



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.

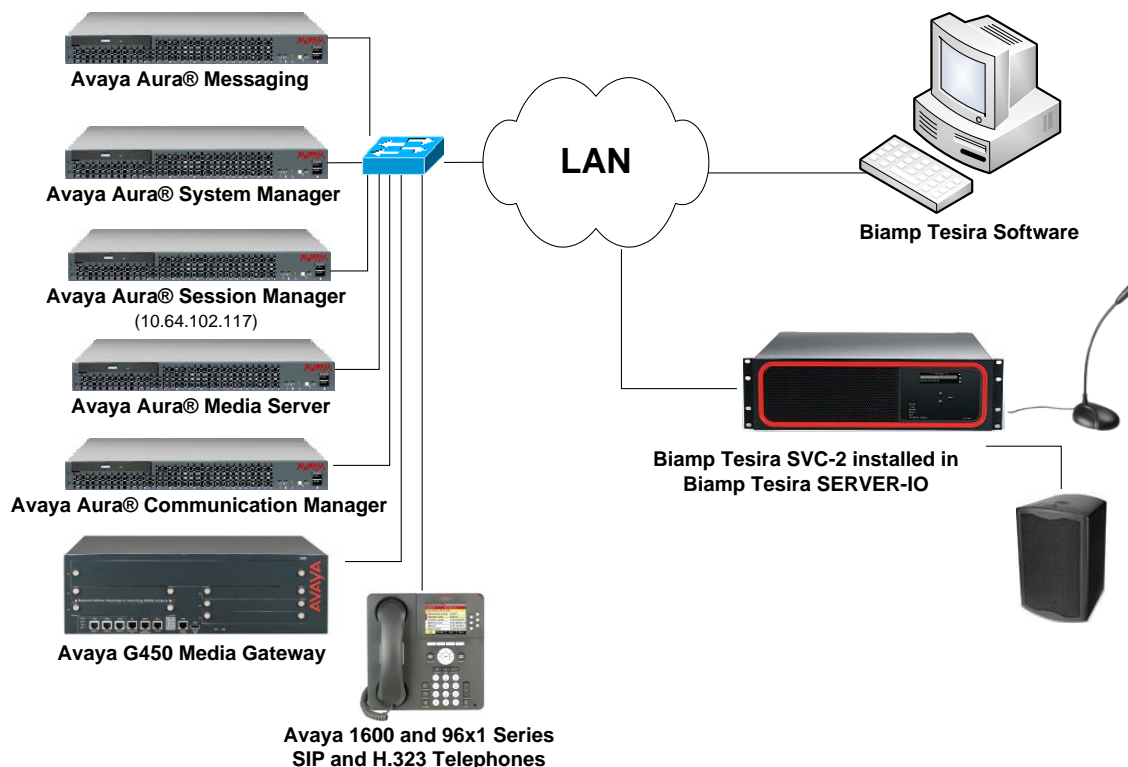


Figure 1: Avaya SIP Network with Biamp Tesira SVC-2 installed in Biamp SERVER-IO

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		
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? y
		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      Req'd
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
                                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.

The screenshot shows the Avaya Aura System Manager 8.0 interface. The left sidebar contains a navigation menu with options: Home, Routing, Domains, Locations, Adaptations, SIP Entities (highlighted), Entity Links, Time Ranges, Routing Policies, Dial Patterns, Regular Expressions, and Defaults. The main content area is titled 'SIP Entity Details' and includes a 'General' tab. The 'General' tab contains the following fields: Name (devcon-sm), IP Address (10.64.102.117), SIP FQDN (empty), Type (Session Manager), Notes (empty), Location (Thornton), Outbound Proxy (empty), Time Zone (America/New_York), Minimum TLS Version (Use Global Setting), and Credential name (empty). At the bottom, there are two monitoring options: SIP Link Monitoring (Use Session Manager Configuration) and CRLF Keep Alive Monitoring (Use Session Manager Configuration). Buttons for 'Commit' and 'Cancel' are visible at the top right of the form.

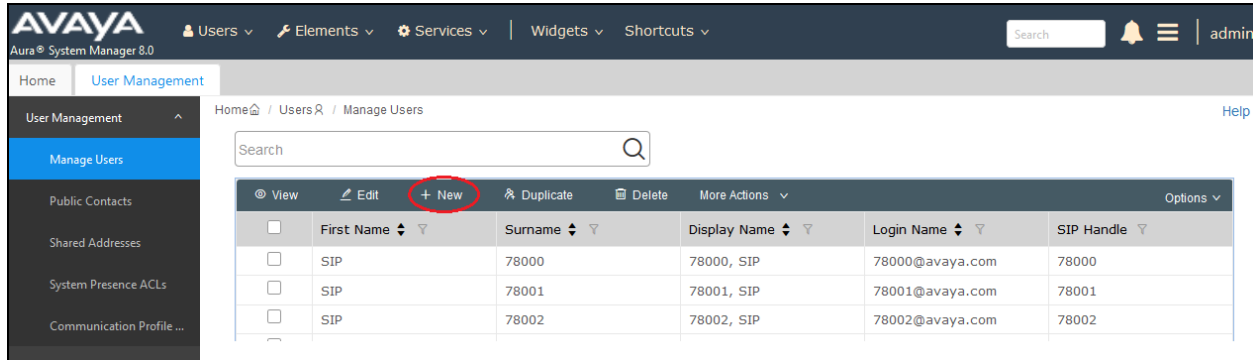
Scroll down to the **Listen Ports** section and verify that the transport network protocol used by Tesira-SVC2 is specified in the list below. For the compliance test, the solution used UDP network transport.

Listen Ports

Add Remove					
3 Items Filter: Enable					
<input type="checkbox"/>	Listen Ports	Protocol	Default Domain	Endpoint	Notes
<input type="checkbox"/>	5060	TCP	avaya.com	<input type="checkbox"/>	
<input type="checkbox"/>	5060	UDP	avaya.com	<input type="checkbox"/>	
<input type="checkbox"/>	5061	TLS	avaya.com	<input type="checkbox"/>	
Select : All, None					

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.

User Profile | Add

Commit & Continue Commit Cancel

Identity Communication Profile Membership Contacts

Basic Info

Address

LocalizedName

User Provisioning Rule: [Rule]

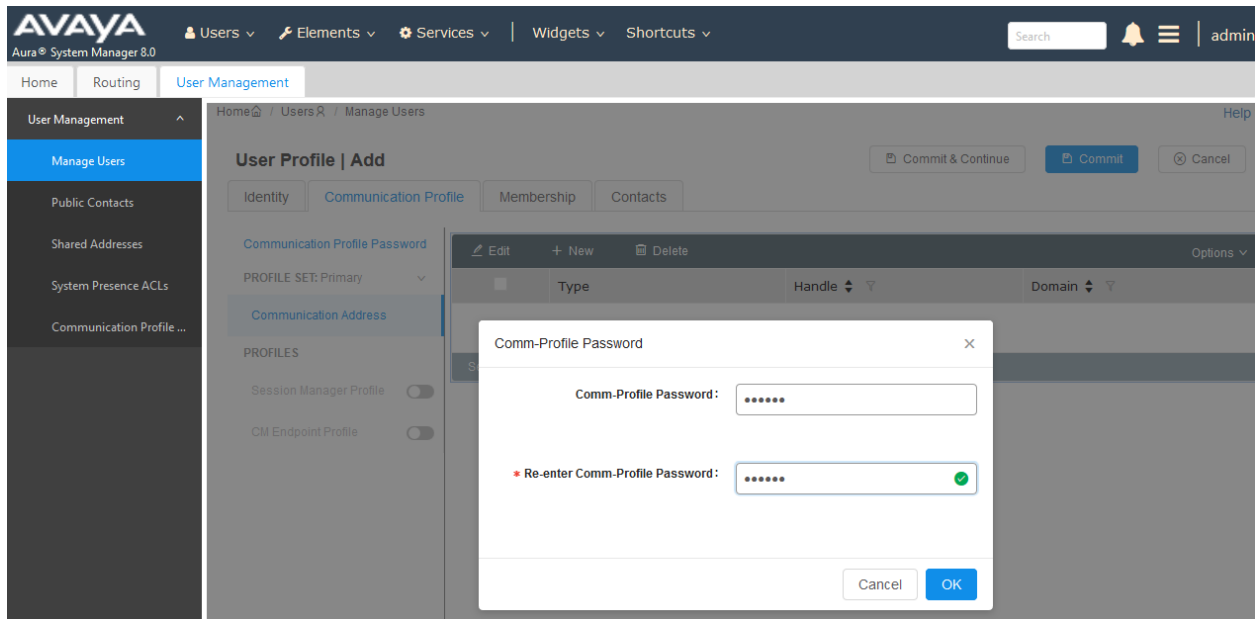
* Last Name: 78010 Last Name (Latin Translation): 78010

* First Name: Tesira First Name (Latin Translation): Tesira

* Login Name: 78010@avaya.com Middle Name: Middle Name Of Use

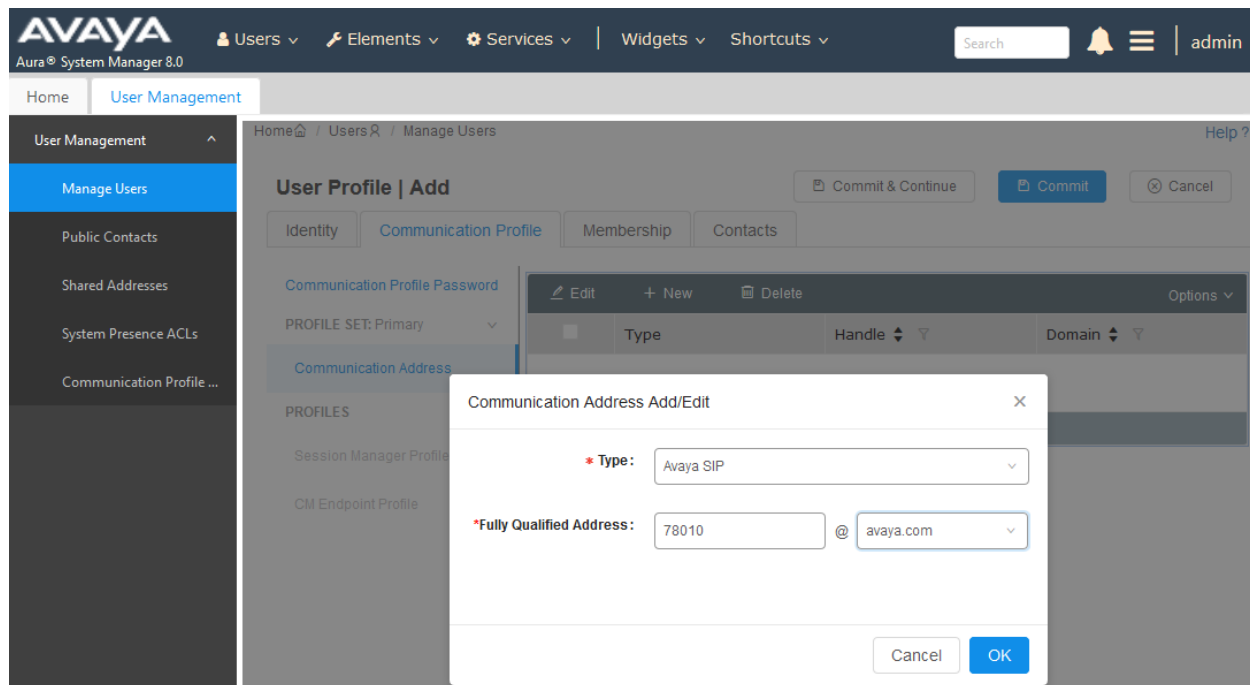
6.3.2. Communication Profile

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.

The screenshot shows the Avaya Aura System Manager 8.0 interface. The top navigation bar includes 'Users', 'Elements', 'Services', 'Widgets', and 'Shortcuts'. The left sidebar shows 'User Management' with a sub-menu 'Manage Users'. The main content area is titled 'User Profile | Add' and has tabs for 'Identity', 'Communication Profile', 'Membership', and 'Contacts'. The 'Communication Profile' tab is active, showing a 'Communication Profile Password' section with a dropdown for 'PROFILE SET: Primary' and a 'Communication Address' field. Below this is a 'PROFILES' section with a 'Session Manager Profile' toggle (turned on) and a 'CM Endpoint Profile' toggle (turned off). The 'SIP Registration' section includes fields for 'Primary Session Manager' (set to 'devcon-sm'), 'Secondary Session Manager' (set to 'Start typing...'), and 'Survivability Server' (set to 'Start typing...'). There is also a 'Max. Simultaneous Devices' dropdown (set to 'Select'). The 'Block New Registration When Maximum Registrations' checkbox is unchecked. The 'Application Sequences' section includes 'Origination Sequence' and 'Termination Sequence' dropdowns, both set to 'DEVCON-CM App Sequ...'. At the bottom, there is a 'Call Routing Settings' section with a 'Home Location' dropdown (set to 'Thornton') and a 'Conference Factory Set' dropdown (set to 'Select').

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

The screenshot shows the 'Call Routing Settings' section. It includes a 'Home Location' dropdown menu with 'Thornton' selected, and a 'Conference Factory Set' dropdown menu with 'Select' selected.

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.

The screenshot displays the Avaya Aura System Manager 8.0 interface. The top navigation bar includes the Avaya logo, navigation links for Users, Elements, Services, Widgets, and Shortcuts, a search bar, and a user profile icon labeled 'admin'. The main content area is titled 'User Profile | Add' and features a sidebar with navigation options: User Management, Manage Users (selected), Public Contacts, Shared Addresses, System Presence ACLs, and Communication Profile... The 'Manage Users' section is active, showing a list of users. The 'User Profile | Add' form is displayed, with the 'Communication Profile' tab selected. The form includes fields for System (devcon-cm), Profile Type (Endpoint), Extension (78010), Template (9630SIP_DEFAULT_CM_8_0), Set Type (9630SIP), Sub Type (Select), System ID (Enter System Id), Port (IP), Preferred Handle (Select), Sip Trunk (aar), Terminal Number, Security Code, Voice Mail Number, Calculate Route Pattern, SIP URI, Enhanced Callr-Info display, Delete on Unassign from User, and Allow H.323 and SIP Endpoint Dual Registration. The 'CM Endpoint Profile' toggle is turned on.

Avaya Aura System Manager 8.0

Users Elements Services Widgets Shortcuts

Search admin

Home User Management

User Management Manage Users Public Contacts Shared Addresses System Presence ACLs Communication Profile...

Home / Users / Manage Users Help ?

User Profile | Add

Commit & Continue Commit Cancel

Identity Communication Profile Membership Contacts

Communication Profile Password

PROFILE SET: Primary

Communication Address

PROFILES

Session Manager Profile

CM Endpoint Profile

System: devcon-cm

Profile Type: Endpoint

Extension: 78010

Set Type: 9630SIP

Template: 9630SIP_DEFAULT_CM_8_0

Sub Type: Select

System ID: Enter System Id

Port: IP

Preferred Handle: Select

Sip Trunk: aar

Terminal Number:

Security Code: Enter Security Code

Voice Mail Number:

Calculate Route Pattern:

SIP URI: Select

Enhanced Callr-Info display for 1-line phones:

Delete on Unassign from User or on Delete User:

Override Endpoint Name and Localized Name:

Allow H.323 and SIP Endpoint Dual Registration:

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).

The screenshot shows the Avaya Aura System Manager 8.0 interface. The 'User Profile | Add' dialog box is open, and the 'Button Assignment' tab is selected. The dialog box contains the following fields and sections:

- Template:** 9630SIP_DEFAULT_CM_8_0
- Port:** IP
- Name:** (empty field)
- Set Type:** 9630SIP
- Security Code:** (empty field)
- General Options (G):** (selected tab)
- Feature Options (F):** (empty field)
- Site Data (S):** (empty field)
- Abbreviated Call Dialing (A):** (empty field)
- Enhanced Call Fwd (E):** (empty field)
- Button Assignment (B):** (selected tab)
- Group Membership (M):** (empty field)
- Main Buttons:**

	call-appr	Auto-A/D	Ring		
1	call-appr	Auto-A/D	Ring		
2	call-appr	Auto-A/D	Ring		
3	call-appr	Auto-A/D	Ring		
4	call-appr				
5	call-appr				
6	call-appr				

At the bottom of the dialog box, there are two checkboxes:

- Enhanced Callr-Info display for 1-line phones:** (unchecked)
- Delete on Unassign from User or on Delete User:** (checked)


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:

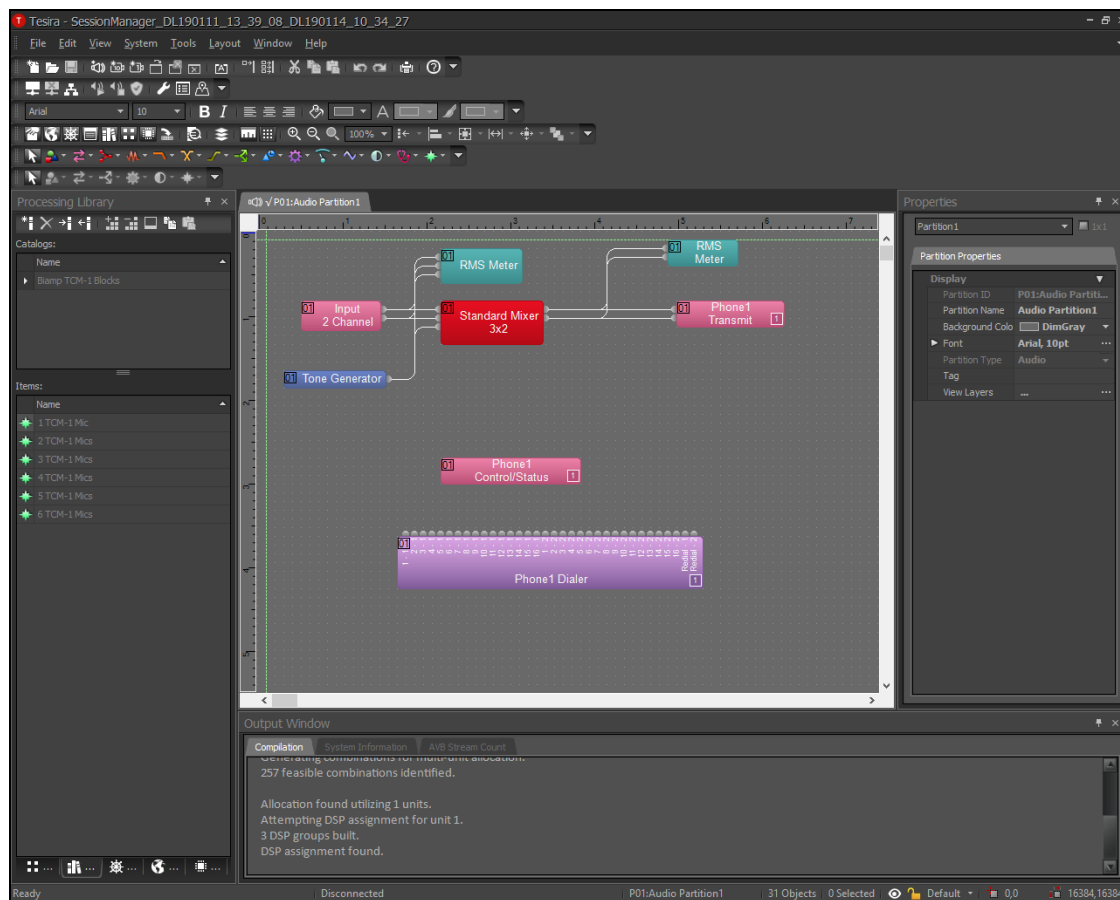
- Launch Tesira Software Application
- Modify the IP Network Settings of the Tesira SERVER-IO Control Network
- Modify the IP Network Settings of Tesira SVC-2
- Configure SIP Parameters of Tesira SVC-2
- Verify Codec Settings
- Save and Send the New Configuration to the System

7.1. Launch Tesira Software Application



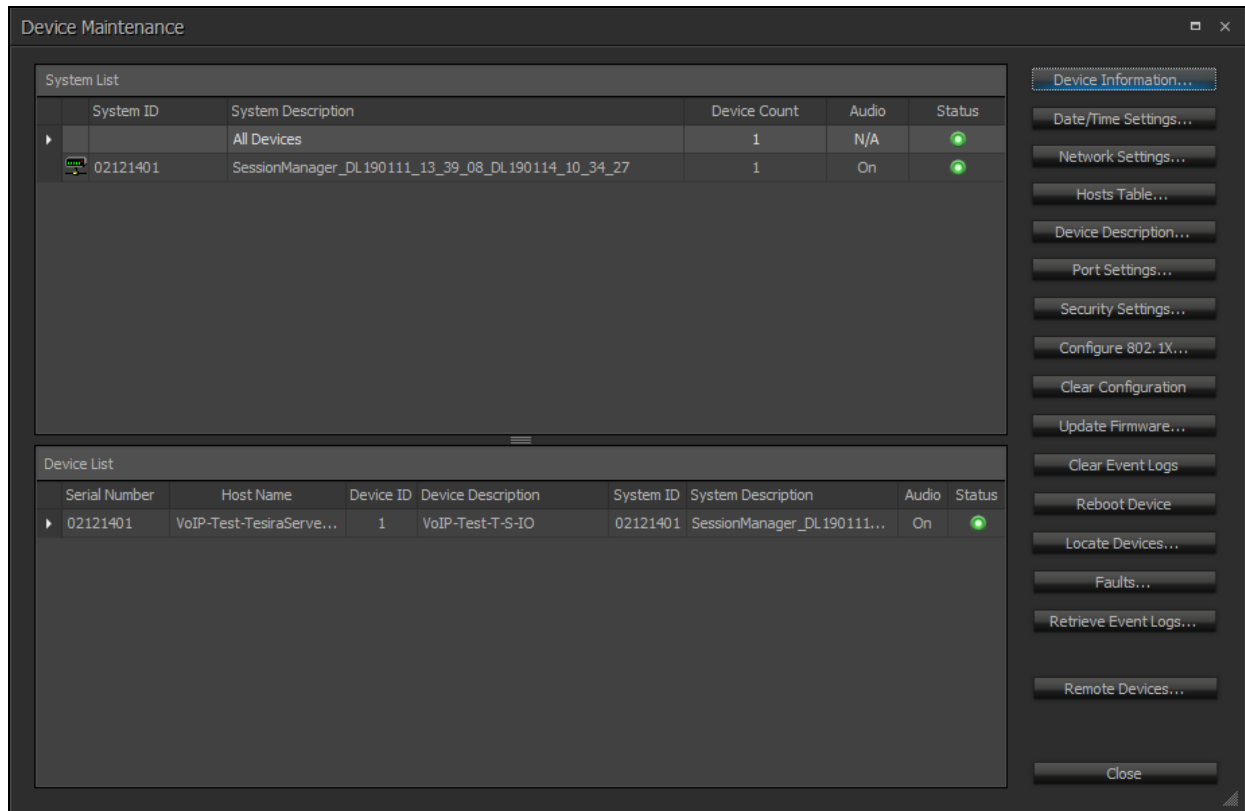
Launch the **Tesira Software** application by clicking on the  icon. The main window is displayed as shown below.

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.

The screenshot shows the 'Network Settings' window with the following sections:

- Control Network** (selected tab)
 - Host Name:** Empty text field. **Current Host Name:** VoIP-Test-TesiraServer02121401
 - DNS Configuration:**
 - Primary DNS Server:** 0 . 0 . 0 . 0. **Current Primary DNS Server:** 0.0.0.0
 - Alternate DNS Server:** 0 . 0 . 0 . 0. **Current Alternate DNS Server:** 0.0.0.0
 - Domain:** Empty text field. **Current Domain:** Empty text field
 - ☒ **Enable Multicast DNS**
 - Services:**
 - ☒ **Enable Telnet**
 - ☒ **Enable SSH**
 - Interface IP Configurations:**
 - ☒ **Enabled**. **Interface ID:** control
 - ☐ **Obtain an IP Address Automatically**
 - ☐ **Use the Following IP Address**
 - IP Address:** 192 . 168 . 100 . 244. **Current IP Address:** 192.168.100.244
 - Net Mask:** 255 . 255 . 255 . 0. **Current Net Mask:** 255.255.255.0
 - Default Gateway:** 192 . 168 . 100 . 1. **Current Default Gateway:** 192.168.100.1 (Active)
- Buttons:** OK, Cancel, Interface Status...

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.

Phone1 Control/Status

Line 1 Line 2 Switch to Standard

General Network Protocol Quality of Service NAT Statistics Normalization

Network

MAC Address

DHCP Server

DHCP

IP Address 192.168.100.245

Netmask 255.255.255.0

Gateway 192.168.100.1

DNS Primary 0.0.0.0 ...

DNS Secondary 0.0.0.0 ...

Domain Name

Detect Duplicated IP

VLAN Tagging

VLAN ID 1

Enable HTTP

Enable HTTPS

Enable Telnet

Ethernet

Speed

Duplex

Pad Short Frame Packets

Accept Short Frame Packets

Provisioning Server

TFTP Server Mode

TFTP Server Address ...

DHCP Custom Option 150

802.1X

Mode

Time

Time Synchronization Mode

Synchronized Time ...

SNTP Address ...

Daylight Savings Time

Time Synchronization Interval

Time Zone ((GMT-08:00) Pacific Time (US & ...))

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*.

The screenshot shows the 'Phone1 Control/Status' window with the 'Protocol' tab selected. The 'SIP' section contains the following fields and values:

Field	Value	Field	Value
Extension	78010	Registration Expiration	3600 seconds
Display Name	78010	Signaling Port	5060
SIP Domain Name		T1 Timer	500 ms
Authen User Name (Ext)	78010	Retransmit Timeout	32000 ms
Password	*****	Session Timer	Enabled
NetBIOS Domain Name		Session Refresher	Auto
Proxy Vendor	Avaya SM	Session Expiration	180 seconds
SIP Proxy Address	10.64.102.117	Minimum Session Expiration	90 seconds
SIP Proxy Port	5060	Prack	None
Registration Status		Transport	UDP

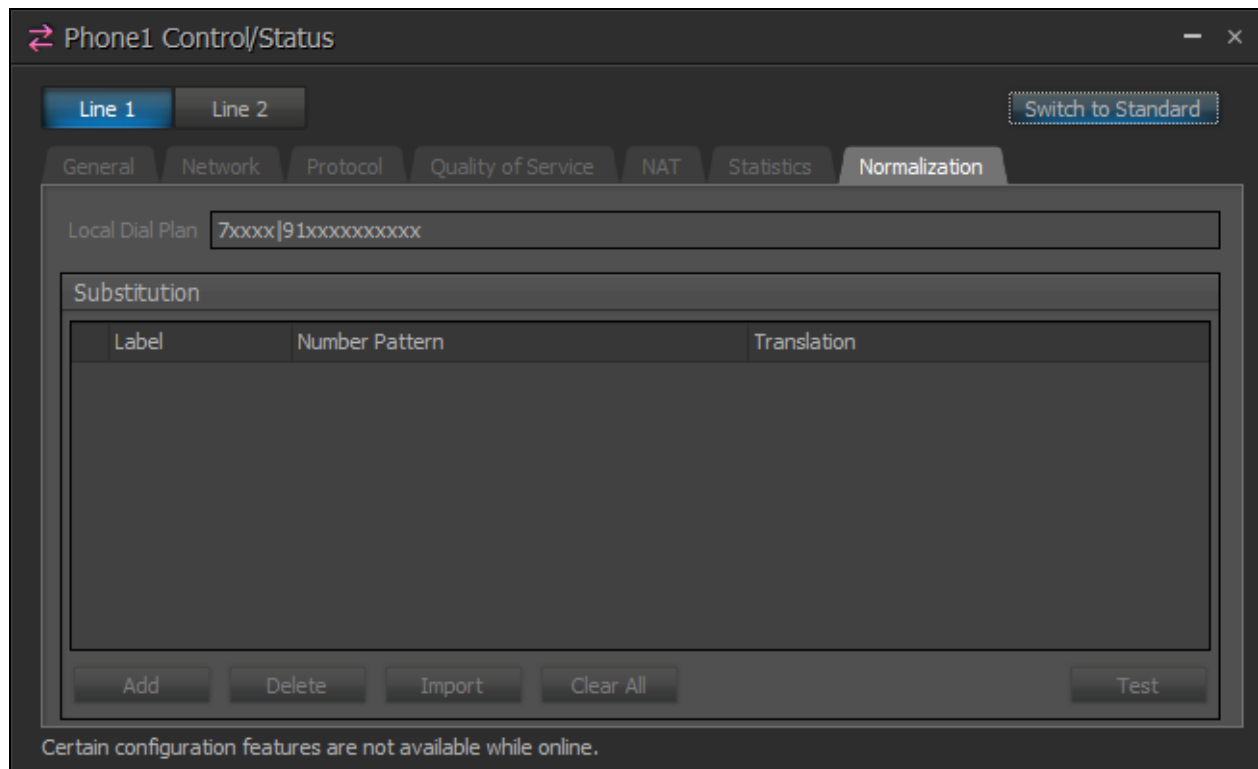
The 'RTP/SRTP' section contains the following fields and values:

Field	Value
Port Start	10000
Port End	14999
Static RTP Port	Enabled
SRTP	Disabled
G.723.1 Encoding Rate	5.3 kbps
Suppress RTCP On Hold	Enable

The 'SIPS' section contains the following fields and values:

Field	Value
Keyword	
SIPS URI	Enable
Certificate Preference	Accept All
Root Certificate File Name	
Customized Certificate File Name	
Certificate File Name	
Private Key File Name	

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|91xxxxxxxxxx` 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.

The screenshot shows the 'Phone1 Control/Status' window with the 'General' tab selected. The window is divided into several sections: 'Dial Plan', 'Tones', 'Call Features', and 'Voice Features'. The 'Voice Features' section contains the 'Voice Codec Priorities' table.

Dial Plan

- Dialing Timeout (s): 3

Tones

- Local DTMF: Mute
- Local DTMF Level: -6.0
- DTMF Transmit Level: -6.0
- Ring Type: Classic
- DTMF On Time (ms): 50
- DTMF Off Time (ms): 50
- Call Progress Tone Level: -20.0
- Out-Of-Band DTMF: Enabled
- Out-Of-Band DTMF Payload Type: 101
- DTMF via SIP Info: Off

Call Features

- Auto Answer: Enable
- Auto Answer Ring Count: 2 Rings
- Redial: Enabled
- Consultative Transfer: Enable
- Caller Id: Enabled
- Do Not Disturb: Enable
- Do Not Disturb Response: Do Not Disturb (480)
- RFC 2543 style Hold: Enable
- Direct URL Dialing: Enable
- Use One Audio Format: Enable
- Refresh Method: UPDATE

Voice Features

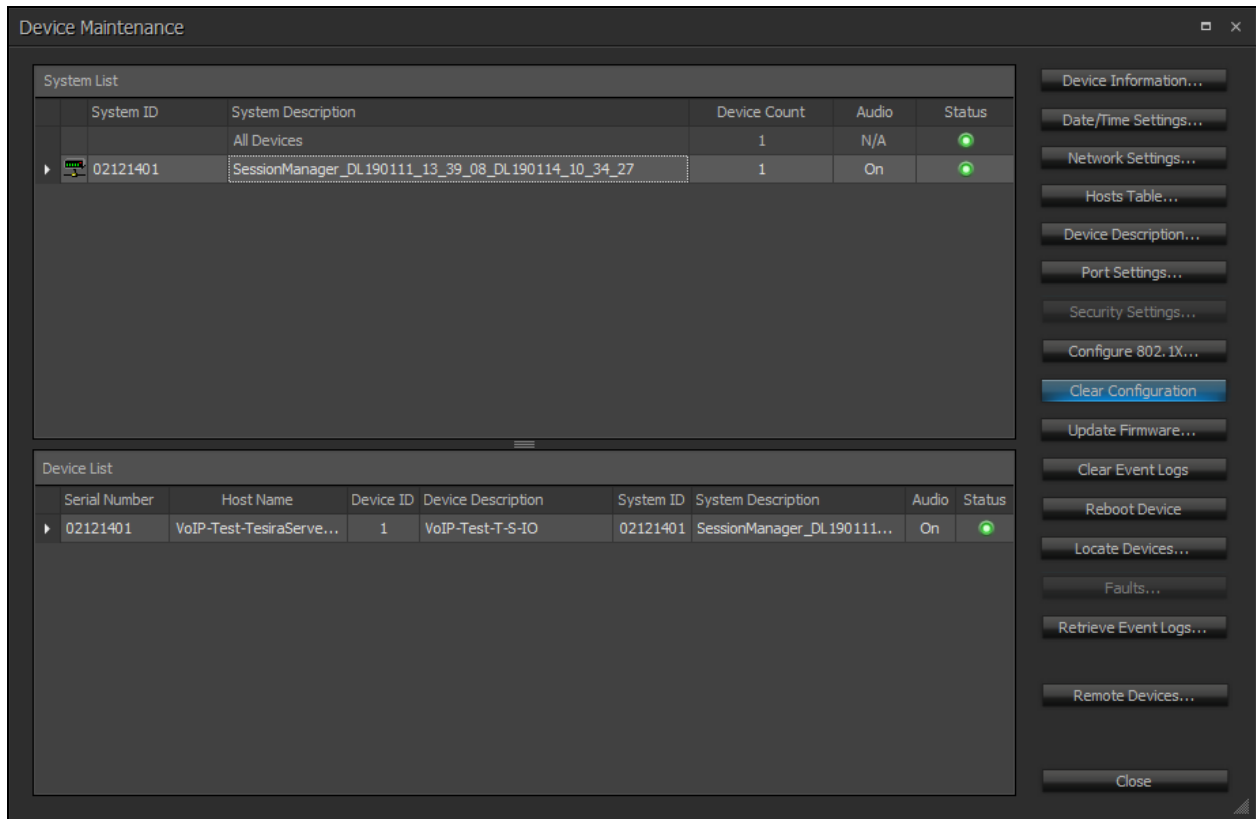
- VAD: Enabled
- VAD Threshold: -40.0
- Voice Codec Priorities: Up, Down

Voice Codec Priorities Table

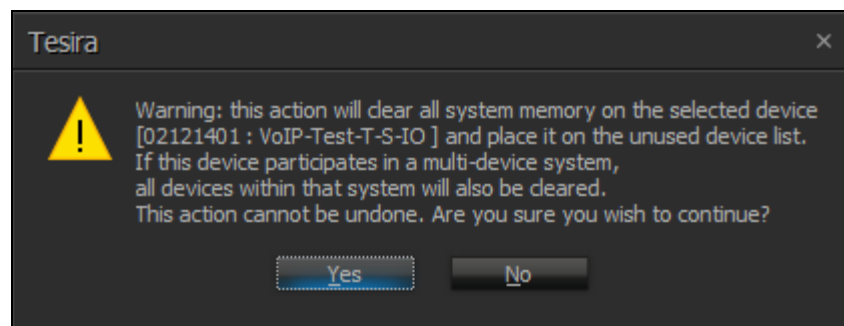
Use	Codec	Jitter Buffer Min	Jitter Buffer Max
<input type="checkbox"/>	G722	40	200
<input checked="" type="checkbox"/>	G711U	40	200
<input type="checkbox"/>	G711A	40	200
<input type="checkbox"/>	G729AB	40	200
<input type="checkbox"/>	G7231	40	200

7.6. Save and Send the New Configuration to the System

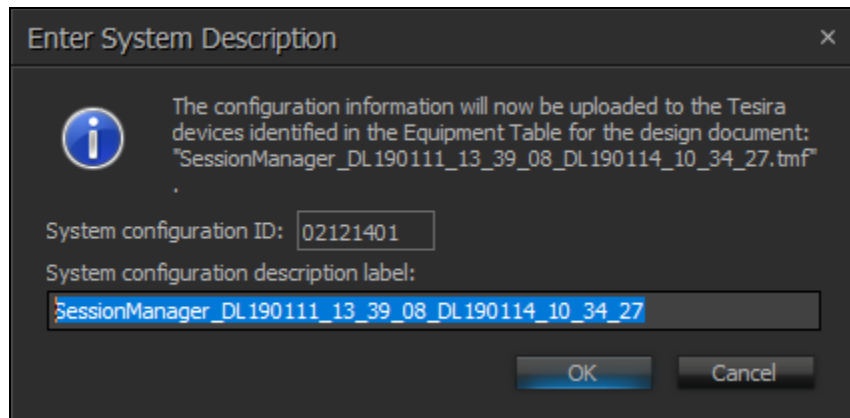
From the **Tesira Software** main window, save the configuration by clicking on **File → Save As** (not shown). Next, clear the configuration from the device before sending the new configuration to the device. Select the device and then click on the **Clear Configuration** button as shown below.



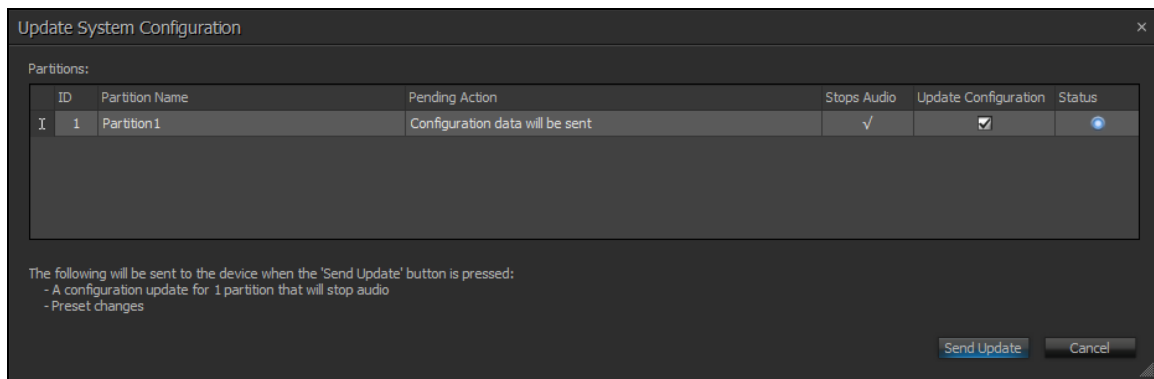
Confirm that the system memory will be cleared.



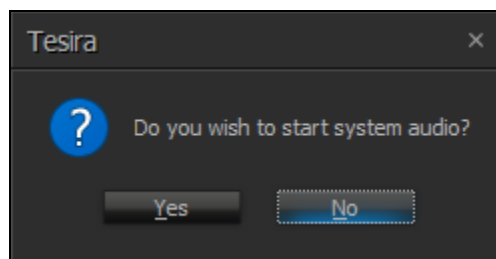
Lastly, click the Send Configuration icon in the main window to send the new configuration to the device (not shown). Confirm the **System configuration description label** by clicking **OK** in the window below.



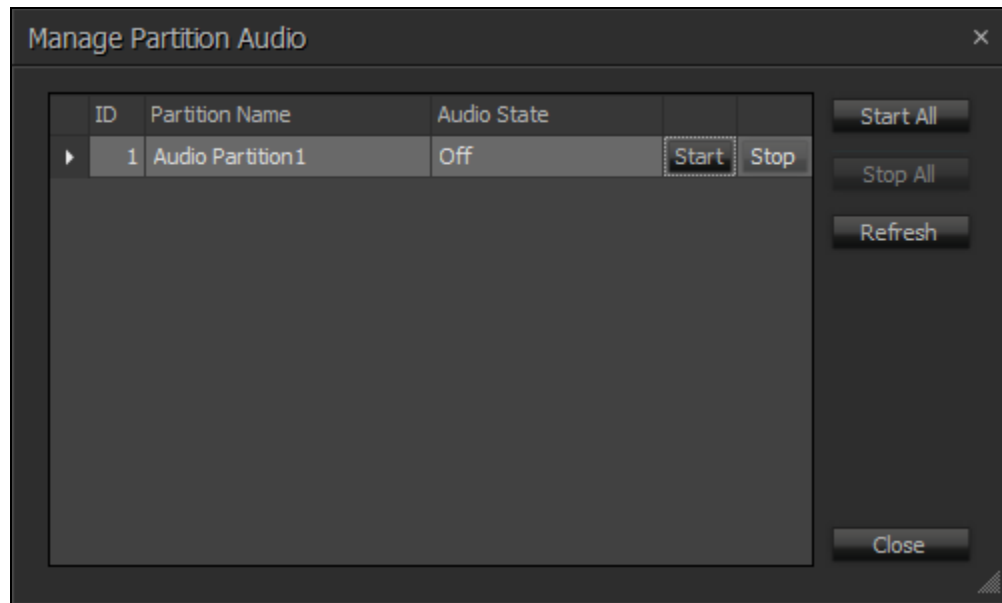
In the **Update System Configuration** window shown below, select the **Update Configuration** checkbox and click the **Send Update** button, if prompted.



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.

AVAYA Aura® System Manager 8.0

Users ▾ Elements ▾ Services ▾ Widgets ▾ Shortcuts ▾ Search [] admin

Home Session Manager

Global Settings
Communication Pro...
Network Configur... ▾
Device and Locati... ▾
Application Confi... ▾
System Status ▴
SIP Entity Monit...
Managed Band...
Security Module...
SIP Firewall Status
Registration Su...
User Registrations
Session Counts

User Registrations

Select rows to send notifications to devices. Click on Details column for complete registration status.

View ▾ Default Export Force Unregister AST Device Notifications: Reboot Reload ▾ Failback As of 12:55 PM Customize ▾ Advanced Search ▾

17 Items Show 15 ▾ Filter: Enable

<input type="checkbox"/>	Details	Address	Login Name	First Name	Last Name ▲	Home Location	IP Address	Simult. Devices	AST Device	Registered	Prim	Sec	Surv
<input type="checkbox"/>	► Show	78000@avaya.com	78000@avaya.com	SIP	78000	Thornton	192.168.100.54	1/1	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/> (AC)	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>
<input type="checkbox"/>	► Show	78001@avaya.com	78001@avaya.com	SIP	78001	Thornton	192.168.100.58	1/1	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/> (AC)	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>
<input type="checkbox"/>	► Show	78002@avaya.com	78002@avaya.com	SIP	78002	Thornton	192.168.100.53	1/1	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/> (AC)	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>
<input type="checkbox"/>	► Show	78010@avaya.com	78010@avaya.com	Tesira	78010	Thornton	192.168.100.245	1/1	<input type="checkbox"/>	<input checked="" type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>
<input type="checkbox"/>	► Show	---	78040@avaya.com	Equinox	78040	Thornton	---	0/1	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>
<input type="checkbox"/>	► Show	---	78400@avaya.com	SIP	78400	Thornton	---	0/1	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>
<input type="checkbox"/>	► Show	---	78401@avaya.com	H175	78401	Thornton	---	0/1	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>

Select : All, None Page 1 of 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*.

The screenshot shows the 'Phone1 Control/Status' window with the 'Protocol' tab selected. The window is divided into three main sections: SIP, RTP/SRTP, and SIPS.

SIP Configuration:

Field	Value	Field	Value
Extension	78010	Registration Expiration	3600 seconds
Display Name	78010	Signaling Port	5060
SIP Domain Name		T1 Timer	500 ms
Authen User Name (Ext)	78010	Retransmit Timeout	32000 ms
Password	*****	Session Timer	Enabled
NetBIOS Domain Name		Session Refresher	Auto
Proxy Vendor	Avaya SM	Session Expiration	180 seconds
SIP Proxy Address	10.64.102.117	Minimum Session Expiration	90 seconds
SIP Proxy Port	5060	Prack	None
Registration Status	Registered	Transport	UDP

RTP/SRTP Configuration:

Field	Value
Port Start	10000
Port End	14999
Static RTP Port	Enabled
SRTP	Disabled
G.723.1 Encoding Rate	5.3 kbps
Suppress RTCP On Hold	Enable

SIPS Configuration:

Field	Value
Keyword	
SIPS URI	Enable
Certificate Preference	Accept All
Root Certificate File Name	
Customized Certificate File Name	
Certificate File Name	
Private Key File Name	

Certain configuration features are not available while online.

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.

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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	<p>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.</p> <ul style="list-style-type: none"> • 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	<p>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</p>

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	<p>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.</p> <ul style="list-style-type: none"> • 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	<p>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.</p> <ul style="list-style-type: none"> • 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	<ul style="list-style-type: none"> • 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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	<p>combined with other TesiraFORTÉ AVB devices and Tesira servers, expanders, endpoints, and controllers.</p> <ul style="list-style-type: none"> • 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	<p>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.</p> <ul style="list-style-type: none"> • 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	<ul style="list-style-type: none"> • 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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	<ul style="list-style-type: none"> • 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	<p>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.</p> <ul style="list-style-type: none"> • 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	<p>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</p>

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	<p>mediumsized rooms that require high-quality audio solutions using VoIP, voice lift, mix-minus, and AEC such as board rooms or distance training facilities.</p> <ul style="list-style-type: none"> • 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	<p>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.</p> <ul style="list-style-type: none"> • 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

AUDIO. VIDEO. CONTROL.



	<ul style="list-style-type: none">• Signal processing via intuitive software allows configuration and control for signal routing, mixing, equalization, filtering, delay, and much more
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Please don't hesitate to contact us if you have further concerns.

Sincerely yours,

A handwritten signature in black ink, appearing to read "Jason Damori".

Jason Damori
Vice President of Engineering

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