



Avaya Solution & Interoperability Test Lab

**Application Notes for VTech 1-Line Hospitality S2x10 SIP
Phones Version 39.3.68.07 with Avaya Aura®
Communication Manager 6.3 and Avaya Aura® Session
Manager 6.3 – Issue 1.0**

Abstract

These Application Notes describe a compliance-tested configuration consisting of Avaya Aura® Session Manager, Avaya Aura® Communication Manager and VTech Hospitality SIP Phones.

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

1. Introduction

The purpose of the document is to summarize configurations, test notes, and issues if any during the compliance test between VTech Hospitality SIP phones and Avaya Aura® Communication Manager and Avaya Aura® Session Manager. The VTech SIP Hospitality Phones are available in a single or dual-line phone, and in two styles as well as offering a corded and cordless option. There is one model VTech 1-Line Corded S2210 presented for the series S2x10 used for the compliance test.

2. General Test Approach and Test Results

The compliance test focused on the interoperability between the VTech Hospitality SIP Phones, Avaya Aura® Session Manager and Avaya Aura® Communication Manager including the ability to make and receive calls from PSTN endpoints and Avaya SIP, H.323, and Digital phones.

As the purpose of these phones is for hotel guest rooms, certain functionality considered to be standard on Avaya endpoints is not supported and therefore was not tested. For example, the VTech phones do not support multiple line appearances. More details on these limitations are described in the Test Results in **Section 2.2**.

DevConnect Compliance Testing is conducted jointly by Avaya and DevConnect members. The jointly-defined test plan focuses on exercising APIs and/or standards-based interfaces pertinent to the interoperability of the tested products and their functionalities. DevConnect Compliance Testing is not intended to substitute full product performance or feature testing performed by DevConnect members, nor is it to be construed as an endorsement by Avaya of the suitability or completeness of a DevConnect member's solution.

2.1. Interoperability Compliance Testing

VTech SIP phones register with Session Manager and thus are able to use the Communication Manager application sequencing in a similar manner to Avaya SIP endpoints. The following were tested in the compliance test:

- Registration of VTech phone to Session Manager.
- Basic call features: Answer, Hold/Resume, Mute/Un-mute, Drop, Decline, Message Waiting Indicator, DTMF, Call Park, Call Pickup, Call Waiting, Call Forward, Transfer, and Conference.
- Codec negotiation, Media Shuffling, Multiple Device Access, and Session Refresh Interval.
- Hospitality features: Automatic Wakeup Call and Housekeeping status.
- Serviceability test, which consisted of the VTech SIP phones re-registering with Session Manager following loss of network connections, and server reboots.

2.2. Test Results

The objectives described in **Section 2.1** were verified. The following observations were made during the testing:

- VTech 1-Line SIP phone S2210 supports the call waiting feature, use the Flash button on the phone to receive second incoming all and switch between two calls.
- 3-way call conference can be initiated by using the Flash button to invite the second party to the conference.

2.3. Support

Information, Documentation and Technical support for VTech Hospitality SIP Phones can be obtained at:

- Phone: +1 (888) 907-2007
- <http://vtechhotelphones.com>

3. Reference Configuration

Figure 1 illustrates the test configuration diagram showing the integration of the VTech SIP phone to the Avaya Solution. VTech Hospitality SIP phone registers to Session Manager via SIP and uses the telephony features from Communication Manager. The system test also had a T1 link from Communication Manager to real PSTN for test cases related to PSTN calls. A SIP trunk was setup to an Avaya Communication Server 1000E (CS1000E) to test that the VTech phone made/received off-net calls. Avaya Aura® Messaging was used as the voice mail system.

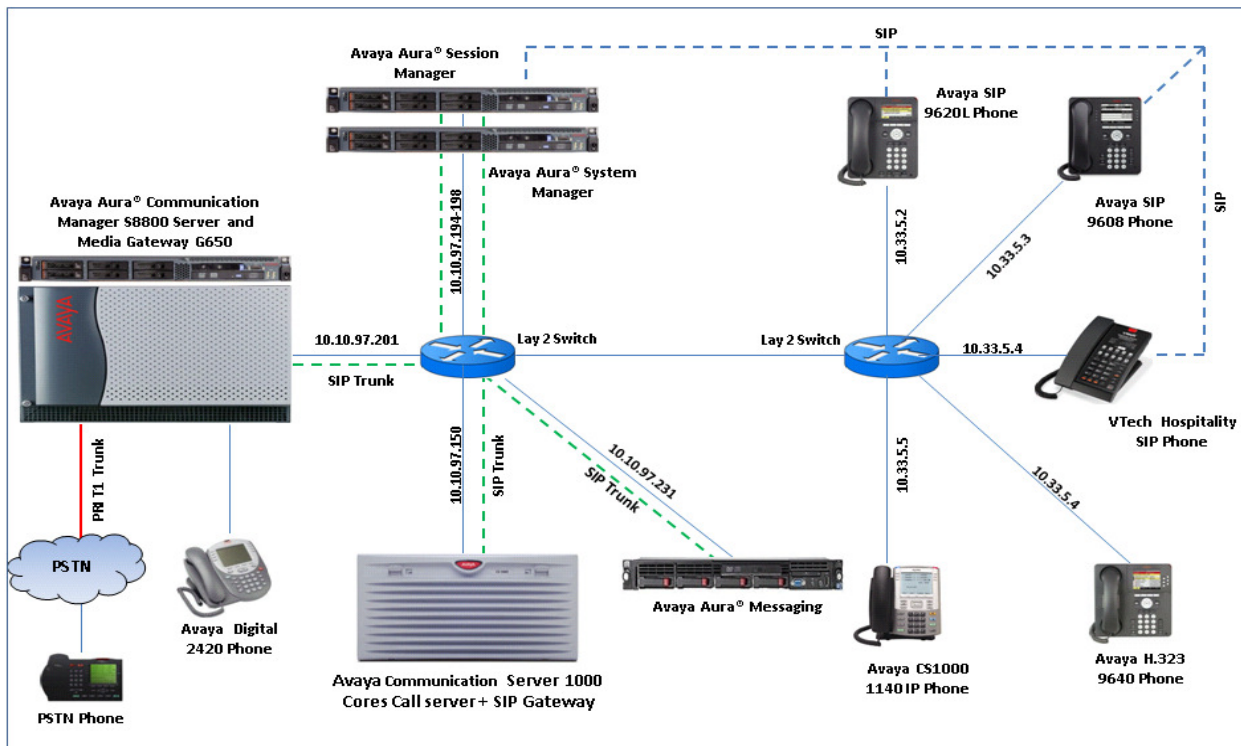


Figure 1: Test configuration diagram

4. Equipment and Software Validated

The following equipment and software were used for the compliance test provided:

Equipment/Software	Release/Version
Avaya Aura® Communication Manager running on an Avaya S8800 Server	R6.3 - FP2 Build R016x.03.0.124.0 Patch 20553
Avaya Aura® System Manager running on an Avaya S8800 Server	R6.3.0 – FP2 Build 6.3.0.8.5682
Avaya Aura® Session Manager running on an Avaya S8800 Server	R6.3.0 – FP2 Build 6.3.2.632001
Avaya Aura® Messaging running on an Avaya S8800 Server	R6.1 SP2 Build R016x.00.1.510.1
Avaya Media Gateway G650 <ul style="list-style-type: none"> • IP Server interface TN2312BP • IP Media Processor TN2302AP 	HW06 - FW043 HW20 - FW117
Avaya Communication Server 1000E running on an Avaya CPPM card	R7.5
Avaya 9611 IP Deskphone (with Avaya one-X® H.323 firmware)	6.2313
Avaya 9620 IP Deskphone (with Avaya one-X® SIP firmware)	2.6.9.1
Avaya 9608 IP Deskphone (with Avaya one-X® H.323 firmware)	6.2313
Avaya Digital Deskphone 2420	N/A
VTech SIP Phone S2420	SIP_39.3.68.07
VTech SIP Phone S2220	SIP_39.3.68.07
VTech SIP Phone S2210	SIP_39.3.68.07

5. Configure Avaya Aura® Communication Manager

It is assumed that Communication Manager is already installed and configured. This section describes the necessary configurations for VTech SIP phone to work with Communication Manager. The steps include setting up an IP codec set, an IP network region, IP node name, a signaling group, a trunk group, and a SIP station. Before a trunk can be configured, it is necessary to verify if there is enough capacity to setup an additional trunk. The highlights in the following screens indicate the values used during the compliance test. Default values may be used for all other fields.

These steps are performed from the Communication Manager System Access Terminal (SAT) interface. VTech SIP phone and other SIP telephones are configured as off-PBX telephones in Communication Manager.

5.1. Capacity Verification

Enter the **display system-parameters customer-options** command. Verify that there are sufficient **Maximum Off-PBX Telephones – OPS** licenses. If not, contact an authorized Avaya account representative to obtain additional licenses.

```
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: 65000 214
                                Maximum Stations: 41000 38
                                Maximum XMOBILE Stations: 41000 0
Maximum Off-PBX Telephones - EC500: 41000 4
Maximum Off-PBX Telephones - OPS: 41000 24
Maximum Off-PBX Telephones - PBFMC: 41000 0
Maximum Off-PBX Telephones - PVFMC: 41000 0
Maximum Off-PBX Telephones - SCCAN: 0 0
                                Maximum Survivable Processors: 313 1
```

On **Page 2** of the form, verify that the **Maximum Administered SIP Trunks** supported by the system is sufficient for the number of SIP trunks needed. If not, contact an authorized Avaya account representative to obtain additional licenses.

display system-parameters customer-options		Page	2 of 11
OPTIONAL FEATURES			
IP PORT CAPACITIES		USED	
	Maximum Administered H.323 Trunks:	12000	0
	Maximum Concurrently Registered IP Stations:	18000	6
	Maximum Administered Remote Office Trunks:	12000	0
	Maximum Concurrently Registered Remote Office Stations:	18000	0
	Maximum Concurrently Registered IP eCons:	414	0
	Max Concur Registered Unauthenticated H.323 Stations:	100	0
	Maximum Video Capable Stations:	41000	0
	Maximum Video Capable IP Softphones:	18000	1
	Maximum Administered SIP Trunks:	24000	130
	Maximum Administered Ad-hoc Video Conferencing Ports:	24000	0
	Maximum Number of DS1 Boards with Echo Cancellation:	522	0
	Maximum TN2501 VAL Boards:	128	1
	Maximum Media Gateway VAL Sources:	250	0
	Maximum TN2602 Boards with 80 VoIP Channels:	128	0
	Maximum TN2602 Boards with 320 VoIP Channels:	128	1
	Maximum Number of Expanded Meet-me Conference Ports:	300	0

5.2. Administer IP Codec Set

This section describes the steps for administering a codec set in Communication Manager. This codec set is used in the IP network region for communications between Communication Manager and Session Manager.

Enter the **change ip-codec-set <c>** command, where **c** is a number between **1** and **7**, inclusive. IP codec sets are used in **Section 5.3** for configuring IP network region to specify which codec sets may be used within and between network regions.

change ip-codec-set 1		Page	1 of 2
		IP Codec Set	
Codec Set: 1			
Audio	Silence	Frames	Packet
Codec	Suppression	Per Pkt	Size (ms)
1: G.711MU	n	2	20
2: G.729	n	2	20
3: G.722-64K		2	20

5.3. Administer IP Network Region

This section describes the steps for administering an IP network region in Communication Manager for communication between Communication Manager and Session Manager. Enter the **change ip-network-region <n>** command, where **n** is a number between **1** and **250** inclusive, and configure the following:

- **Authoritative Domain** – Enter the appropriate name for the Authoritative Domain. During the compliance test, the authoritative domain is set to **bvwdev.com**. This should match the SIP Domain value on Session Manager, in **Section 6.1**.
- **Intra-region IP-IP Direct Audio** – Set to **yes** to allow direct IP-to-IP audio connectivity between endpoints registered to Communication Manager or Session Manager in the same IP network region. The default value for this field is **yes**.
- **Codec Set** – Set the codec set number as provisioned in **Section 5.2**.
- **Inter-region IP-IP Direct Audio** – Set to **yes** to allow direct IP-to-IP audio connectivity between endpoints registered to Communication Manager or Session Manager in different IP network regions. The default value for this field is **yes**.

```
change ip-network-region 1                               Page 1 of 20
                                                         IP NETWORK REGION
Region: 1
Location: 1      Authoritative Domain: bvwdev.com
Name:           Stub Network Region: n
MEDIA PARAMETERS      Intra-region IP-IP Direct Audio: yes
                    Codec Set: 1      Inter-region IP-IP Direct Audio: yes
                    UDP Port Min: 2048      IP Audio Hairpinning? n
                    UDP Port Max: 3329
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
```

5.4. Administer IP Node Name

This section describes the steps for setting IP node name for Session Manager's SIP signaling interface in Communication Manager. Enter the **change node-names ip** command, and add a node name for Session Manager along with its IP address.

```
change node-names ip                                     Page 1 of 2
                                     IP NODE NAMES
Name                                IP Address
SM63                                10.10.97.198
GW                                  10.10.97.193
default                             0.0.0.0
procr                                10.10.97.201
```

5.5. Administer SIP Signaling IP

This section describes the steps for administering a signaling group in Communication Manager for signaling between Communication Manager and Session Manager. Enter the **add signaling-group <s>** command, where **s** is an available signaling group and configure the following:

- **Group Type** – Set to **sip**.
- **Transport Method** – Set to **tcp**.
- **Near-end Node Name** - Set to **procr** as displayed in **Section 5.4**.
- **Far-end Node Name** - Set to the **Session Manager** name configured in **Section 5.4**.
- **Far-end Network Region** - Set to the region configured in **Section 5.3**.
- **Far-end Domain** - Set to **bvwdev.com**. This should match the SIP Domain value in **Section 6.1**.

```
add signaling-group 1                                     Page 1 of 2
                                     SIGNALING GROUP
Group Number: 1                                         Group Type: sip
IMS Enabled? n                                         Transport Method: tcp
Q-SIP? n
IP Video? y                                           Priority Video? n           Enforce SIPS URI for SRTP? y
Peer Detection Enabled? y Peer Server: SM
Prepend '+' to Outgoing Calling/Alerting/Diverting/Connected Public Numbers? y
Remove '+' from Incoming Called/Calling/Alerting/Diverting/Connected Numbers? n

Near-end Node Name: procr                               Far-end Node Name: SM63
Near-end Listen Port: 5060                             Far-end Listen Port: 5060
Far-end Network Region: 1

Far-end Domain: bvwdev.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
30                                                    Alternate Route Timer(sec):
```

5.6. Administer SIP Trunk

This section describes the steps for administering a trunk between Communication Manager and Session Manager. Enter the **add trunk-group <t>** command, where **t** is an unallocated trunk group and configure the following:

- **Group Type** – Set the Group Type field to **sip**.
- **Group Name** – Enter a descriptive name.
- **TAC (Trunk Access Code)** – Set to any available trunk access code.
- **Service Type** – Enter **tie**.
- **Signaling Group** – Set to the Group Number field value configured in **Section 5.5**.
- **Number of Members** – Allowed value is between 0 and 255. Set to a value large enough to accommodate the number of SIP telephone extensions being used.

```
add trunk-group 1                                     Page 1 of 21
                                                    TRUNK GROUP
Group Number: 1                                     Group Type: sip          CDR Reports: n
  Group Name: SIP trunk to SM                       COR: 1                  TN: 1          TAC: #001
  Direction: two-way                               Outgoing Display? y
  Dial Access? n                                    Night Service:
  Queue Length: 0
  Service Type: tie                                Auth Code? n
                                                    Member Assignment Method: auto
                                                    Signaling Group: 1
                                                    Number of Members: 15
```

6. Configure Avaya Aura® Session Manager

This section provides the procedures for configuring Session Manager. All SIP call provisioning for Session Manager is performed through System Manager using a Web interface to make changes which are then downloaded to Session Manager.

Note: This section assumes that Session Manager and System Manager have been installed, and network connectivity exists between the two platforms.

The following steps describe the configuration of Session Manager.

- Administer SIP Domain
- Administer Locations
- Administer SIP Entities
- Administer Entity Links
- Administer Routing Policy
- Administer Dial Patterns
- Administer Manage Element
- Administer Applications
- Administer Application Sequence
- Administer User Management

6.1. Administer SIP Domain

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

AVAYA Avaya Aura® System Manager 6.3

Home / Log On

Log On

This system is restricted solely to authorized users for legitimate business purposes only. The actual or attempted unauthorized access, use, or modification of this system is strictly prohibited.

Unauthorized users are subject to company disciplinary procedures and or criminal and civil penalties under state, federal, or other applicable domestic and foreign laws.

The use of this system may be monitored and recorded for administrative and

User ID:

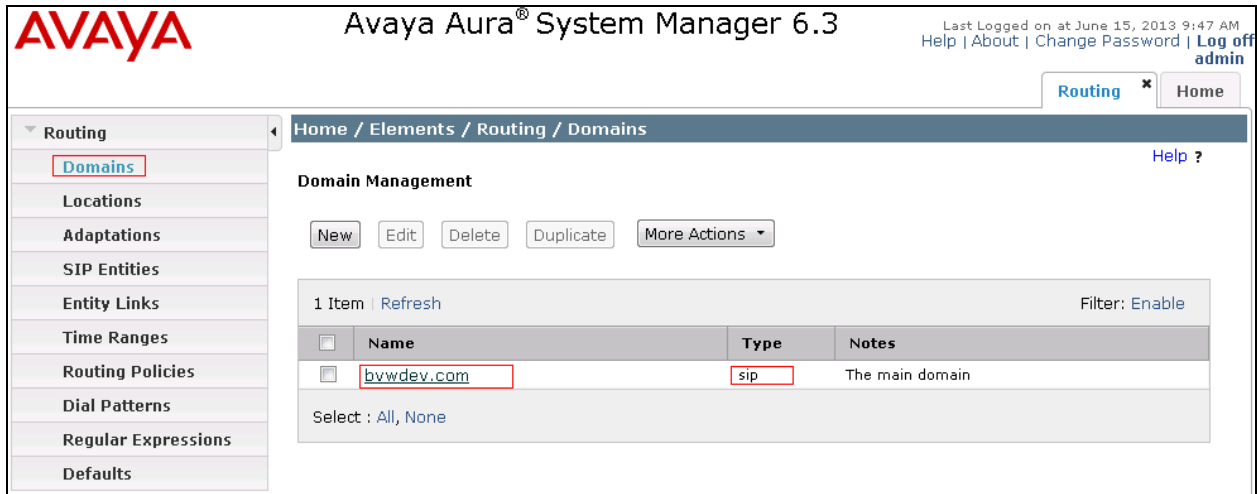
Password:

Supported Browsers: Internet Explorer 8.x, 9.x or 10.x or Firefox 15.0, 16.0 or 17.0.

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

- **Name** –Enter the Authoritative Domain name specified in **Section** Error! Reference source not found., which is **bvwddev.com**.
- **Type** – Select **SIP**.

Click **Commit** to save (not shown). The following screen shows the Domains page used during the compliance test.



The screenshot shows the Avaya Aura System Manager 6.3 interface. The top left features the AVAYA logo. The top right shows the user is logged in as 'admin' and last logged on at June 15, 2013 9:47 AM. The breadcrumb navigation is 'Home / Elements / Routing / Domains'. The left sidebar lists various configuration options, with 'Domains' selected. The main content area is titled 'Domain Management' and includes buttons for 'New', 'Edit', 'Delete', 'Duplicate', and 'More Actions'. Below these is a table with one item: 'bvwddev.com' of type 'sip' with the note 'The main domain'. The table has columns for 'Name', 'Type', and 'Notes'. A 'Filter: Enable' option is visible on the right. The bottom of the table shows 'Select : All, None'.

<input type="checkbox"/>	Name	Type	Notes
<input type="checkbox"/>	bvwddev.com	sip	The main domain

6.2. Administer Locations

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

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

General section

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

- Enter a descriptive Location in the **Name** field (e.g. **Subnet 10.10.97.0**).
- Enter a description in the **Notes** field if desired.

Location Pattern section

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

- The IP address information for the **IP address Pattern** (e.g. **10.10.97.0**).
- A description in the **Notes** field if desired.

Repeat these steps in the Location Pattern section if the Location has multiple IP segments. Modify the remaining values on the form, if necessary; otherwise, use all the default values. Click on the **Commit** button (not shown).

Repeat all the steps for each new Location. The following screen shows the Location page used during the compliance test.

The screenshot shows the Avaya Aura System Manager 6.3 interface. The top navigation bar includes the Avaya logo, the title 'Avaya Aura® System Manager 6.3', and user information: 'Last Logged on at June 15, 2013 9:47 AM', 'Help | About | Change Password | Log off admin'. The main content area is titled 'Locations' and shows a table with 2 items. The table has columns for 'Name' and 'Notes'. The first row is 'Belleville' with notes 'Belleville DevConnect Location'. The second row is 'Subnet 10.10.97.0', which is highlighted with a red box. Below the table, there is a 'Select : All, None' option.

Name	Notes
Belleville	Belleville DevConnect Location
Subnet 10.10.97.0	

6.3. Administer SIP Entities

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

- Session Manager.
- Communication Manager (Avaya S8800 Server).

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

General section

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

- **Name:** Enter a descriptive name.
- **FQDN or IP Address:** Enter the IP address of the signaling interface on each:
 - Communication Manager
 - Session Manager virtual SM-100
- From the **Type** drop down menu, select a type that best matches the SIP Entity:
 - For Communication Manager, select **CM**
 - For Session Manager, select **Session Manager**
- 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 (not shown). The following screen shows the SIP Entities page used during the compliance test.

Repeat all the steps for each new entity.

The screenshot shows the Avaya Aura System Manager 6.3 interface. The top navigation bar includes the Avaya logo, the product name 'Avaya Aura® System Manager 6.3', and user information: 'Last Logged on at June 15, 2013 9:47 AM', 'Help | About | Change Password | Log off admin'. The main content area is titled 'SIP Entities' and features a sidebar on the left with a menu containing 'Routing', 'Domains', 'Locations', 'Adaptations', 'SIP Entities' (highlighted), 'Entity Links', 'Time Ranges', 'Routing Policies', 'Dial Patterns', 'Regular Expressions', and 'Defaults'. The main area has a breadcrumb trail 'Home / Elements / Routing / SIP Entities' and a 'Help ?' link. Below the breadcrumb are buttons for 'New', 'Edit', 'Delete', 'Duplicate', and 'More Actions'. A table displays 18 items with a 'Refresh' button and a 'Filter: Enable' option. The table has the following data:

<input type="checkbox"/>	Name	FQDN or IP Address	Type	Notes
<input type="checkbox"/>	DevCM	10.10.97.201	CM	CM SIP Entity in the main lab
<input type="checkbox"/>	DevCM3_62	10.33.4.9	CM	Phuong CM
<input type="checkbox"/>	DevSM	10.10.97.198	Session Manager	SIP Entity for Session Manager
<input type="checkbox"/>	ESNA	10.10.98.115	Other	ESNA Office LinX
<input type="checkbox"/>	IP_Office_Bottom	10.10.97.39	Other	SIP entity for bottom IP Office
<input type="checkbox"/>	IP_Office_Top	10.10.97.36	Other	SIP entity for top IP Office

At the bottom of the table, there is a 'Select : All, None' option and a pagination control showing '< Previous | Page 1 of 2 | Next >'.

6.4. Administer Entity Links

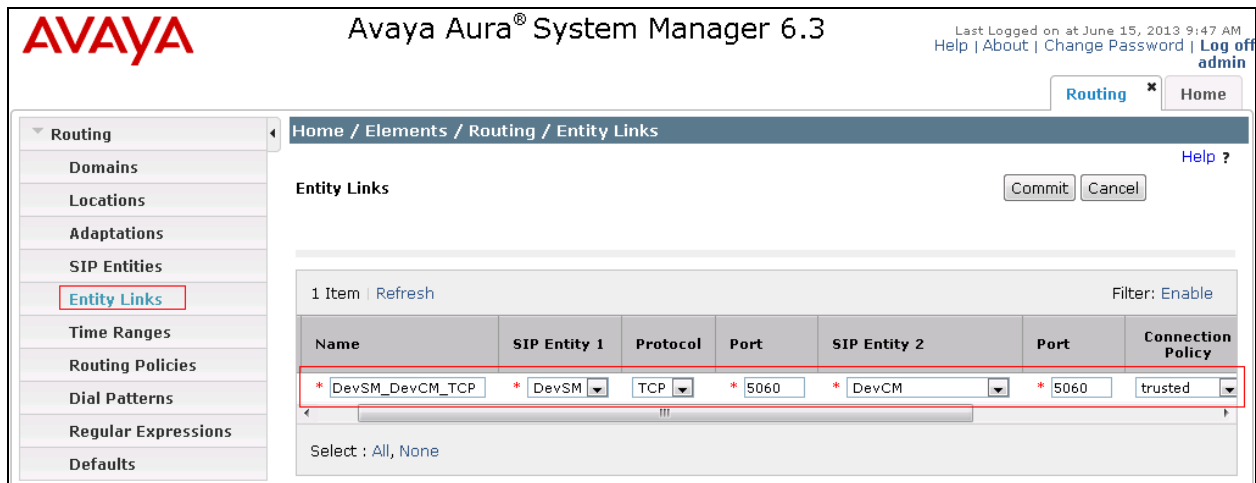
Entity Links define the connections between the SIP Entities and Session Manager. In the compliance test, the following entity links are defined from Session Manager.

- Session Manager ↔ Communication Manager (Avaya S8800 Server).

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

- **Name:** Enter a descriptive name.
- In the **SIP Entity 1** drop down menu, select the Session Manager SIP Entity created in **Section Error! Reference source not found.** (e.g. **DevSM**).
- In the **Protocol** drop down menu, select the protocol to be used.
- In the **Port** field, enter the port to be used (e.g. **5060** or **5061**).
 - TLS – 5061
 - TCP – 5060
- In the **SIP Entity 2** drop down menu, select one of the two entities in the bullet list above (which were created in **Section Error! Reference source not found.**, e.g. **DevCM**).
- In the **Port** field, enter the port to be used (e.g. **5060** or **5061**).
- Check the **Trusted** box.
- Enter a description in the **Notes** field if desired.

Click on the **Commit** button to save each Entity Link definition (not shown). The following screen shows an Entity Links page used during the compliance test.



The screenshot shows the Avaya Aura System Manager 6.3 interface. The left sidebar contains a navigation menu with 'Entity Links' highlighted. The main content area shows the 'Entity Links' configuration page. At the top right, there are 'Commit' and 'Cancel' buttons. Below the buttons, there is a table with one row of data. The table has the following columns: Name, SIP Entity 1, Protocol, Port, SIP Entity 2, Port, and Connection Policy. The data row is: * DevSM_DevCM_TCP, * DevSM, TCP, * 5060, * DevCM, * 5060, trusted. Below the table, there is a 'Select : All, None' option.

Name	SIP Entity 1	Protocol	Port	SIP Entity 2	Port	Connection Policy
* DevSM_DevCM_TCP	* DevSM	TCP	* 5060	* DevCM	* 5060	trusted

6.5. Administer Routing Policy

Routing Policies associate destination SIP Entities (**Section Error! Reference source not found.**) with Time of Day admission control parameters (**Section Error! Reference source not found.**) and Dial Patterns (**Section Error! Reference source not found.**). In the reference configuration, Routing Policies are defined for:

- Inbound calls to Communication Manager.

To add a Routing Policy, navigate to **Routing**→**Routing Policies** and click on the **New** button on the right (screen not shown). 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 a SIP Entity that will be the destination for this call.
- 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. Repeat the steps for each new Routing Policy. The following screen shows the Routing Policy used for Communication Manager during the compliance test.

The screenshot displays the Avaya Aura System Manager 6.3 interface. The top navigation bar includes the Avaya logo, the title 'Avaya Aura® System Manager 6.3', and user information: 'Last Logged on at June 15, 2013 9:47 AM', 'Help | About | Change Password | Log off', and 'admin'. The main content area is titled 'Routing Policy Details' and is divided into two sections: 'General' and 'SIP Entity as Destination'. The 'General' section contains fields for '* Name:' (To-DevCM), 'Disabled:' (checkbox), '* Retries:' (input field), and 'Notes:' (Route to DevCM with G650). The 'SIP Entity as Destination' section has a 'Select' button and a table listing available SIP entities. The table has columns for Name, FQDN or IP Address, Type, and Notes. One entity, 'DevCM', is highlighted with a red box, showing its FQDN as 10.10.97.201, Type as CM, and Notes as 'CM SIP Entity in the main lab'. The interface also features a left-hand navigation menu with 'Routing Policies' selected, and buttons for 'Commit' and 'Cancel'.

Name	FQDN or IP Address	Type	Notes
DevCM	10.10.97.201	CM	CM SIP Entity in the main lab

6.6. Administer Dial Pattern

Dial Patterns define digit strings to be matched for inbound and outbound calls. In the compliance test, the following dial patterns are defined from Session Manager.

- 53 – SIP endpoints in Avaya S8800 Server.

To add a Dial Pattern, select **Routing→Dial Patterns** (not shown) and click on the **New** button (not shown) on the right pane. Provide the following information:

General section

- Enter a unique pattern in the **Pattern** field (e.g. **53**).
- 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** drop down menu select the domain that will be contained in the Request URI received by Session Manager from Communication Manager.

Originating Locations and Routing Policies section

- Click on the **Add** button and a window will open (not shown).
- Click on the box for the appropriate Originating Locations, and Routing Policies (see **Section 6.5**) that pertain to this Dial Pattern.
 - Select the Originating Location to apply the selected routing policies to **All**.
 - Select Routing Policies to **To-DevCM**.
 - Click on the **Select** button and return to the Dial Pattern window.

Click the **Commit** button to save the new definition (not shown). The following screen shows the dial pattern used for **53xxx** during the compliance test. Repeat steps for the remaining Dial Patterns.

Avaya Aura® System Manager 6.3

Last Logged on at June 15, 2013 9:47 AM
Help | About | Change Password | Log off admin

Routing * Home

Home / Elements / Routing / Dial Patterns

Dial Pattern Details Commit Cancel Help ?

General

* Pattern: 53
* Min: 5
* Max: 5

Emergency Call:
Emergency Priority: 1
Emergency Type:

SIP Domain: bvwddev.com
Notes: Dial Pattern for DevCM with G650

Originating Locations and Routing Policies

Add Remove

1 Item Refresh Filter: Enable

<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"/>	-ALL-		To-DevCM		<input type="checkbox"/>	DevCM	Route to DevCM with G650

6.7. Administer Manage Element

To define a new Manage Element, navigate to **Elements**→**Inventory**→**Manage Elements**. Click on the **New** button (screen not shown) to open the **New Elements** page; in the **General** tab select **Communication Manager** in the **Type** dropdown menu.

The screenshot shows the 'New Elements' page in Avaya Aura System Manager 6.3. The 'Type' dropdown menu is open, and 'Communication Manager' is selected. The page includes a navigation menu on the left, a breadcrumb trail, and 'Commit' and 'Cancel' buttons.

The **Add Communication Manager** page is displayed. In the **General Attribute** tab:

- **Name** - enter a descriptive name, e.g. **DevCM**.
- **Host name or IP Address** - enter IP address of CM **10.10.97.201**.
- **Login** - enter the user name "**cus**". Note that the user name "**cus**" was created while installing Communication Manager, this can be any name.
- **Authentication Type** - select **Password**.
- **Password** - enter a password for username "**cus**".
- **Confirm Password** - enter the password again.
- Keep other field at default.
- Click **Commit** button to save the element.

The screenshot shows the 'Add Communication Manager' page in Avaya Aura System Manager 6.3. The 'General Attributes (G)' tab is active, showing fields for Name, Hostname or IP Address, Login, Authentication Type, Password, Confirm Password, SSH Connection, RSA SSH Fingerprint, Port, and Location. The 'Commit' button is visible.

6.8. Administer Applications

To define a new Application, navigate to **Elements** → **Session Manager** → **Application Configuration** → **Applications**. Click **New** (not shown) to open the Applications Editor page:

- Application Editor section
 - **Name** – Enter name for the application.
 - **SIP Entity**–Select the SIP Entity for Communication Manager defined in **Section Error! Reference source not found.**
 - **CM System for SIP Entity** –Select the name of the Managed Element defined for Communication Manager in **Section 6.7.**
 - **Description**– Enter description if desired.
- Leave the fields in the Application Attributes (optional) section blank.
- Click **Commit** button to save the Application.

The screenshot displays the Avaya Aura System Manager 6.3 interface. The top navigation bar includes the Avaya logo, the product name 'Avaya Aura® System Manager 6.3', and user information: 'Last Logged on at June 15, 2013 9:47 AM', 'Help | About | Change Password | Log off admin'. The breadcrumb trail is 'Home / Elements / Session Manager / Application Configuration / Applications'. The left sidebar shows a tree view with 'Applications' selected. The main content area is titled 'Application Editor' and contains the following form fields:

- Name:** Text input field containing 'DevCM-APP'.
- SIP Entity:** Dropdown menu with 'DevCM' selected.
- CM System for SIP Entity:** Dropdown menu with 'DevCM' selected, a 'Refresh' button, and a link 'View/Add CM Systems'.
- Description:** Text input field containing 'Application for DevCM with G650'.

Buttons for 'Commit' and 'Cancel' are located at the top right of the form area.

6.9. Administer Application Sequence

Navigate to **Elements** → **Session Manager** → **Application Configuration** → **Application Sequences**. Click **New** (screen not shown) and provide the following information:

- Sequence Name section
 - **Name** – The name for the application.
 - **Description** – Enter description, if desired.

Application Sequence Editor

Commit Cancel

Application Sequence

*Name

Description

- Available Applications section
 - Click **+** icon associated with the Application for Communication Manager defined in **Section** Error! Reference source not found. to select this application.
 - Verify a new entry is added to the Applications in this Sequence table as shown below.

Click the **Commit** button (screen not shown) to save the new Application Sequence.

Available Applications

3 Items | Refresh Filter: Enable

	Name	SIP Entity	Description
+	CM5_APP	CMRIs5	Application for CM Release 5
+	DevCM3_APP	DevCM3_62	Phuong CM
+	DevCM-APP	DevCM	Application for DevCM with G650

The screen below shows the Application Sequence, **DevCM-SEQ**, defined during the compliance test.

AVAYA Avaya Aura® System Manager 6.3 Last Logged on at June 15, 2013 9:47 AM
Help | About | Change Password | Log off
admin

Session Manager * Home

Home / Elements / Session Manager / Application Configuration / Application Sequences [Help ?](#)

Application Sequences

This page allows you to add, edit, or remove sequences of applications.

[Application Sequences](#)

3 Items | Refresh Filter: Enable

<input type="checkbox"/>	Name	Description
<input type="checkbox"/>	CM5_Seq	Sequence for CM 5.2
<input type="checkbox"/>	DevCM3_Seq	DevCM3Seq
<input type="checkbox"/>	DevCM-SEQ	Sequence for DevCM

Select : All, None

6.10. Administer SIP User

When adding new SIP user, use the option to automatically generate the SIP station in Communication Manager, after adding a new SIP user.

To add new SIP users, Navigate to **Home → Users → User Manage → Manage Users** (not shown). Click **New** (not shown) and provide the following information:

- Identity section
 - **Last Name** – Enter last name of user.
 - **First Name** – Enter first name of user.
 - **Login Name** – Enter extension number@sip domain. The sip domain is defined as Authoritative Domain in **Section 6.1**.
 - **Authentication Type** – Verify **Basic** is selected.
 - **Password** – Enter password to be used to log into System Manager.
 - **Confirm Password** – Repeat value entered above.

New User Profile

Identity * **Communication Profile *** **Membership** **Contacts**

Identity ▾

* **Last Name:**

* **First Name:**

Middle Name:

Description:

* **Login Name:**

* **Authentication Type:** ▾

Password:

Confirm Password:

Localized Display Name:

Endpoint Display Name:

Title:

Language Preference: ▾

Time Zone: ▾

Employee ID:

- Communication Profile section
 - **Communication Profile Password** – Type Communication profile password in this field.
 - **Confirm Password** – Repeat value entered above.

New User Profile

Identity *

Communication Profile *

Membership

Contacts

Communication Profile ▾

Communication Profile Password:

Confirm Password:

	Name
<input checked="" type="radio"/>	Primary
Select : None	

* Name:

Default :

- Communication Profile sub-section
 - **Fully Qualified Address** – Enter the extension of the user.
 - Click **Add** button.

Communication Address ▾

<input type="checkbox"/>	Type	Handle	Domain
No Records found			

Type:

* Fully Qualified Address: @

- Session Manager Profile section
 - **Primary Session Manager** – Select one of the Session Managers.
 - **Secondary Session Manager** – Select **(None)** from drop-down menu.
 - **Origination Application Sequence** – Select Application Sequence defined in **Section 6.8** for Communication Manager.
 - **Termination Application Sequence** – Select Application Sequence defined in **Section 6.8** for Communication Manager.
 - **Survivability Server** – Select **(None)** from drop-down menu.
 - **Max. Simultaneous** – Select number of devices can use this SIP user. The Session Manager supports up to 10 devices to register to one SIP user and this feature is only available from Release 6.3.
 - **Home Location** – Select Location defined in **Section 6.2**.

Session Manager Profile ▼

SIP Registration

* **Primary Session Manager** ▼

Primary	Secondary
31	0

Secondary Session Manager ▼

Survivability Server ▼

Max. Simultaneous Devices ▼

Block New Registration When Maximum Registrations Active?

Application Sequences

Origination Sequence ▼

Termination Sequence ▼

Call Routing Settings

* **Home Location** ▼

Conference Factory Set ▼

- CM Endpoint Profile section
 - **System** – Select Manage Element defined in **Section 6.7**.
 - **Profile Type** – Select **Endpoint**.
 - **Use Existing Endpoints** - Leave unchecked to automatically create new endpoint when new user is created. Or else, check the box if endpoint is already defined in Communication Manager.
 - **Extension** - Enter same extension number used in this section.
 - **Template** – Select template for type of SIP phone.
 - **Security Code** – Leave it as default (blank).
 - **Port** – Select **IP** from drop down menu.
 - **Voice Mail Number** – Enter **Pilot Number** for Avaya Modular Messaging if installed. Or else, leave field blank. This feature is not used during the compliance test.
 - **Delete Station on Unassign of Endpoint** – Check the box to automatically delete station when Endpoint Profile is un-assigned from user.
 - **Override Endpoint Name** – Checked.

CM Endpoint Profile ▼

* **System** ▼

* **Profile Type** ▼

Use Existing Endpoints

* **Extension**

* **Template** ▼

Set Type

Security Code

Port

Voice Mail Number

Preferred Handle ▼

Enhanced Callr-Info display for 1-line phones


Delete Endpoint on Unassign of Endpoint from User or on Delete User

Override Endpoint Name

7. Configure VTech SIP Phone

VTech SIP Hotel Phones are configured using a web browser. The phones use DHCP by default and are powered over their Ethernet port. In the tested configuration, the phones were connected to the LAN via an Avaya BayStack 5510-PWR network switch on a segment with a DHCP server. Using the admin tools on the DHCP server provided a way to discover IP Addresses.

Enter the URL of the phone such as <http://<host or IP address>>. When prompted, login using 'root' for the user account, and the appropriate password (not shown). The initial screen is shown below and all navigation is via the navigation tree on the left panel. Some of the links lead to configuration settings that are not yet supported. See the VTech documentation for more details [4]. The home page of the VTech phone is displayed in the screen below.

 VTech SIP Phone Web Portal Basic Phone Information Hotel Information System Configuration Network Configuration Network Security Static IP Mapping Phone Configuration SIP Account Settings Advanced SIP Settings Audio Codec Advanced Call Features Ring Tone Speed Dial Other Phone Settings	Basic Phone Information													
	<table><tr><td>Model Number</td><td><input type="text" value="S2220"/></td></tr><tr><td>MAC Address</td><td><input type="text" value="00:12:2a:19:3e:b9"/></td></tr><tr><td>Hardware Version</td><td><input type="text" value="7800008"/></td></tr><tr><td>Boot Version</td><td><input type="text" value="VTechBoot 1.02.00"/></td></tr><tr><td>Firmware Version</td><td><input type="text" value="SIP_39.3.68.07"/></td></tr><tr><td>Release Date</td><td><input type="text" value="May 16 2013 - 15:17:43"/></td></tr><tr><td>Audio Profile Version</td><td><input type="text" value="S2100 S2210 S2220 0007"/></td></tr></table>	Model Number	<input type="text" value="S2220"/>	MAC Address	<input type="text" value="00:12:2a:19:3e:b9"/>	Hardware Version	<input type="text" value="7800008"/>	Boot Version	<input type="text" value="VTechBoot 1.02.00"/>	Firmware Version	<input type="text" value="SIP_39.3.68.07"/>	Release Date	<input type="text" value="May 16 2013 - 15:17:43"/>	Audio Profile Version
Model Number	<input type="text" value="S2220"/>													
MAC Address	<input type="text" value="00:12:2a:19:3e:b9"/>													
Hardware Version	<input type="text" value="7800008"/>													
Boot Version	<input type="text" value="VTechBoot 1.02.00"/>													
Firmware Version	<input type="text" value="SIP_39.3.68.07"/>													
Release Date	<input type="text" value="May 16 2013 - 15:17:43"/>													
Audio Profile Version	<input type="text" value="S2100 S2210 S2220 0007"/>													

To register VTech SIP phone to Session Manager, click on the **SIP Account Settings** under the **Phone Configuration** category in the left navigation pane. The **SIP Account Settings** page is displayed in the right. Enter the username and its Communication Profile password as configured in **Section 6.10**. Click the **Save** button to save changes.

The screenshot shows the VTech SIP Phone Web Portal interface. On the left is a navigation pane with the VTech logo and several menu items: 'VTech SIP Phone Web Portal', 'Basic Phone Information', 'Hotel Information', 'System Configuration' (with sub-items 'Network Configuration', 'Network Security', and 'Static IP Mapping'), and 'Phone Configuration' (with sub-items 'SIP Account Settings', 'Advanced SIP Settings', 'Audio Codec', 'Advanced Call Features', 'Ring Tone', and 'Speed Dial'). The 'SIP Account Settings' page is highlighted with a red border. It features a 'Line 1' section with the following fields: 'Extension' (53104), 'Authentication Name' (53104), 'Password' (masked with four dots), 'DTMF Method' (RFC 2833), 'External Call Prefix' (9), and 'SIP Registration Status' (Unregistered). A 'Save' button is located below these fields.

SIP Account Settings	
Line 1	
Extension	<input type="text" value="53104"/>
Authentication Name	<input type="text" value="53104"/>
Password	<input type="password" value="••••"/>
DTMF Method	<input type="text" value="RFC 2833"/>
External Call Prefix	<input type="text" value="9"/>
SIP Registration Status	<input type="text" value="Unregistered"/>
<input type="button" value="Save"/>	

Click on **Advanced SIP Settings**, the **Advance SIP Settings** is displayed in the right. Enter the Session Manager IP **10.10.97.198** in the **Registrar Server Address** and **Messaging Waiting Server** fields with port **5060**. Note that in order Session Manager sends MWI message to VTech phone, the VTech phone must subscribe MWI feature to Session Manager therefore the Session Manager IP **10.10.97.198** must be inputted in the **Message Waiting Server** field. Keep other fields at default.

Click the **Save** button to save the change. The VTech SIP phone needs a reboot for changes take effect. Reboot the phone by click on Reboot Phone link in the left navigation pane.

The screenshot shows the VTech SIP Phone Web Portal interface. On the left is a navigation menu with categories: VTech SIP Phone Web Portal, System Configuration, Phone Configuration, and System Resources. The 'Advanced SIP Settings' link under Phone Configuration is highlighted with a red box. The main content area is titled 'Advanced SIP Settings' and contains a list of configuration fields. A red box highlights the 'Registrar Server Address : Port', 'Proxy Server Address : Port', and 'Message Waiting Server : Port' fields. The 'Registrar Server Address : Port' field contains '10.10.97.198' and '5060'. The 'Message Waiting Server : Port' field also contains '10.10.97.198' and '5060'. Other fields include 'Backup Registrar Server' (Disable), 'Backup Registrar Server Address : Port' (empty), 'Backup Registrar Retry Count' (2), 'SIP Transport' (UDP), 'Registration Timeout (second)' (300), 'Registration Retry Limit (attempt)' (10), 'Message Waiting Subscribe Timeout (second)' (300), 'PRACK' (Disable), 'Dial Plan' (.T), 'Interdigit Timeout (second)' (5), and 'On Hold Timeout (minute)' (15). A 'Save' button is located at the bottom left of the settings area.

Field	Value
Registrar Server Address : Port	10.10.97.198 : 5060
Proxy Server Address : Port	: 5060
Message Waiting Server : Port	10.10.97.198 : 5060
Backup Registrar Server	Disable
Backup Registrar Server Address : Port	:
Backup Registrar Retry Count	2
SIP Transport	UDP
Registration Timeout (second)	300
Registration Retry Limit (attempt)	10
Message Waiting Subscribe Timeout (second)	300
PRACK	Disable
Dial Plan	.T
Interdigit Timeout (second)	5
On Hold Timeout (minute)	15

After the VTech phone rebooted, if the VTech phone is successfully registered to the Session Manager the SIP Registration Status will show as “Registered” as the screen below.

The screenshot displays the VTech SIP Phone Web Portal interface. On the left is a navigation menu with the VTech logo and various configuration links. The main content area is titled 'SIP Account Settings' and shows settings for 'Line 1'. The 'SIP Registration Status' field is highlighted with a red border and contains the text 'Registered'. Below the settings is a 'Save' button.

vtech	
VTech SIP Phone Web Portal	
Basic Phone Information	
Hotel Information	
System Configuration	
Network Configuration	
Network Security	
Static IP Mapping	
Phone Configuration	
SIP Account Settings	
Advanced SIP Settings	
Audio Codec	
Advanced Call Features	
Ring Tone	
Speed Dial	

SIP Account Settings	
Line 1	
Extension	<input type="text" value="53104"/>
Authentication Name	<input type="text" value="53104"/>
Password	<input type="password" value="••••"/>
DTMF Method	<input type="text" value="RFC 2833"/>
External Call Prefix	<input type="text" value="9"/>
SIP Registration Status	<input type="text" value="Registered"/>
<input type="button" value="Save"/>	

To configure audio codec, click on **Audio Codec** link in the left navigation pane. The **Audio Codec** page is displayed in the right. During the compliance test, the codec **G.711u** was configured as first priority for calls as shown in the screen below.

vtech

VTech SIP Phone Web Portal

[Basic Phone Information](#)
[Hotel Information](#)

System Configuration

[Network Configuration](#)
[Network Security](#)
[Static IP Mapping](#)

Phone Configuration

[SIP Account Settings](#)
[Advanced SIP Settings](#)
[Audio Codec](#)
[Advanced Call Features](#)
[Ring Tone](#)
[Speed Dial](#)

[Other Phone Settings](#)

Audio Codec

Line 1

Audio Codec 1

Audio Codec 2

Audio Codec 3

Audio Codec 4

SRTP Mode

8. Verification Steps

Calls were placed to and from the VTech phones manually. Confirmation of functionality was generally observed by listening for audio on connected calls. Tracing was used on Avaya Aura® Session Manager, and using Wireshark on a locally connected PC to review SIP messages to and from the phones.

9. Conclusion

The VTech 1-Line Hospitality SIP Phones successfully interoperated with the Avaya Aura® Communication Manager and Avaya Aura® Session Manager as described in these Application Notes. There were some observations noted in **Section 2.2**.

10. Additional References

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

[1] *Implementing Avaya Aura® Session Manager*, Issue 2, Release 6.3, May 2013.

[2] *Administering Avaya Aura® Communication Manager*, Issue 8, Release 6.3, May 2013.

[3] *Administering Avaya Aura® System Manager*, Release 6.3, Issue 2, May 2013.

Product information for VTech SIP Hotel Phones may be found at <http://vtechhotelphones.com>.

[4] *VTech SIP Phone System Integration Guide*, Document ID 91-004252-010-100, Version 5

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