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.

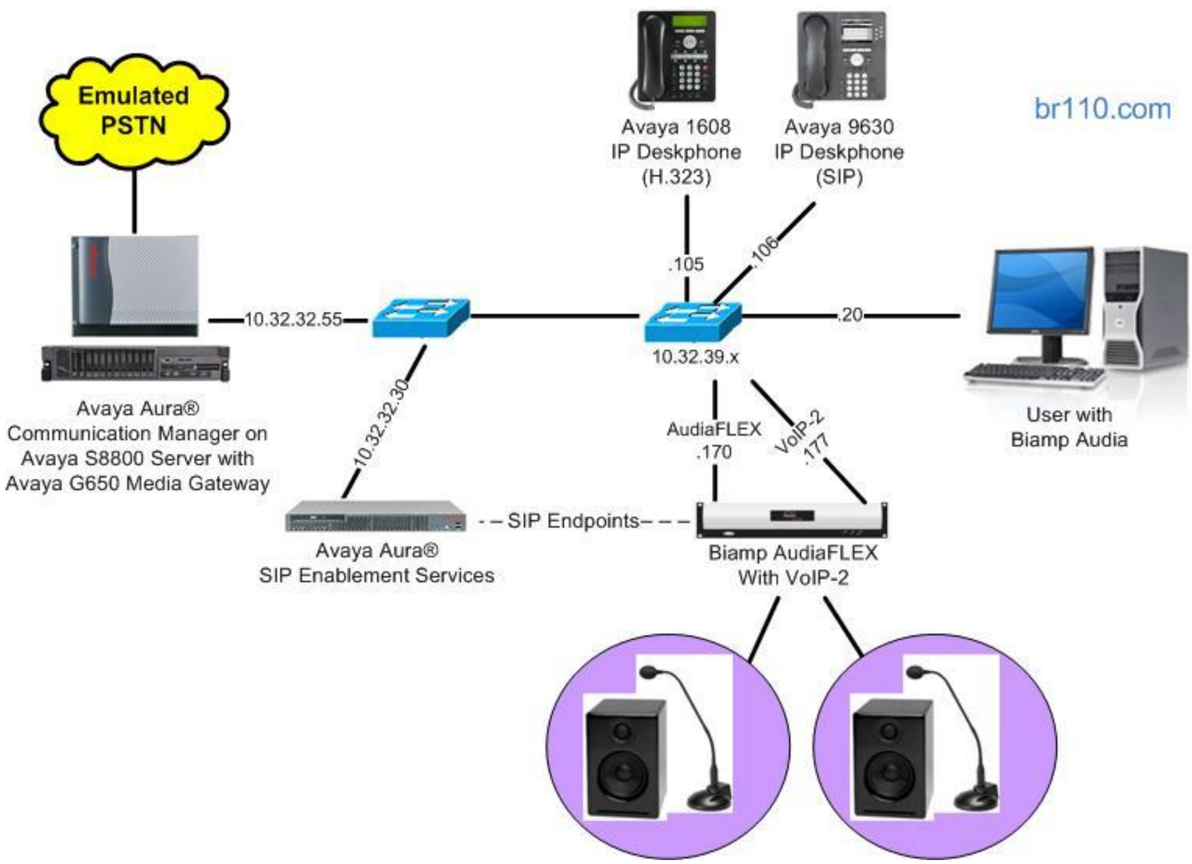


Figure 1: Compliance Testing Configuration

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                               Software Package: Standard
       Location: 1                              RFA System ID (SID): 1
       Platform: 12                             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/          Port/   Name/                        Room/       Cv1/ COR/   Cable/
 Hunt-to       Type      Surv GK NN         Move    Data Ext    Cv2  COS 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       Application Dial   CC  Phone Number    Trunk       Config  Dual
 Extension                 Prefix                     Selection   Set     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        Appl   CC  Phone Number     Config Trunk   Mapping      Calls
Extension                                  Set    Select  Mode         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

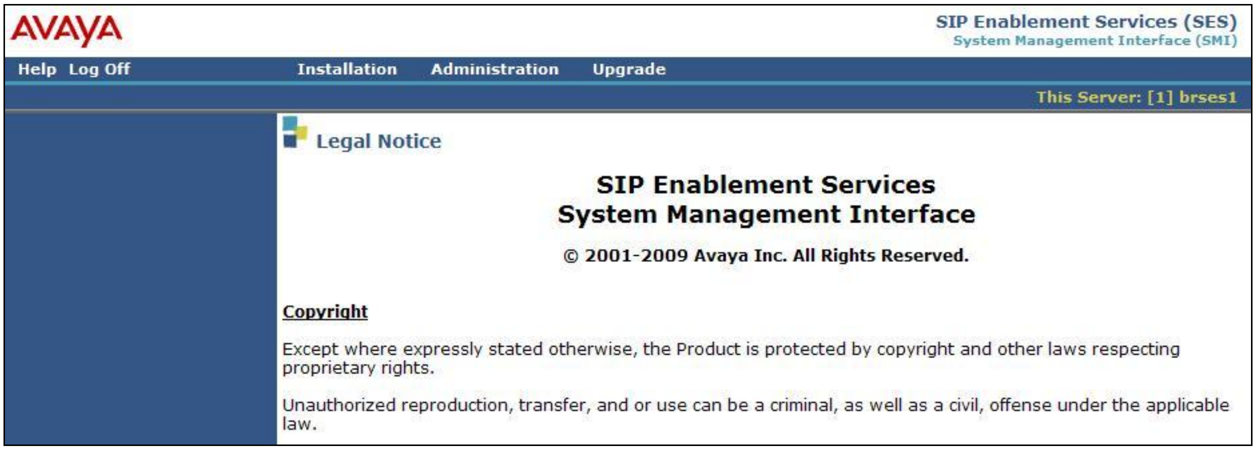
This section provides the procedures for configuring SIP Enablement Services. The procedures include the following areas:

- 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.

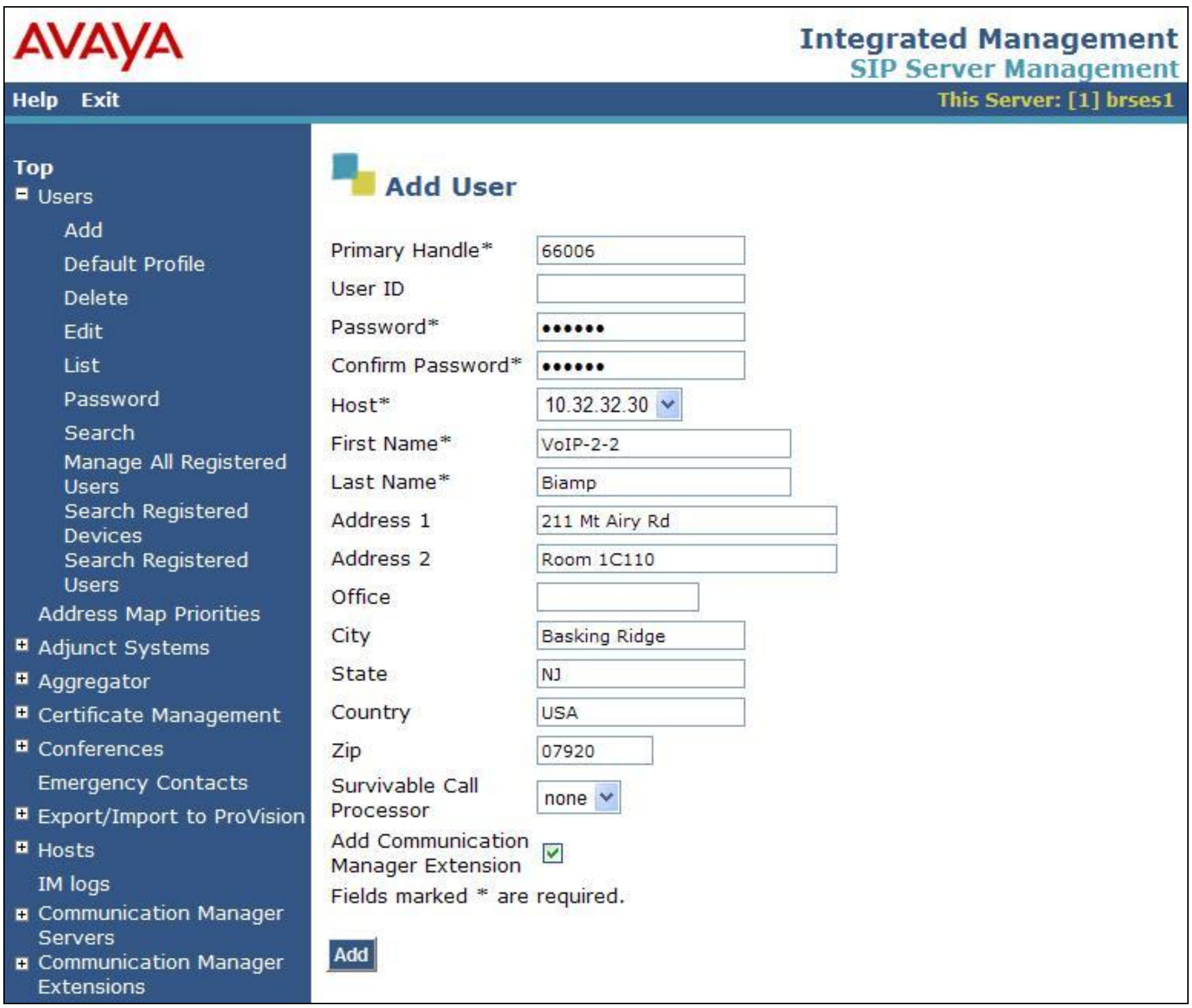


The **Top** screen is displayed next.

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.



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.

7. Configure Biamp AudiaFLEX VoIP-2

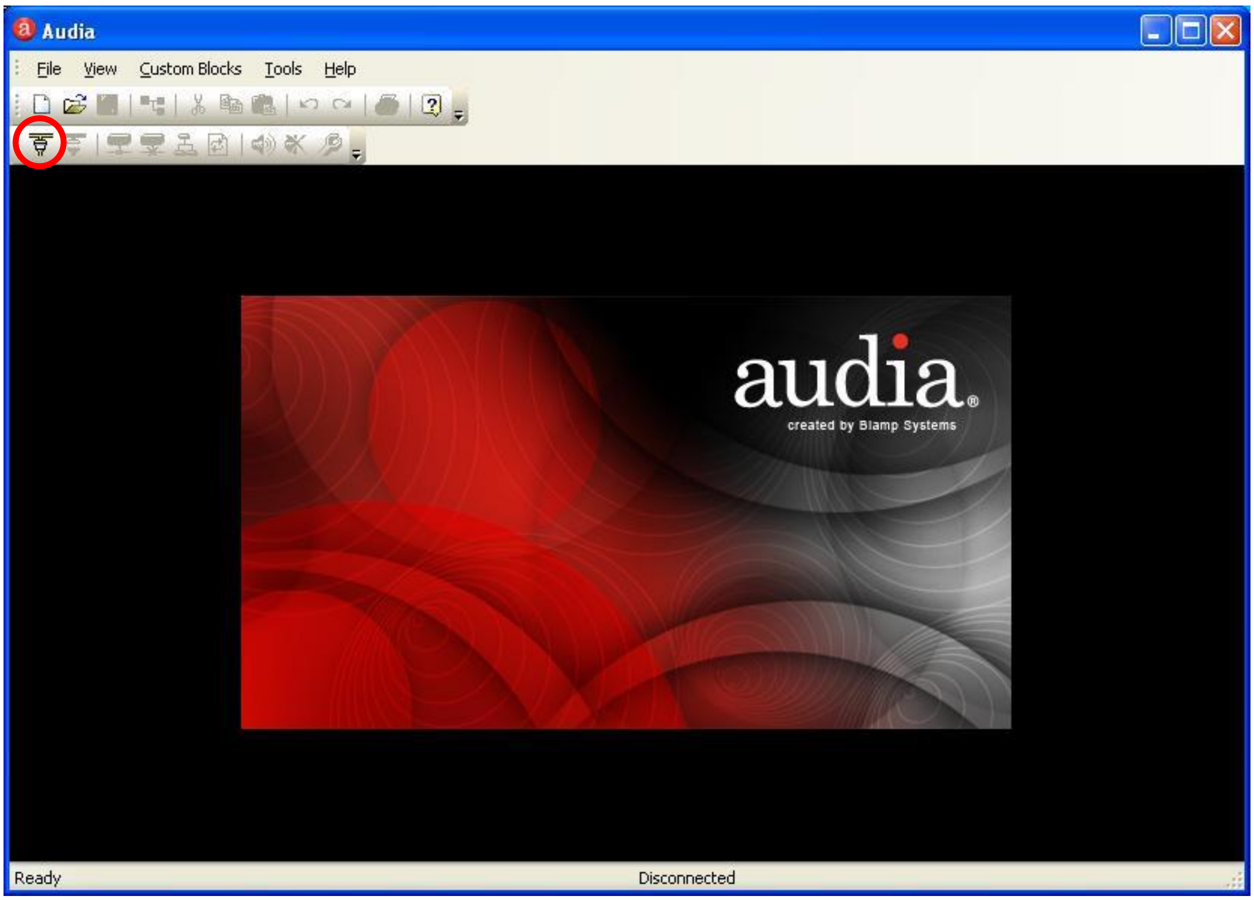
This section provides the procedures for configuring AudiaFLEX VoIP-2. The procedures include the following areas:

- 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.

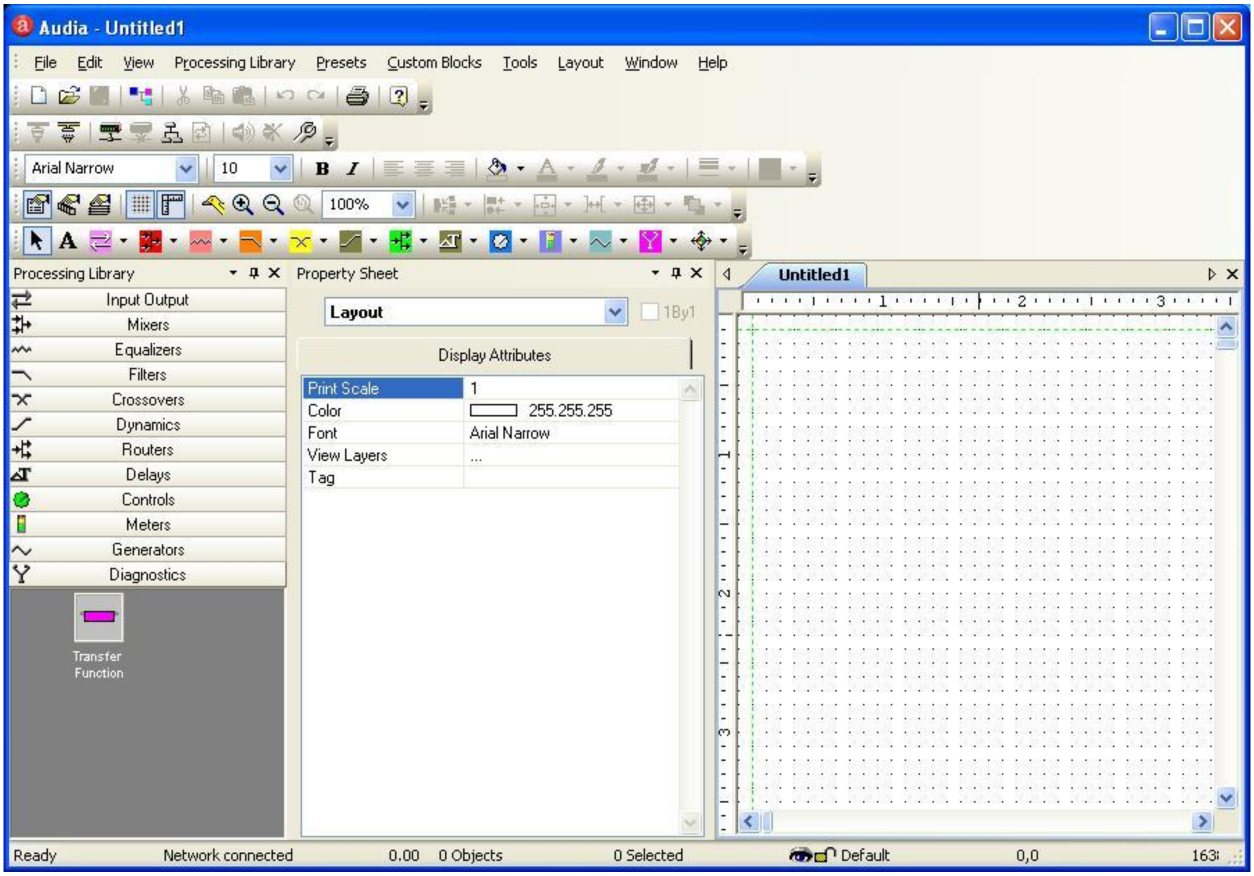
7.1. Launch Audia

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

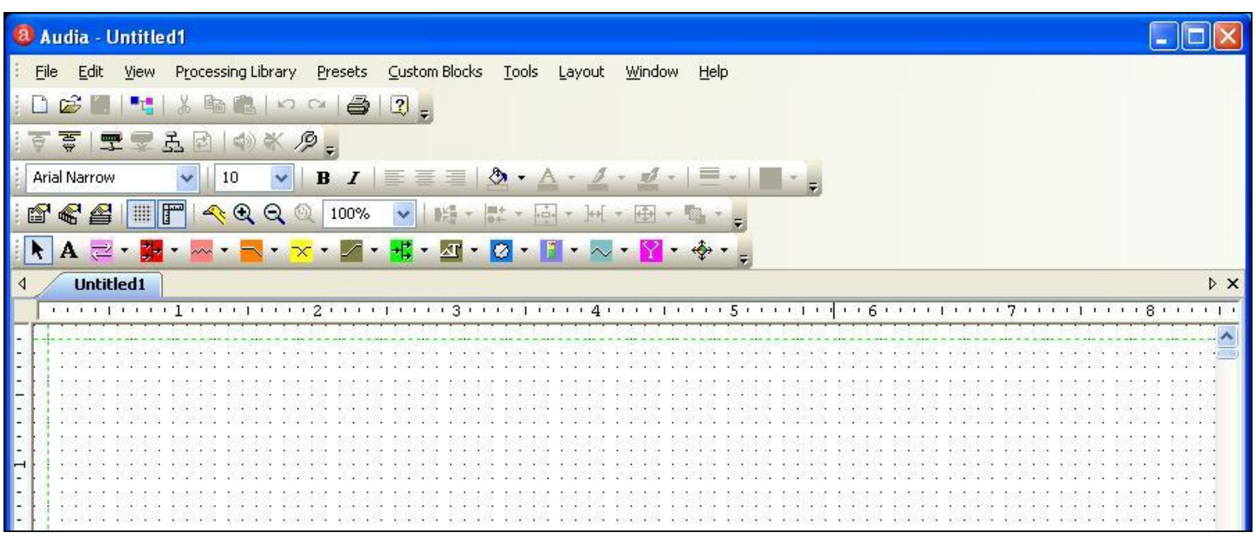


7.2. Administer Design Components

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.

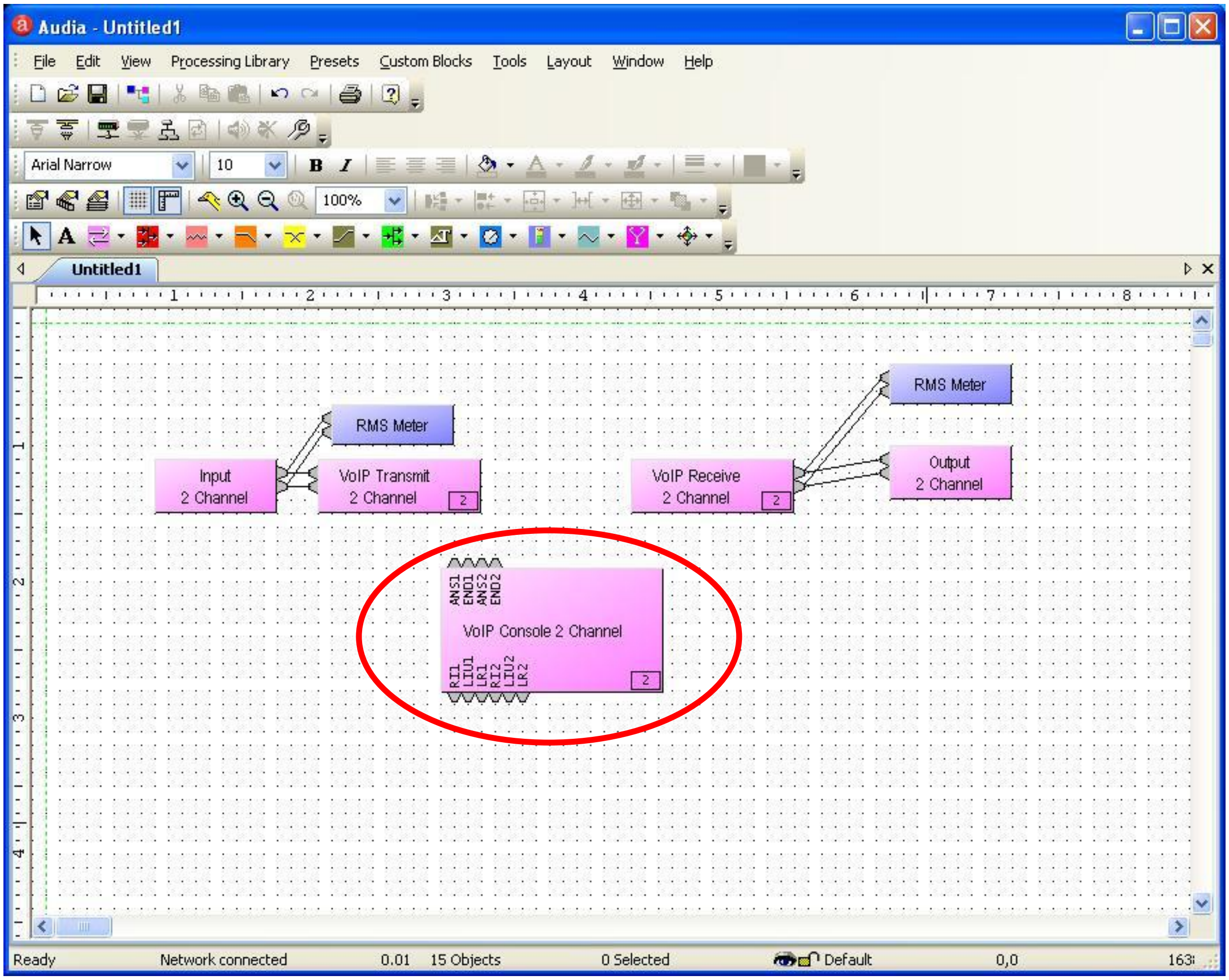


The **Audia** screen is updated as shown below.



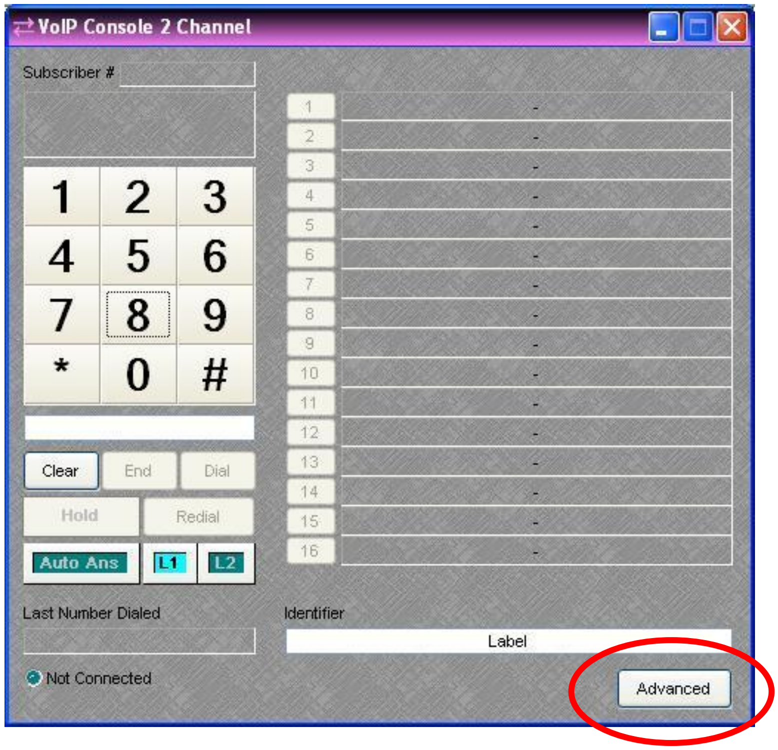
Follow [3] to place relevant component objects into the layout to match the system design. Below is the resultant layout used in the compliance testing. Note that both channels of the VoIP-2 card were used, as shown below.

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

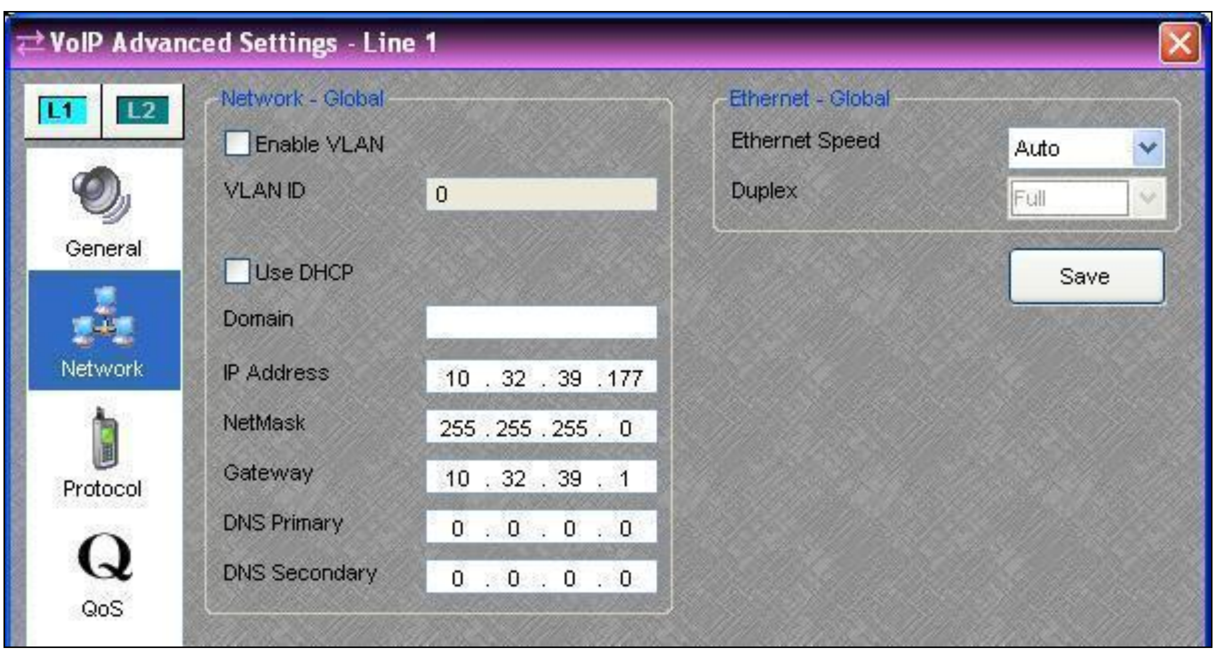


7.3. Administer VoIP Console

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

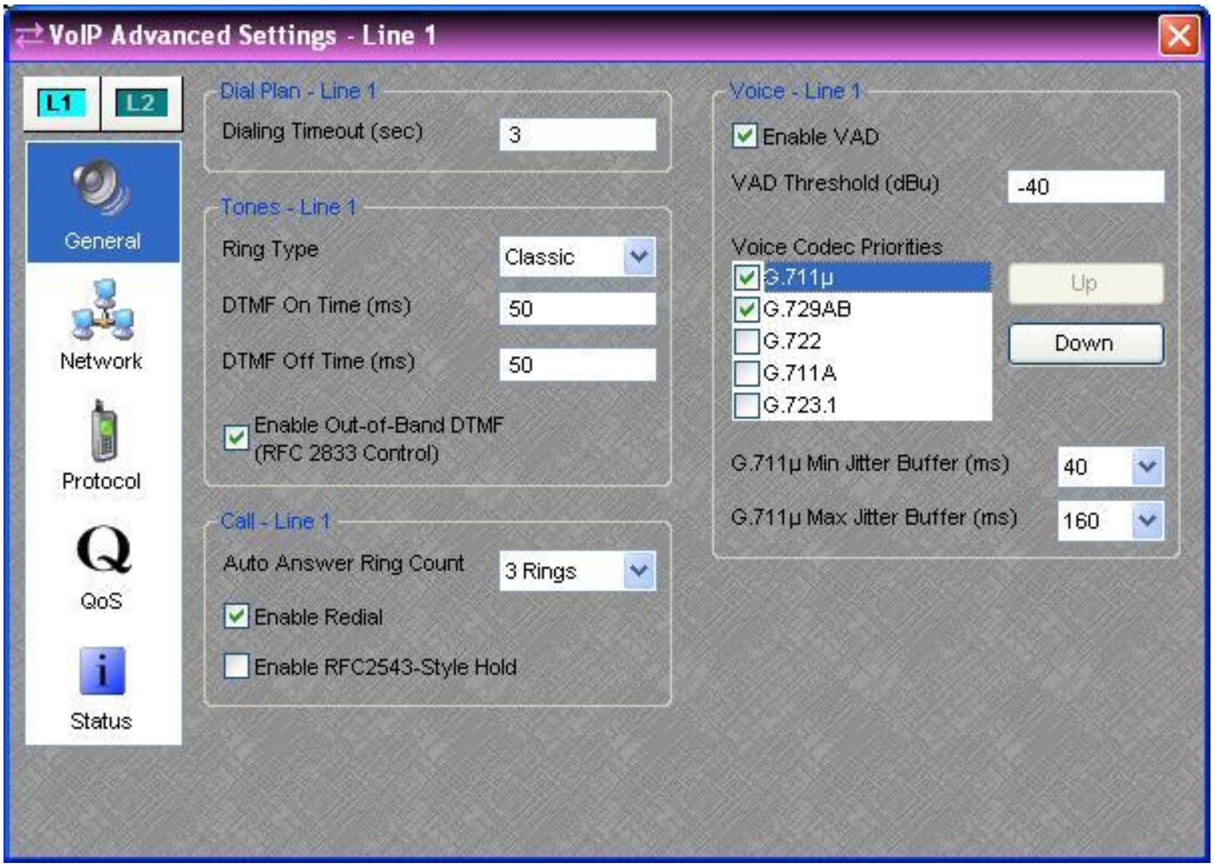


The **VoIP Advanced Settings – Line 1** screen is displayed next. Select **Network** in the left pane, and modify the **Network – Global** section as desired to match the network configuration. Note that the network setting is global and applies to both channels.



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.

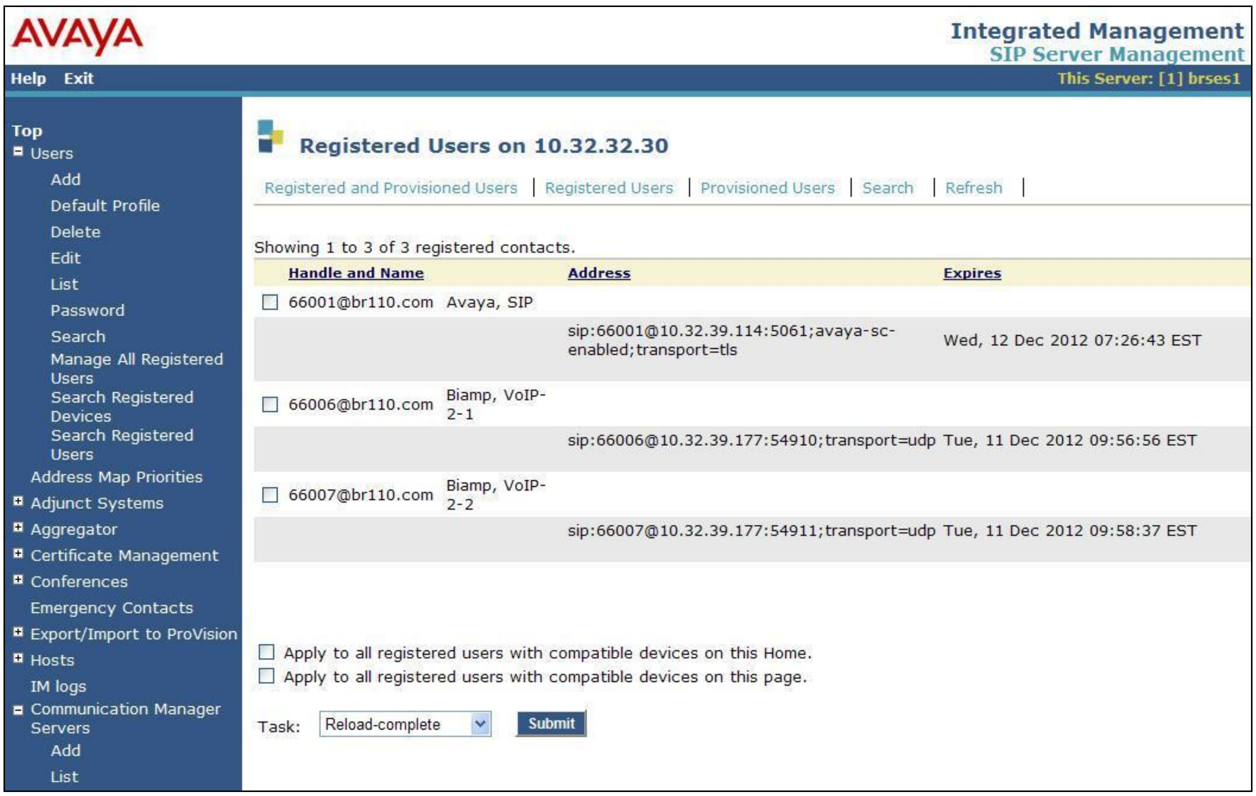
Select **L2** in the upper left portion of the screen, and repeat similar procedure for the second channel.

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.

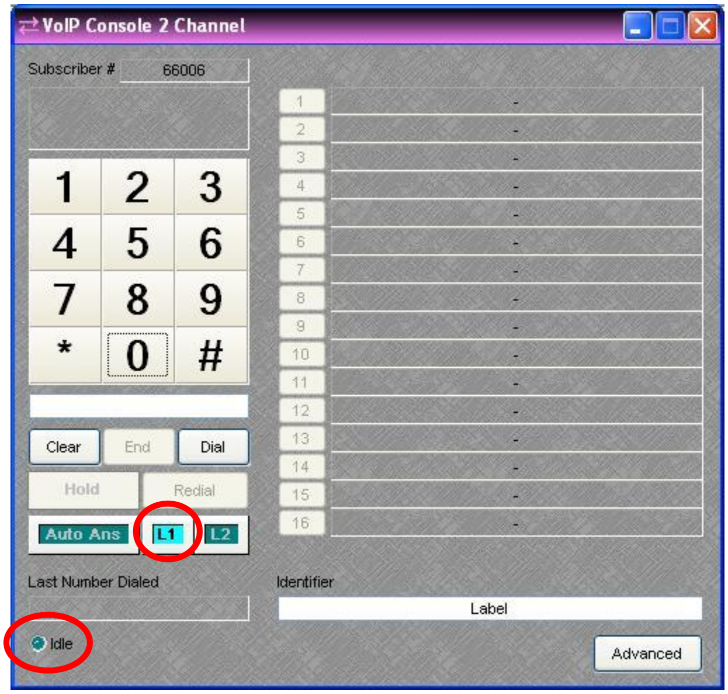
8.1. Verify Communication Manager and SIP Enablement Services

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.



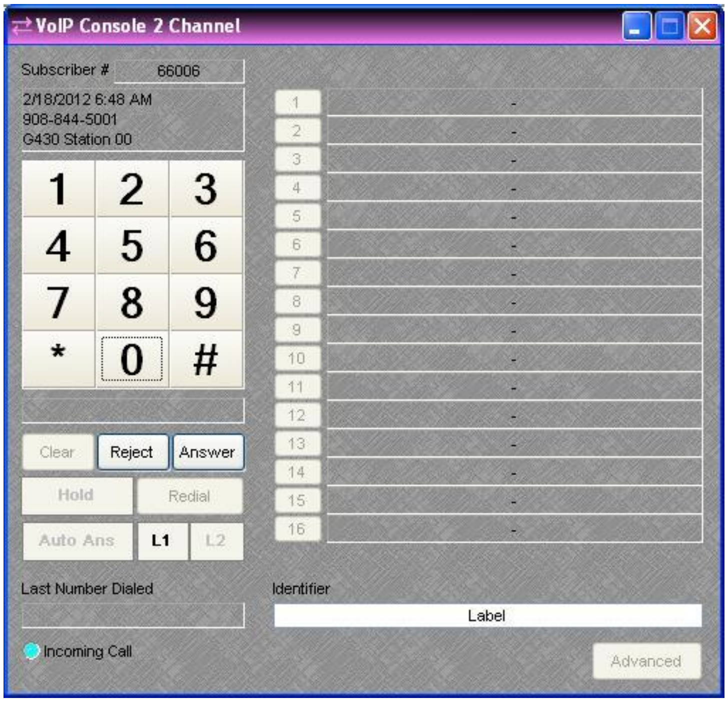
8.2. Verify Biamp AudiaFLEX VoIP-2

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**.



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

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 Aura^{TM} 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.

Please e-mail any questions or comments pertaining to these Application Notes along with the full title name and filename, located in the lower right corner, directly to the Avaya DevConnect Program at devconnect@avaya.com.