



Avaya Solution & Interoperability Test Lab

Application Notes for Biamp AudiaFLEX VoIP-2 with Avaya Aura® Communication Manager Using Avaya Aura® SIP Enablement Services – Issue 1.0

Abstract

These Application Notes describe the configuration steps required for Biamp AudiaFLEX VoIP-2 to interoperate with Avaya Aura® Communication Manager using Avaya Aura® SIP Enablement Services.

Biamp AudiaFLEX is a digital audio platform, and the VoIP-2 card allows connection to IP-based phone systems. In the compliance testing, the VoIP-2 card registered as two SIP endpoints to Avaya Aura® SIP Enablement Services.

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

These Application Notes describe the configuration steps required for Biamp AudiaFLEX VoIP-2 to interoperate with Avaya Aura® Communication Manager using Avaya Aura® SIP Enablement Services.

Biamp AudiaFLEX is a digital audio platform, and the VoIP-2 card allows connection to IP-based phone systems. In the compliance testing, the VoIP-2 card registered as two SIP endpoints to Avaya Aura® SIP Enablement Services.

Biamp AudioFLEX VoIP-2 is typically controlled by custom third party applications, developed using the Biamp API. The compliance test used the default out-of-the-box Biamp Audia application to configure and control the VoIP-2 card, along with microphones and speakers to test the audio connections. Any customized application developed using the Biamp API is outside the scope of this compliance test.

2. General Test Approach and Test Results

The feature test cases were performed manually. Calls were manually established between AudiaFLEX VoIP-2 users with Avaya SIP, Avaya H.323, and/or PSTN users. Call controls were performed from the various users to verify the call scenarios.

The serviceability test cases were performed manually by disconnecting and reconnecting the Ethernet cable to AudiaFLEX VoIP-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

The interoperability compliance test included feature and serviceability testing.

The feature testing included registration, basic call, display, mute/unmute, hold/reconnect, drop, media shuffling, G.711, G.729, codec negotiation, music on hold, DTMF, long hold with held call reminder, long duration, coverage, simultaneous calls at both AudiaFLEX VoIP-2 channels, call progress tones and treatment of reorder and busy.

The serviceability testing focused on verifying the ability of AudiaFLEX VoIP-2 to recover from adverse conditions, such as disconnecting/reconnecting the Ethernet cable to AudiaFLEX VoIP-2.

2.2. Test Results

All test cases were executed. The following were observations on AudiaFLEX VoIP-2 from the compliance testing.

- Only one call appearance is supported by each VoIP-2 channel, therefore features such as call park, transfer, and conference are not applicable.
- For outbound calls, only dialed number is provided.
- The Message Waiting Indicator is not supported.

2.3. Support

Technical support on AudiaFLEX VoIP-2 can be obtained through the following:

- **Phone:** (800) 826-1457
- **Email:** support@biamp.com
- **Web :** <http://www.biamp.com/support/index.aspx>

3. Reference Configuration

The configuration used for the compliance testing is shown below. The Biamp Audia application was installed on a PC to configure and control the AudiaFLEX VoIP-2 card. Two separate sets of microphone and speaker were used and physically connected to the AudiaFLEX server to verify the audio connections. As shown in the test configuration below, the domain name used in the testing was “br110.com”.

The detailed administration of basic connectivity between Communication Manager and SIP Enablement Services is not the focus of these Application Notes and will not be described.

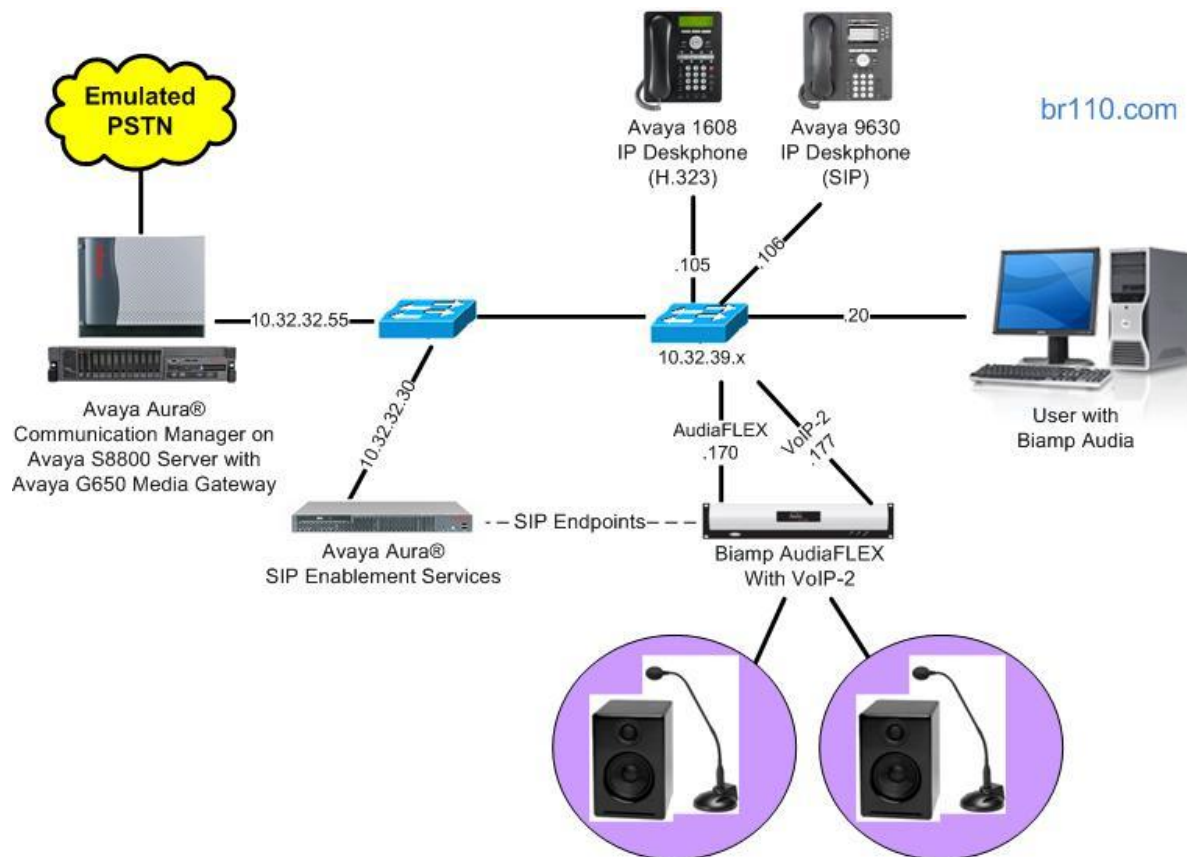


Figure 1: Compliance Testing Configuration

4. Equipment and Software Validated

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

Equipment/Software	Release/Version
Avaya Aura® Communication Manager on Avaya S8800 Server with Avaya G650 Media Gateway	5.2.1 SP13 (R015x.02.1.016.4-19880)
Avaya Aura® SIP Enablement Services	5.2.1 SP7
Avaya 1608 IP Deskphone (H.323)	1.302S
Avaya 9630 IP Deskphone (SIP)	2.6.8
Biamp AudiaFLEX • VoIP-2	3.401-2.3-4.830 1.201
Biamp Audia on Microsoft Windows XP Professional	5.3 2002 SP3

5. Configure Avaya Aura® Communication Manager

This section provides the procedures for configuring Communication Manager. The procedures include the following areas:

- Verify license
- Administer IP codec set
- Administer stations
- Administer off-pbx stations

5.1. Verify License

Log in to the System Access Terminal (SAT) to verify that the Communication Manager license has proper permissions for features illustrated in these Application Notes. Use the “display system-parameters customer-options” command to verify that there is sufficient capacity for SIP stations by comparing the **Maximum Off-PBX Telephones - OPS** field value with the corresponding value in the **USED** column. The license file installed on the system controls the maximum permitted. If there is insufficient capacity, contact an authorized Avaya sales representative to make the appropriate changes.

display system-parameters customer-options				Page	1 of	11
OPTIONAL FEATURES						
G3 Version: V15						
Location: 1						
Platform: 12						
Software Package: Standard						
RFA System ID (SID): 1						
RFA Module ID (MID): 1						
USED						
Platform Maximum Ports: 44000 242						
Maximum Stations: 36000 38						
Maximum XMOBILE Stations: 0 0						
Maximum Off-PBX Telephones - EC500: 36000 0						
Maximum Off-PBX Telephones - OPS: 36000 8						
Maximum Off-PBX Telephones - PBFMC: 0 0						

5.2. Administer IP Codec Set

Use the “change ip-codec-set n” command, where “n” is the existing codec set number associated with the SIP trunk group to SIP Enablement Services. Update the audio codec types in the **Audio Codec** fields as desired. The screenshot below shows the codec used in the compliance testing.

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.729AB	n	2	20		

5.3. Administer Stations

Add a station for each VoIP-2 channel by using the “add station n” command, where “n” is an available extension number. Enter “x” for **Port** to indicate no hardware associated with the station. Enter a descriptive **Name**, and retain the default values for the remaining fields. Note that there is no need to set the security code, as this will be configured on SIP Enablement Services.

```

add station 66006

```

		Page 1 of 5
STATION		
Extension: 66006	Lock Messages? n	BCC: 0
Type: 6408D+	Security Code:	TN: 1
Port: x	Coverage Path 1:	COR: 1
Name: Biamp VoIP-2 #1	Coverage Path 2:	COS: 1
	Hunt-to Station:	
STATION OPTIONS		
	Time of Day Lock Table:	
Loss Group: 2	Personalized Ringing Pattern: 1	
Data Module? n	Message Lamp Ext: 66006	
Speakerphone: 2-way	Mute Button Enabled? y	
Display Language: english		
Survivable COR: internal	Media Complex Ext:	
Survivable Trunk Dest? y	IP SoftPhone? n	
	Remote Office Phone? N	

Repeat this section to add a station for each channel on each VoIP-2 card. For the compliance testing, two stations were administered for the VoIP-2 card, as shown below.

```

list station 66006 count 2

```

STATIONS									
Ext/ Hunt-to	Port/ Type	Name/ Surv GK NN	Move	Room/ Data Ext	Cv1/ Cv2	COR/ COS	Cable/ TN Jack		
66006	X	Biamp VoIP-2 #1				1			
	6408D+		no			1	1		
66007	X	Biamp VoIP-2 #2				1			
	6408D+		no			1	1		

5.4. Administer Off-PBX Stations

Use the “change off-pbx-telephone station-mapping n” command, where “n” is the first station extension number from **Section 5.3**, to specify routing of calls for the station to SIP Enablement Services. Enter the following values for the specified fields, and retain the default values for the remaining fields.

- **Application:** Enter “OPS” to indicate off-PBX station.
- **Phone Number:** Same digits from the **Station Extension** field.
- **Trunk Selection:** The existing trunk group to reach SIP Enablement Services.
- **Config Set:** An existing configuration set to be used for the off-pbx call treatment.

change off-pbx-telephone station-mapping 66006							Page 1 of 3
STATIONS WITH OFF-PBX TELEPHONE INTEGRATION							
Station Extension	Application	Dial Prefix	CC	Phone Number	Trunk Selection	Config Set	Dual Mode
66006	OPS	-		66006	5	1	

Repeat this section for all stations from **Section 5.3**. For the compliance testing, two off-pbx stations were administered, as shown below.

list off-pbx-telephone station-mapping							
STATION TO OFF-PBX TELEPHONE MAPPING							
Station Extension	Appl	CC	Phone Number	Config Set	Trunk Select	Mapping Mode	Calls Allowed
66001	OPS		66001	1 /	5	both	all
66006	OPS		66006	1 /	5	both	all
66007	OPS		66007	1 /	5	both	all

6. Configure Avaya Aura® SIP Enablement Services

This section provides the procedures for configuring SIP Enablement Services. The procedures include the following areas:

- Launch web interface
- Administer users

6.1. Launch Web Interface

Access the web interface by using the URL “http://ip-address/admin” in an Internet browser window, where “ip-address” is the IP address of SIP Enablement Services. Log in using the appropriate credentials.



The **SIP Enablement Services System Management Interface** screen is displayed. Select **Administration → SIP Enablement Services** from the top menu.

AVAYA SIP Enablement Services (SES) System Management Interface (SMI)

Help Log Off Installation Administration Upgrade This Server: [1] brses1

Legal Notice

SIP Enablement Services System Management Interface

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The **Top** screen is displayed next.

AVAYA Integrated Management SIP Server Management

Help Exit This Server: [1] brses1

Top

Users

- Add
- Default Profile
- Delete
- Edit
- List
- Password
- Search
- Manage All Registered Users
- Search Registered Devices
- Search Registered Users
- Address Map Priorities

Adjunct Systems

- Add
- List

Top

Manage Users	Add and delete Users.
Manage Address Map Priorities	Adjust Address Map Priorities.
Manage Adjunct Systems	Add and delete Adjunct Systems.
Manage Event Aggregators	Add/Delete Event Aggregators.
Certificate Management	Manage Certificates.
Manage Conferencing	Add and delete Conference Extensions.
Manage Emergency Contacts	Add and delete Emergency Contacts.
Export Import to ProVision	Export and import data using ProVision on this host.
Manage Hosts	Add and delete Hosts.

6.2. Administer Users

Select **Users** → **Add** from the left pane to display the **Add User** screen. Enter the following values for the specified fields, and retain the default values in the remaining fields.

- **Primary Handle:** The first station extension from **Section 5.3**.
- **Password:** A desired password for user registration.
- **Confirm Password:** Re-enter the same password.
- **Host:** Select the applicable host.
- **First Name:** A desired first name.
- **Last Name:** A desired last name.
- **Add Communication Manager Extension:** Check the box.

AVAYA Integrated Management SIP Server Management
Help Exit This Server: [1] brses1

Add User

Primary Handle* 66006

User ID

Password*

Confirm Password*

Host* 10.32.32.30

First Name* VoIP-2-2

Last Name* Biamp

Address 1 211 Mt Airy Rd

Address 2 Room 1C110

Office

City Basking Ridge

State NJ

Country USA

Zip 07920

Survivable Call Processor none

Add Communication Manager Extension ☒

Fields marked * are required.

Add

Click **Continue** in the subsequent screen (not shown), to display the **Add Communication Manager Extension** screen below.

For **Extension**, enter the same station extension. For **Communication Manager Server**, select the appropriate server, in this case “CM-G650”.

Repeat this section to create a user for each station in **Section 5.3**. For the compliance testing, two users were administered.



The screenshot displays the Avaya Integrated Management SIP Server Management web interface. The top header features the Avaya logo on the left and the text 'Integrated Management SIP Server Management' on the right, with a status indicator 'This Server: [1] brses1'. Below the header is a navigation menu on the left with options: Top, Users (selected), Add, Default Profile, Delete, Edit, List, Password, Search, Manage All Registered Users, Search Registered, and Devices. The main content area is titled 'Add Communication Manager Extension' and contains the following text: 'Add Communication Manager extension for user 66006.' Below this, there are two input fields: 'Extension' with the value '66006' and 'Communication Manager Server' with a dropdown menu showing 'CM-G650'. A note states 'Fields marked * are required.' and an 'Add' button is located at the bottom left of the form area.

7. Configure Biamp AudiaFLEX VoIP-2

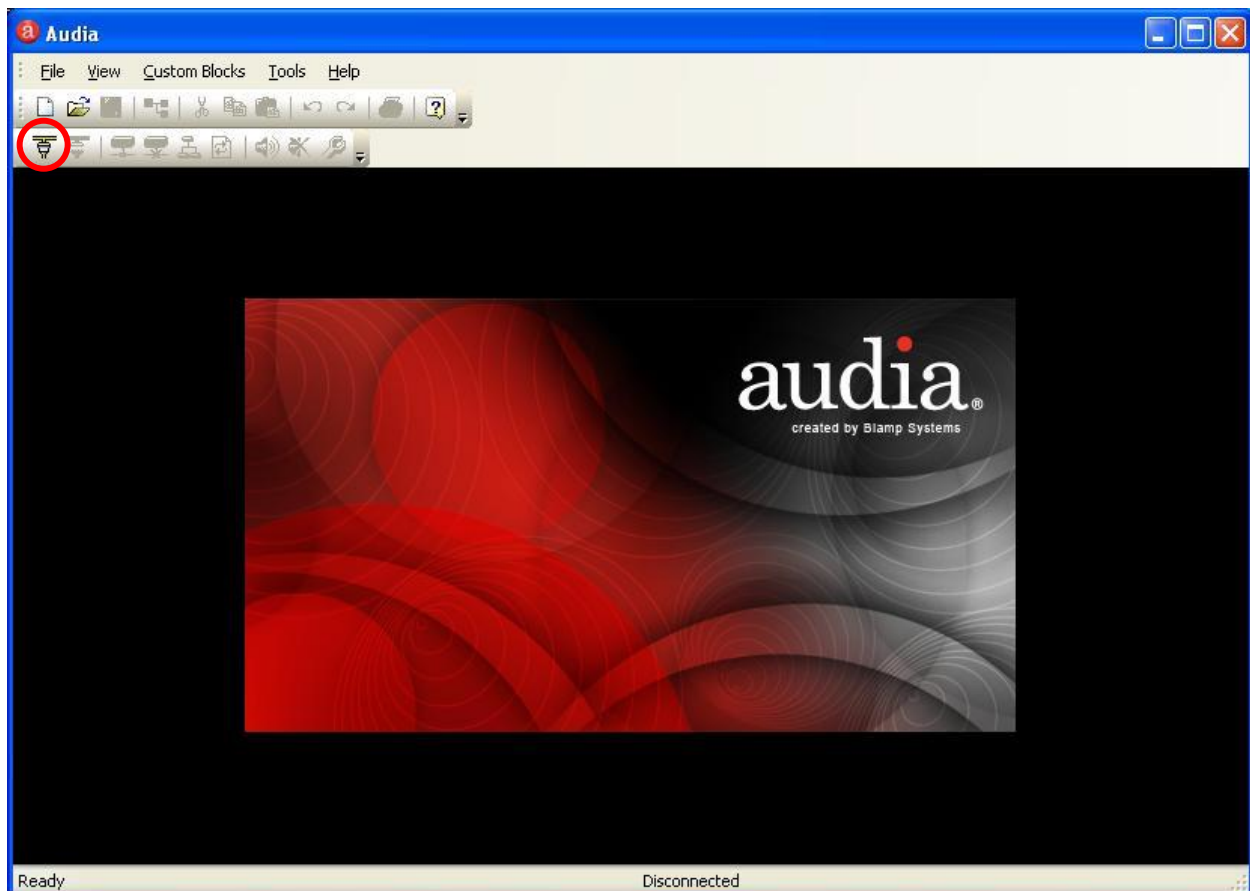
This section provides the procedures for configuring AudiaFLEX VoIP-2. The procedures include the following areas:

- Launch Audia
- Administer design components
- Administer VoIP console

The configuration of AudiaFLEX VoIP-2 is typically performed by authorized third party integrators. The procedural steps are presented in these Application Notes for informational purposes.

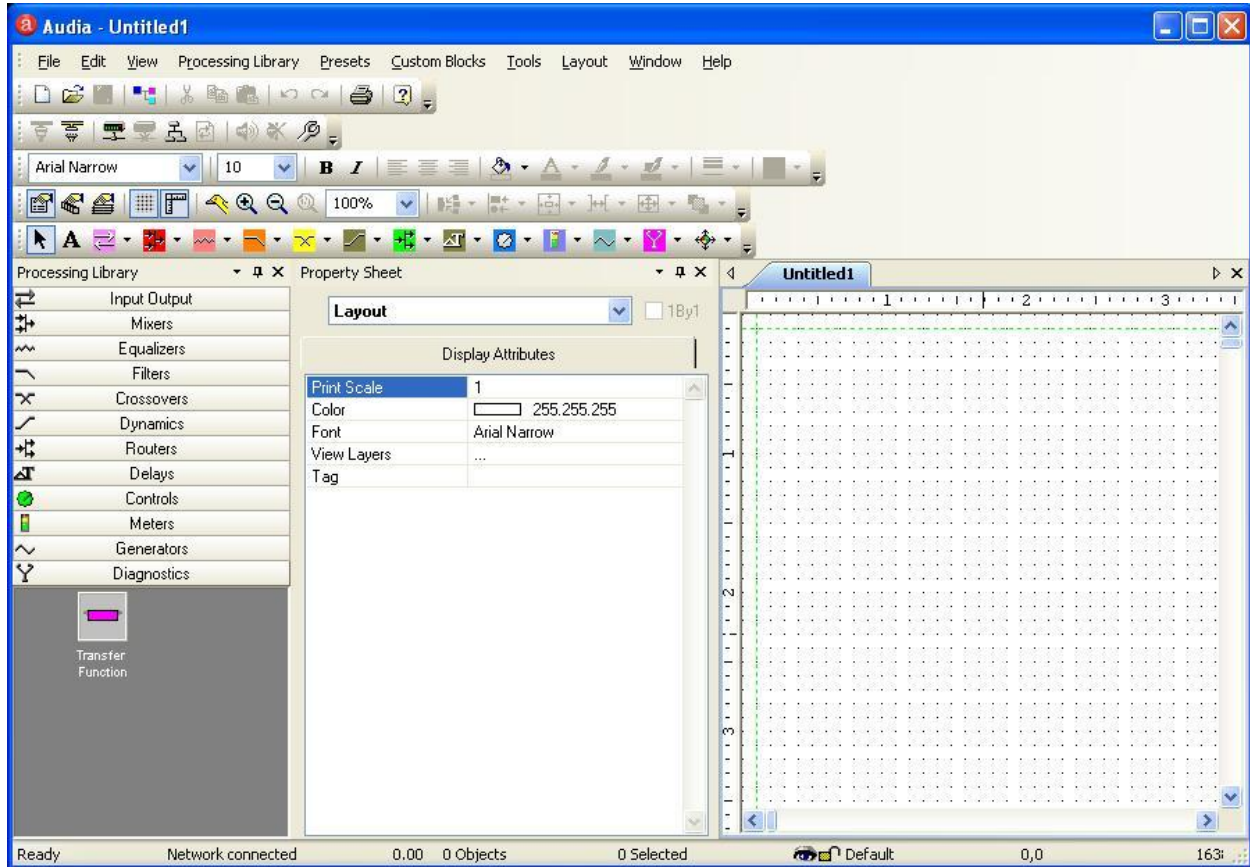
7.1. Launch Audia

From a PC running the Audia application, select **Start → All Program → Audia → Audia** to launch the application. Click on the **Connect** icon show below.

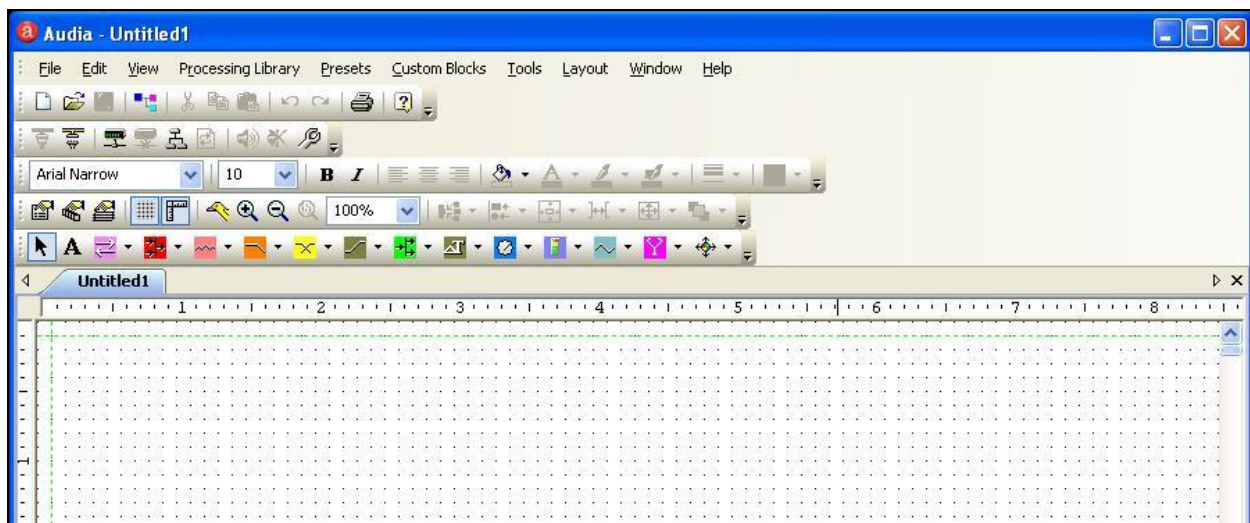


7.2. Administer Design Components

The **Audia** screen is displayed, as shown below. Close the **Processing Library** in the lower left pane, and **Property Sheet** in the lower middle pane.

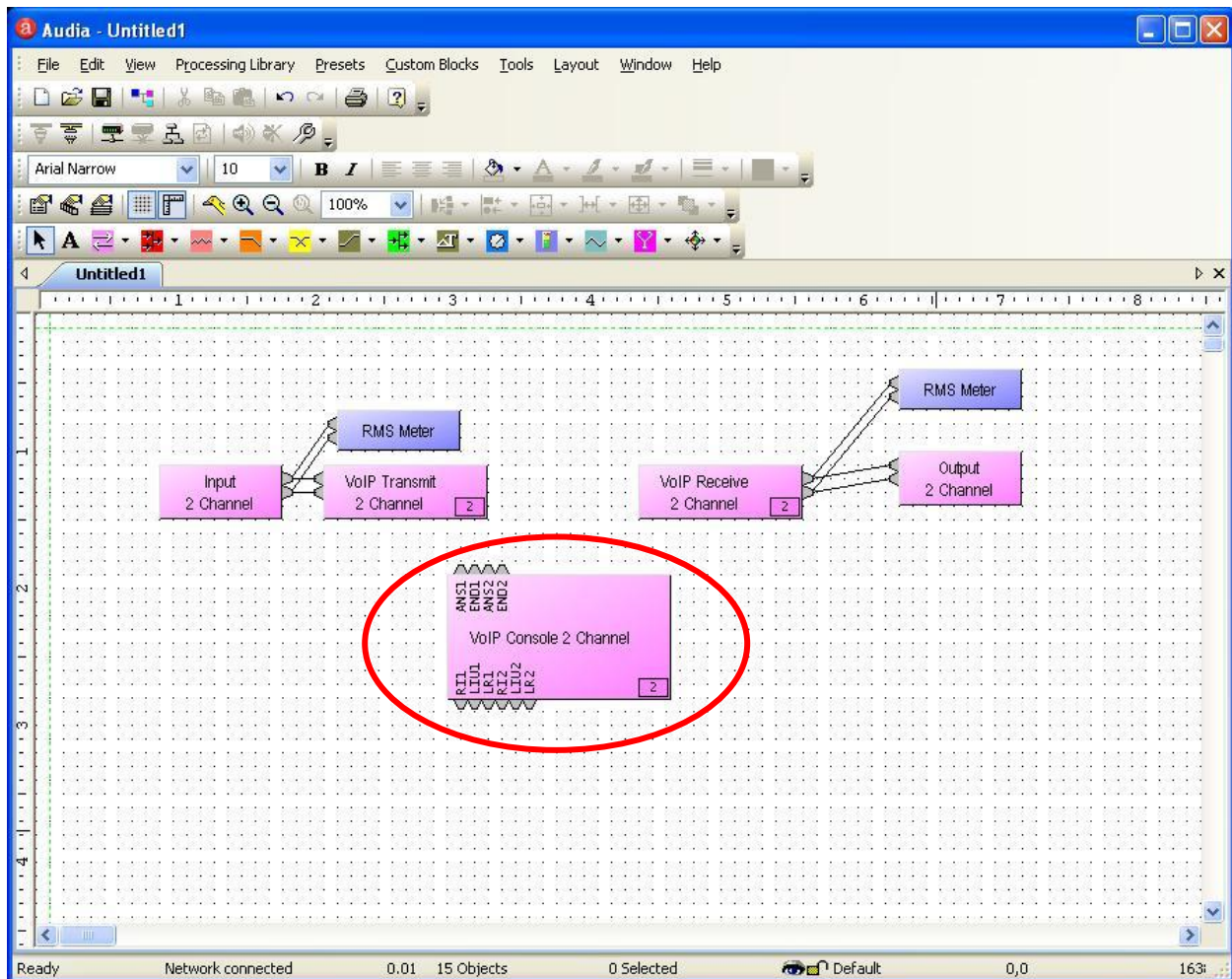


The **Audia** screen is updated as shown below.



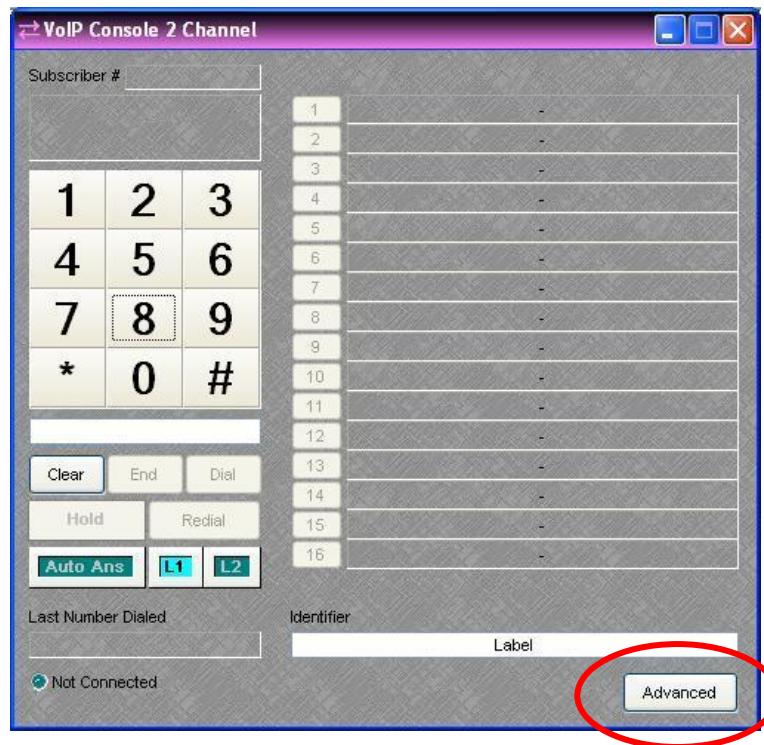
Follow [3] to place relevant component objects into the layout to match the system design. Below is the resultant layout used in the compliance testing. Note that both channels of the VoIP-2 card were used, as shown below.

Double click on the **VoIP Console 2 Channel** object.

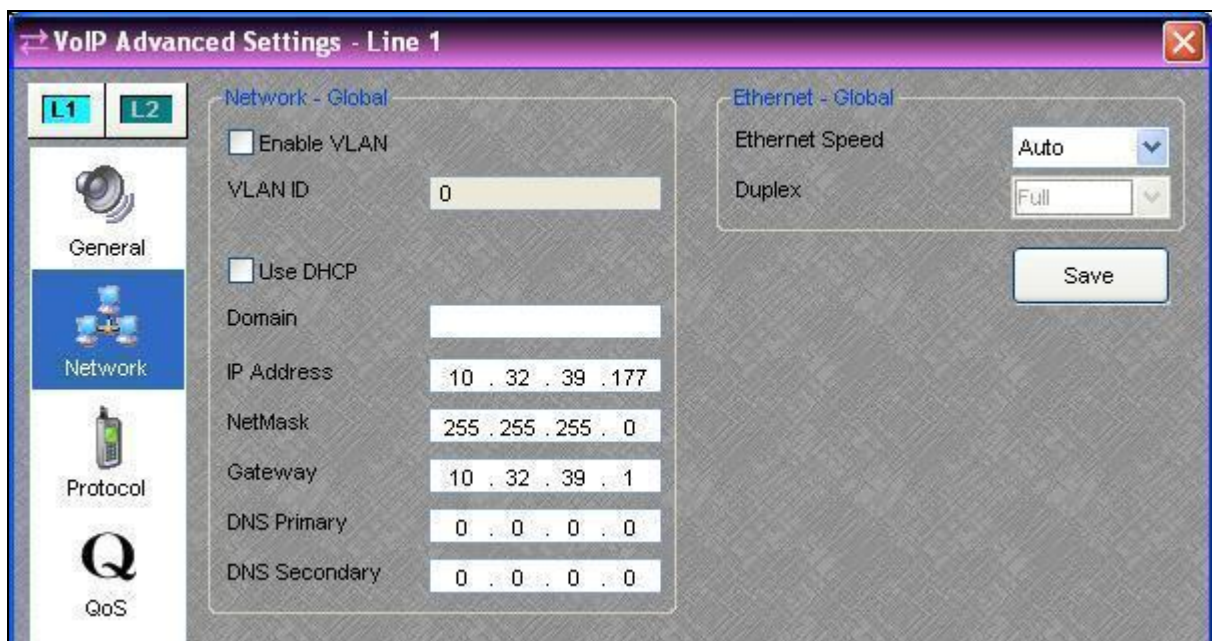


7.3. Administer VoIP Console

The VoIP Console 2 Channel screen is displayed. Click **Advanced**.

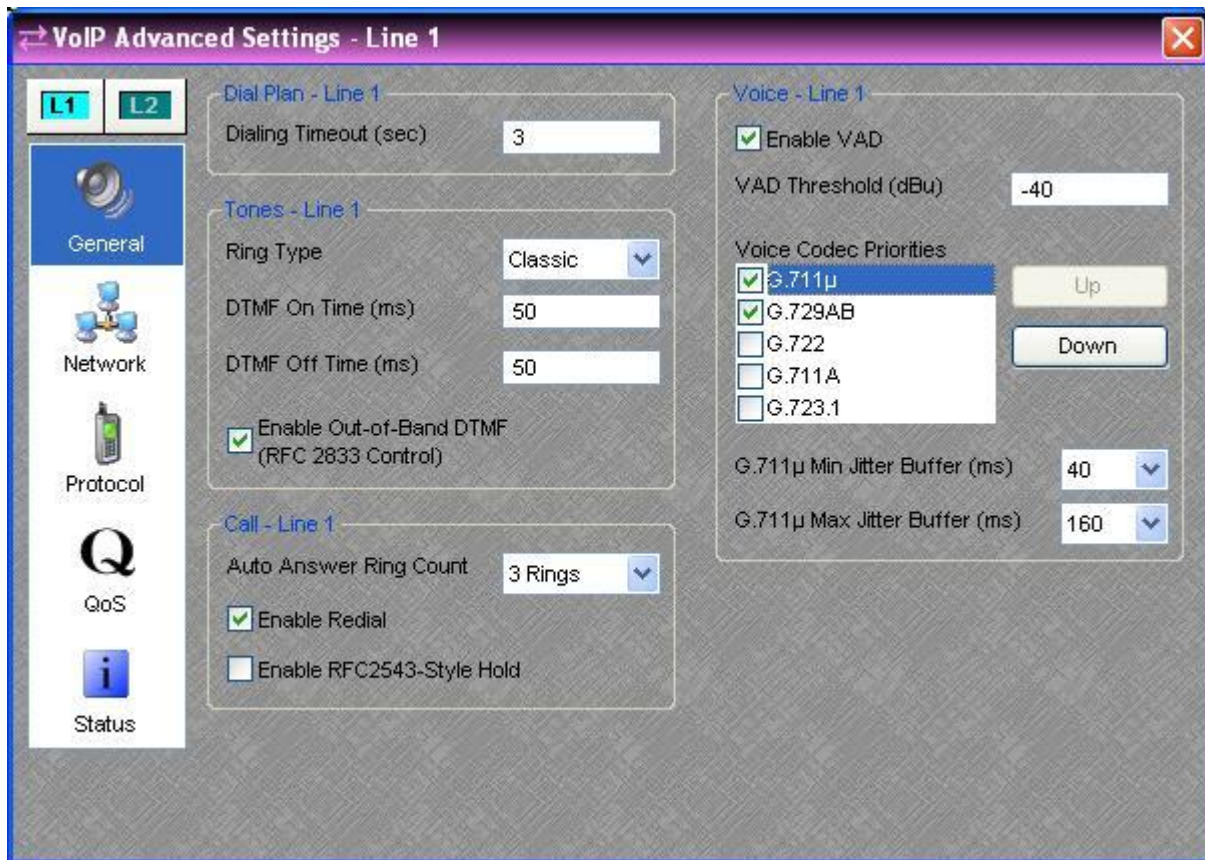


The **VoIP Advanced Settings – Line 1** screen is displayed next. Select **Network** in the left pane, and modify the **Network – Global** section as desired to match the network configuration. Note that the network setting is global and applies to both channels.



Select **General** in the left pane. For **Voice Codec Priorities**, select and rearrange the desired codec. The screenshot below shows the codec configuration used in the compliance testing.

Select **L2** in the upper left portion of the screen, and repeat the same procedure for the second channel.



Select **Protocol** in the left pane. Enter the following values for the specified fields, and retain the default values for the remaining fields.

- **Subscriber Name:** The first user primary handle from **Section 6.2**.
- **Proxy Username:** The first user primary handle from **Section 6.2**.
- **Proxy Password:** The first user password from **Section 6.2**.
- **Proxy Address:** The IP address of the SIP Enablement Services signaling interface.

Select **L2** in the upper left portion of the screen, and repeat similar procedure for the second channel.

VoIP Advanced Settings - Line 1

Protocol - Line 1

Subscriber Number: 66006

Proxy Server - Line 1

Proxy Username: 66006

Proxy Password: ••••••

Reg. Expiration (sec): 3600

Proxy Discovery: Static

Proxy Address: 10.32.32.30

Proxy Port: 5060

Outbound Proxy Server - Line 1

Outbound Proxy Address:

Outbound Proxy Port: 5060

Save

8. Verification Steps

This section provides the tests that can be performed to verify proper configuration of Communication Manager, SIP Enablement Services, and AudiaFLEX VoIP-2.

8.1. Verify Communication Manager and SIP Enablement Services

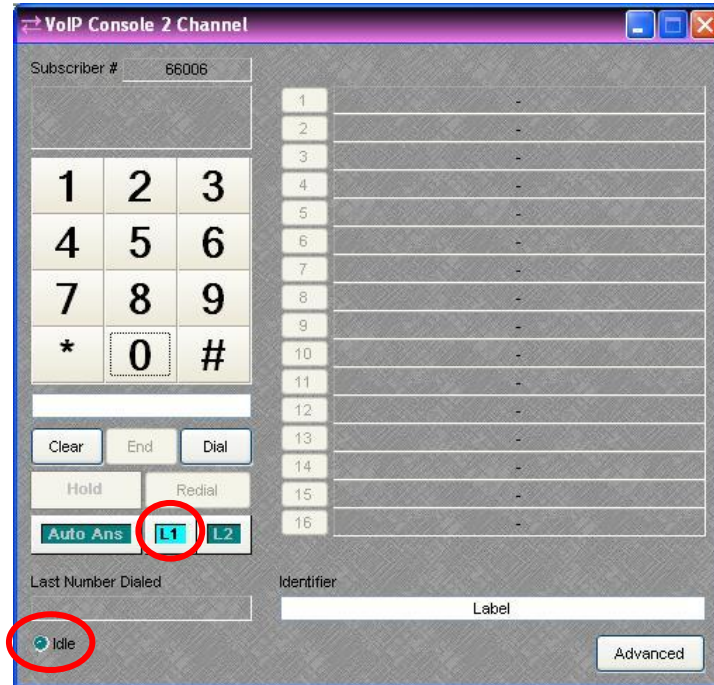
From the web interface of SIP Enablement Services, use Users → Search Registered Users from the left pane, to display the Registered Users screen. Verify that the users from **Section 6.2** are listed, as shown below.

The screenshot displays the Avaya Integrated Management SIP Server Management web interface. The top header shows the Avaya logo and the title "Integrated Management SIP Server Management". Below the header, there is a navigation pane on the left with a "Top" section containing "Users" and a "Communication Manager" section containing "Servers". The "Users" section is expanded, showing a list of options: Add, Default Profile, Delete, Edit, List, Password, Search, Manage All Registered Users, Search Registered Devices, Search Registered Users, Address Map Priorities, Adjunct Systems, Aggregator, Certificate Management, Conferences, Emergency Contacts, Export/Import to ProVision, Hosts, IM logs, and Servers. The "Search Registered Users" option is selected. The main content area is titled "Registered Users on 10.32.32.30" and shows a table of registered contacts. The table has three columns: "Handle and Name", "Address", and "Expires". There are three rows of data, each with a checkbox in the first column. The first row shows a user with handle 66001@br110.com, name Avaya, SIP, and address sip:66001@10.32.39.114:5061;avaya-sc-enabled;transport=tls. The second row shows a user with handle 66006@br110.com, name Biamp, VoIP-2-1, and address sip:66006@10.32.39.177:54910;transport=udp. The third row shows a user with handle 66007@br110.com, name Biamp, VoIP-2-2, and address sip:66007@10.32.39.177:54911;transport=udp. Below the table, there are two checkboxes: "Apply to all registered users with compatible devices on this Home." and "Apply to all registered users with compatible devices on this page." At the bottom, there is a "Task:" label, a dropdown menu set to "Reload-complete", and a "Submit" button.

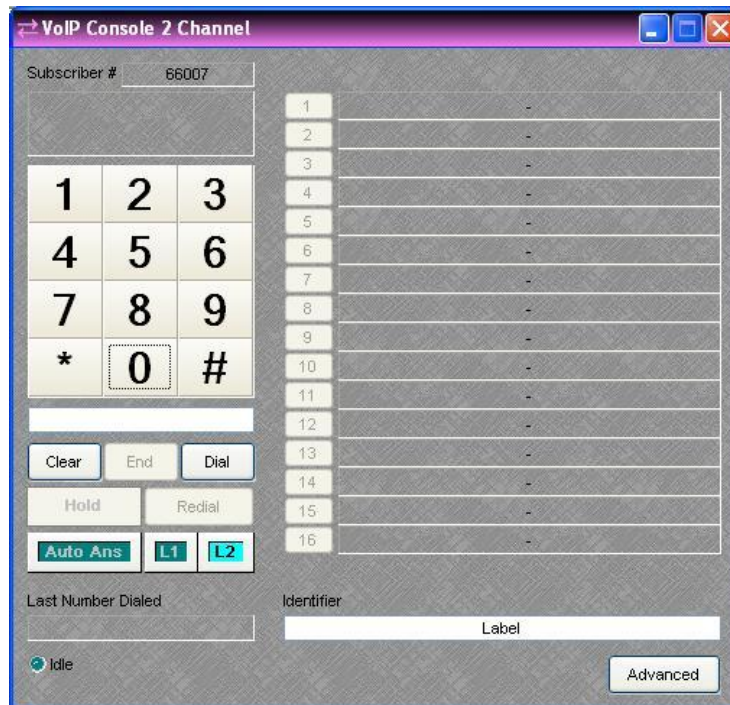
Handle and Name	Address	Expires
<input type="checkbox"/> 66001@br110.com Avaya, SIP	sip:66001@10.32.39.114:5061;avaya-sc-enabled;transport=tls	Wed, 12 Dec 2012 07:26:43 EST
<input type="checkbox"/> 66006@br110.com Biamp, VoIP-2-1	sip:66006@10.32.39.177:54910;transport=udp	Tue, 11 Dec 2012 09:56:56 EST
<input type="checkbox"/> 66007@br110.com Biamp, VoIP-2-2	sip:66007@10.32.39.177:54911;transport=udp	Tue, 11 Dec 2012 09:58:37 EST

8.2. Verify Biamp AudiaFLEX VoIP-2

Follow the procedures in **Section 7.2** to launch the **VoIP Console 2 Channel** screen. Click **L1**, and verify that the status is “Idle”, indicating successful registration.



Click **L2**, and verify that the status is also “Idle”, as shown below.



Make an incoming trunk call from the PSTN to one of the AudiaFLEX VoIP-2 channels. Verify that the display for the corresponding channel shows the calling party information, and that the status shows “Incoming Call”, as shown below. Click **Answer**.

The screenshot shows the 'VoIP Console 2 Channel' window. At the top, it displays 'Subscriber # 66006', '2/18/2012 6:48 AM', '908-844-5001', and 'G430 Station 00'. Below this is a numeric keypad with buttons for 1-9, *, 0, and #. To the right of the keypad is a vertical list of 16 channels, each with a number and a status indicator. The status indicators for all channels are currently dashes (-). Below the keypad are buttons for 'Clear', 'Reject', 'Answer', 'Hold', 'Redial', 'Auto Ans', 'L1', and 'L2'. At the bottom, there are fields for 'Last Number Dialed' and 'Identifier', and a 'Label' field. A status indicator at the bottom left shows a green light and the text 'Incoming Call'. An 'Advanced' button is located at the bottom right.

Verify that the call is connected with two-way talk paths, and that the status is updated to “Connected”.

The screenshot shows the 'VoIP Console 2 Channel' window after the call has been answered. The status indicator at the bottom left now shows a green light and the text 'Connected'. The 'Answer' button is now disabled, and the 'End' button is visible. The 'Hold' button is highlighted in red. The 'Auto Ans' button is also highlighted in green. The 'Last Number Dialed' field now contains the number '908-844-5001'. The 'Identifier' field is empty, and the 'Label' field contains the text 'Label'. The 'Advanced' button remains at the bottom right.

9. Conclusion

These Application Notes describe the configuration steps required for Biamp AudiaFLEX VoIP-2 to successfully interoperate with Avaya Aura® Communication Manager using Avaya Aura® SIP Enablement Services. All feature and serviceability test cases were completed with observations noted in **Section 2.2**.

10. Additional References

This section references the product documentation relevant to these Application Notes.

1. *Administering Avaya AuraTM Communication Manager*, Document 03-300509, Issue 5.2, Release 5.2, May 2009, available at <http://support.avaya.com>.
2. *Installing, Administering, Maintaining, and Troubleshooting Avaya Aura® SIP Enablement Services*, Document Number 03-600768, Issue 9.0, January 2011, available at <http://support.avaya.com>.
3. *AUDIA Help*, available as part of the Biamp Audia application.

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