Avaya Solution & Interoperability Test Lab

Application Notes for Biamp Tesira SVC-2 with Avaya Aura® Communication Manager and Avaya Aura® Session Manager - Issue 1.0

Abstract

These Application Notes describe the configuration steps required to integrate the Biamp Tesira SVC-2 with Avaya Aura® Communication Manager and Avaya Aura® Session Manager. Biamp Tesira SVC-2 is a modular VoIP card for use with Biamp Tesira SERVER-IO, a conferencing platform. Biamp Tesira SVC-2 allows a Tesira system to connect directly to IP-based telephone systems. Biamp Tesira SVC-2 supports a range of telephony functions, including dial, hold, resume, transfer, and conference, and registers with Avaya Aura® Session Manager as a SIP endpoint.

Readers should pay attention to **Section 2**, in particular the scope of testing as outlined in **Section 2.1** as well as the observations noted in **Section 2.2**, to ensure that their own use cases are adequately covered by this scope and results.

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 to integrate the Biamp Tesira SVC-2 with Avaya Aura® Communication Manager and Avaya Aura® Session Manager. Biamp Tesira SVC-2 is a modular VoIP card for use with Biamp Tesira SERVER-IO, a conferencing platform. Biamp Tesira SVC-2 allows a Tesira system to connect directly to IP-based telephone systems. Biamp Tesira SVC-2 supports a range of telephony functions, including dial, hold, resume, transfer, and conference, and registers with Avaya Aura® Session Manager as a SIP endpoint.

With the Biamp Tesira SVC-2 card, Biamp Tesira SERVER-IO can establish or participate in an audio conference with local stations or PSTN users via Avaya Aura® Communication Manager and Avaya Aura® Session Manager. Other participants in a meeting room or class room, where Biamp Tesira SERVER-IO is located, could then communicate with the conference participants via a microphone and speakerphone connected to Biamp Tesira SERVER-IO with the Biamp Tesira SVC-2.

Biamp has indicated that other products in the Tesira family share the same SIP stack and software version as v3.8.0.24, which was compliance tested. The differences between the other products are capacities. Therefore, this testing also applies to those products. See **Attachment 1** for details, or contact Tesira Support, as noted in **Section 2.3**.

2. General Test Approach and Test Results

The interoperability compliance test included feature and serviceability testing. The feature testing focused on establishing calls between Tesira SVC-2 card installed in Tesira SERVER-IO, Avaya SIP / H.323 IP Deskphones, and the PSTN, and exercising basic telephony features such as hold, mute, transfer and conference. Additional telephony features such as call forward, call coverage, and call pickup were also verified using Communication Manager Feature Access Codes (FACs).

The serviceability testing focused on verifying that the Tesira SVC-2 card came back into service after re-connecting the Ethernet cable to Tesira SERVER-IO.

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.

Avaya recommends our customers implement Avaya solutions using appropriate security and encryption capabilities enabled by our products. The testing referenced in this DevConnect Application Note included the enablement of supported encryption capabilities in the Avaya products. Readers should consult the appropriate Avaya product documentation for further information regarding security and encryption capabilities supported by those Avaya products.

Support for these security and encryption capabilities in any non-Avaya solution component is the responsibility of each individual vendor. Readers should consult the appropriate vendor-supplied product documentation for more information regarding those products.

For the testing associated with this Application Note, the interface between Avaya systems and Biamp Tesira SVC-2 did not include use of any specific encryption features as requested by Biamp.

2.1. Interoperability Compliance Testing

Interoperability compliance testing covered the following features and functionality:

- SIP registration of Tesira SVC-2 with Session Manager
- Calls between Tesira SVC-2 and Avaya SIP/H.323 Deskphones with Direct IP Media (Shuffling) enabled and disabled.
- Calls between Tesira SVC-2 and the PSTN.
- UDP transport protocol.
- Support of G.711, G.729, and G.722 codecs.
- Proper recognition of DTMF tones.
- Basic telephony features including hold, mute, redial, multiple calls, blind/attended transfer, attended conference, and long duration calls.
- Extended telephony features using Communication Manager FACs for Call Forward, Call Coverage, Call Park/Unpark, and Call Pickup.
- Proper system recovery after a restart of the Tesira SERVER-IO and loss of IP network connectivity.

2.2. Test Results

All test cases passed with the following observations:

- Blind conference is not supported but attended/supervised conference is supported.
- There should only be one codec enabled on Tesira SVC-2 to avoid audio problems (i.e., audio noise) after a conference call is established.
- With Direct IP Media (i.e., Shuffling) enabled, there is a moment of audio noise heard on Tesira SVC-2 side when an Avaya SIP Deskphone resumes a held call. This occurs initially after the call is resumed and the audio problem clears. This issue isn't encountered if Shuffling is disabled.

2.3. Support

For technical support and information on Biamp Tesira SVC-2, contact Biamp Support at:

- Phone: +1 (503) 718-9257
- Website: https://support.biamp.com/Tesira
- Email: support@biamp.com

3. Reference Configuration

Figure 1 illustrates a sample configuration with an Avaya SIP-based network that includes the following products:

- Avaya Aura® Communication Manager running in a virtual environment with an Avaya G450 Media Gateway. Avaya G450 Media Gateway was connected to the PSTN via an ISDN-PRI trunk (not shown).
- Media resources in the Avaya G450 Media Gateway and Avaya Aura® Media Server.
- Avaya Aura® Session Manager connected to Communication Manager via a SIP trunk and acting as a Registrar/Proxy for SIP deskphones.
- Avaya Aura® System Manager used to configure Session Manager.
- Avaya 96x1 Series H.323 and SIP Deskphones.
- Avaya 1600 Series H.323 Deskphones.
- Biamp Tesira SVC-2 card installed in Biamp SERVER-IO.

Biamp Tesira SVC-2 registered with Session Manager and was configured as an Off-PBX Station (OPS) on Communication Manager.

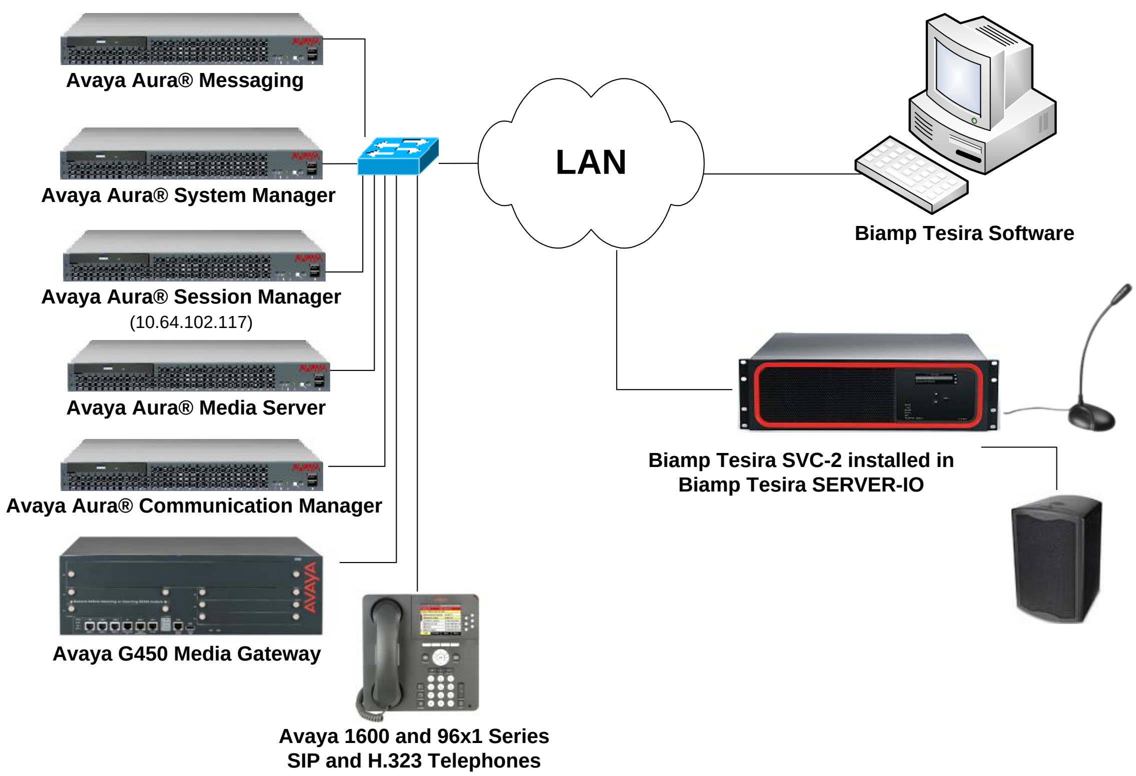


Figure 1: Avaya SIP Network with Biamp Tesira SVC-2 installed in Biamp SERVER-IO

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	8.0.1.0.0-FP1 (R018x.00.0.822.0 with Patch 25031)
Avaya G450 Media Gateway	FW 38.21.1
Avaya Aura® Media Server	v.7.8.0.393
Avaya Aura® Session Manager	8.0.1.0801007
Avaya Aura® System Manager	8.0.1.0 Build No. – 8.0.0.0.931077 Software Update Revision No: 8.0.1.0.038826 Feature Pack 1
Avaya Aura® Messaging	7.1.3.1.0-FP3SP1
Avaya 96x1 Series IP Deskphones	6.7104 (H.323) 7.1.4.0.11 (SIP)
Avaya 1600 Series IP Deskphones	1.3120 (H.323)
Biamp Tesira Software	v3.8.0.14
Biamp Tesira SVC-2 installed in Biamp Tesira SERVER-IO	v3.8.0.24

5. Configure Avaya Aura® Communication Manager

This section provides the procedure for configuring Communication Manager. The procedure includes the following areas:

- Verify Communication Manager license
- Administer IP Node Names
- Administer IP Network Region and IP Codec Set
- Administer SIP Trunk to Session Manager
- Administer AAR Call Routing

Use the System Access Terminal (SAT) to configure Communication Manager and log in with appropriate credentials.

Note: The SIP station configuration for Biamp Tesira SVC-2 is configured through Avaya Aura® System Manager in **Section 6.3**.

5.1. Verify Communication Manager License

Using the SAT, verify that the Off-PBX Telephones (OPS) option is enabled on the **system-parameters customer-options** form. The license file installed on the system controls these options. If a required feature is not enabled, contact an authorized Avaya sales representative.

On **Page 1**, verify that the number of OPS stations allowed in the system is sufficient for the number of SIP endpoints that will be deployed.

```
display system-parameters customer-options                      Page   1 of  12
                                OPTIONAL FEATURES

     G3 Version: V18                                 Software Package: Enterprise
       Location: 2                                  System ID (SID): 1
       Platform: 28                                 Module ID (MID): 1

                                                              USED
                                Platform Maximum Ports: 48000    87
                                      Maximum Stations: 36000 26
                              Maximum XMOBILE Stations: 36000 0
                    Maximum Off-PBX Telephones - EC500: 41000 0
                    Maximum Off-PBX Telephones -   OPS: 41000 17
                    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   0




        (NOTE: You must logoff & login to effect the permission changes.)
```

5.2. Administer IP Node Names

In the **IP Node Names** form, assign an IP address and host name for Communication Manager (*procr*) and Session Manager (*devcon-sm*). The host names will be used in other configuration screens of Communication Manager.

```
change node-names ip                                            Page   1 of   2
                                  IP NODE NAMES
    Name              IP Address
default             0.0.0.0
devcon-aes          10.64.102.119
devcon-ams          10.64.102.118
devcon-sm           10.64.102.117
procr               10.64.102.115
procr6              ::

( 6  of 6    administered node-names were displayed )
Use 'list node-names' command to see all the administered node-names
Use 'change node-names ip xxx' to change a node-name 'xxx' or add a node-name
```

5.3. Administer IP Network Region and IP Codec Set

In the **IP Network Region** form, the **Authoritative Domain** field is configured to match the domain name configured on Session Manager. In this configuration, the domain name is *avaya.com*. By default, **IP-IP Direct Audio** (shuffling) is enabled to allow audio traffic to be sent directly between IP endpoints without using media resources in the Avaya G450 Media Gateway or Avaya Aura® Media Server. The **IP Network Region** form also specifies the **IP Codec Set** to be used for calls routed over the SIP trunk to Session Manager.

```
change ip-network-region 1                                      Page   1 of  20
                               IP NETWORK REGION
  Region: 1
Location: 1       Authoritative Domain: avaya.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: 50999
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
```

In the **IP Codec Set** form, select the audio codec type supported for calls routed over the SIP trunk to Tesira SVC-2. The form is accessed via the **change ip-codec-set 1** command. Note that IP codec set '1' was specified in IP Network Region '1' shown above. The default settings of the **IP Codec Set** form are shown below. Tesira SVC-2 was tested using G.711, G.722 and G.729 codecs. Specify the desired codecs in the **IP Codec Set** form as per customer requirements.

```
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:
 3:
```

5.4. Administer SIP Trunk to Session Manager

Prior to configuring a SIP trunk group for communication with Session Manager, a SIP signaling group must be configured. Configure the **Signaling Group** form as follows:

- Set the **Group Type** field to *sip*.
- Set the **IMS Enabled** field to *n*.
- The **Transport Method** field was set to *tls*.
- Set the **Enforce SIPS URI for SRTP** field to *n*.
- Specify Communication Manager (*procr*) and the Session Manager as the two ends of the signaling group in the **Near-end Node Name** field and the **Far-end Node Name** field, respectively. These field values are taken from the **IP Node Names** form.
- Ensure that the TLS port value of *5061* is configured in the **Near-end Listen Port** and the **Far-end Listen Port** fields.
- The preferred codec for the call will be selected from the IP codec set assigned to the IP network region specified in the **Far-end Network Region** field.
- Enter the domain name of Session Manager in the **Far-end Domain** field. In this configuration, the domain name is *avaya.com*.
- The **Direct IP-IP Audio Connections** field was enabled on this form.
- The **DTMF over IP** field should be set to the default value of *rtp-payload*.
- Enable **Initial IP-IP Direct Media**.

Communication Manager supports DTMF transmission using RFC 2833. The default values for the other fields may be used.

```
add signaling-group 10                                        Page   1 of   2
                                SIGNALING GROUP

 Group Number: 10              Group Type: sip
  IMS Enabled? n        Transport Method: tls
        Q-SIP? n
     IP Video? n                                   Enforce SIPS URI for SRTP? n
  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
Alert Incoming SIP Crisis Calls? n
   Near-end Node Name: procr                 Far-end Node Name: devcon-sm
 Near-end Listen Port: 5061                Far-end Listen Port: 5061
                                        Far-end Network Region: 1

Far-end Domain: avaya.com
                                             Bypass If IP Threshold Exceeded? n
Incoming Dialog Loopbacks: eliminate                  RFC 3389 Comfort Noise? n
         DTMF over IP: rtp-payload             Direct IP-IP Audio Connections? y
Session Establishment Timer(min): 3                     IP Audio Hairpinning? n
         Enable Layer 3 Test? y                    Initial IP-IP Direct Media? y
H.323 Station Outgoing Direct Media? n             Alternate Route Timer(sec): 6
```

Configure the **Trunk Group** form as shown below. This trunk group is used for SIP calls to/from Tesira SVC-2, Avaya SIP Deskphones, and Avaya Aura® Messaging. Set the **Group Type** field to *sip*, set the **Service Type** field to *tie*, specify the signaling group associated with this trunk group in the **Signaling Group** field, and specify the **Number of Members** supported by this SIP trunk group. Configure the other fields in bold and accept the default values for the remaining fields.

```
add trunk-group 10                                                  Page   1 of  22
                                TRUNK GROUP

Group Number: 10                   Group Type: sip            CDR Reports: y
  Group Name: To devcon-sm                COR: 1        TN: 1        TAC: 1010
   Direction: two-way        Outgoing Display? n
 Dial Access? n                                     Night Service:
Queue Length: 0
Service Type: tie                   Auth Code? n
                                              Member Assignment Method: auto
                                                       Signaling Group: 10
                                                     Number of Members: 10
```

5.5. Administer AAR Call Routing

SIP calls to Session Manager are routed over a SIP trunk via AAR call routing. Configure the AAR analysis form and enter add an entry that routes digits beginning with '78' to route pattern 10 as shown below.

```
change aar analysis 78                                                 Page   1 of   2
                             AAR DIGIT ANALYSIS TABLE
                                   Location: all           Percent Full: 1

          Dialed           Total      Route     Call   Node   ANI
          String          Min  Max   Pattern    Type   Num    Reqd
    78                    5    5      10        lev0          n
```

Configure a preference in **Route Pattern** 10 to route calls over SIP trunk group 10 as shown below.

```
change route-pattern 10                                                Page   1 of   3
                    Pattern Number: 10     Pattern Name: To devcon-sm
    SCCAN? n    Secure SIP? n     Used for SIP stations? n

    Grp FRL NPA Pfx Hop Toll No.  Inserted                              DCS/ IXC
    No          Mrk Lmt List Del  Digits                                QSIG
                             Dgts                                       Intw
 1: 10   0                                                               n   user
 2:                                                                      n   user
 3:                                                                      n   user
 4:                                                                      n   user
 5:                                                                      n   user
 6:                                                                      n   user

     BCC VALUE  TSC CA-TSC    ITC BCIE Service/Feature PARM Sub  Numbering LAR
    0 1 2 M 4 W     Request                                 Dgts Format
 1: y y y y y n  n            rest                               unk-unk   none
 2: y y y y y n  n            rest                                         none
```

6. Configure Avaya Aura® Session Manager

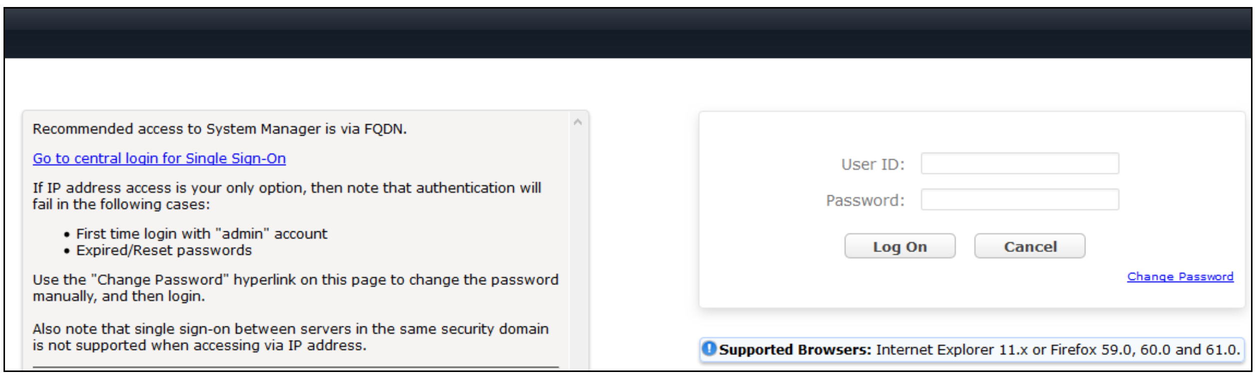
This section provides the procedure for configuring Session Manager. The procedures include the following areas:

- Launch System Manager
- Set Network Transport Protocol
- Administer SIP User

Note: It is assumed that basic configuration of Session Manager has already been performed. This section will focus on the configuration of a SIP user for Biamp Tesira SVC-2.

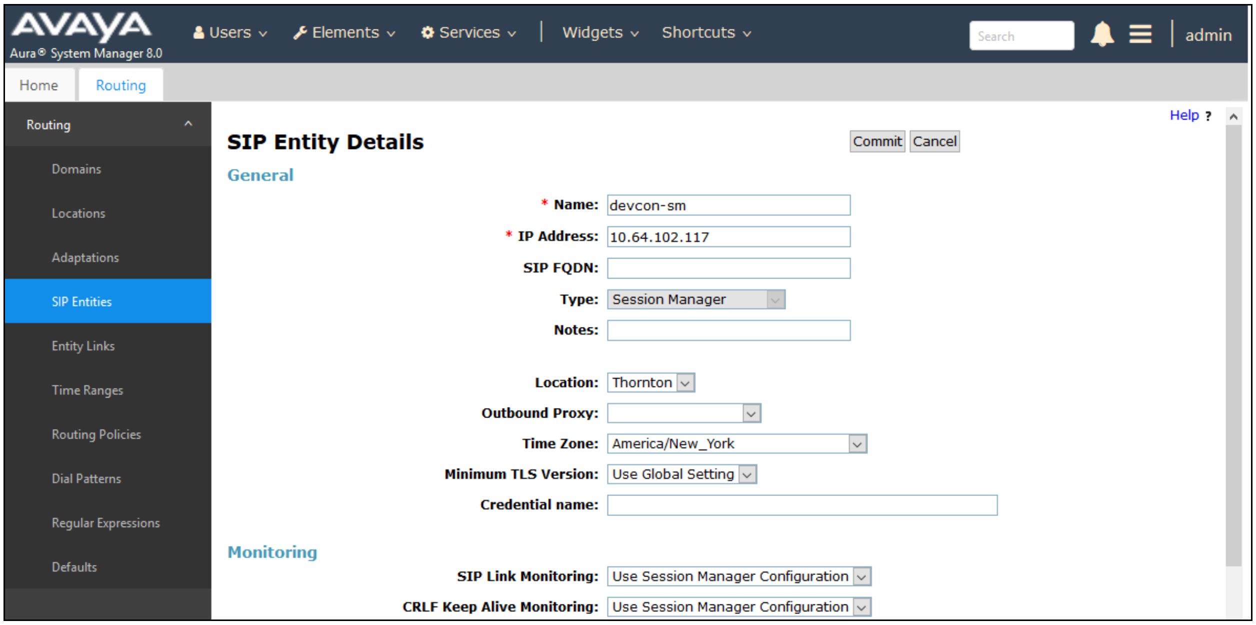
6.1. Launch System Manager

Access the System Manager Web interface by using the URL "https://ip-address" in an Internet browser window, where "ip-address" is the IP address of the System Manager server. Log in using the appropriate credentials.

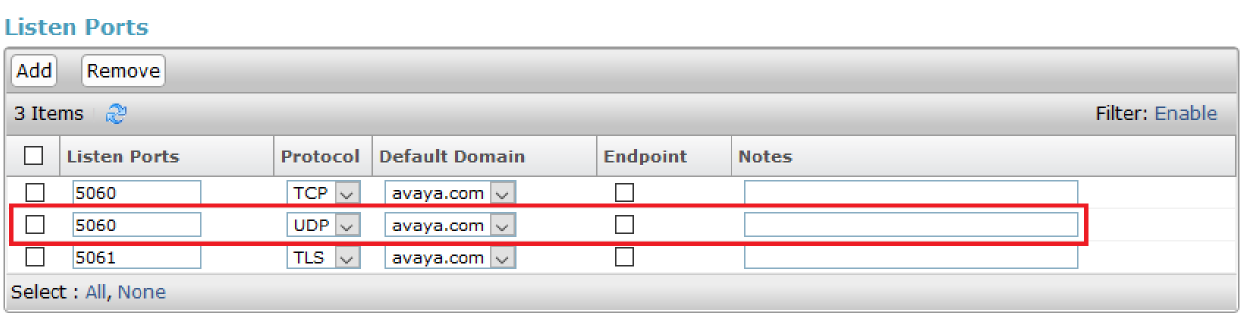


6.2. Set Network Transport Protocol

From the System Manager **Home** screen, select **Elements → Routing → SIP Entities** and edit the SIP Entity for Session Manager shown below.

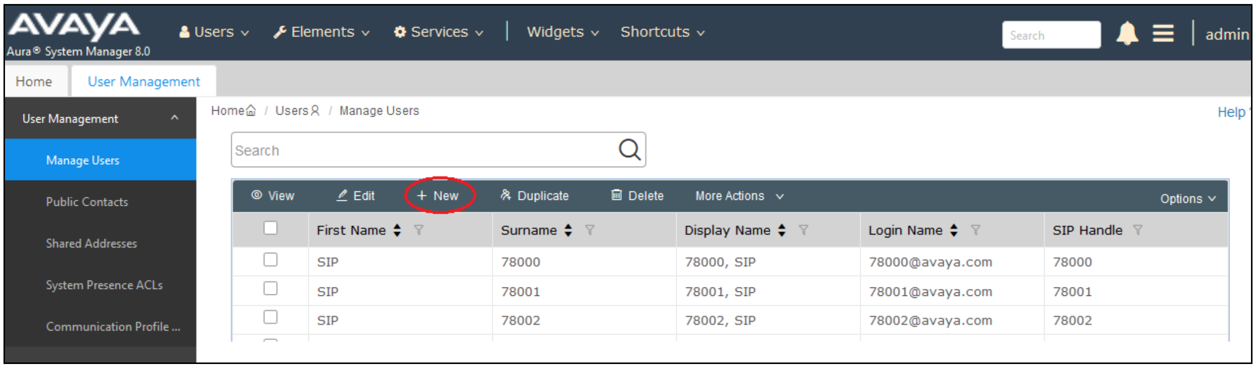


Scroll down to the **Listen Ports** section and verify that the transport network protocol used by Tesira-SVC2 is specified in the list below. For the compliance test, the solution used UDP network transport.



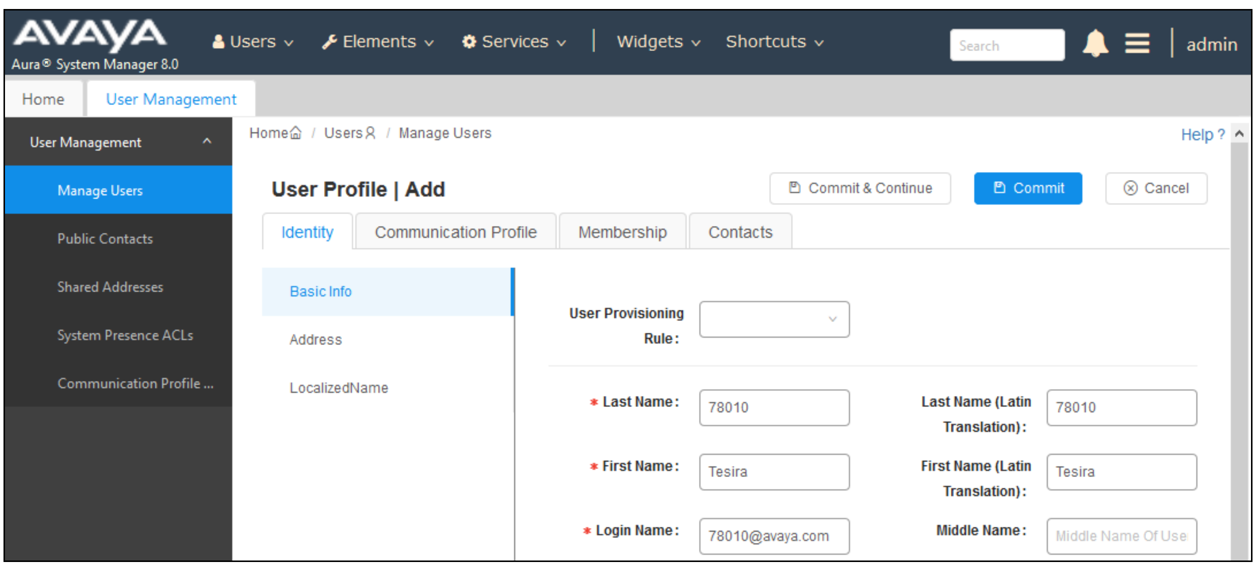
6.3. Administer SIP User

In the **Home** screen (not shown), select **Users → User Management → Manage Users** to display the **User Management** screen below. Click **New** to add a user.



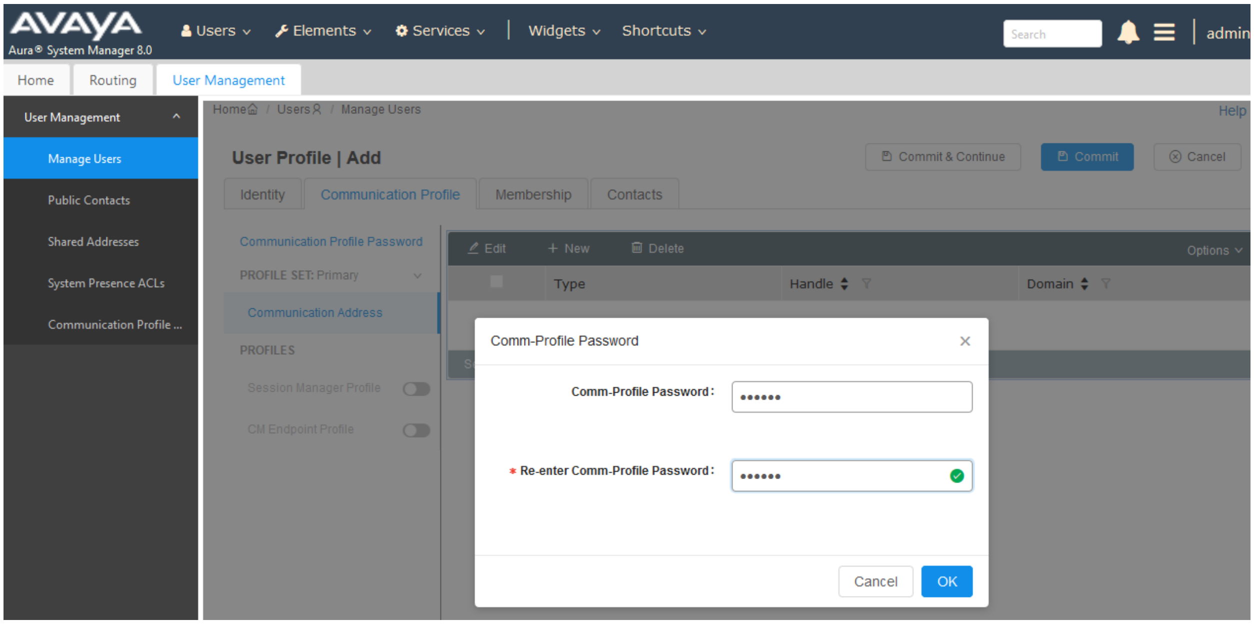
6.3.1. Identity

The **New User Profile** screen is displayed. Enter desired **Last Name** and **First Name**. For **Login Name**, enter "*<ext>*@*<domain>*", where "*<ext>*" is the desired Tesira SVC-2 SIP extension and "*<domain>*" is the applicable SIP domain name from **Section 5.3**. Retain the default values in the remaining fields.



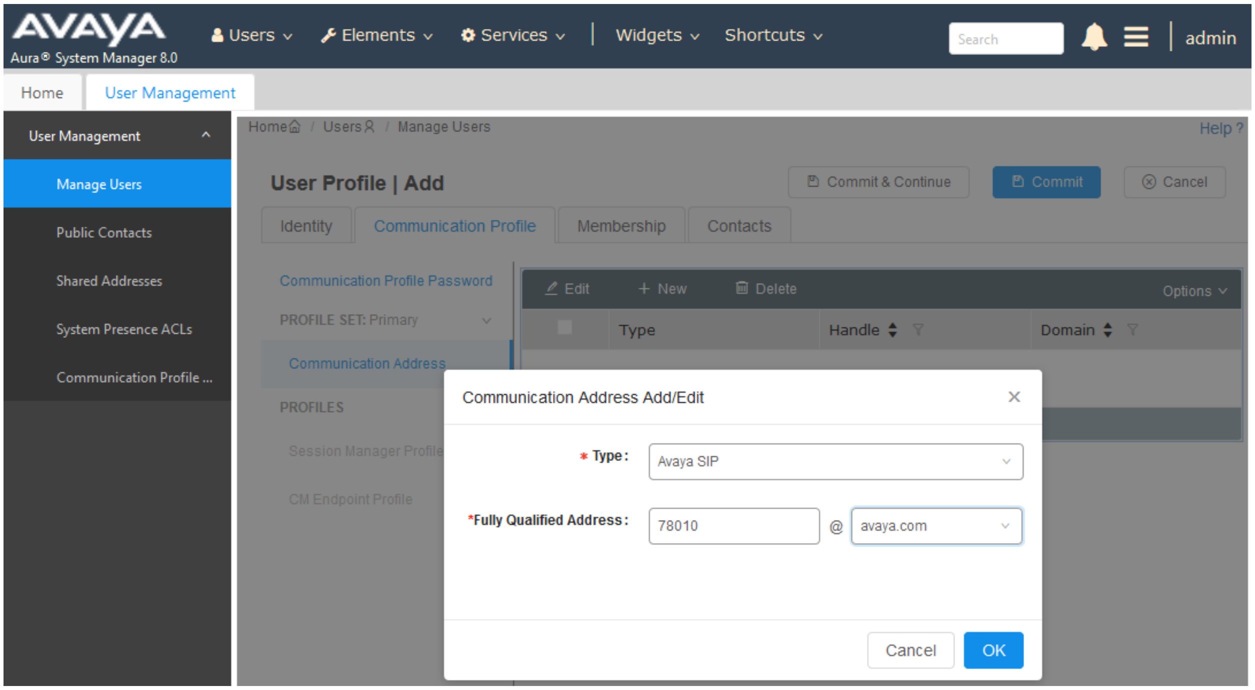
6.3.2. Communication Profile

Select the **Communication Profile** tab. Next, click on **Communication Profile Password**. For **Comm-Profile Password** and **Re-enter Comm-Profile Password**, enter the desired password for the SIP user to use for registration. Click **OK**.



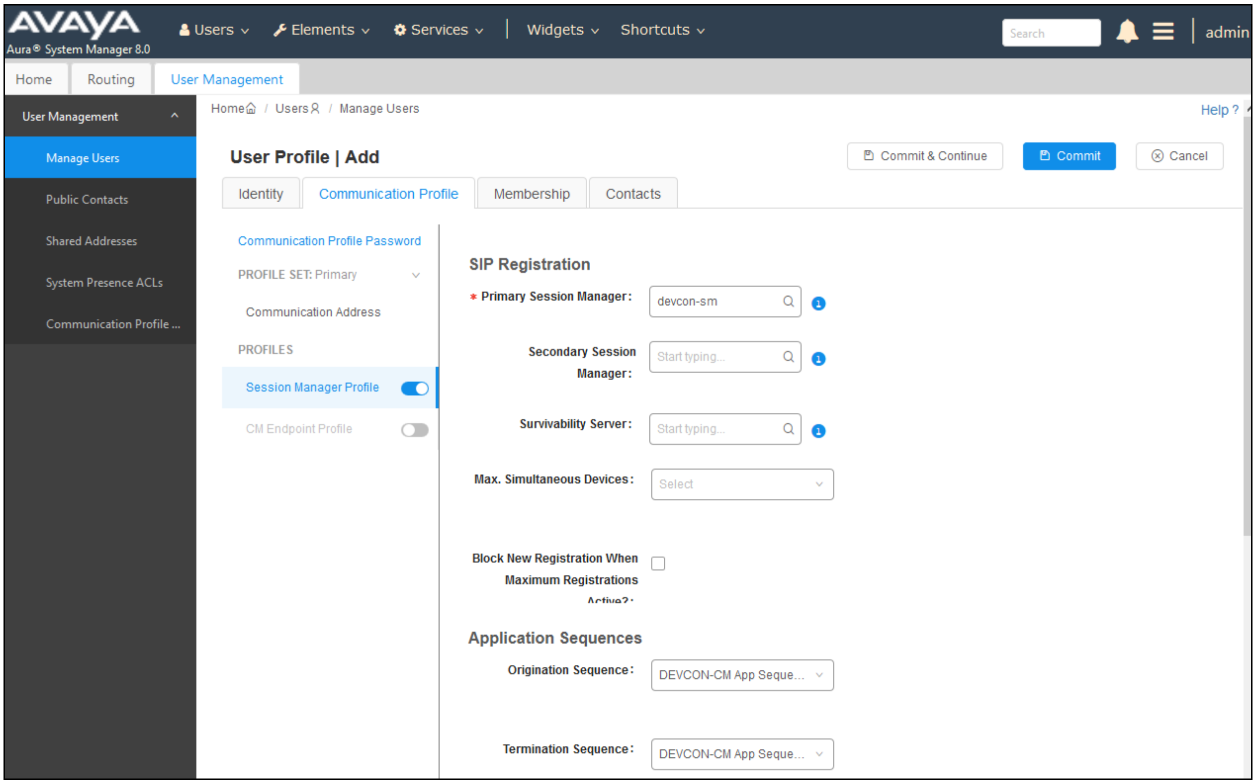
6.3.3. Communication Address

Click on **Communication Address** and then click **New** to add a new entry. The **Communication Address Add/Edit** dialog box is displayed as shown below. For **Type**, select *Avaya SIP*. For **Fully Qualified Address**, enter the SIP user extension and select the domain name to match the login name from **Section 6.3.1**. Click **OK**.

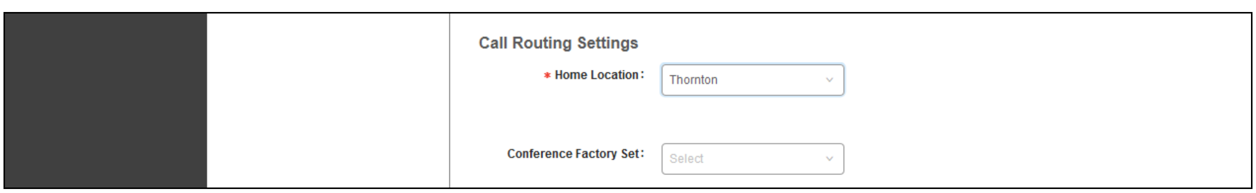


6.3.4. Session Manager Profile

Click on toggle button by **Session Manager Profile**. For **Primary Session Manager**, **Origination Application Sequence**, **Termination Application Sequence**, and **Home Location**, select the values corresponding to the applicable Session Manager and Communication Manager. Retain the default values in the remaining fields.

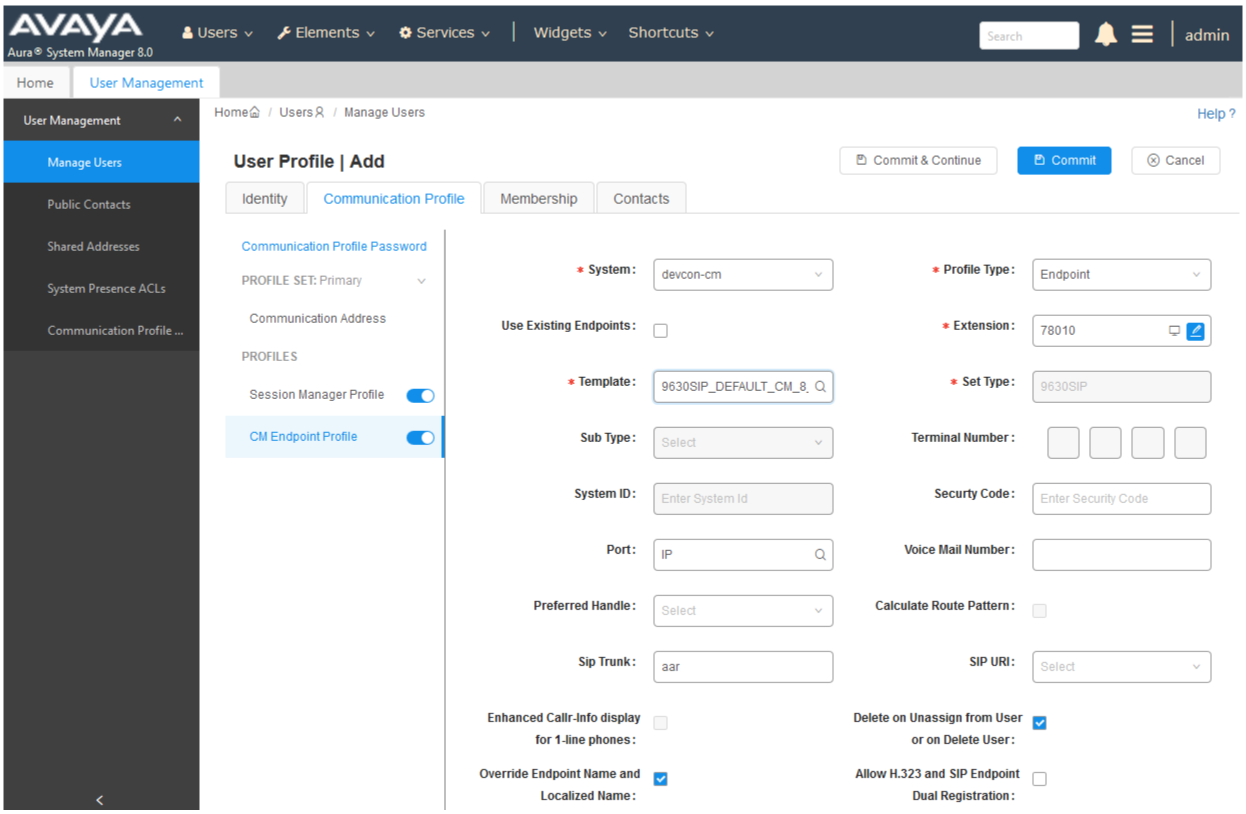


Scroll down to the **Call Routing Settings** section to configure the **Home Location**.

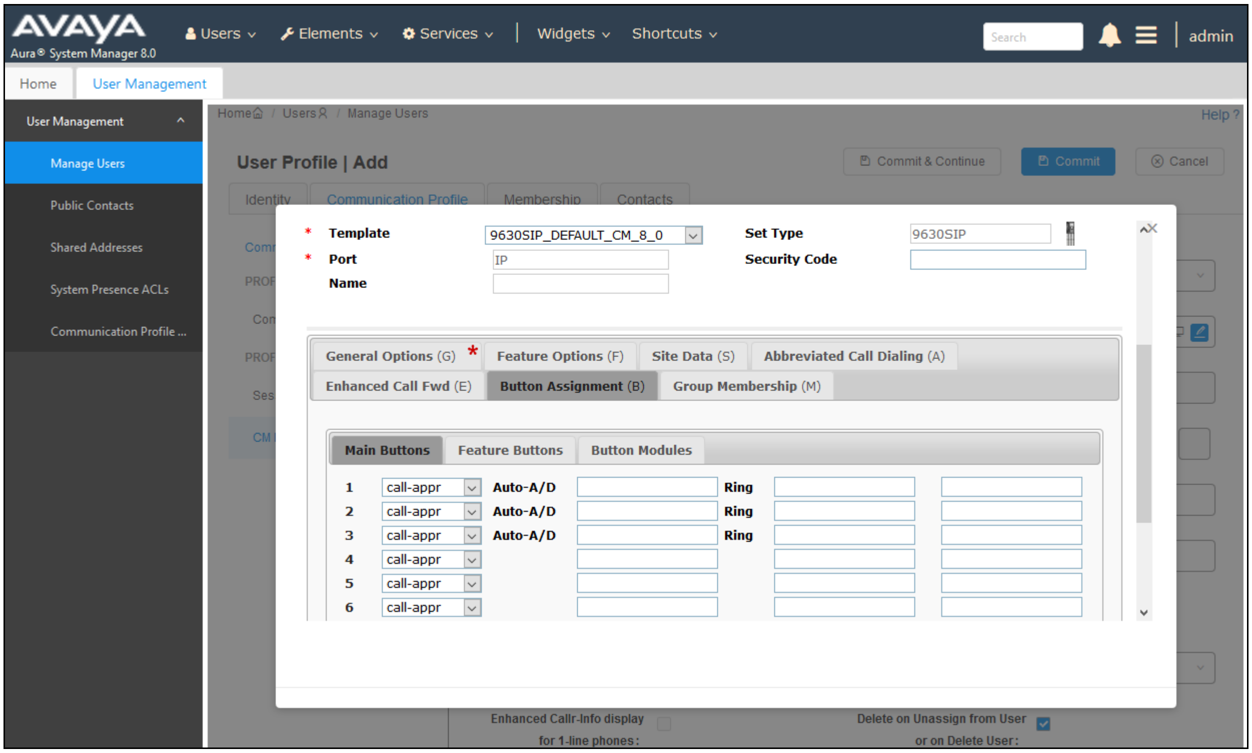


6.3.5. CM Endpoint Profile

Click on the toggle button by **CM Endpoint Profile**. For **System**, select the value corresponding to the applicable Communication Manager. For **Extension**, enter the SIP user extension from **Section 6.3.1**. For **Template**, select *9630SIP_DEFAULT_CM_8_0*. For **Port**, click and select *IP*. Retain the default values in the remaining fields. Click on the **Endpoint Editor** (i.e, edit icon in Extension field) to configure additional call appearances.



Navigate to the **Button Assignment** tab and add up to six call appearance buttons to allow Tesira SVC-2 to support up to six simultaneous calls. Click **Done** (not shown) to return to the previous web page and then **Commit** to save the configuration (not shown).



7. Configure Biamp Tesira SVC-2

This section covers the configuration of the Tesira SVC-2 card using the Tesira Software application. The configuration covers the following areas:

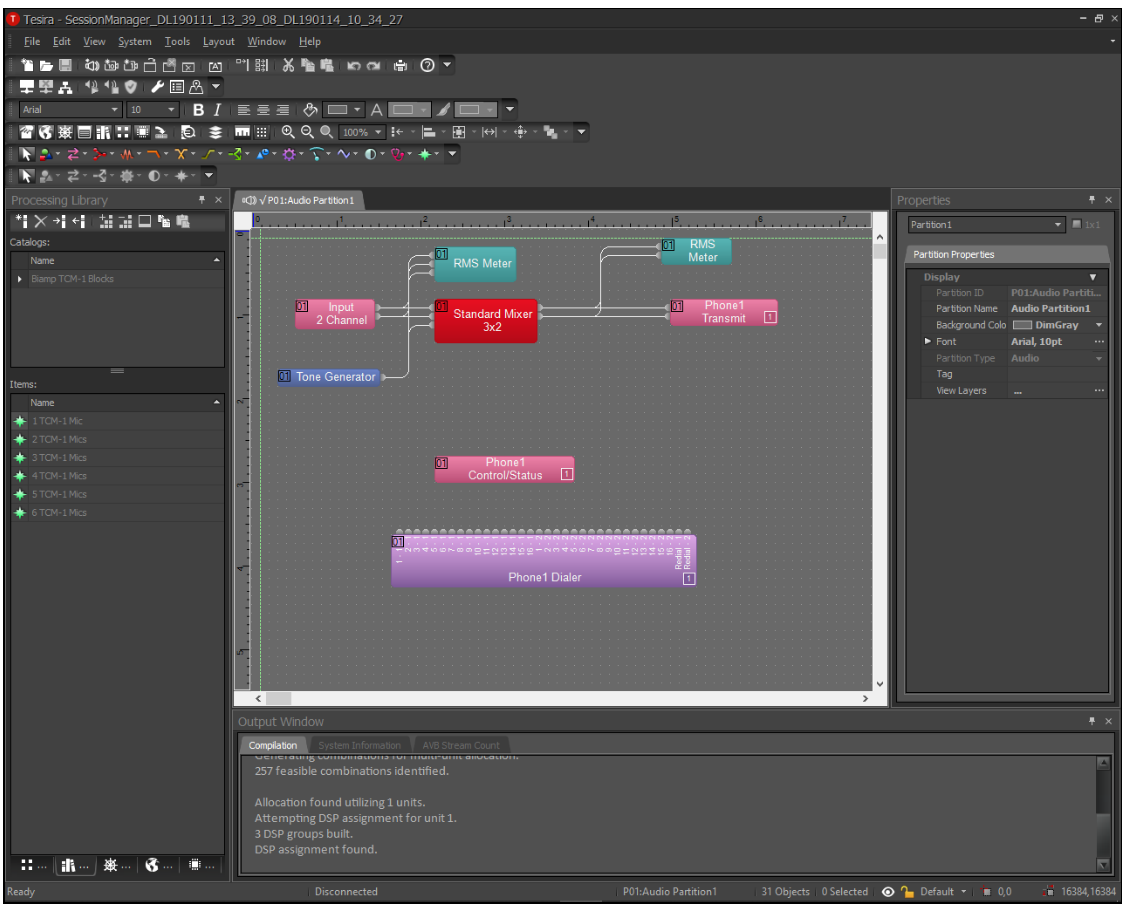
- Launch Tesira Software Application
- Modify the IP Network Settings of the Tesira SERVER-IO Control Network
- Modify the IP Network Settings of Tesira SVC-2
- Configure SIP Parameters of Tesira SVC-2
- Verify Codec Settings
- Save and Send the New Configuration to the System

7.1. Launch Tesira Software Application

Launch the **Tesira Software** application by clicking on the 

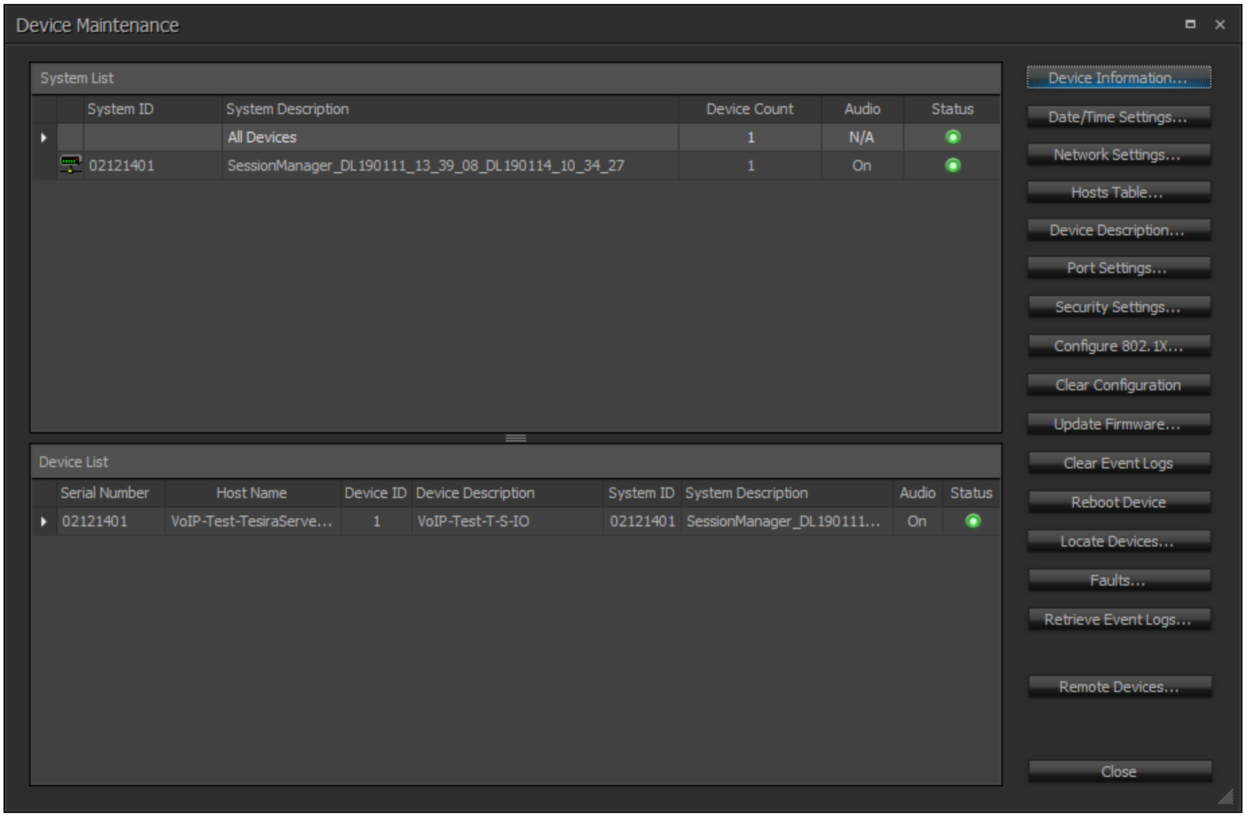
icon. The main window is displayed as shown below.

Note: The Tesira Software must be disconnected from the device to make changes to the configuration.

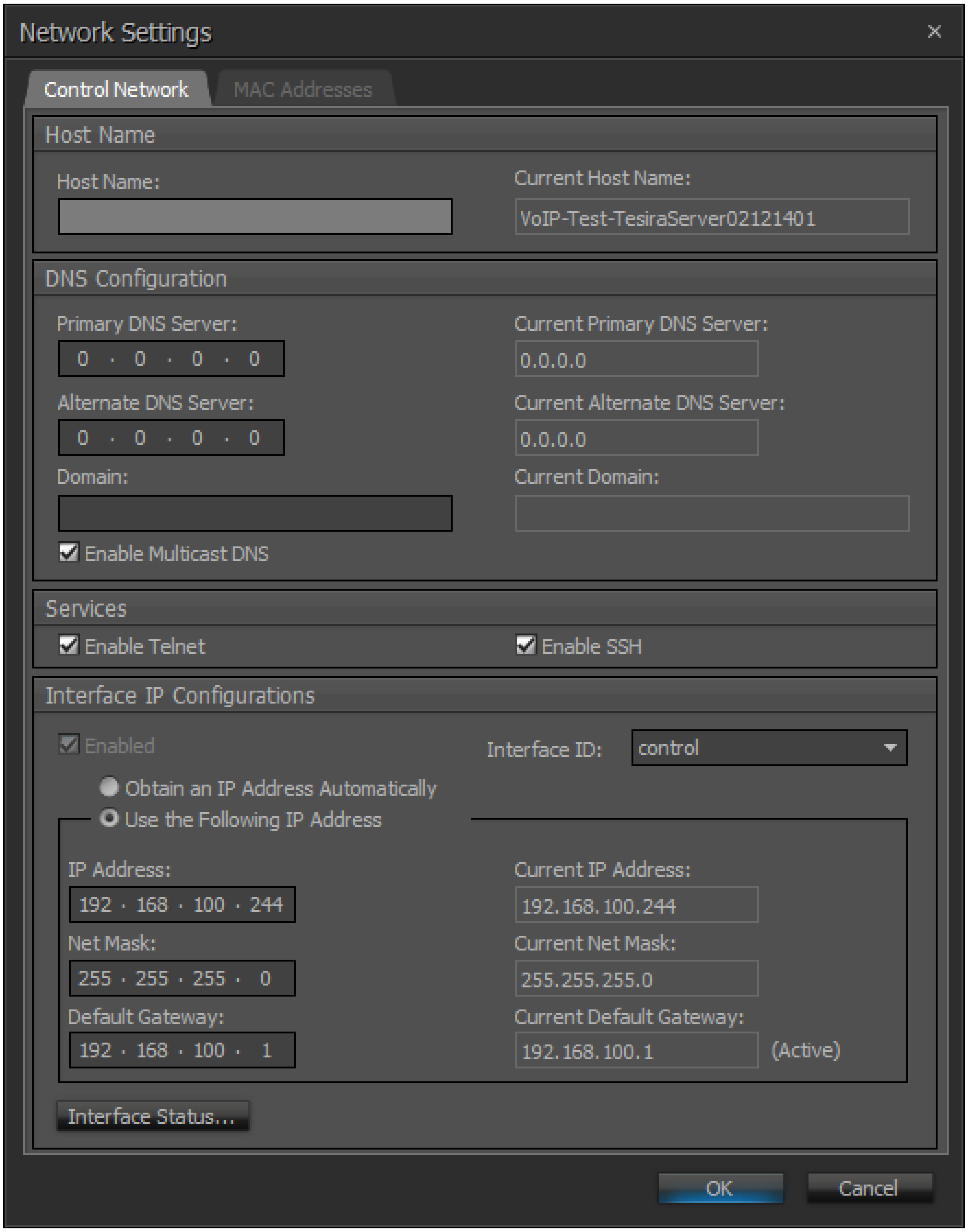


7.2. Modify the IP Network Settings of Biamp Tesira SERVER-IO Control Network

Click on the **Perform Device Maintenance** icon (not shown) to modify the network settings of the Tesira SERVER-IO control network. The **Device Maintenance** window shown below is displayed. Select the device and then click on the **Network Settings** in the right pane.

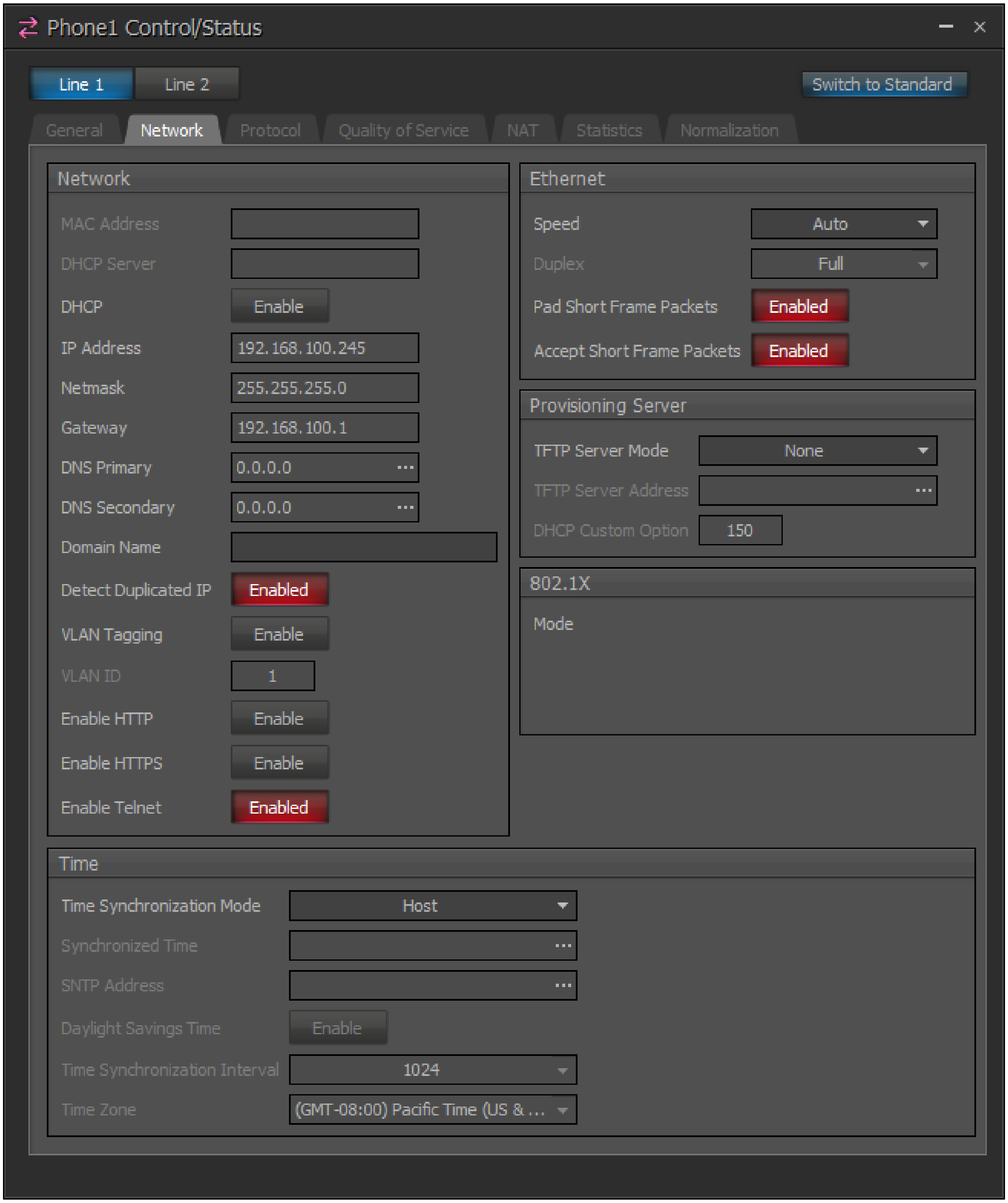


The **Network Settings** window is displayed as shown below. Tesira SERVER-IO supports DHCP or a static IP address. For this configuration, a static IP address (i.e., *192.168.100.244*) was assigned to the system in the **Interface IP Configurations** section. Click **OK**. Follow the procedure in **Section 7.6** to save and send the configuration to the system.



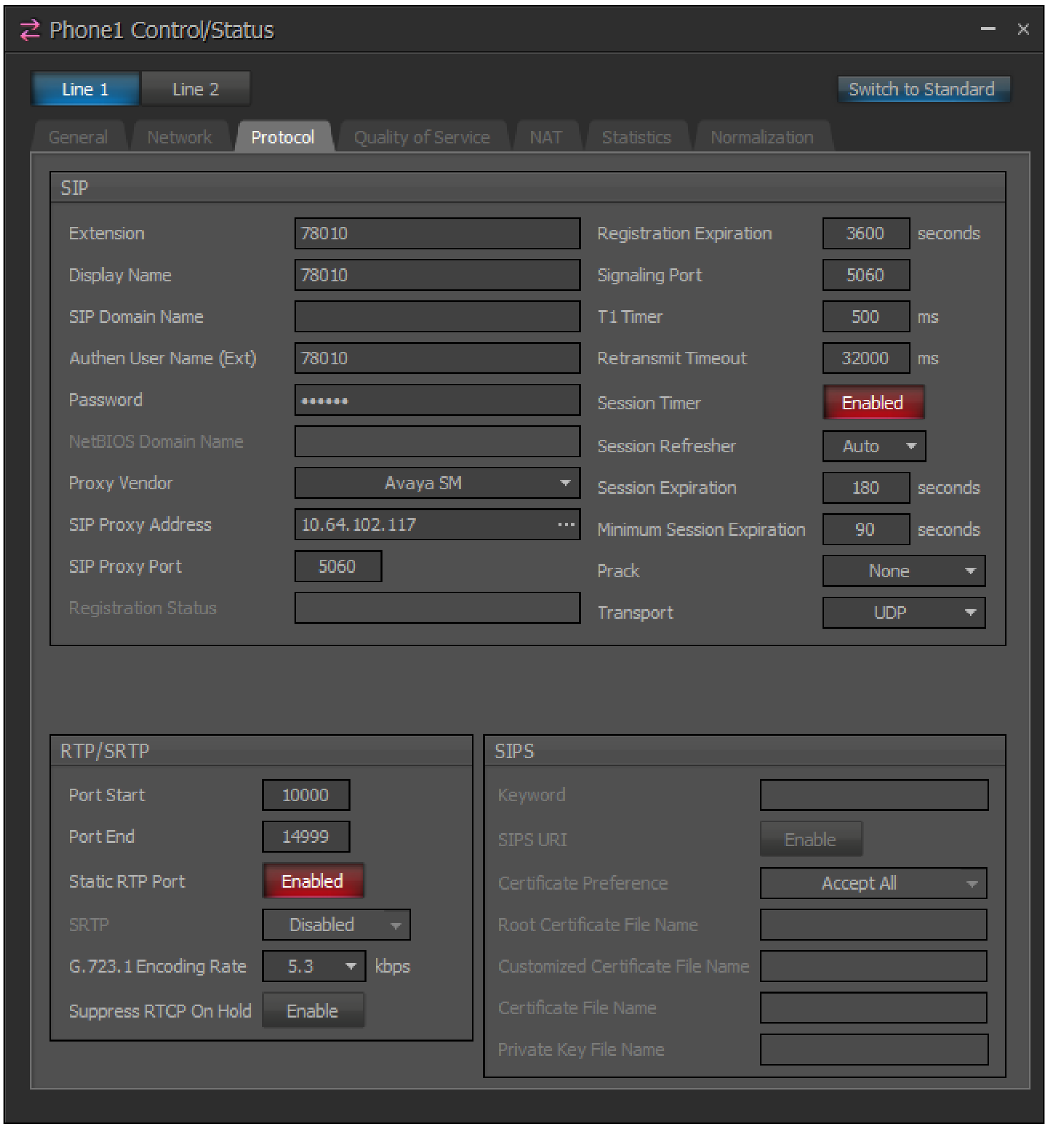
7.3. Modify the IP Network Settings of Biamp Tesira SVC-2

From the **Tesira Software** main window, double-click on **Phone1 Control Status** to display the **Phone1 Control/Status** window. Click on the **Switch to Advanced** button to display all the configuration options. Navigate to the Network tab to configure the IP network settings (i.e., *192.168.100.245*) of the Tesira SVC-2 card. Follow the procedure in **Section 7.6** to save and send the configuration to the system.

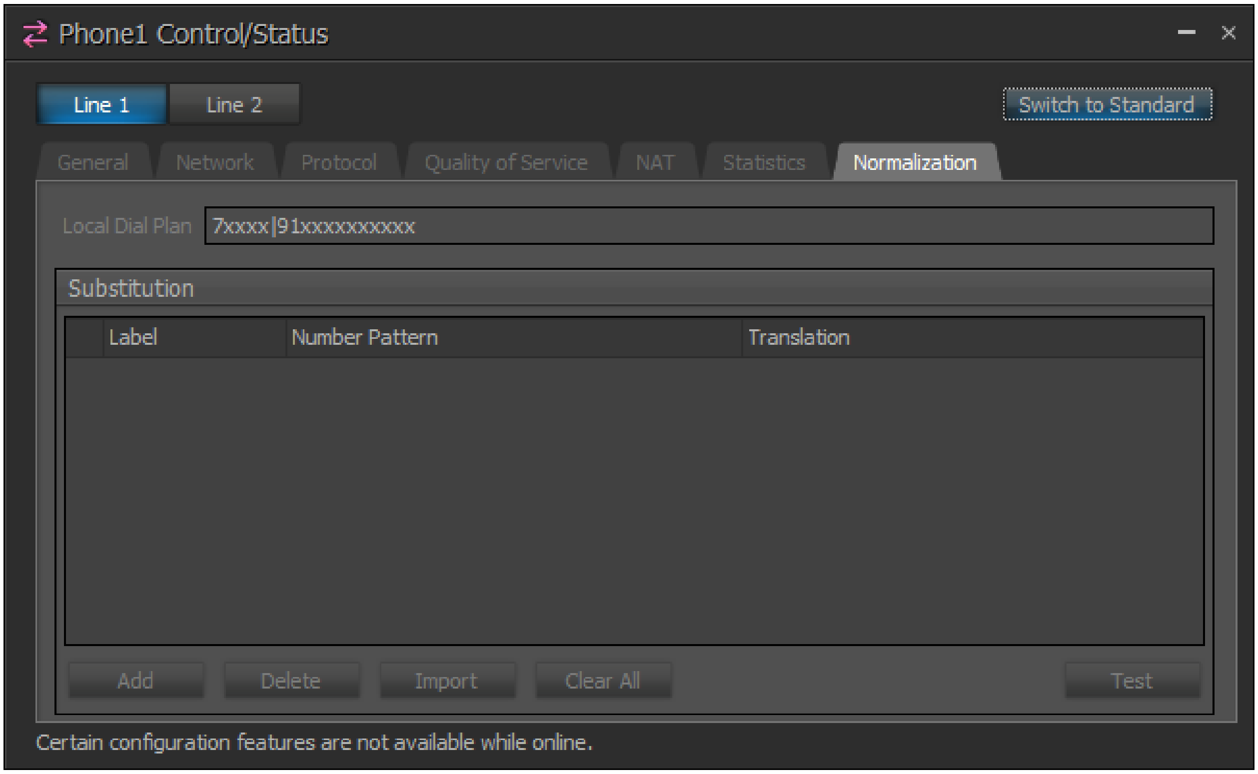


7.4. Configure SIP Parameters of BiampTesira SVC-2

From the **Tesira Software** main window, double-click on **Phone1 Control/Status** and select **Protocol** tab in the **Phone1 Control/Status** window. Set the **Extension** and **Display Name** fields to desired values. In this configuration, the SIP extension was used. Next, set the **Authen User Name (Ext)** to the SIP extension and the **Password** to the password used to register Tesira SVC-2 with Session Manager. The SIP username and password were configured on Session Manager in **Section 6.3**. Set the **Proxy Vendor** to *Avaya SM* and specify the **SIP Proxy Address** to the Session Manager IP address (i.e., *10.64.102.117*) noted in **Section 5.2**. Specify **SIP Proxy Port** *5060* and the **Transport** to *UDP*.

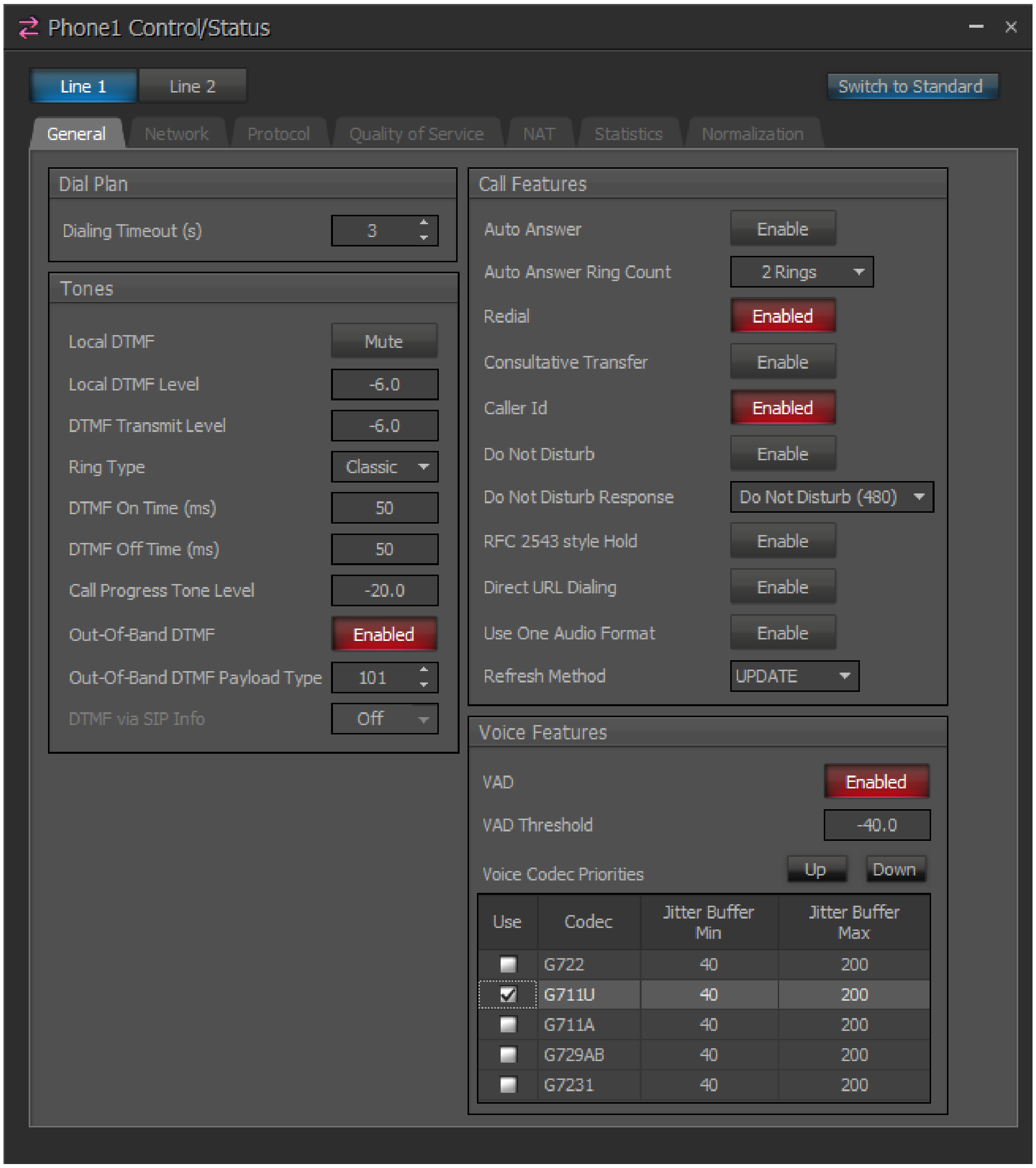


Navigate to the **Normalization** tab to set the dial plan. In this configuration, 5-digit extensions starting with '7' and 11-digit PSTN number prepended with a '91', the short code for routing external calls, were specified. If the dialed digit format is not specified in the dial plan, Tesira SVC-2 would have to wait for the inter-digit timeout to expire to determine when dialing has ended. The **Local Dial Plan** field was set to *7xxxx|91xxxxxxxxxxx* as shown below.



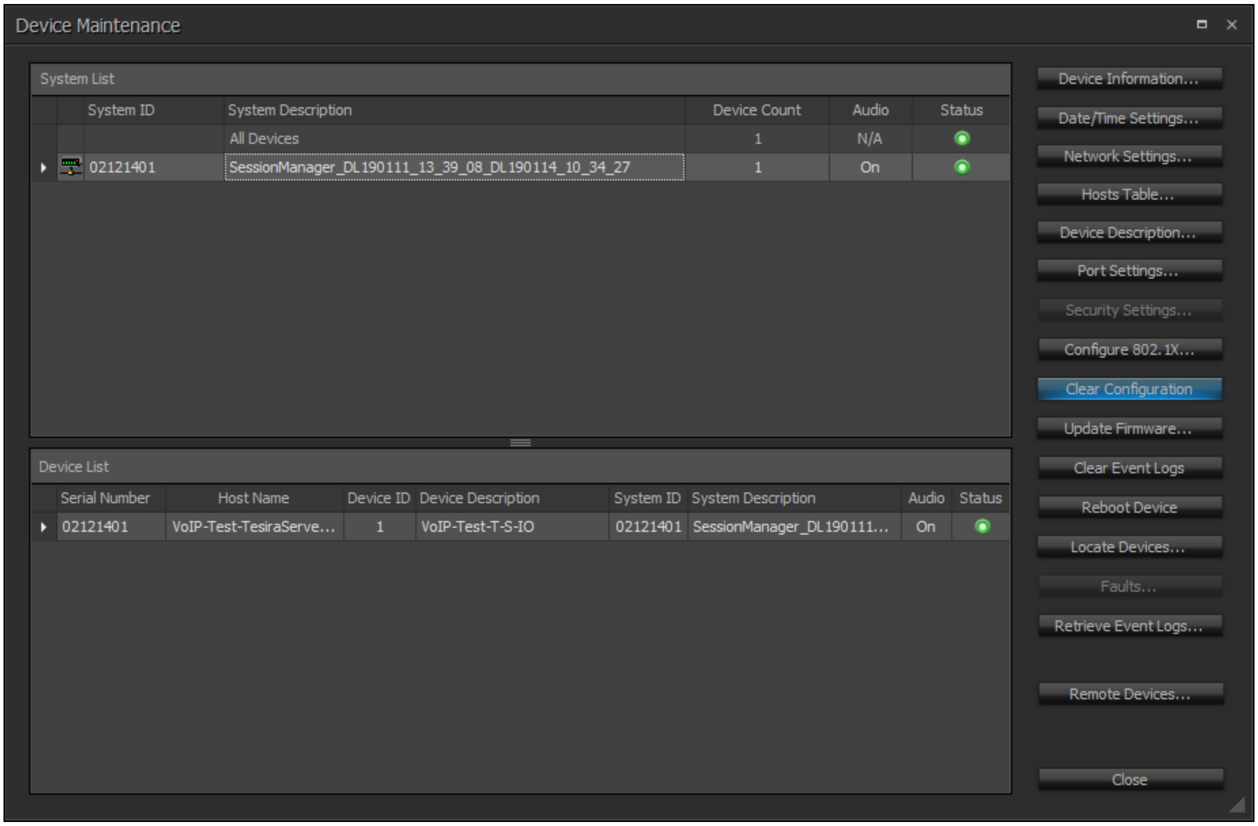
7.5. Verify Codec Settings

Navigate to the **General** tab shown below. In the **Voice Codec Priorities** section, select a single codec. Selecting multiple codecs will cause audio problems for conference calls as mentioned in **Section 2.2**. Also, select whether blind or consultative transfers should be supported. In the configuration below, the **Consultative Transfer** field is not selected indicating that blind transfers are supported.

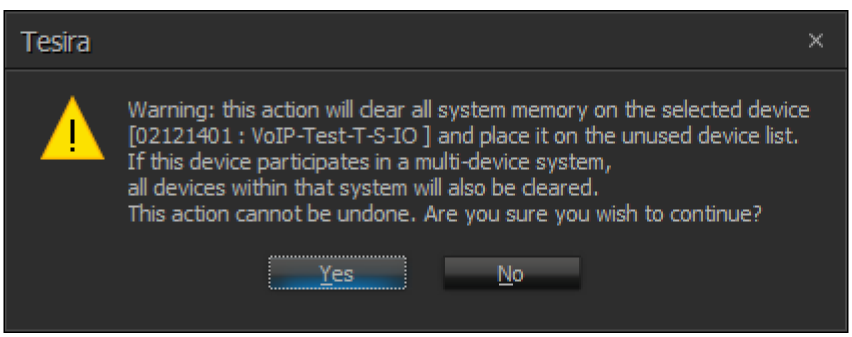


7.6. Save and Send the New Configuration to the System

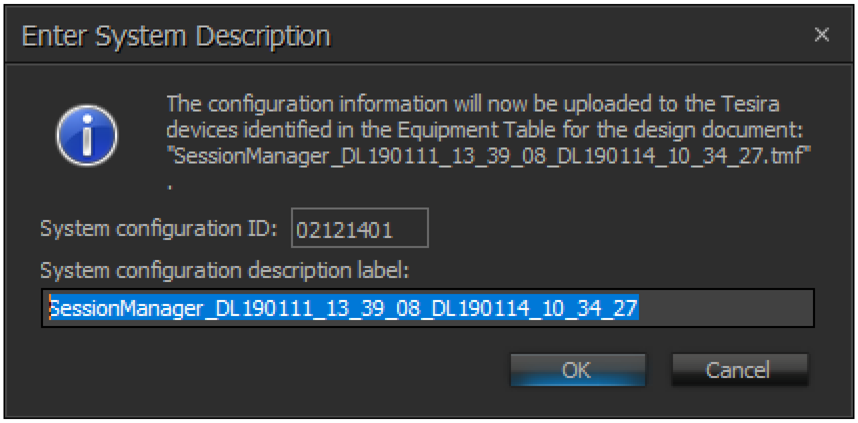
From the **Tesira Software** main window, save the configuration by clicking on **File → Save As** (not shown). Next, clear the configuration from the device before sending the new configuration to the device. Select the device and then click on the **Clear Configuration** button as shown below.



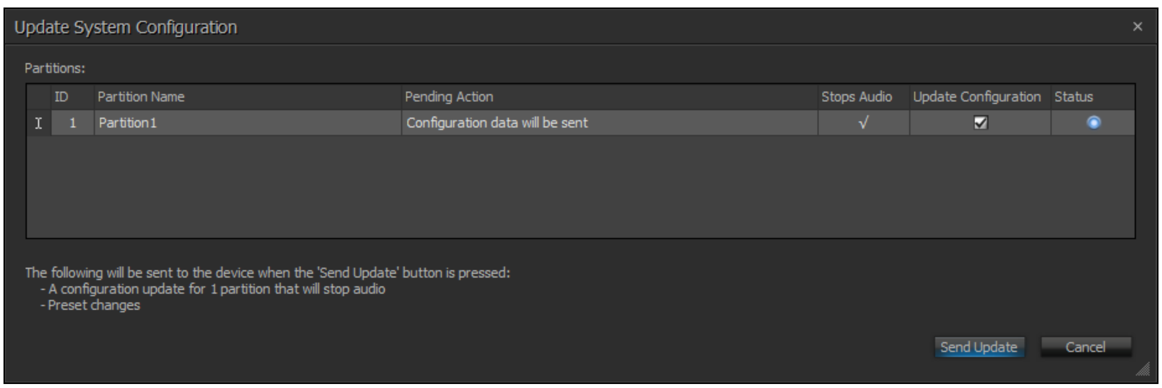
Confirm that the system memory will be cleared.



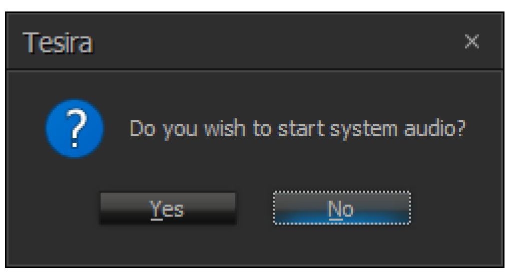
Lastly, click the Send Configuration icon in the main window to send the new configuration to the device (not shown). Confirm the **System configuration description label** by clicking **OK** in the window below.



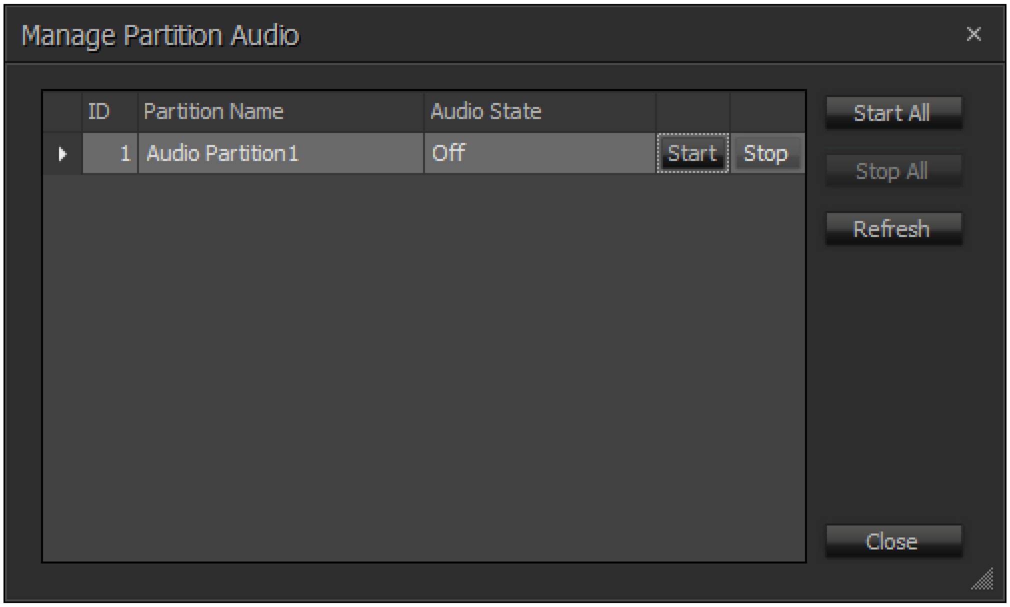
In the **Update System Configuration** window shown below, select the **Update Configuration** checkbox and click the **Send Update** button, if prompted.



Respond **Yes** when asked whether to start system audio.



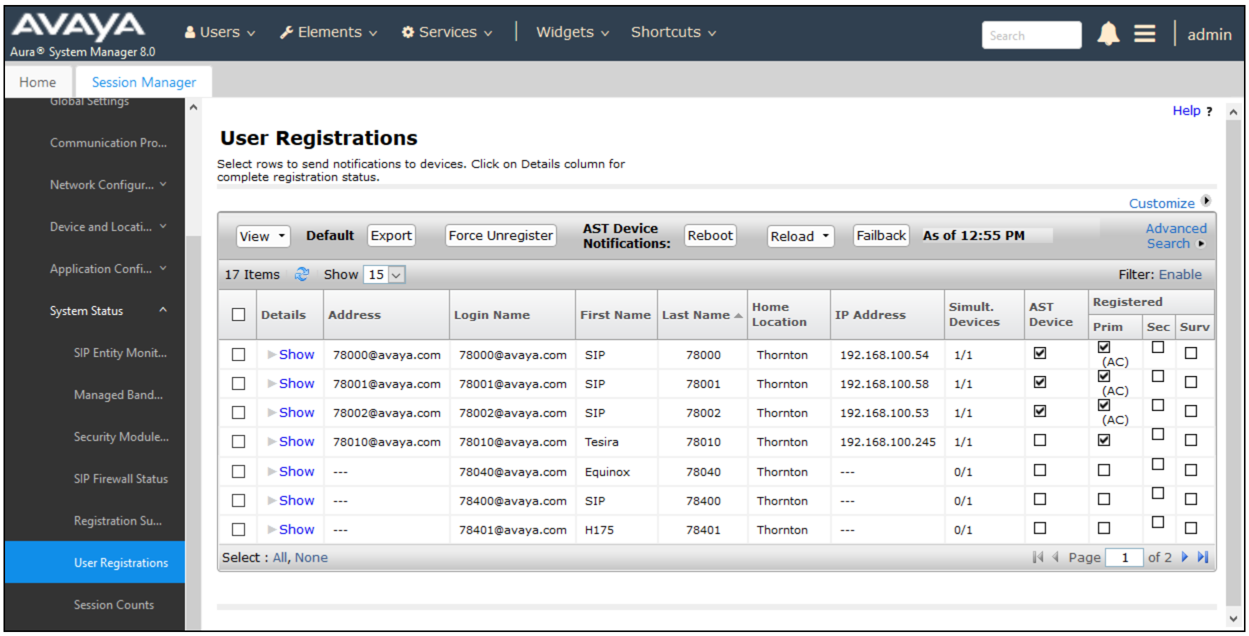
In the **Manage Partition Audio** window shown below, click on **Start All** and then click the **Close** button.



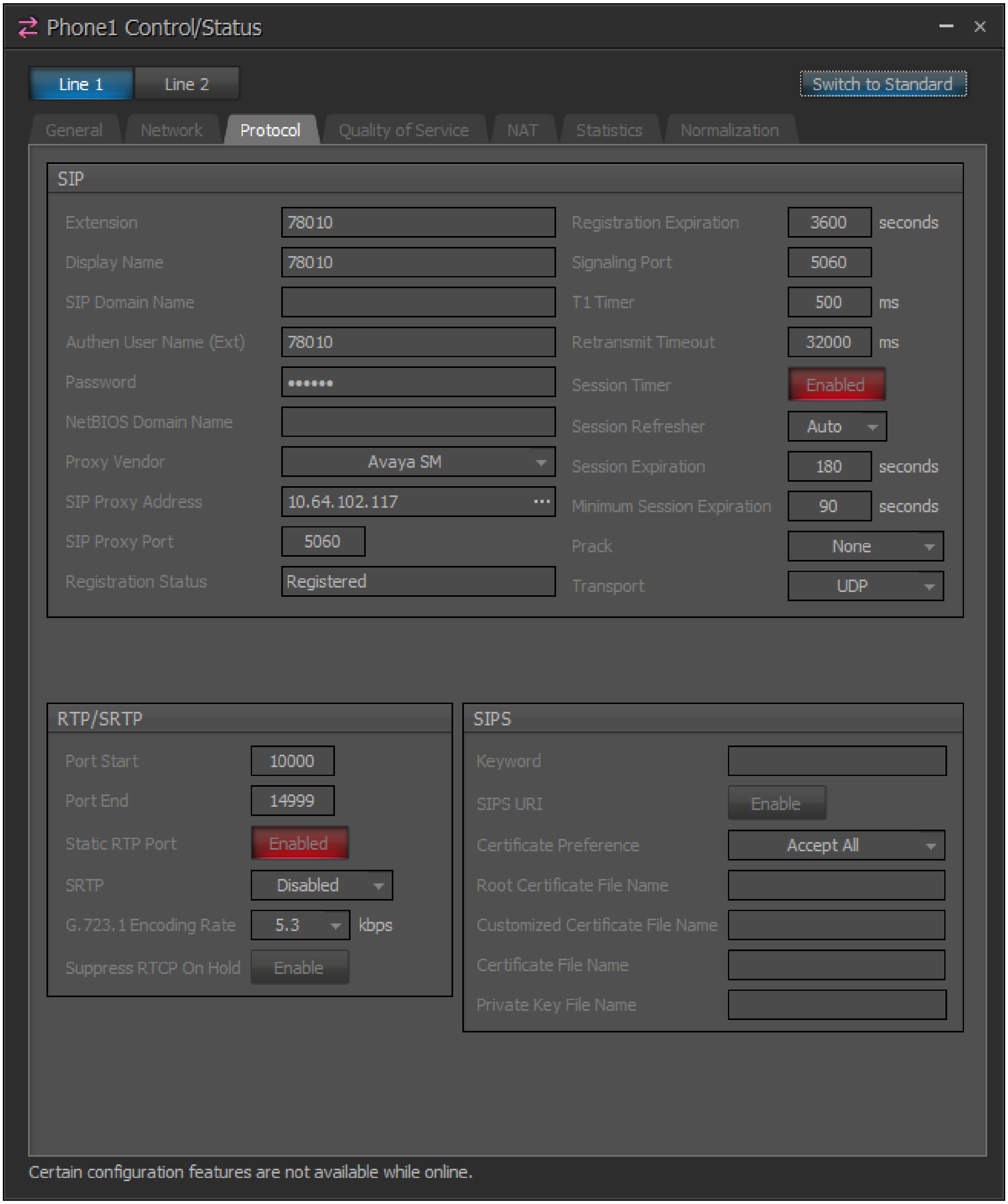
8. Verification Steps

This section provides the tests that may be performed to verify proper configuration of Biamp Tesira SVC-2 with Avaya Aura® Session Manager and Avaya Aura® Communication Manager.

1. Verify that Tesira SVC-2 has successfully registered with Session Manager. In System Manager, navigate to **Elements → Session Manager → System Status → User Registrations** to check the registration status.



2. Alternatively, verify that Tesira SVC-2 has successfully registered with Session Manager. Double-click on **VoIP Control/Status** in the Tesira Software main window and navigate to the **Protocol** tab. Note the **Registration Status**, which should be *Registered*.



3. Verify basic telephony feature by establishing calls with Tesira SVC-2. Verify two-way audio, that the call can be placed on hold, and that a 3rd party can be joined into a conference.

9. Conclusion

These Application Notes described the configuration steps required to integrate Biamp Tesira SVC-2 with Avaya Aura® Communication Manager and Avaya Aura® Session Manager. Biamp Tesira SVC-2 was able to establish calls with H.323 / SIP deskphones and the PSTN. In addition, basic telephony features were verified. All feature and serviceability test cases were completed successfully with observations noted in **Section 2.2**.

10. References

This section references the Avaya documentation relevant to these Application Notes. The Avaya product documentation is available at http://support.avaya.com. Biamp Tesira SVC-2 documentation is available through online help via the Tesira Software.

[1] *Administering Avaya Aura® Communication Manager,* Release 8.0, Issue 1, July 2018.

[2] *Administering Avaya Aura® System Manager for Release 8.0*, Release 8.0, Issue 4, September 2018.

[3] *Administering Avaya Aura® Session Manager,* Release 8.0, Issue 2, July 2018.

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.

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January 18, 2019

To Whom It May Concern

Tesira SVC-2 on Tesira Server and TesiraServerIO, TesiraForte VI, TesiraForte AVB VI, TesiraForte DAN VI, TesiraForte VT, Tesira AVB VT, Tesira DAN VT, Tesira AVB VT4, and Tesira DAN VT4 share the same SIP stack and VoIP firmware version. Tesira Server can have one SVC-2 card. Tesira ServerIO can have up to 6 SVC-2 cards. TesiraForte VI, TesiraForte AVB VI, TesiraForte DAN VI, TesiraForte VT, Tesira AVB VT, Tesira DAN VT, Tesira AVB VT4, and Tesira DAN VT4 have only one VoIP application. The Tesira firmware version under the test is 3.8.0.24. Please refer to the table below for non-VoIP differences.

Product	Specification
Tesira SERVER	The Tesira SERVER is a digital network server. It is factory configured with one DSP-2 card and can accept up to a total of eight DSP-2 cards. The SERVER is also factory configured with one AVB-1 card and has a second slot that can be outfitted with an additional AVB-1 card, a 32 x 32 channel SCM-1 CobraNet card, a 64 x 64 channel DAN-1 Dante™ card, or a standard I/O card for four channels of local I/O. The SERVER is the core of a Tesira digital audio system and can be used with Tesira expanders to form a highly scalable audio network. Two Tesira SERVERs can also be designed as a redundant pair, carrying identical processing and card configurations. The secondary SERVER stays 'live' with the primary, updating runtime parameters. If the primary SERVER should need maintenance, the secondary takes over with no loss of continuity or downtime. • Supports up to 8 DSP-2 cards • Up to 420 x 420 channels of digital I/O over AVB • Supports optional 32 x 32 CobraNet audio networking • Supports optional 64 x 64 Dante audio networking • System configuration and control via Ethernet or serial connection • Front panel OLED display for device and system information • SpeechSense™ and AmbientSense™ processing algorithms • Signal processing via intuitive software allows configuration and control for: signal routing and mixing, equalization, filtering, dynamics, delay and much more • Extensive input, output and logic expansion devices supported as part of the Tesira digital audio networking platform
Tesira SERVER-IO	The Tesira SERVER-IO is a digital network server. It is factory configured with one DSP-2 card and is capable of handling up to two additional DSP-2 cards. The SERVER-IO has capacity for up to three total audio networking cards per server. The combinations of networking cards may include one AVB-1 Audio Video Bridging network cards, up to two SCM-1 CobraNet network cards, and up to

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	two DAN-1 Dante network cards in any configuration. The SERVER-IO can support up to 12 standard Tesira I/O cards for up to 48 channels of audio I/O. The SERVER-IO can be used as a standalone device or with Tesira expanders to form a highly scalable audio network. • Supports up to 3 DSP-2 cards • Supports up to 12 I/O cards with a maximum of 48 channels of analog audio • Up to 420 x 420 channels of digital I/O over AVB • Supports optional 32 x 32 CobraNet audio networking • Supports optional 64 x 64 Dante audio networking • System configuration and control via Ethernet or serial connection • Front panel OLED display for device and system information • SpeechSense™ and AmbientSense™ processing algorithms • Signal processing via intuitive software allows configuration and control for: signal routing and mixing, equalization, filtering, dynamics, delay and much more • Extensive input, output and logic expansion devices supported as part of the Tesira digital audio networking platform
TesiraFORTÉ VT	The TesiraFORTÉ VT is a digital audio server with 12 analog inputs and 8 analog outputs, and includes Acoustic Echo Cancellation (AEC) technology on all 12 inputs. It includes up to 8 channels of configurable USB audio, a 2-channel VoIP interface, and a standard FXO telephone interface. • 12 mic/line level inputs with AEC, 8 mic/line level outputs • Gigabit Ethernet port • RS-232 serial port • 4-pin GPIO • 2-line OLED display with capacitive-touch navigation • System configuration and control via Ethernet • Internal universal power supply • SIP VoIP interface via RJ-45 connector • Standard FXO telephone interface via RJ-11 connector • Signal processing via intuitive software allows configuration and control for signal routing, mixing, equalization, filtering, and delay
TesiraFORTÉ AVB VT	• The TesiraFORTÉ AVB VT is a digital audio server with 12 analog inputs and 8 analog outputs, and includes Acoustic Echo Cancellation (AEC) technology on all 12 inputs. It includes up to 8 channels of configurable USB audio, a 2-channel VoIP interface, and a standard FXO telephone interface. TesiraFORTÉ AVB VT utilizes AVB/TSN for digital audio networking and can be used as a standalone device or

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	combined with other TesiraFORTÉ AVB devices and Tesira servers, expanders, endpoints, and controllers. • 128 x 128 channels of AVB • 12 mic/line level inputs with AEC, 8 mic/line level outputs • Gigabit Ethernet port • RS-232 serial port • 4-pin GPIO • 2-line OLED display with capacitive-touch navigation • System configuration and control via Ethernet • Internal universal power supply • SIP VoIP interface via RJ-45 connector • Standard FXO telephone interface via RJ-11 connector • Signal processing via intuitive software allows configuration and control for signal routing, mixing, equalization, filtering, and delay
TesiraFORTÉ DAN VT	The TesiraFORTÉ DAN VT is a digital audio server with 12 analog inputs and 8 analog outputs, and includes Acoustic Echo Cancellation (AEC) technology on all 12 inputs. It includes up to 8 channels of configurable USB audio, a 2-channel VoIP interface, a standard FXO telephone interface, and Dante digital audio networking. • 32 x 32 channels of Dante • 12 mic/line level inputs with AEC, 8 mic/line level outputs • Gigabit Ethernet port • RS-232 serial port • 4-pin GPIO • 2-line OLED display with capacitive-touch navigation • System configuration and control via Ethernet • Internal universal power supply • SIP VoIP interface via RJ-45 connector • Standard FXO telephone interface via RJ-11 connector • Signal processing via intuitive software allows configuration and control for signal routing, mixing, equalization, filtering, and delay
TesiraFORTÉ AVB VT4	• The TesiraFORTÉ AVB VT4 is a digital audio server with 4 analog inputs and 4 analog outputs, and includes Acoustic Echo Cancellation (AEC) technology on all 4 inputs. It includes up to 8 channels of configurable USB audio, a 2-channel VoIP interface, and a standard FXO telephone interface. TesiraFORTÉ AVB VT4 utilizes AVB/TSN digital audio networking, and can be used as a standalone device or combined with other TesiraFORTÉ AVB devices and Tesira servers, expanders, endpoints, and controllers. • 128 x 128 channels of AVB • 4 mic/line level inputs with AEC, 4 mic/line level outputs

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	• Gigabit Ethernet port • RS-232 serial port • 4-pin GPIO • 2-line OLED display with capacitive-touch navigation • System configuration and control via Ethernet • Internal universal power supply • SIP VoIP interface via RJ-45 connector • Standard FXO telephone interface via RJ-11 connector • Signal processing via intuitive software allows configuration and control for signal routing, mixing, equalization, filtering, and delay
TesiraFORTÉ DAN VT4	The TesiraFORTÉ DAN VT4 is a digital audio server with 4 analog inputs and 4 analog outputs, and includes Acoustic Echo Cancellation (AEC) technology on all 4 inputs. It includes up to 8 channels of configurable USB audio, a 2-channel VoIP interface, a standard FXO telephone interface, and Dante digital audio networking. • 32 x 32 channels of Dante • 4 mic/line level inputs with AEC, 4 mic/line level outputs • Gigabit Ethernet port • RS-232 serial port • 4-pin GPIO • 2-line OLED display with capacitive-touch navigation • System configuration and control via Ethernet • Internal universal power supply • SIP VoIP interface via RJ-45 connector • Standard FXO telephone interface via RJ-11 connector • Signal processing via intuitive software allows configuration and control for signal routing, mixing, equalization, filtering, and delay
TesiraFORTÉ VI TesiraFORTÉ AVB VI	The TesiraFORTÉ VI is a digital audio server with 12 analog inputs and 8 analog outputs and includes Acoustic Echo Cancellation (AEC) technology on all 12 inputs. It also includes up to 8 channels of configurable USB audio, and a 2-channel VoIP interface via a RJ-45 connector. USB audio allows TesiraFORTÉ to interface directly with USB audio hosts, as well as to take full advantage of today's most sophisticated conferencing solutions. TesiraFORTÉ AVB VI adds Audio Video Bridging (AVB) digital audio networking. The AVB model can be used as a standalone device or can be combined with other TesiraFORTÉ devices and Tesira servers, expanders, and controllers. TesiraFORTÉ VI also provides extensive audio processing, including but not limited to: AEC technology, signal routing and mixing, equalization, filtering, dynamics, and delay, as well as control, monitoring, and diagnostic tools; all configured through the Tesira configuration software. TesiraFORTÉ VI is best-suited for small- to

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	mediumsized rooms that require high-quality audio solutions using VoIP, voice lift, mix-minus, and AEC such as board rooms or distance training facilities. • 128 x 128 channels of AVB (AVB model only) • 12 mic/line level inputs with AEC, 8 mic/line level outputs • Gigabit Ethernet port • Up to 8 channels of configurable USB audio • RS-232 serial port • 4-pin GPIO • 2-line OLED display with capacitive-touch navigation • System configuration and control via Ethernet • Internal universal power supply • SIP VoIP interface via a RJ-45 connector • Fully compatible with Tesira servers, endpoints, expanders, and controllers (AVB model) • Signal processing via intuitive software allows configuration and control for signal routing, mixing, equalization, filtering, delay and much more
TESIRA FORTÉ DAN VI	The TesiraFORTÉ DAN VI is a digital audio server with 32 bi-directional channels of Dante™ digital audio, 12 analog inputs with Acoustic Echo Cancellation (AEC), and 8 analog outputs. It also includes up to 8 channels of configurable USB audio, and a 2-channel SIP VoIP interface via a RJ-45 connector. USB audio allows TesiraFORTÉ to interface directly with USB audio hosts, as well as to take full advantage of today's most sophisticated conferencing solutions. TesiraFORTÉ DAN VI provides extensive audio processing, including but not limited to: signal routing and mixing, equalization, filtering, dynamics, and delay; as well as control, monitoring, and diagnostic tools; all configured through the Tesira configuration software. TesiraFORTÉ DAN VI is best suited for room requiring AEC, voice lift, and mix-minus, such as conference rooms or distance learning environments that use VoIP. • 32x32 channels of digital audio networking via the Dante protocol • 12 mic/line level inputs with AEC, 8 mic/line level outputs • 2 Gigabit Ethernet ports: Dante digital audio and Tesira control • Up to 8 channels of configurable USB audio • RS-232 serial port • 4-pin GPIO • 2-line OLED display with capacitive-touch navigation • System configuration and control via Ethernet • Internal universal power supply • SIP VoIP interface via a RJ-45 connector

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	• Signal processing via intuitive software allows configuration and control for signal routing, mixing, equalization, filtering, delay, and much more

Please don't hesitate to contact us if you have further concerns.

Sincerely yours,

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