test greeting should be heard.

Configuration of Call Out Test

Telephone number:

Port number (optional):

Results

```

Test: Call-out Time: 4:09:01 PM
Usage: testCALL extensionNumber [portNumber]
Checking Call-out ... calling 22237 ... [ OK ]
Line:100 (irapi100) Got dial tone Dialing is done Connected Near End disconnected CP=NEAR_END_DISCONNECT

```

10.2.3. Message Waiting Indicator (MWI) light on phones.

To perform this test, select *Administration* → *Messaging*. In the left panel, under *Diagnostics* select *Diagnostics (Application)*. In the right panel fill in the following:

- **Select the test(s) to run:** Select **MWI** from the drop down menu.
- **Extension number:** Enter the number of the phone to test.

Click on **Run Tests** to start the test. The phones MWI light will turn on and off. The **Results** section of the page will update with information about the test as shown below.

Diagnostics (Application)

Selection & Configuration

Select the test(s) to run:

MWI

This tests the Message Waiting Indicator, MWI. It will turn on and off (few times) the MWI light for the extension.

Configuration of MWI Test

Extension number:

22232

MWI port number (optional):

Run Tests

Reset Page

Results

Test: MWI

Time: 9:20:47 AM

Checking MWI ...

PLEASE NOTE:

- An [OK] result does not necessarily confirm that the Message Waiting Indicator (MWI) was successfully set. Always verify on the actual phone that the MWI was turned on and off twice.
- Some PBXs require that the MWI is switched off from the same line that was used to switch it on. On these PBXs, perform the test on a phone for which MWI is not switched on.

```
Turn MWI 1 for 22232 (1 second) ...[ OK ]
Turn MWI 0 for 22232 (1 second) ...[ OK ]
Turn MWI 1 for 22232 (1 second) ...[ OK ]
Turn MWI 0 for 22232 (1 second) ...[ OK ]
```

10.2.4. Other call scenarios

- Call to Forward All(forward to messaging access number) endpoint
 - Call to Busy endpoint (messaging access number is set if this endpoint busy)
- All the call is forwarded to pilot number.

11. Conclusion

Interoperability testing of Avaya Aura® Messaging 6.1 Single Server as a voice mail solution for Avaya Business Communication Manager with SIP Trunking through Avaya Aura® Session Manager R6.2 was successful. See section 2.2 for detail of test result and observation.

12. Additional References

This section provides references to the product documentation relevant to this Application Note.

Documentation for Avaya products may be found at <http://support.avaya.com>.

Avaya Aura® Session Manager

- 1) Avaya Aura® Session Manager Overview, Doc ID 03-603323
- 2) Installing and Configuring Avaya Aura® Session Manager
- 3) Avaya Aura® Session Manager Case Studies
- 4) Maintaining and Troubleshooting Avaya Aura® Session Manager, Doc ID 03-603325
- 5) Administering Avaya Aura® Session Manager, Doc ID -3-603324

Avaya Aura® Messaging

- 6) Administering Avaya Aura® Messaging 6.1 CID: 151610 December 2011
- 7) Using Avaya Aura® Messaging 6.1 December 2011
- 8) Implementing Avaya Aura® Messaging 6.1 CID: 150976 October 2011

Avaya Business Communication Manager 450

- 9) Avaya Business Communication Manager 6.0 Planning and Engineering NN40170-200
- 10) Avaya Business Communication Manager 6.0 Configuration - Telephony NN40170-502

Avaya Communication Manager

- 11) Avaya Aura® Communication Manager Screen Reference Release 6.2 03-602878 Issue 3.0 February 2012

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