



## Avaya Solution & Interoperability Test Lab

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# Application Notes for Biamp Tesira SVC-2 with Avaya IP Office Server Edition 11.0 – Issue 1.0

### Abstract

These Application Notes describe the configuration steps required to integrate the Biamp Tesira SVC-2 with Avaya IP Office Server Edition. 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 IP Office Server Edition 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 IP Office Server Edition. 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 IP Office Server Edition 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 IP Office Server Edition. 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 and 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.

The serviceability testing focused on verifying that the Tesira SVC-2 card came back into service after re-connecting the Ethernet cable or rebooting the 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 card with IP Office Server Edition and IP Office 500 V2 Expansion System.
- Calls between Tesira SERVER-IO with Tesira SVC-2 card and Avaya SIP/H.323 IP Deskphones on IP Office Server Edition and IP Office 500 V2 Expansion System.
- Calls between Tesira SVC-2 and the PSTN.
- G.711, G.729, and G.722 codec support.
- Proper recognition of DTMF tones.
- Basic telephony features including hold, mute, redial, multiple calls, blind and attended transfer calls, and 3-party conference.
- Extended telephony features using IP Office short codes for Call Forward and Call Pickup.
- Use of programmable buttons on Tesira SVC-2.
- Proper system recovery after a restart of Tesira SERVER-IO with Tesira SVC-2 and loss of IP connectivity.

## **2.2. Test Results**

All test cases passed. Note that blind conference is not supported but attended/supervised conference is supported.

## **2.3. Support**

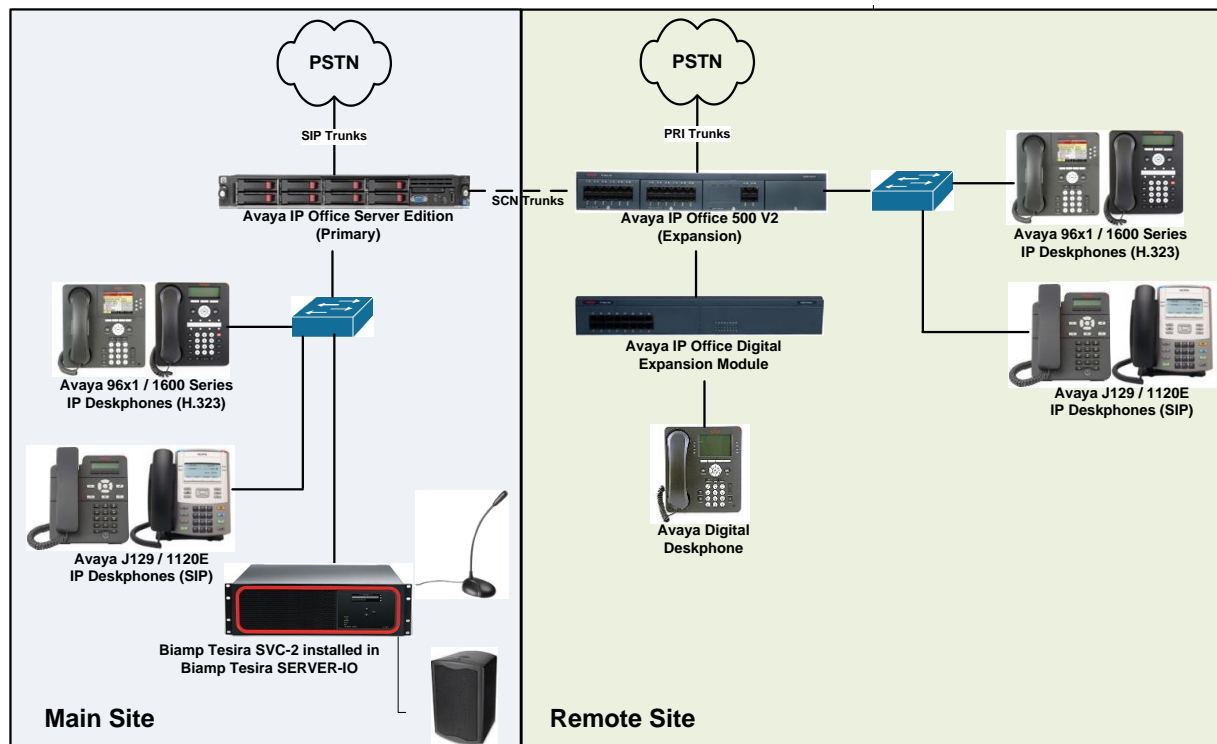
For technical support and information on Biamp Tesira SVC-2, contact Biamp Support at:

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

### 3. Reference Configuration

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

- Avaya IP Office Server Edition and Avaya IP Office 500 V2 Expansion connected via a SCN trunk and configured via Avaya IP Office Manager.
- PSTN connectivity provided by a SIP trunk on Avaya IP Office Server Edition and an ISDN-PRI trunk on Avaya IP Office 500 V2 Expansion System.
- Avaya 96x1 Series H.323 Deskphones, Avaya J129 SIP Deskphones, and Avaya 1100/1200 Series SIP Deskphones registered to Avaya IP Office Server Edition and Avaya IP Office 500 V2 Expansion.
- Biamp Tesira SVC-2, installed in Biamp Tesira SERVER-IO, provided connectivity to Avaya IP Office Server Edition. Biamp Tesira SVC-2 registered with Avaya IP Office Server Edition as a SIP endpoint. Tesira Software application was used to configure Biamp Tesira products.



**Figure 1: Avaya IP Office Server Edition with Biamp Tesira SVC-2 installed in Biamp Tesira SERVER-IO**

## 4. Equipment and Software Validated

The following equipment and software were used for the test configuration.

Equipment/Software	Release/Version
Avaya IP Office Server Edition	11.0.0.2.0 Build 23
Avaya IP Office 500 V2 Expansion	11.0.0.2.0 Build 23
Avaya 96x1 Series IP Deskphones	6.6604 (H.323)
Avaya 1100/1200 Series IP Deskphones	04.04.26.00 (SIP)
Avaya J129 SIP Deskphones	3.0.0.1.6 (6)
Biamp Tesira SVC-2 installed in Biamp Tesira SERVER-IO	v3.8.0.24
Biamp Tesira Software	v3.8.0.14

*Compliance Testing is applicable when the tested solution is deployed with a standalone IP Office 500 V2 and also when deployed with IP Office Server Edition in all configurations.*

## 5. Configure Avaya IP Office Server Edition

This section provides the procedures for configuring Avaya IP Office Server Edition. The procedures include the following areas:

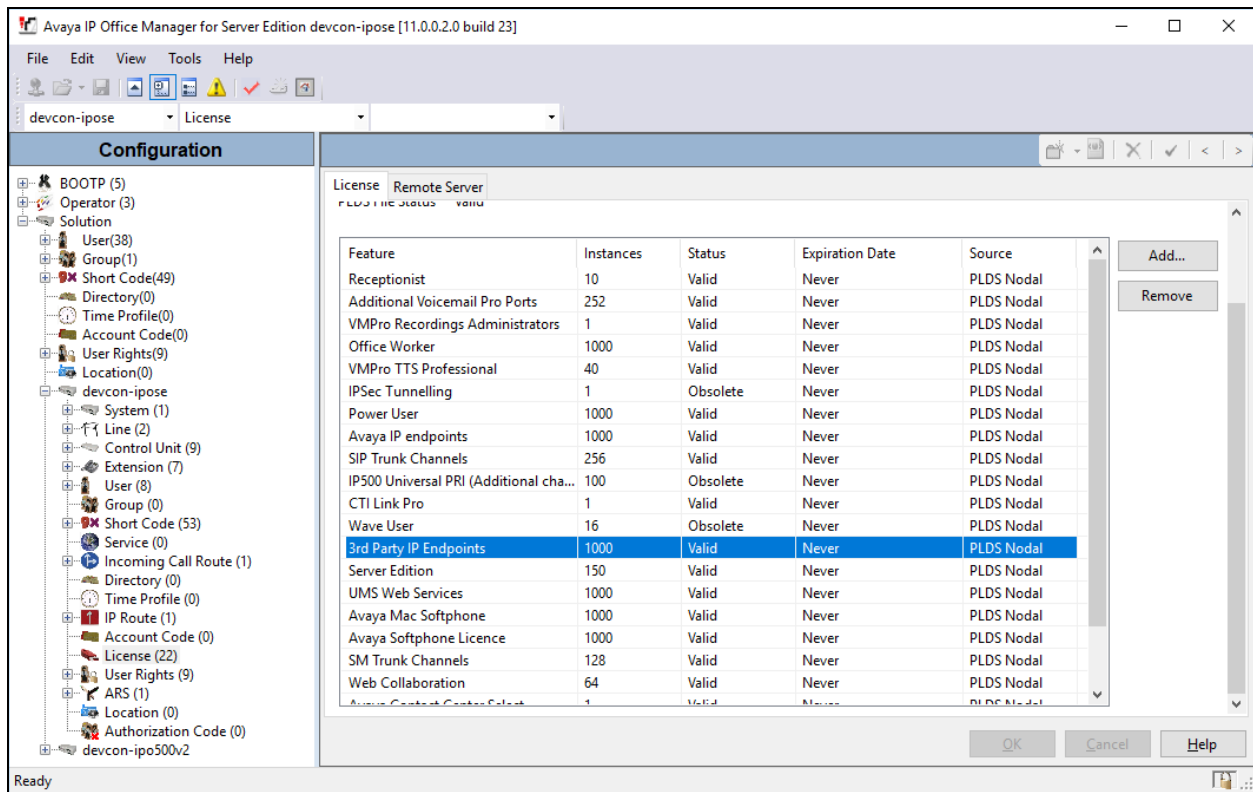
- Verify IP Office License
- Obtain LAN IP Address
- Administer SIP Registrar
- Administer SIP Extension
- Administer SIP User

**Note:** Integration of IP Office 500 V2 Expansion and call routing to the PSTN are outside the scope of these Application Notes.

### 5.1. Verify IP Office License

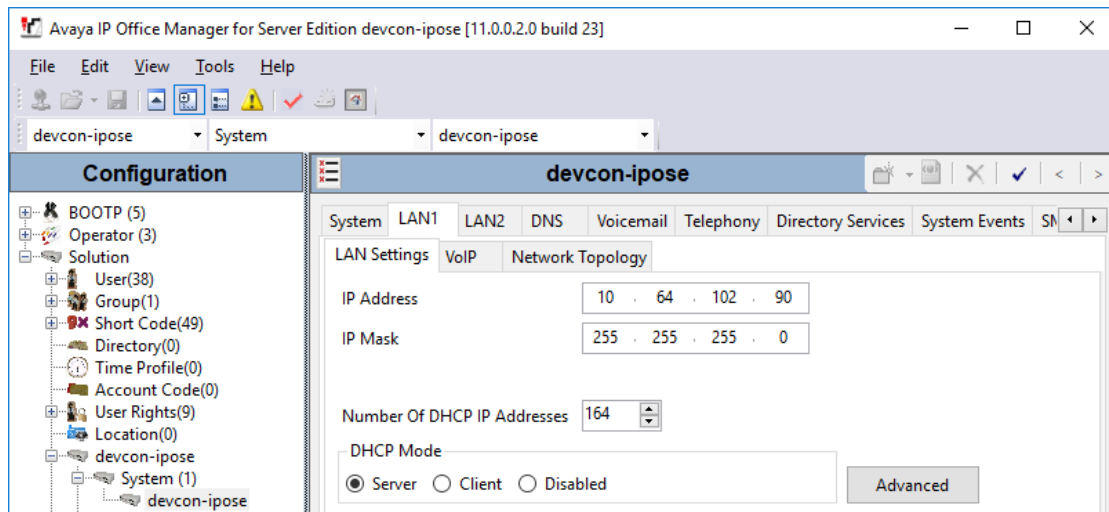
From a PC running the Avaya IP Office Manager application, select **Start** → **Programs** → **IP Office** → **Manager** to launch the Manager application. Select the required IP Office system and log in with the appropriate credentials.

The **Avaya IP Office Manager for Server Edition** screen is displayed. From the configuration tree in the left pane, select **License** to display the license screen in the right pane. Verify that the **License Status** is *Valid* for **3rd Party IP Endpoints**.



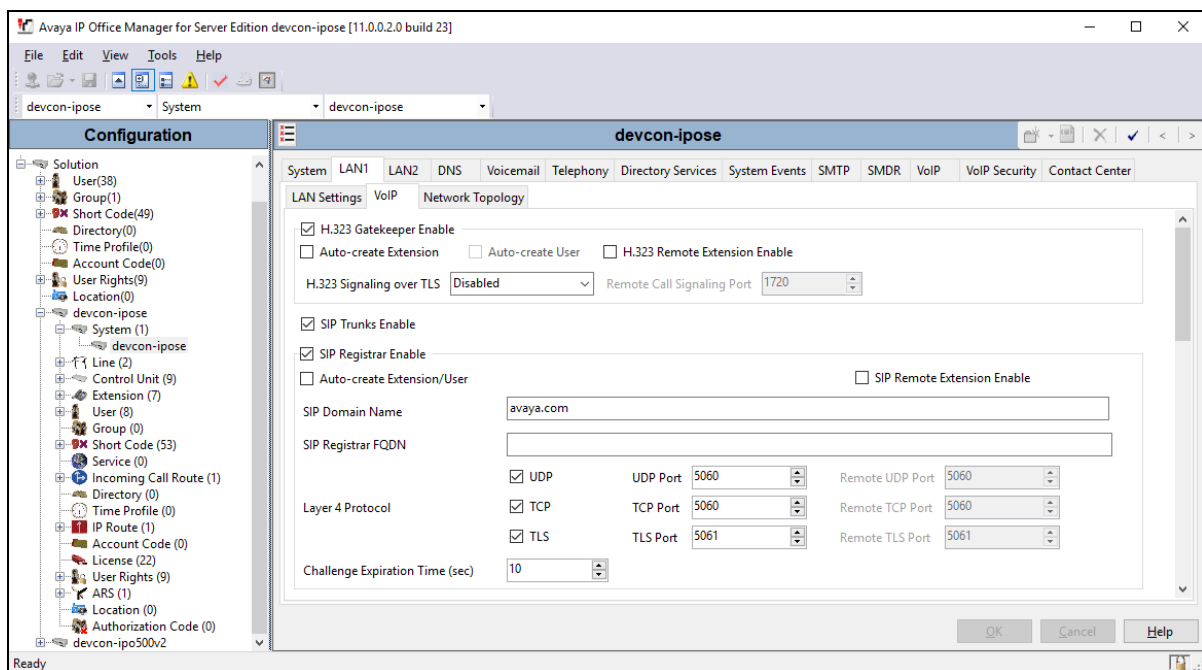
## 5.2. Obtain LAN IP Address

From the configuration tree in the left pane, select **System** to display the **System** screen for the IP Office Server Edition in the right pane. Select the **LAN1** tab, followed by the **LAN Settings** sub-tab in the right pane. Make a note of the **IP Address**, which will be used later to configure Tesira SVC-2.

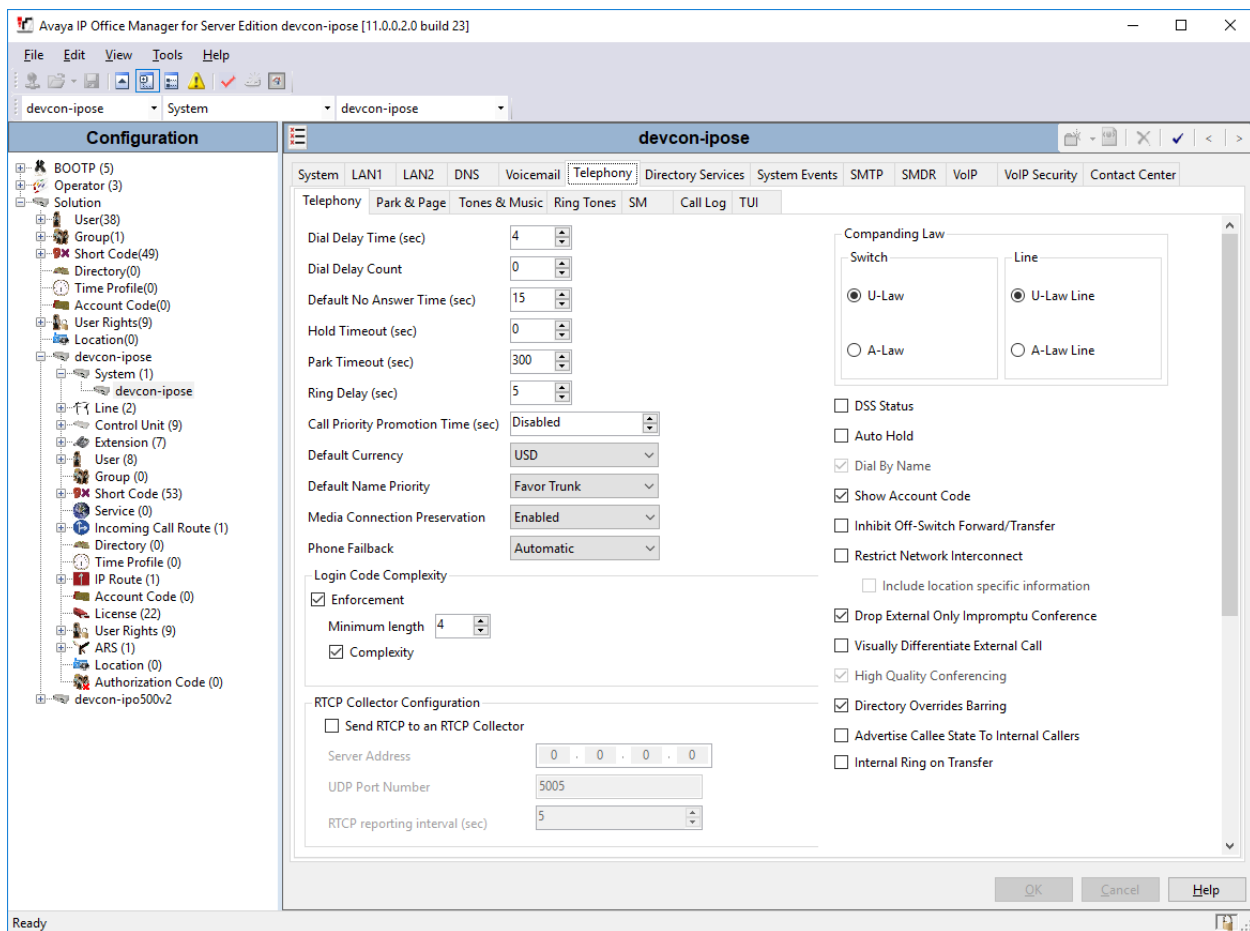


## 5.3. Administer SIP Registrar

Select the **VoIP** sub-tab. Ensure that **SIP Registrar Enable** and that UDP transport is selected, which will be used by Tesira SVC-2. Also, enter a valid **SIP Domain Name**.



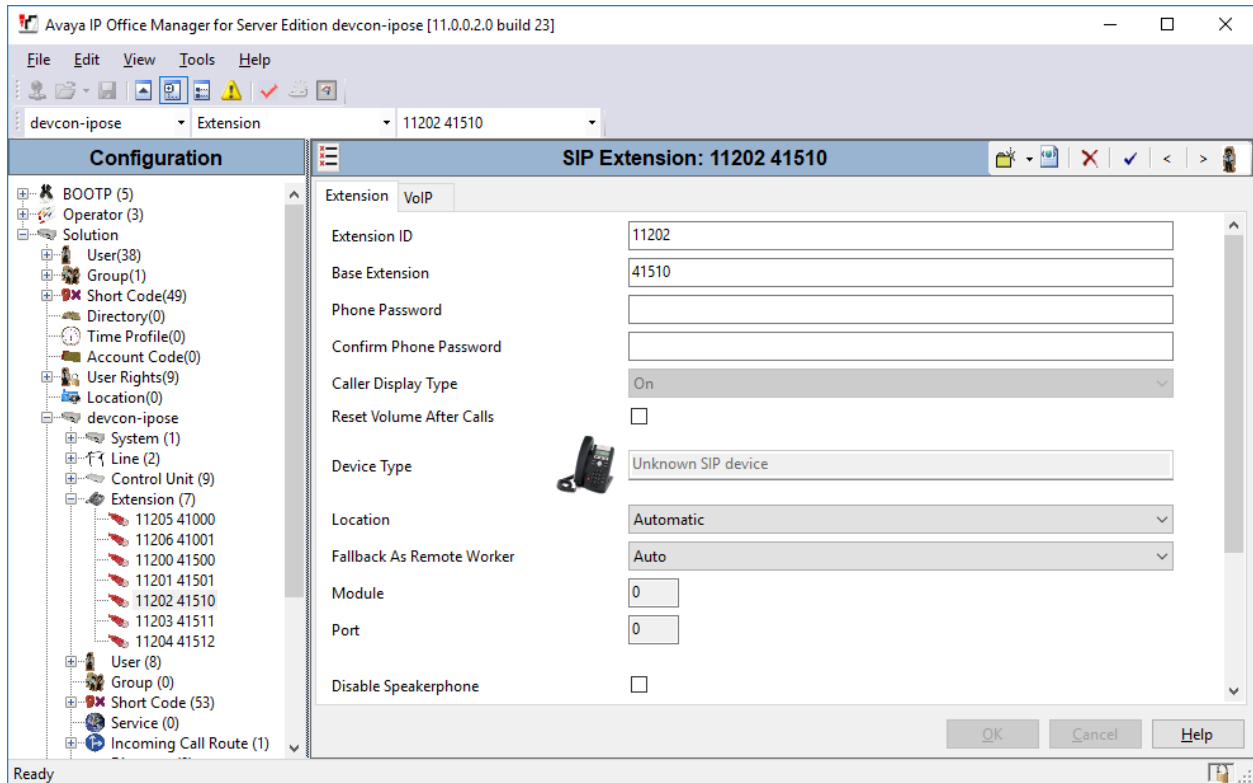
Select the **Telephony** tab followed by the **Telephony** sub-tab as shown below. Verify that **Inhibit Off-Switch Forward/Transfer** is not checked so that transfers and conferences with the PSTN is allowed.



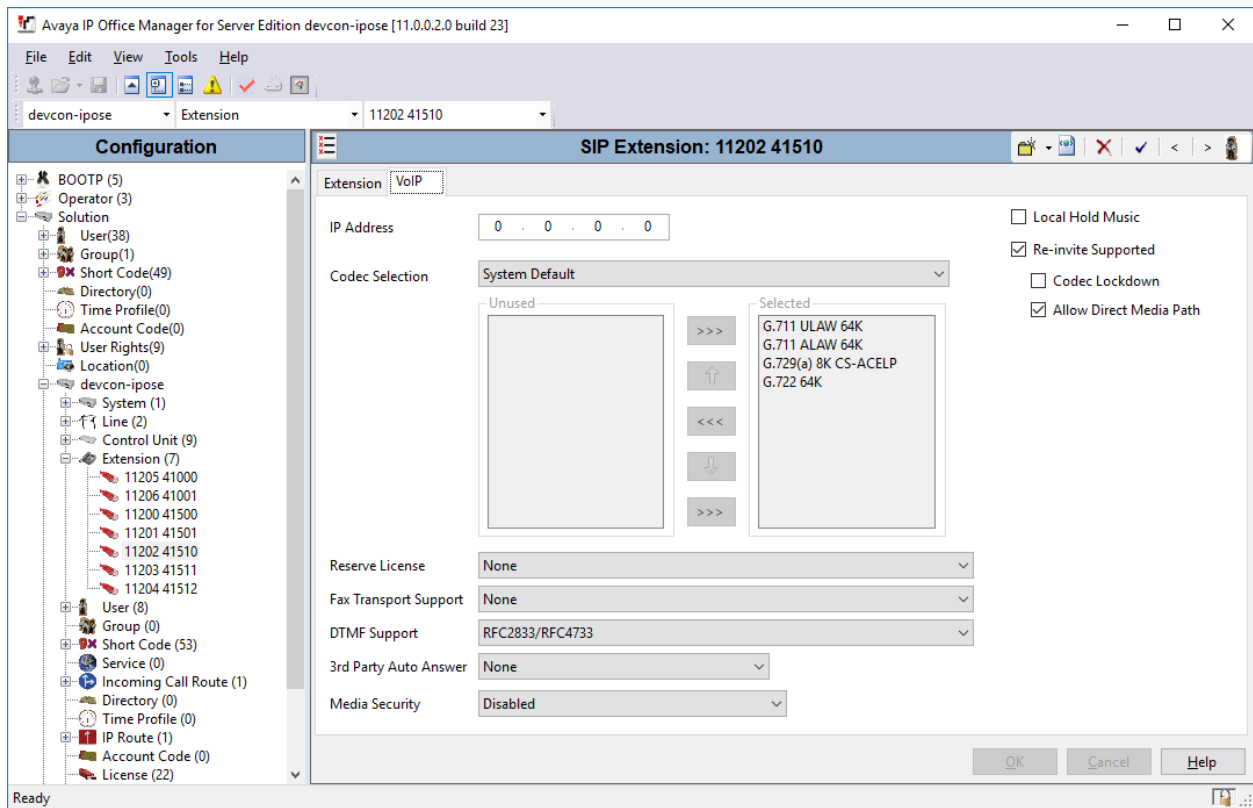


## 5.4. Administer SIP Extension

From the configuration tree in the left pane, right-click on **Extension** and select **New → SIP** from the pop-up list to add a new SIP extension (not shown). Enter the desired extension for the **Base Extension** field as shown below. In this example, Tesira SVC-2 was assigned extension *41510*.



Select the **VoIP** tab and retain the default values in the all fields. During the compliance test, Tesira SVC-2 was configured to support G.711, G.729, and G.722 codecs.



## 5.5. Administer SIP User

From the configuration tree in the left pane, right-click on **User** and select **New** from the pop-up list. Enter desired values for the **Name** and **Full Name** fields. For the **Extension** field, enter the SIP extension created in **Section 5.4**. The **Extension** field specifies the username that will be used by Tesira SVC-2 to register with IP Office Server Edition. Note that the SIP authentication password was configured in the **Telephony** tab below instead of in the **Password** field shown below.

The screenshot shows the Avaya IP Office Manager for Server Edition configuration window. The left pane displays a configuration tree with the following structure:

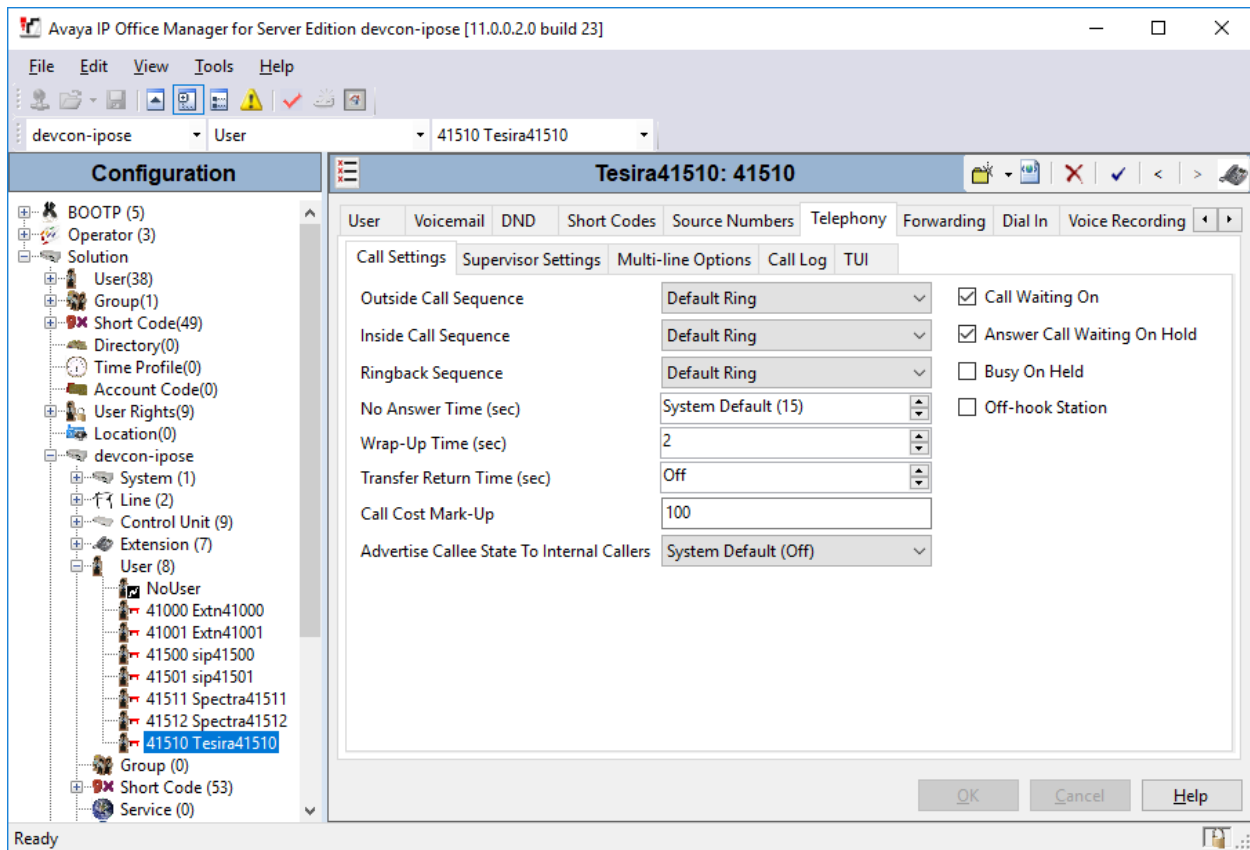
- BOOTP (5)
- Operator (3)
- Solution
  - User(38)
  - Group(1)
  - Short Code(49)
  - Directory(0)
  - Time Profile(0)
  - Account Code(0)
  - User Rights(9)
  - Location(0)
- devcon-ipose
  - System (1)
  - Line (2)
  - Control Unit (9)
  - Extension (7)
    - User (8)
      - NoUser
      - 41000 Extn41000
      - 41001 Extn41001
      - 41500 sip41500
      - 41501 sip41501
      - 41511 Spectra41511
      - 41512 Spectra41512
      - 41510 Tesira41510**
    - Group (0)
    - Short Code (53)
    - Service (0)
    - Incoming Call Route (1)
    - Directory (0)
    - Time Profile (0)
    - IP Route (1)
    - Account Code (0)
    - License (22)
    - User Rights (9)
    - ARS (1)
    - Location (0)
    - Authorization Code (0)
- devcon-ipo500v2

The right pane shows the configuration for the selected user, **Tesira41510: 41510**. The configuration is organized into tabs: **User**, **Voicemail**, **DND**, **Short Codes**, **Source Numbers**, **Telephony**, **Forwarding**, **Dial In**, **Voice Recording**, and **Button Program**. The **User** tab is active, showing the following fields:

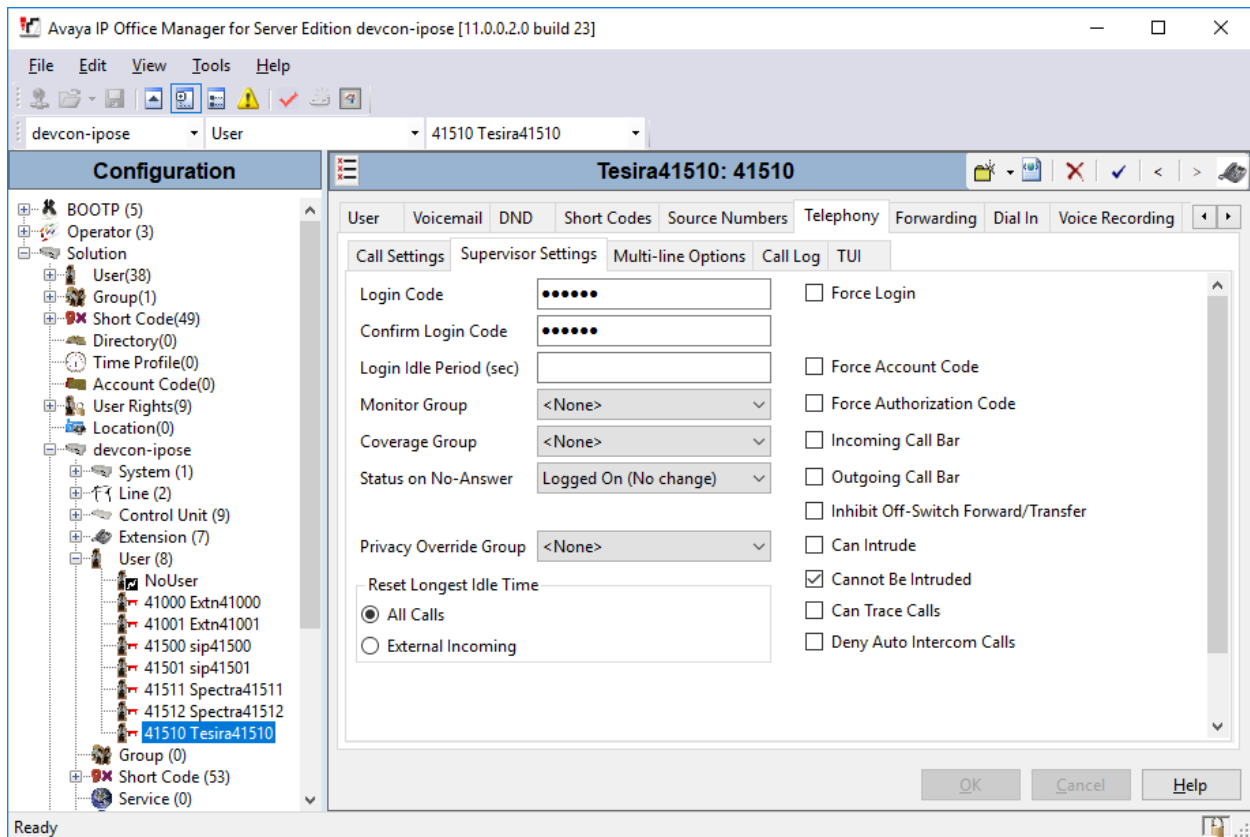
- Name: Tesira41510
- Password: (empty)
- Confirm Password: (empty)
- Unique Identity: (empty)
- Conference PIN: (empty)
- Confirm Audio Conference PIN: (empty)
- Account Status: Enabled (dropdown)
- Full Name: Tesira
- Extension: 41510
- Email Address: (empty)
- Locale: (empty)
- Priority: 5 (dropdown)
- System Phone Rights: None (dropdown)
- Profile: Basic User (dropdown)
- Receptionist: ☐
- Enable Softphone: ☐
- Enable one-X Portal Services: ☐
- Enable one-X TeleCommuter: ☐
- Enable Remote Worker: ☐
- Enable Desktop/Tablet VoIP client: ☐
- Enable Mobile VoIP Client: ☐
- Send Mobility Email: ☐
- Web Collaboration: ☐
- Exclude From Directory: ☐
- Device Type: Unknown SIP device
- User Rights
  - User Rights view: User data (dropdown)
  - Working hours time profile: <None> (dropdown)
  - Working hours User Rights: (empty)
  - Out of hours User Rights: (empty)

The bottom of the window shows the status bar with the text "Ready" and the "OK", "Cancel", and "Help" buttons.

Select the **Telephony** tab followed by the **Call Settings** sub-tab. Note the default settings were used for the user.



Select the **Supervisor Settings** sub-tab and enter a desired **Login Code**. The **Login Code** is the password that will be used by Tesira SVC-2 to register with IP Office Server Edition.




## 6. 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:

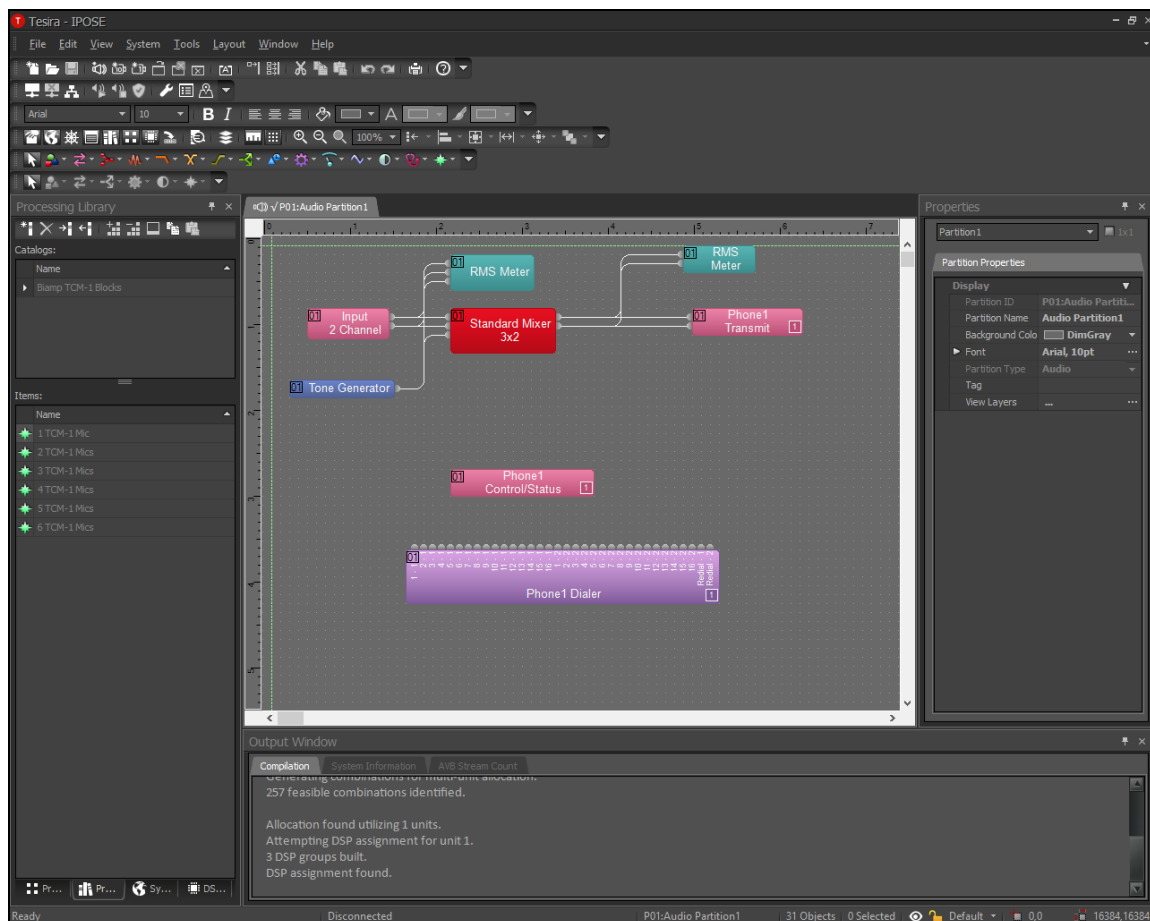
- Launch the Tesira Software Application
- Modify the IP Network Settings of the Tesira SERVER-IO Control Network
- Modify the IP Network Settings of Tesira SVC-2
- Configure SIP Parameters of Tesira SVC-2
- Verify Codec Settings
- Save and Send the New Configuration to the System

### 6.1. Launch Tesira Software Application



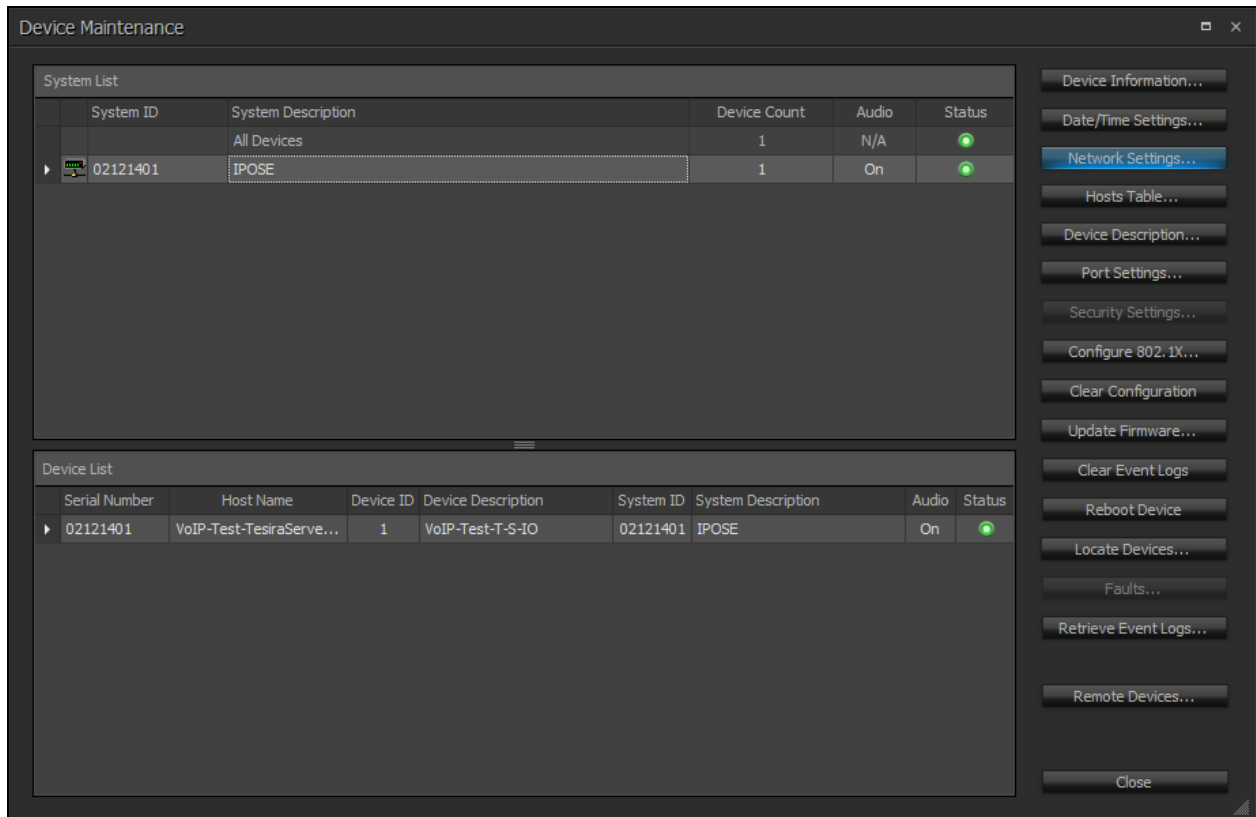
Launch the **Tesira Software** application by clicking on the  icon. The main window is displayed as shown below.

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



## 6.2. 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 6.6** to save and send the configuration to the system.

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

- Control Network** (selected tab) and **MAC Addresses** (inactive tab).
- Host Name**: Host Name field is empty; Current Host Name is 'VoIP-Test-TesiraServer02121401'.
- DNS Configuration**: Primary and Alternate DNS Servers are set to '0 . 0 . 0 . 0'; Current Primary and Alternate DNS Servers are '0.0.0.0'; Domain field is empty; 'Enable Multicast DNS' is checked.
- Services**: 'Enable Telnet' and 'Enable SSH' are both checked.
- Interface IP Configurations**:
  - 'Enabled' is checked.
  - 'Interface ID' is set to 'control'.
  - 'Obtain an IP Address Automatically' is selected (radio button).
  - 'Use the Following IP Address' is selected (radio button).
  - 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)'.
  - 'Interface Status...' button.

At the bottom right are 'OK' and 'Cancel' buttons.



### 6.3. 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 6.6** to save and send the configuration to the system.

The screenshot shows the 'Phone1 Control/Status' window with the 'Network' tab selected. The window has a title bar with a back arrow, the title 'Phone1 Control/Status', and standard window controls. Below the title bar are tabs for 'Line 1', 'Line 2', and a 'Switch to Standard' button. The 'Network' tab is active, showing various configuration sections: 'Network' (MAC Address, DHCP Server, DHCP Enable, IP Address: 192.168.100.245, Netmask: 255.255.255.0, Gateway: 192.168.100.1, DNS Primary/Secondary: 0.0.0.0, Domain Name, Detect Duplicated IP: Enabled, VLAN Tagging: Enable, VLAN ID: 1, Enable HTTP/HTTPS/Telnet: Enabled), 'Ethernet' (Speed: Auto, Duplex: Full, Pad Short Frame Packets: Enabled, Accept Short Frame Packets: Enabled), 'Provisioning Server' (TFTP Server Mode: None, TFTP Server Address, DHCP Custom Option: 150), '802.1X' (Mode), and 'Time' (Time Synchronization Mode: Host, Synchronized Time, SNTP Address, Daylight Savings Time: Enable, Time Synchronization Interval: 1024, Time Zone: (GMT-08:00) Pacific Time (US & ...)).

Section	Parameter	Value
Network	MAC Address	
	DHCP Server	
	DHCP	Enable
	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	Enabled
VLAN	VLAN Tagging	Enable
	VLAN ID	1
Enable	Enable HTTP	Enable
	Enable HTTPS	Enable
	Enable Telnet	Enabled
Ethernet	Speed	Auto
	Duplex	Full
	Pad Short Frame Packets	Enabled
	Accept Short Frame Packets	Enabled
Provisioning Server	TFTP Server Mode	None
	TFTP Server Address	
	DHCP Custom Option	150
802.1X	Mode	
Time	Time Synchronization Mode	Host
	Synchronized Time	
	SNTP Address	
	Daylight Savings Time	Enable
	Time Synchronization Interval	1024
	Time Zone	(GMT-08:00) Pacific Time (US & ...)

## 6.4. 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 IP Office Server Edition. The SIP username and password were configured on IP Office Server Edition in **Section 5.5**. Set the **Proxy Vendor** to *Avaya IP Office* and specify the **SIP Proxy Address** to the IP Office Server Edition IP address (i.e., *10.64.102.90*) 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	41510	Registration Expiration	3600 seconds
Display Name	41510	Signaling Port	5060
SIP Domain Name		T1 Timer	500 ms
Authen User Name (Ext)	41510	Retransmit Timeout	32000 ms
Password	*****	Session Timer	Enabled
NetBIOS Domain Name		Session Refresher	Auto
Proxy Vendor	Avaya IP Office	Session Expiration	1800 seconds
SIP Proxy Address	10.64.102.90	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	Enable
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 '4' 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 *4xxxx/91xxxxxxxxxx* as shown below.

The screenshot shows the 'Phone1 Control/Status' window with the 'Normalization' tab selected. The 'Local Dial Plan' field contains the text '4xxxx/91xxxxxxxxxx'. Below this is a 'Substitution' section with a table that has three columns: 'Label', 'Number Pattern', and 'Translation'. The table is currently empty. At the bottom of the window, there are buttons for 'Add', 'Delete', 'Import', 'Clear All', and 'Test'.

Label	Number Pattern	Translation
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## 6.5. Verify Codec Settings

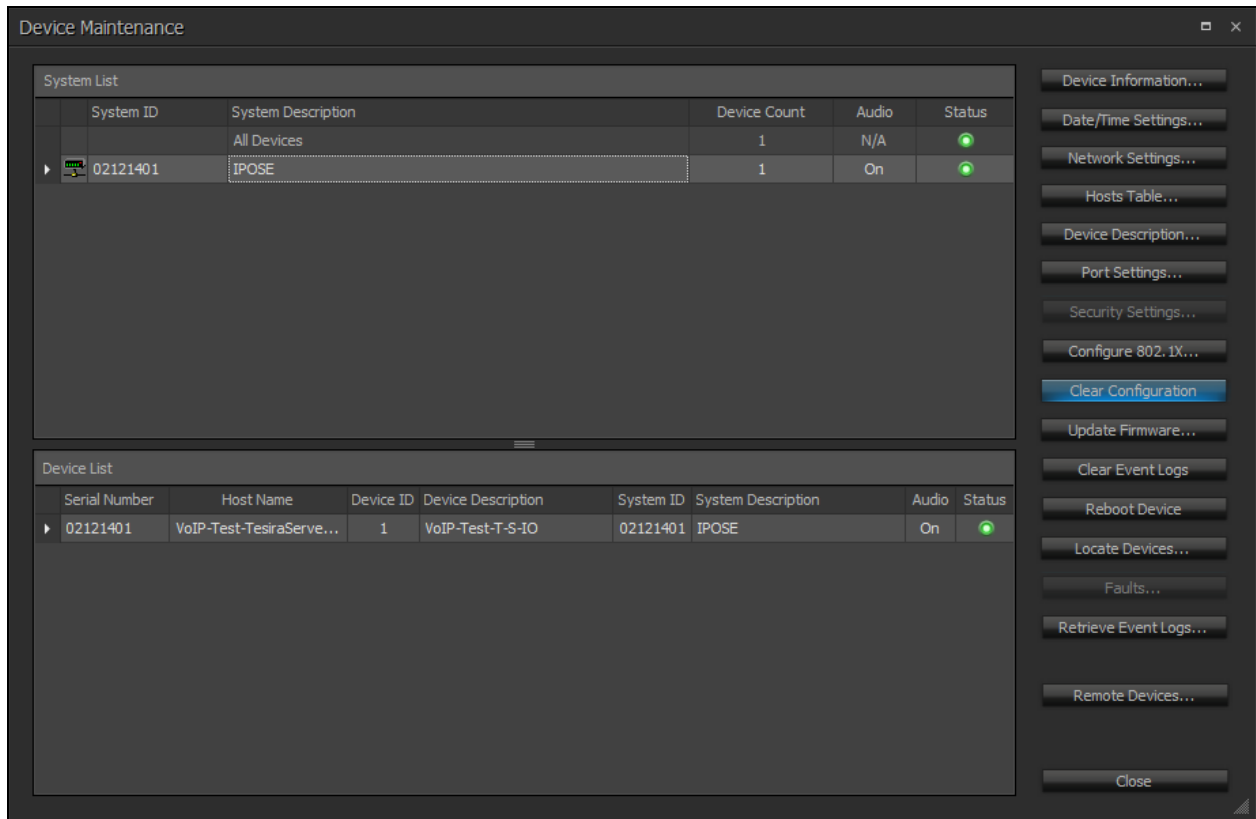
Navigate to the **General** tab shown below. In the **Voice Codec Priorities** section and select the desired codecs to be supported. Also, select whether blind or consultative transfers should be supported. In the configuration below, the **Consultative Transfer** field is not selected indicating that blind transfers are supported.

The screenshot shows the 'Phone1 Control/Status' configuration window. The 'General' tab is selected. The 'Dial Plan' section has 'Dialing Timeout (s)' set to 3. The 'Tones' section has 'Local DTMF' set to Mute, 'Local DTMF Level' and 'DTMF Transmit Level' set to -6.0, 'Ring Type' set to Classic, 'DTMF On Time (ms)' and 'DTMF Off Time (ms)' set to 50, 'Call Progress Tone Level' set to -20.0, 'Out-Of-Band DTMF' set to Enabled, 'Out-Of-Band DTMF Payload Type' set to 101, and 'DTMF via SIP Info' set to Off. The 'Call Features' section has 'Auto Answer' set to Enable, 'Auto Answer Ring Count' set to 2 Rings, 'Redial' set to Enabled, 'Consultative Transfer' set to Enable, 'Caller Id' set to Enabled, 'Do Not Disturb' set to Enable, 'Do Not Disturb Response' set to Do Not Disturb (480), 'RFC 2543 style Hold' set to Enable, 'Direct URL Dialing' set to Enable, 'Use One Audio Format' set to Enable, and 'Refresh Method' set to UPDATE. The 'Voice Features' section has 'VAD' set to Enabled and 'VAD Threshold' set to -40.0. The 'Voice Codec Priorities' section has 'Up' and 'Down' buttons. Below these buttons is a table with 4 columns: Use, Codec, Jitter Buffer Min, and Jitter Buffer Max. The table lists 5 codecs: G722, G711U, G711A, G729AB, and G7231, all with 'Use' checked, 'Jitter Buffer Min' set to 40, and 'Jitter Buffer Max' set to 200.

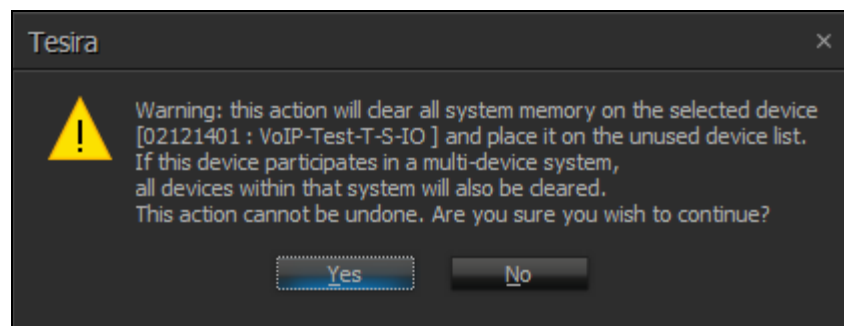
Use	Codec	Jitter Buffer Min	Jitter Buffer Max
<input checked="" type="checkbox"/>	G722	40	200
<input checked="" type="checkbox"/>	G711U	40	200
<input checked="" type="checkbox"/>	G711A	40	200
<input checked="" type="checkbox"/>	G729AB	40	200
<input checked="" type="checkbox"/>	G7231	40	200

## 6.6. Save and Send the New Configuration to the System

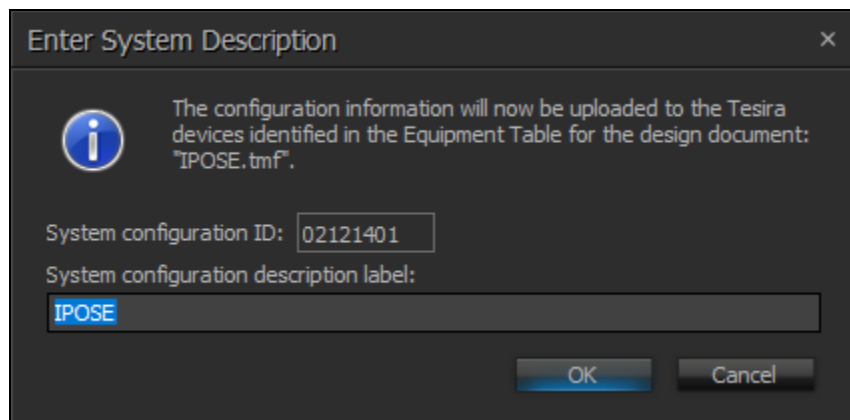
From the **Tesira Software** main window, save the configuration by clicking on **File → Save As** (not shown). Next, clear the configuration from the device before sending the new configuration to the device. Select the device and then click on the **Clear Configuration** button as shown below.



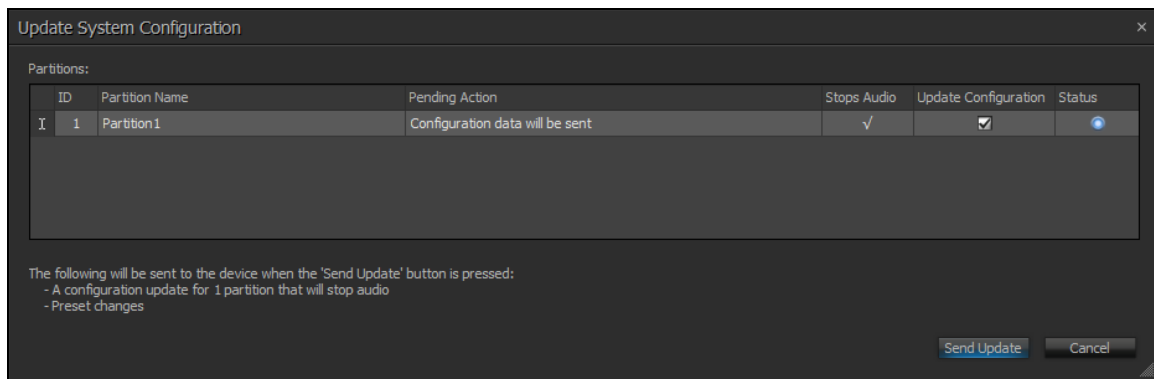
Confirm that the system memory will be cleared.



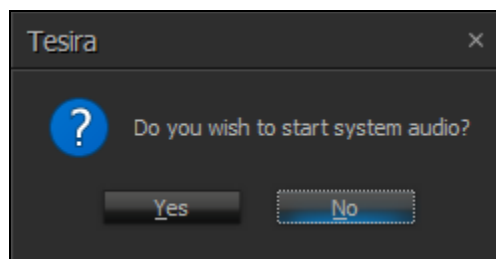
Lastly, click the Send Configuration icon in the main window to send the new configuration to the device (not shown). Confirm the **System configuration description label** by clicking **OK** in the window below.



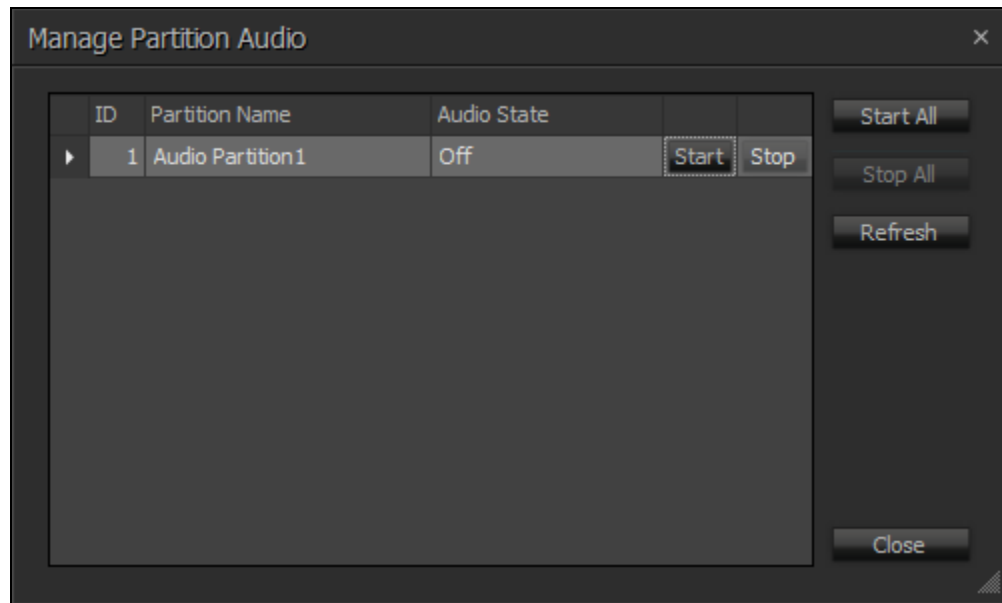
In the **Update System Configuration** window shown below, select the **Update Configuration** checkbox and click the **Send Update** button, if prompted.



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.



## 7. Verification Steps

This section provides the tests that may be performed to verify proper configuration of the Biamp Tesira SVC-2 with Avaya IP Office Server Edition.

1. Verify that Tesira SVC-2 has successfully registered with IP Office Server Edition. Launch **IP Office System Status** and navigate to **Extensions** → **<SIP Extension>**, where **<SIP Extension>** is the Tesira SVC-2 extension. Verify that the **Current State** is *Idle* as shown below.

The screenshot shows the Avaya IP Office System Status application window. The title bar reads "Avaya IP Office System Status - devcon-ipose (10.64.102.90) - IP Office Linux PC 11.0.0.2.0 build 23". The Avaya logo is in the top left, and the title "IP Office System Status" is in the top right. Below the title bar is a menu bar with "Help", "Snapshot", "LogOff", "Exit", and "About".

The left sidebar contains a tree view with the following items: "System", "Alarms (1)", "Extensions (4)", "Trunks (2)", "Active Calls", "Resources", "Voicemail", "IP Networking", and "Locations". The "Extensions (4)" item is expanded, showing a list of extensions: "41000", "41001", "41501", and "41510". The "41510" extension is selected.

The main content area is titled "Extension Status" and displays the following information for extension 41510:

- Extension Number: 41510
- IP address: 192.168.100.245
- Standard Location: None
- Registrar: Primary
- Telephone Type: Unknown SIP Device
- User-Agent SIP header: Tesira/1.10.0.16
- Media Stream: RTP
- Layer 4 Protocol: UDP
- Current User Extension Number: 41510
- Current User Name: Tesira41510
- Forwarding: Off
- Twinning: Off
- Do Not Disturb: Off
- Message Waiting: Off
- Phone Manager Type: None
- SIP Device Features: REFER,UPDATE
- License Reserved: No
- Last Date and Time License Allocated: 1/18/2019 12:04:47 PM
- Packet Loss Fraction: Connection Type:
- Jitter: Codec:
- Round Trip Delay: Remote Media Address:

Below this information is a table with the following columns: "Call Ref", "Current State", "Time in State", "Calling Number or Called Number", "Direction", and "Other Party on Call". The table contains one row with the following data:

Call Ref	Current State	Time in State	Calling Number or Called Number	Direction	Other Party on Call
	Idle	00:15:44			

At the bottom of the main content area are several buttons: "Trace", "Trace All", "Pause", "Ping", "Call Details", "Print...", and "Save As...".

The bottom status bar shows the time "12:20:31 PM", the status "Online", and a small icon.



2. Alternatively, the Tesira Software can be used to verify the registration status of Tesira SVC-2 with Avaya IP Office Server Edition. Double-click on **Phone1 Control/Status** in the Tesira Software and navigate to the Protocol tab. Note the **Registration Status** indicates *Registered*.

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

SIP	
Extension	41510
Display Name	41510
SIP Domain Name	
Authen User Name (Ext)	41510
Password	*****
NetBIOS Domain Name	
Proxy Vendor	Avaya IP Office
SIP Proxy Address	10.64.102.90
SIP Proxy Port	5060
Registration Status	Registered

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 3<sup>rd</sup> party can be joined into a conference.

## 8. Conclusion

These Application Notes have described the administration steps required to integrate the Biamp Tesira SVC-2 Card (installed in Biamp Tesira SERVER-IO) with Avaya IP Office Server Edition. Biamp Tesira SVC-2 successfully registered with Avaya IP Office Server Edition and basic and telephony features were verified. All test cases passed with observations noted in **Section 2.2**.

## 9. Additional References

This section references the documentation relevant to these Application Notes. The following Avaya product documentation is available at [support.avaya.com](http://support.avaya.com) and the Tesira product documentation is available at [www.biamp.com/downloads](http://www.biamp.com/downloads). Additional Tesira documentation is available via online help in the Tesira Software.

- [1] *Administering Avaya IP Office Platform with Manager*, Release 11.0, Issue 17a, August 2018.
- [2] *Tesira® Server IO Operation Manual*, 585.027890C, March 2016.

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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](mailto:devconnect@avaya.com).



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"> <li>• Supports up to 8 DSP-2 cards</li> <li>• Up to 420 x 420 channels of digital I/O over AVB</li> <li>• Supports optional 32 x 32 CobraNet audio networking</li> <li>• Supports optional 64 x 64 Dante audio networking</li> <li>• System configuration and control via Ethernet or serial connection</li> <li>• Front panel OLED display for device and system information</li> <li>• SpeechSense™ and AmbientSense™ processing algorithms</li> <li>• Signal processing via intuitive software allows configuration and control for: signal routing and mixing, equalization, filtering, dynamics, delay and much more</li> <li>• Extensive input, output and logic expansion devices supported as part of the Tesira digital audio networking platform</li> </ul>
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"> <li>• Supports up to 3 DSP-2 cards</li> <li>• Supports up to 12 I/O cards with a maximum of 48 channels of analog audio</li> <li>• Up to 420 x 420 channels of digital I/O over AVB</li> <li>• Supports optional 32 x 32 CobraNet audio networking</li> <li>• Supports optional 64 x 64 Dante audio networking</li> <li>• System configuration and control via Ethernet or serial connection</li> <li>• Front panel OLED display for device and system information</li> <li>• SpeechSense™ and AmbientSense™ processing algorithms</li> <li>• Signal processing via intuitive software allows configuration and control for: signal routing and mixing, equalization, filtering, dynamics, delay and much more</li> <li>• Extensive input, output and logic expansion devices supported as part of the Tesira digital audio networking platform</li> </ul>
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"> <li>• 12 mic/line level inputs with AEC, 8 mic/line level outputs</li> <li>• Gigabit Ethernet port</li> <li>• RS-232 serial port</li> <li>• 4-pin GPIO</li> <li>• 2-line OLED display with capacitive-touch navigation</li> <li>• System configuration and control via Ethernet</li> <li>• Internal universal power supply</li> <li>• SIP VoIP interface via RJ-45 connector</li> <li>• Standard FXO telephone interface via RJ-11 connector</li> <li>• Signal processing via intuitive software allows configuration and control for signal routing, mixing, equalization, filtering, and delay</li> </ul>
TesiraFORTÉ AVB VT	<ul style="list-style-type: none"> <li>• 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</li> </ul>

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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"> <li>• 128 x 128 channels of AVB</li> <li>• 12 mic/line level inputs with AEC, 8 mic/line level outputs</li> <li>• Gigabit Ethernet port</li> <li>• RS-232 serial port</li> <li>• 4-pin GPIO</li> <li>• 2-line OLED display with capacitive-touch navigation</li> <li>• System configuration and control via Ethernet</li> <li>• Internal universal power supply</li> <li>• SIP VoIP interface via RJ-45 connector</li> <li>• Standard FXO telephone interface via RJ-11 connector</li> <li>• Signal processing via intuitive software allows configuration and control for signal routing, mixing, equalization, filtering, and delay</li> </ul>
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"> <li>• 32 x 32 channels of Dante</li> <li>• 12 mic/line level inputs with AEC, 8 mic/line level outputs</li> <li>• Gigabit Ethernet port</li> <li>• RS-232 serial port</li> <li>• 4-pin GPIO</li> <li>• 2-line OLED display with capacitive-touch navigation</li> <li>• System configuration and control via Ethernet</li> <li>• Internal universal power supply</li> <li>• SIP VoIP interface via RJ-45 connector</li> <li>• Standard FXO telephone interface via RJ-11 connector</li> <li>• Signal processing via intuitive software allows configuration and control for signal routing, mixing, equalization, filtering, and delay</li> </ul>
TesiraFORTÉ AVB VT4	<ul style="list-style-type: none"> <li>• 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.</li> <li>• 128 x 128 channels of AVB</li> <li>• 4 mic/line level inputs with AEC, 4 mic/line level outputs</li> </ul>

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	<ul style="list-style-type: none"> <li>• Gigabit Ethernet port</li> <li>• RS-232 serial port</li> <li>• 4-pin GPIO</li> <li>• 2-line OLED display with capacitive-touch navigation</li> <li>• System configuration and control via Ethernet</li> <li>• Internal universal power supply</li> <li>• SIP VoIP interface via RJ-45 connector</li> <li>• Standard FXO telephone interface via RJ-11 connector</li> <li>• Signal processing via intuitive software allows configuration and control for signal routing, mixing, equalization, filtering, and delay</li> </ul>
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"> <li>• 32 x 32 channels of Dante</li> <li>• 4 mic/line level inputs with AEC, 4 mic/line level outputs</li> <li>• Gigabit Ethernet port</li> <li>• RS-232 serial port</li> <li>• 4-pin GPIO</li> <li>• 2-line OLED display with capacitive-touch navigation</li> <li>• System configuration and control via Ethernet</li> <li>• Internal universal power supply</li> <li>• SIP VoIP interface via RJ-45 connector</li> <li>• Standard FXO telephone interface via RJ-11 connector</li> <li>• Signal processing via intuitive software allows configuration and control for signal routing, mixing, equalization, filtering, and delay</li> </ul>
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"> <li>• 128 x 128 channels of AVB (AVB model only)</li> <li>• 12 mic/line level inputs with AEC, 8 mic/line level outputs</li> <li>• Gigabit Ethernet port • Up to 8 channels of configurable USB audio</li> <li>• RS-232 serial port</li> <li>• 4-pin GPIO</li> <li>• 2-line OLED display with capacitive-touch navigation</li> <li>• System configuration and control via Ethernet</li> <li>• Internal universal power supply</li> <li>• SIP VoIP interface via a RJ-45 connector</li> <li>• Fully compatible with Tesira servers, endpoints, expanders, and controllers (AVB model)</li> <li>• Signal processing via intuitive software allows configuration and control for signal routing, mixing, equalization, filtering, delay and much more</li> </ul>
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"> <li>• 32x32 channels of digital audio networking via the Dante protocol</li> <li>• 12 mic/line level inputs with AEC, 8 mic/line level outputs</li> <li>• 2 Gigabit Ethernet ports: Dante digital audio and Tesira control</li> <li>• Up to 8 channels of configurable USB audio</li> <li>• RS-232 serial port</li> <li>• 4-pin GPIO • 2-line OLED display with capacitive-touch navigation</li> <li>• System configuration and control via Ethernet</li> <li>• Internal universal power supply</li> <li>• SIP VoIP interface via a RJ-45 connector</li> </ul>

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	<ul style="list-style-type: none"><li>• Signal processing via intuitive software allows configuration and control for signal routing, mixing, equalization, filtering, delay, and much more</li></ul>
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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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