LINEAR ACOUSTIC®
SDI XNODE

Livewire+/AES67 Audio over IP Interface with Dual HD/SD-SDI Ports

USER'S MANUAL

Manual for software version 1.6.4e
January 2019

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User Warnings and Cautions

The installation and service instructions in this manual are for use by qualified personnel only. To avoid electric shock, do not perform any servicing other than that contained in the operating instructions unless you are qualified to do so. Refer all servicing to qualified personnel

This instrument has an autoranging line voltage input. Ensure the power voltage is within the specified range of 100-240VAC. The ~ symbol, if used, indicates an alternating current supply.

This symbol, wherever it appears, alerts you to the presence of uninsulated, dangerous voltage inside the enclosure – voltage which may be sufficient to constitute a risk of shock.

This symbol, wherever it appears, alerts you to important operating and maintenance instructions. Read the manual.

CAUTION: HAZARDOUS VOLTAGES

The instrument power supply incorporates an internal fuse. Hazardous voltages may still be present on some of the primary parts even when the fuse has blown. If fuse replacement is required, replace fuse only with same type and value for continued protection against fire.

WARNING:

The product's power cord is the primary disconnect device. The socket outlet should be located near the device and easily accessible. The unit should not be located such that access to the power cord is impaired. If the unit is incorporated into an equipment rack, an easily accessible safety disconnect device should be included in the rack design.

To reduce the risk of electrical shock, do not expose this product to rain or moisture. This unit is for indoor use only.

This equipment requires the free flow of air for adequate cooling. Do not block the ventilation openings on the rear and sides of the unit. Failure to allow proper ventilation could damage the unit or create a fire hazard. Do not place the units on a carpet, bedding, or other materials that could interfere with any panel ventilation openings.

If the equipment is used in a manner not specified by the manufacturer, the protection provided by the equipment may be impaired.

USA CLASS A COMPUTING DEVICE INFORMATION TO USER.

WARNING:

This equipment generates, uses, and can radiate radio-frequency energy. If it is not installed and used as directed by this manual, it may cause interference to radio communication. This equipment complies with the limits for a Class A computing device, as specified by FCC rules, part 15, subpart j, which are designed to provide reasonable protection against such interference when this type of equipment is operated in a commercial environment. Operation of this equipment in a residential area is likely to cause interference. If it does, the user will be required to eliminate the interference at the user's expense. Note: objectionable interference to TV or radio reception can occur if other devices are connected to this device without the use of shielded interconnect cables. FCC rules require the use of shielded cables.

CANADA WARNING:

"This digital apparatus does not exceed the Class A limits for radio noise emissions set out in the radio interference regulations of the Canadian department of communications."
"Le présent appareil numérique n'émet pas de bruits radioélectriques dépassant les limites applicables aux appareils numériques (de Class A) prescrites dans le règlement sur le brouillage radioélectrique édicté par le ministère des communications du Canada."

CE CONFORMANCE INFORMATION:

This device complies with the requirements of the EEC council directives:
- ♦ 93/68/EEC (CE MARKING)
- ♦ 73/23/EEC (SAFETY – LOW VOLTAGE DIRECTIVE)
- ♦ 89/336/EEC (ELECTROMAGNETIC COMPATIBILITY)

Conformity is declared to those standards: EN50081-1, EN50082-1.

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Patent information can be found at www.TelosAlliance.com/legal

Updates

Linear Acoustic SDI xNode features and operations are determined largely by software. The Telos Alliance strives to provide the most stable and feature-rich software available. We encourage you to check for software updates from time to time by visiting our website or by contacting us directly.

Feedback

We welcome feedback on any aspect of our products or this manual. In the past, many good ideas from users have made their way into software revisions or new products. Please contact us with your comments or suggestions.

We support you…

By Phone/Fax

You may reach our Telos Support Team in emergencies by calling +1 216-622-0247. For billing questions or other non-emergency technical questions, call +1 216-241-7225 between 9:00 AM to 5:00 PM USA Eastern Time, Monday through Friday.

By Email.

Non-emergency technical support is available at Support@TelosAlliance.com.

By Web

The Linear Acoustic Web site has a variety of information that may be useful for product selection and support. The URL is https://www.telosalliance.com/Linear .

SERVICE

You must contact Telos Alliance before returning any equipment for factory service. We will need your unit's serial number, located on the back of the unit. We will issue a return authorization number, which must be written on the exterior of your shipping container. Please do not include cables or accessories unless specifically requested by the Technical Support Engineer. Be sure to adequately insure your shipment for its replacement value. Packages without proper authorization may be refused. US customers, please contact Telos Alliance Technical Support at +1-216-622-0247. All other customers should contact local representative to make arrangements for service.

Warranty

For the latest Telos Alliance warranty, visit: telosalliance.com/warranty

Register your product

Register your product today to get the full benefits of our warranty, support, and product updates. telosalliance.com/product-registration/

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Creating the Most Exciting and Engaging Audio Experiences Imaginable

Congratulations on your new Telos Alliance product!

The gang here at Telos is committed to shaping the future of audio by delivering innovative, intuitive solutions that inspire our customers to create the most exciting and engaging audio experiences imaginable.

We're grateful that you have chosen audio tools from Telos® Systems, Omnia® Audio, Axia® Audio, Linear Acoustic®, 25-Seven Systems®, and Minnetonka Audio®. We're here to help you make your work truly shine. We hope that you enjoy your Telos Alliance product for many years to come and won't hesitate to let us know if we can help in any way.

<div align="right">The Telos Alliance</div>

1   Introduction

Introduction

Congratulations on your purchase of the Linear Acoustic SDI xNode. The SDI xNode is a compact and powerful HD/SD-SDI to Livewire+/AES-67 Audio over IP interface. It is capable of managing dual, independent HD/SD-SDI paths and can de-embed, shuffle, and re-embed up to 16 channels of audio in an SDI path.

The SDI xNode features:

♦ Two HD/SD-SDI inputs and outputs with compensating video delay.

♦ de-embed up to 16-channels of audio from SDI and output them across a Livewire+/AES67 (AoIP) network.

♦ Receive up to 16-channels of audio from a Livewire+ network and embed them into one or two SDI streams.

♦ Autoranging power supply plus input for a back-up power supply.

♦ Network configuration, control, and monitoring

Includes all standard xNode controls including front panel and web interface.

The SDI xNode provides up to 16-channels of audio to the Livewire+ network (called a Livewire+ "Source") and receives up to 16-channels of audio from the Livewire+ network (Livewire+ "Destination") .

xNode Family

The xNodes are a family of compact 1U high, one-half space wide Livewire+ audio and logic interfaces with advanced features. Many of the xNode models are offered by Axia Audio, one of our sister companies in the Telos Alliance.

Livewire+ is a patented AoIP protocol, developed by Axia Audio, which enables high-reliability, low-delay uncompressed digital audio over a standard Ethernet network.

xNodes are fully AES67 compliant, and include support for Ravenna stream routing and IEEE 1588 synchronization.

All xNodes, including the SDI xNode, include dual 100BT Ethernet ports which are user configurable as either mirroring redundant ports or as a single Livewire+ port and separate management port.

A front panel OLED display for setup and status is included, along with a http server for full configuration, control, and monitoring from a web browser.

Other members of the xNode family, which are offered by Axia Audio, include:

Analog xNode:

4 Stereo Analog Line level Inputs, 4 Stereo Analog Line-level Outputs

AES/EBU xNode:

4 Stereo Digital AES-3 Inputs, 4 Stereo Digital AES-3 Outputs

Microphone xNode:

4 Mic Inputs, 4 Stereo Analog Line-level Outputs

GPIO Logic xNode:

6 GPIO Ports, each with 5 optically isolated inputs and 5 optically isolated outputs

Mixed Signal xNode:

1 Mic/Line Analog Input

2 Analog Line Inputs

1 AES-3 Input

3 Analog Line Outputs

1 AES-3 Output

2 GPIO Ports, each with 5 inputs and 5 outputs.

xSelector:

Includes Livewire+ I/O, Analog, and AES-3 I/O along with a Headphone output. Six "radio buttons" allow instant monitoring of favorite sources.

xSwitch:

The world's only zero-configuration Ethernet switch optimized for Livewire+. Features eight (8) 100BT Ethernet ports, along with two (2) 1000BT ports for trunking. 1000BT ports include both RJ-45 and SFP fiber connections.

About Livewire+

This manual covers the details of the SDI xNode. To learn more about Livewire+ and AoIP, you may wish to read *Introduction to Livewire+: System Design Reference and Primer* which is available on the Telos Alliance website (www.telosalliance.com) and *Audio Over IP: Building Pro AoIP Systems with Livewire+* by Steve Church and Skip Pizzi, available from Elsevier Press.

In these publications we explain the ideas that motivated Livewire+ and how you can use and benefit from it, as well as nitty-gritty details about wiring, connectors, and the like. Since Livewire+ is built on standard networks, we also help you to understand general network engineering so that you have the full background for Livewire+'s fundamentals.

Warranty and Feedback

For the latest Telos Alliance warranty, visit: telosalliance.com/warranty.

2 Overview and Connections

Unpacking and Inspection

Before unpacking the unit, inspect the outer carton for shipping damage. If the carton shows damage, inspect the unit in those areas. Please save the carefully designed shipping carton and packing materials. In the unlikely event that the unit needs to be returned to the factory, alternate cartons or packing materials may not be adequate and can cause damage not covered by warranty.

The following essential items are provided with the unit:

♦ IEC power cord (style matches country of order)

♦ Quick-start sheet to get you up and running

♦ USB memory stick containing documentation, including this manual

♦ rack mount hardware kit

♦ a handy black pen.

Unit Overview

This section introduces you to SDI xNode and its views:

Block diagram

The SDI xNode is capable of:

♦ handling two independent SDI streams

♦ de-embedding up to 16-channels of audio from SDI input stream(s) and outputting them onto a Livewire+/AES67 AoIP network

♦ receiving up to 16-channels of audio from a Livewire+ network and embedding them into the SDI output stream(s)

♦ pass-through of input SDI audio and pair shuffling to the SDI output

♦ compensating video delay of up to 1 second for each SDI stream

The block diagram of the SDI xNode is shown below:
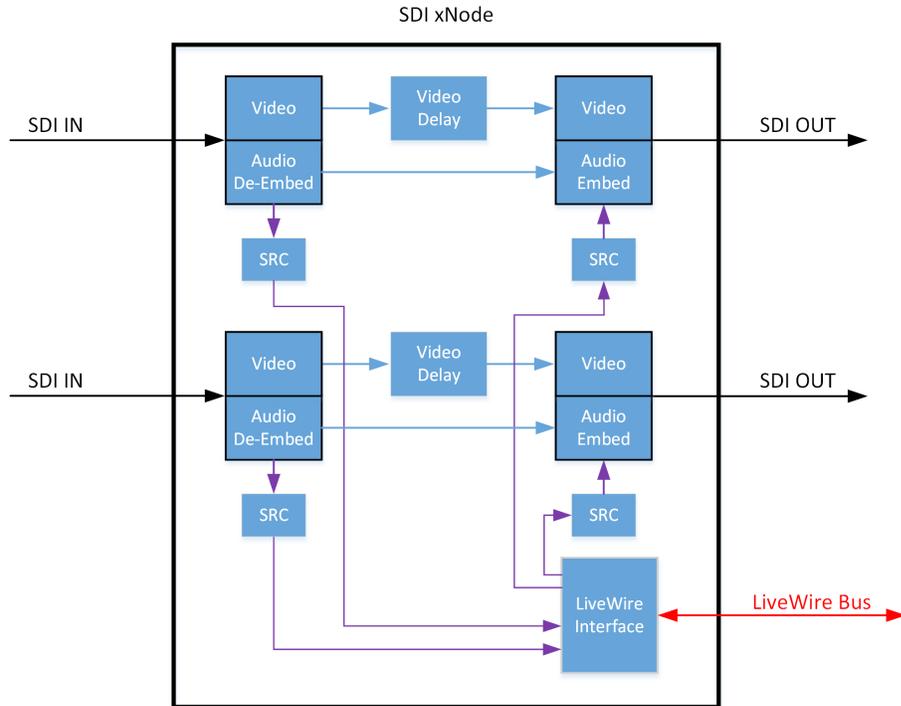
SDI xNode

Figure 1 - SDI xNode Block Diagram

Front panel

On the front panel of the SDI xNode, you will find:

- ◆ OLED display
- ◆ Two pushbutton switches for:
    - ◊ Display page selection（Screen button）
    - ◊ Edit Mode selection（Pencil button）
- ◆ Power LED

The OLED display provides the user with visual feedback and interaction as to what the SDI xNode is doing, parameter settings, and input signal status.

Figure 2 - SDI xNode front panel

A closer view of the front panel and controls is provided below:
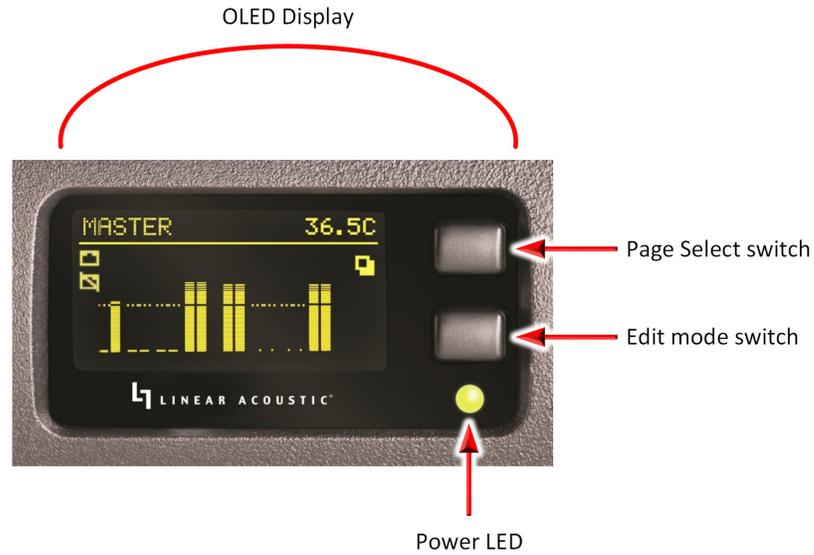
Figure 3 - SDI xNode display and controls

Rear panel

The rear panel of the SDI xNode is where all connections are made.

The rear panel includes:

♦ Power inlets

♦ Two 100 BT Ethernet ports

♦ Two 75-ohm BNC connectors for SDI inputs

♦ Two 75-ohm BNC connectors for SDI outputs

Connections

This section covers all connections

Rear Panel

The rear panel of the SDI xNode contains the connections which will be used.

Figure 4 - SDI xNode Rear Panel

AC Power Connection

An IEC universal power inlet is provided. This accepts an AC input from 95v up through 240v, at either 50 or 60 Hz.

Backup Power Connection

A 12VDC power connector is provided for a redundant, or back-up, power supply.

Network Connections

Two 100BT Ethernet ports are provided. They are configurable as either mirroring redundant ports or as a single Livewire+ port and separate management port.

To be precise, the mode of the Ethernet ports when configured as redundant ports is called "active-passive Ethernet port bonding". In active-passive mode, only one port is used, and the other one is a fail-over backup.

SDI Input

Two autosensing SD/HD-SDI input connectors are included on the SDI xNode.

All 16-channels of an SDI stream are available for de-embedding, embedding, and pair shuffling.

SDI Output

Two SD/HD-SDI output connections are included on the rear of the SDI xNode. All 16 audio channels in the SDI stream are available for embedding and pair shuffling.

Installation and Mounting

Installation

The xNode is a fanless device, so it may be located in a control room or other location where fan noise cannot be tolerated. Because this unit is passively cooled, it is recommended to leave an empty space above the unit when mounted in a rack or other enclosure. An internal thermal sensor is designed in to the product, so that the operating temperature of the xNode may be monitored remotely

Mounting

The SDI xNode includes a rack mounting kit. The mounting hole location and pattern is consistent with all xNodes, so any xNode may be mounted using the available kit, and two of them may be mounted side by side.

Rack Mount kit

The rack mount kit provides the ability to secure the xNode into an equipment rack. The kit contains two short rack ears, one long rack ear, and a spacer plate.
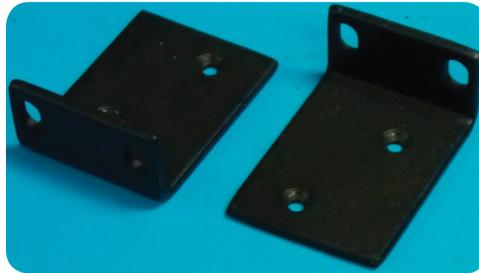

Figure 5 - Short rack mount ears

Figure 6 - Long rack mount ear

Single xNode

To rack mount a single xNode, secure the short rack ear to one side of the xNode and secure the long ear to the opposing side of the xNode.

Figure 7 - xNode with long and short rack ear installed

Two xNodes

It is possible to install two xNodes in a side-by-side configuration. The mounting hole locations and pattern are consistent with all xNodes, so any xNode may be mounted next to the SDI xNode.

To mount two units side by side:

Remove the top lid from both xNodes. Place two xNodes side by side. Place the spacer plate between the two xNodes
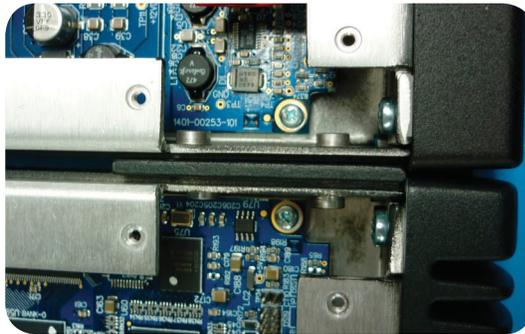
Figure 8 - Spacer plate in between two xNodes, top view

Use the four (4) screws provided to secure the two nodes together.
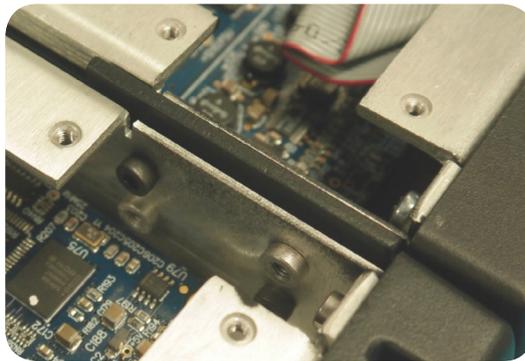
Figure 9 - Spacer plate in between two xNodes, angled view

Replace the lids on the xNodes. Secure the short rack ears to either side of the two xNodes

Front Panel Displays and Indicators

The front panel OLED provides several different status and configuration screens, along with the Power and Status LED. The description and definition of those displays are below.

Power and Status LED

The Power LED located on the front panel is a multicolor LED and changes its color to indicate the status of the power source (s) applied to the SDI xNode.

Color	Description
Green	Powered by the primary AC Mains power supply (IEC inlet)
Yellow	Powered by both AC Mains and redundant power input
Red	Powered by external 12 VDC redundant power

Home Screen

The initial screen displayed after boot up is this home screen. It identifies the unit as the SDI xNode, the software version in use, and the internal temperature of the unit. After approximately 20 seconds, the display will change over to the Audio Meter display for Source/Destination 1 – 4.
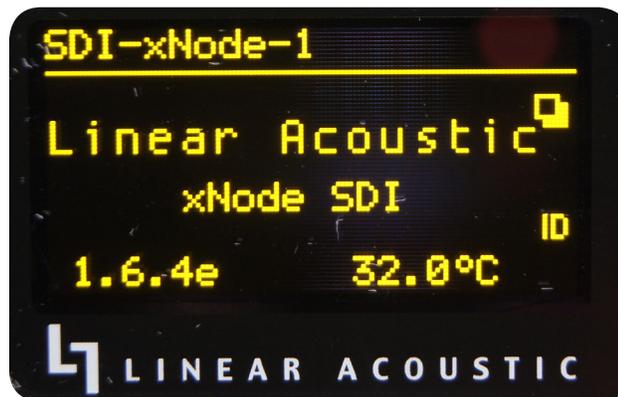
Figure 10 - Home Screen

Audio Meters Source/Destination 1-4

Audio meters for Sources 1-4 and Destinations 1-4 are provided.   Sources 1-4 are shown on the left, and Destinations 1-4 are shown on the right.
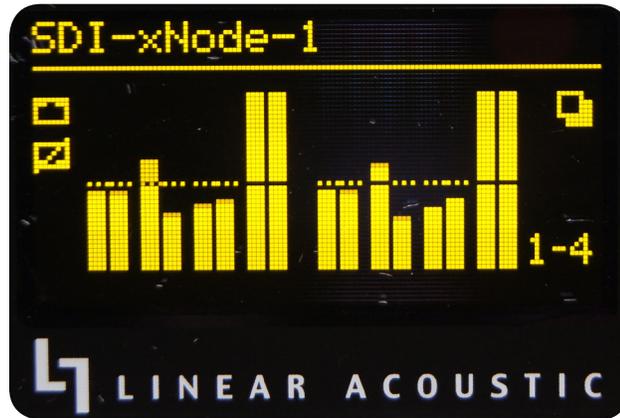

Figure 11 – Meters for I/O pairs 1 - 4

Audio Meters Source/Destination 5-8

Audio meters for Sources 5-8 and Destinations 5-8 are provided.   Sources are shown on the left, and Destinations are shown on the right
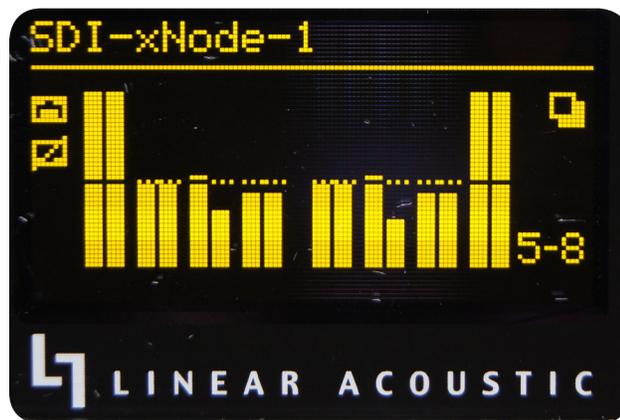

Figure 12 - Meters for I/O pairs 5 - 8

SDI Input Format

This screen provides status information regarding the input status and video format and frame rate of the input signal.
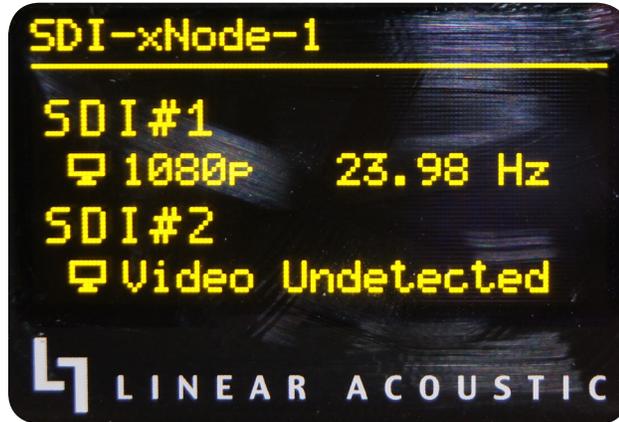
Figure 13 - SDI input status screen

xNode ID Screen

The Node number, source channel numbers, and IP address are displayed on this screen.
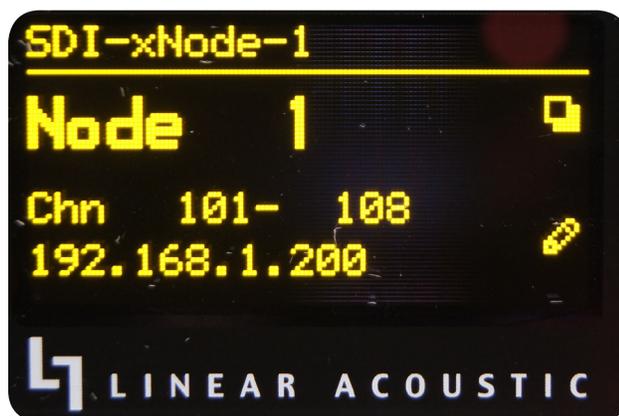
Figure 14 - Node ID screen

IP Address and Subnet Mask

The IP address, Subnet Mask, and status of active network ports are displayed on this screen.
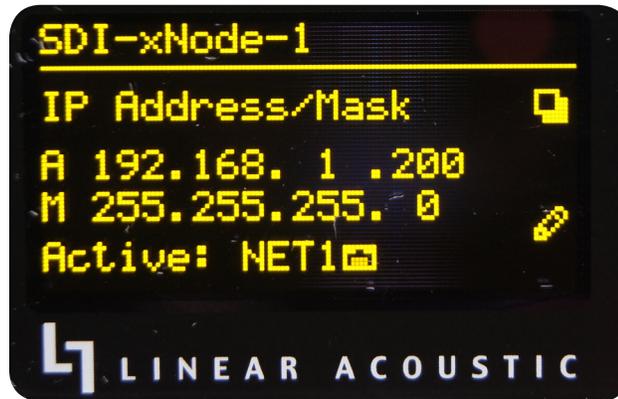
Figure 15 - IP configuration screen

3 Initial Configuration

Unit Configuration

This chapter discusses how to initialize and configure the SDI xNode to sit on the Livewire+ network. Configuring the SDI xNode to handle and route SDI and audio signals will be discussed in a later chapter.

There are different options for how the xNode can be configured:

♦ assigning an ID to the xNode

♦ manually configuring the IP address.

♦ iProbe software application

In the fast setup, we will use the ID method. Following subsections of this chapter will cover the other methods in detail.

Fast Setup

Apply power to the xNode and wait for the boot process to complete showing you a home screen identifying the xNode and version.

Press the top button, the screen button, four (4) times to reach the ID page which shows you a Node ID with no value, Channel numbers 1-8 and no IP address (shown in)

Press the second button, pencil icon, for 10 seconds to enter into edit mode.

Figure 16 - Front Panel Display

With the cursor shown, you can use the top button (I+) to increment value and the bottom button (>) to move the cursor to the next position. Press "next" button twice and press the increment button once to give the ID value of 1 to the xNode. (You'll give a different ID number to each xNode you install.)

Pressing the move button one additional time completes edit mode and automatically configures the xNode with IP address 10.216.0.101 and channel numbers of 101 – 108.

That's actually the end of fast setup! With IP and channel numbers configured, you can attach the xNode to your network and begin connecting sources. But let's continue on for some quick customization.

Connect a cross over cable between a computer and the xNode or connect the xNode to the appropriately configured switch.

Set a static IP address to your computer that has a link to the xNode directly or through a network switch. (an IP address of 10.216.0.100 would be suitable)

Open a browser and type in the IP address of the xNode (10.216.0.101) into the URL field.

Default Authentication is:

Username: user

Password: (there is no password)

Select the Simple Setup button to enter into a simple configuration screen for the xNode.

Enter descriptive text in the Source Name fields which describe what devices are connected to the xNode (e.g. server, control room, on-air feed, Studio A)

Enter descriptive text into the Name field under the Destination section which documents what is connected to the xNode's outputs (e.g. Control Room Monitors, TX, BOC, uplink)
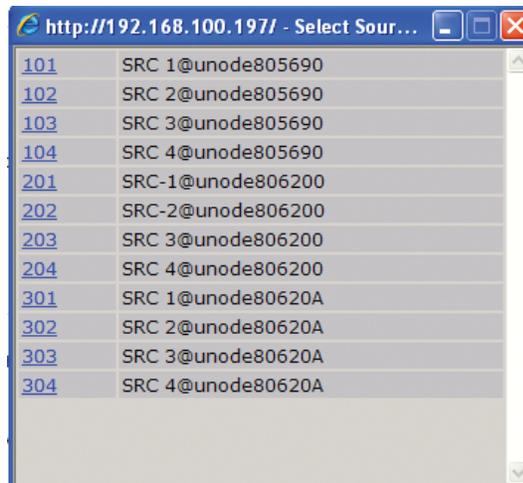
Figure 17 - Source Selection

The next step would be to define the audio that is present at the output. This is done by choosing the browse button next the channel text box. A popup will appear with a list of sources that are present on the network. A sample of three xNodes freshly installed into a network is shown in Figure 17. At this point, not many sources have been configured with valuable descriptive names. Hopefully you see the value of the good practice in using the naming fields. Once you have configured all your xNode sources and other Livewire+ devices, this browse popup will have more descriptive information which will help you locate the correct source to assign to an output (Destination).

If you notice in Figure 17, the three xNodes are showing a default host name which provides little assistance to the end user. To resolve this you can select the System link under the Advanced Options to change the host name to something more descriptive (e.g. Studio1, BOC, SATRCK-1). With a descriptive host name applied to the xNode, a listing of available audio streams will be displayed in a more descriptive and intuitive manner.

You're done! These steps were designed to get you up and running quickly, but if you need more details on other configuration options, please continue to read this chapter.

Assigning an IP address manually

If the IP value assigned in the previous steps do not fit within your Livewire+ network plan or you wish to configure by manual process, you can assign an IP address from the front panel by pressing the page button (top button) until the IP page is shown.

Press and hold the edit button (bottom button) for 10 seconds.

Figure 18 - IP Address Configuration

A cursor will appear in the Address field. Use the top button to increment and the bottom button to move the cursor to the next position. After 30 seconds of no activity the Edit view will cancel back to the status view. Edit the IP address to your needs. Press the next button (bottom) to move the cursor to the next line which is the Netmask setting. Use the top button to increment the mask to your needs. Press the next button to move to the next line. The NET is highlighted. Pressing the top button will toggle between the two ports allowing you to select which port is active. Press the next button (bottom) one more time to complete the edit process. If the cursor returns to the address field, then an incorrect value has been entered.

> **Note!**
>
> **You can change the node ID but maintain the same IP address by pressing the increment button when the IP address is highlighted in the Node ID page. This will toggle between a custom address or the new address created though the auto configuration process**

Once the xNode has the desired IP value, you can log into the HTML interface with a PC that has a static IP value within the same subnet as the xNode and connected to the same network. Later in this chapter the Simple Setup HTML interface is covered.

Assigning an ID

The method of setting an xNode with an ID was covered in the Fast Setup. Some details of this option were left out. To review the ID method: After the xNode has booted, the title page should show on the OLED display, showing the name of the node and the software version running. The top button acts as the page toggle function. Press the top button until the Node ID page is shown. Press and hold the lower button (represented by the pencil icon) for 10 seconds A cursor appears following the word Node. Use the top button to increment the cursor value and use the lower button to move to the next position. After 30 seconds of no activity, the edit page will reset back to the status page. Select a unique ID for the xNode. Press the lower button one last time to finalize the edit and return to the status view. The node has now been given an IP address and the inputs have been given unique channel numbers.

More on ID Values

Number ranges of 1-99 (nn) support small-to-medium networks with up to 99 nodes. The node ID determines the IP address (10.216.0.1nn) and the source channel numbers (nn01-nn08).

Number ranges of 100-319 (abc) support large networks with up to 2299 nodes. When the 3 digit ID is being used, a second level identifier becomes available with range of d=<0-9>. The node ID determines IP address in range: 10.216.ab.1cd. Channel numbers: abcd1-abcd8.

Once the xNode has been assigned an ID value, the device has an IP address that can be used to access its HTML configuration interface. Each input has been assigned a unique Livewire+ channel number. The next configuration steps are to enter in descriptive names to the inputs and outputs and to assign Livewire+ sources to the destinations. The Simple Setup page is covered later in the chapter.

iProbe configuration

Axia iProbe is a stand-alone Windows software application that provides management, support, and control of a Livewire+ network and devices (SDI xNodes, other xNodes, Axia Control Surfaces, and other supported Livewire+ devices) in a friendly, unified environment.
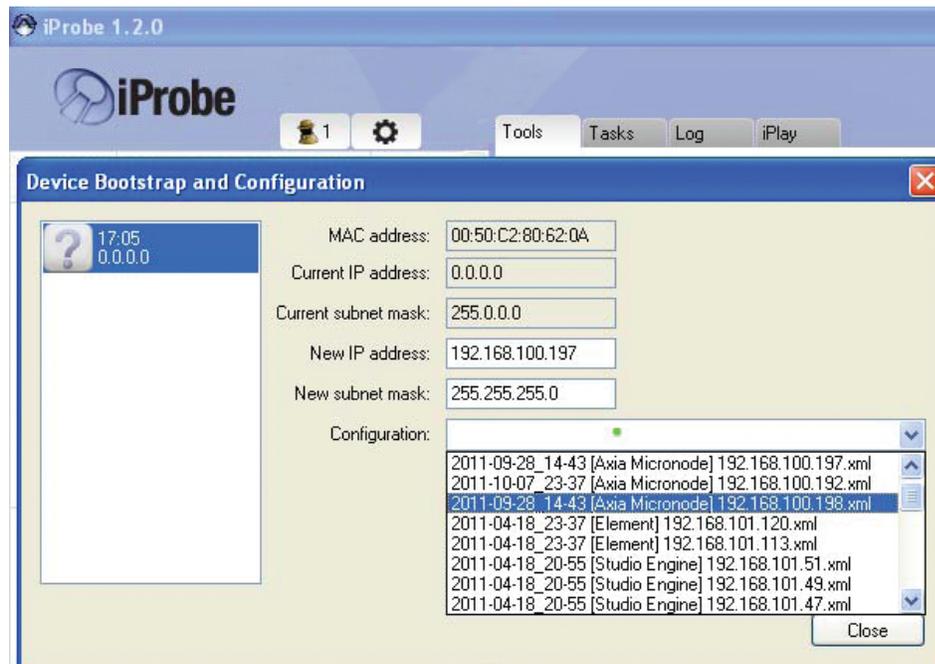
Figure 19 - iProbe Screen

When an xNode boots and it has no configuration, it will broadcast a bootp request within 4 seconds. The broadcasts will continue with an increasing delay between messages and randomization until a configuration is set. iProbe includes a bootp function which will observe the request and provide an option to respond or ignore.

A counter at the bootstrap configuration button will appear when a bootp request is received by iProbe. Clicking on the button will open the Device Bootstrap and Configuration window.

From the window view, select the device that has issued a bootp request. The configuration options will appear allowing you to push a New IP address, New subnet mask, and a Configuration. If you have no backup configuration to send, then apply an IP address and proceed to log into the device through the iProbe interface, and configure it per the "web interface configuration".

Web interface configuration ("Simple setup")

Initially accessing the user interface of the xNode will require authentication. The default authentication is:

Username: user

Password: (none)

The initial page is the Home page. Select the "Simple setup" button.

In the Sources section you will provide a name value to help identify the audio source in network advertisements. If a unique ID was assigned to the node, the sources were automatically given channel numbers. Sources are the xNode's audio inputs. These inputs are known within the network as sources and go to Destinations.

Destinations of the xNode are the audio outputs. The Name field is used for documenting what is connected to the audio output port. Channel field is where you select an audio stream from the network to provide audio for the output. If you don't know the channel number, use the browse button immediately to the right to search the network for the desired source. Selecting the source from the pop-up window will populate the channel value of the Destination field.

Factory Reset / Restoring Defaults

Performing factory reset will clear all configurations and return the xNode to a "blank canvas". To restore default, execute the following steps.

Disconnect all power from the xNode, then press and hold the two front panel buttons.

Apply power to the xNode while continuing to press the buttons.

xNode will show a countdown timer for factory reset. When the timer reaches 0, release the two buttons.

4 Web Interface

Interface

The SDI xNode includes an http server which enables full configuration, control, and monitoring of the unit from a web browser.

Access to the unit is controlled via user name and password in order to prevent unauthorized or unintended changes during normal operation

Note!

The default login details for the SDI xNode are:

Username = user (all lower case letters)

Password = (no password)

System Option Pages

Home Page

After logging in to the SDI xNode, the Home Page will appear.   The home page shows system information and acts as the default page for xNodes.

Figure 20   Home Page

The home page displays some high level information about the unit, such as:

Node ID

An xNode can be assigned an ID which is used to assist with IP addressing and channel numbering. If an xNode has an ID assigned, this field will be shown.

Software Version

The code version which is used by the xNode.

Temperature

The temperature reading from the internal sensor. Alarm status will appear if the temperature is above acceptable limits.

Sync

A solid Sync indication represents synchronization with the network.

Net1 and Net2

Shows the status of the xNode's Ethernet ports and indicates which port(s) is active, along with which port(s) is configured to carry Livewire+.

System Uptime

The amount of time since the last boot up.

Simple Setup

There is a "Simple Setup" button near the top of the home page, which links to the Simple Setup page.

Simple Setup Page

The Simple Setup Page allows for quick and easy selection of sources and destinations, along with setting the xNode's ID.

Figure 21   Simple Setup page

Node ID:

At the top of the Simple Setup page, the ID of the SDI xNode can be set, along with forcing the unit's host name to the Node ID, and the channel numbers for the sources to match that of the Node ID.

Sources

Sources define the audio inputs of the xNode coming from the SDI Input (s) . An input of the xNode is given a name and a unique channel number that is used to distribute through the AoIP network.

Destinations

The Destination page defines the audio outputs of the xNode. These are streams coming from the AoIP network to be embedded in the SDI output (s) .

Unicast Link Page

Unicast Link allows using xNodes to create a multichannel audio link between locations connected with an Ethernet network without support for IP multicasting. In particular it can be used as a studio transmitter link.
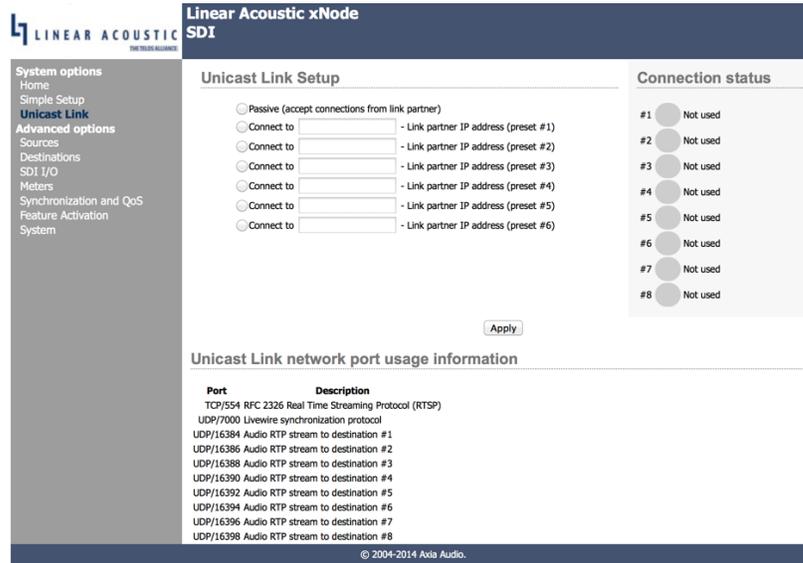

Figure 22 - Unicast Link page

Unicast Link can be quickly established using a dedicated WEB page of the xNode. This page shows current configuration. The page functions as a configuration wizard and applying the settings makes changes in audio routing setup and other options presented on the Sources, Destinations and QoS pages.

Unicast Link is by default bidirectional. The connection protocol supporting Unicast Link is Real Time Streaming Protocol (RTSP), which is a TCP/IP protocol. Audio is sent using unicast UDP/IP protocol.

Basic Setup

The most basic setup involves setting one xNode as Passive. This is also the default configuration of an xNode if user never configures Livewire+ Sources. On a brand new node, the only required step is setting IP address using the IP Setup screen on the front panel.

If the xNode is connected to a network with firewall, in order to allow Unicast Link connections, TCP and UDP ports listed on the WEB page need to be configured. The following ports are used:

Port	Description
TCP/554	RFC 2326 Real Time Streaming Protocol (RTSP)
UDP/7000	Livewire+ synchronization protocol
UDP/16384	Audio RTP stream to destination #1
UDP/16386	Audio RTP stream to destination #2
UDP/16388	Audio RTP stream to destination #3
UDP/16390	Audio RTP stream to destination #4

In order to connect, enter IP address of the remote "Passive" node in the "Connect to" field as shown below.

Figure 23 – Unicast Link Setup

Connection Status area provides diagnostic information of the established link. When the connection is successful, the icons turn green and Stream active / RTSP OK status is displayed.

Custom Setup

In order to limit number of channels, set Stream Mode to Disabled on the Sources page:
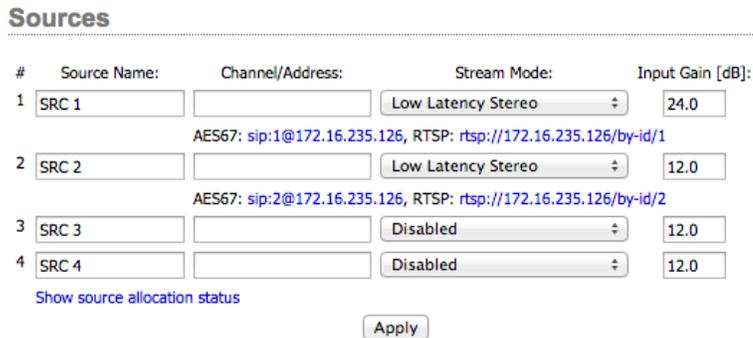

Figure 24 - Source configuration for Unicast Link

After doing this configuration on both link partners, the result is a 2-channel bidirectional link which status is indicated as shown below:

Figure 25 - 2x stereo link: Passive link partner

Unicast Link Setup

- ○ Passive (accept connections from link partner)
- ⦿ Connect to `172.16.235.126` — Link partner IP address (preset #1)
- ○ Connect to _____ — Link partner IP address (preset #2)
- ○ Connect to _____ — Link partner IP address (preset #3)
- ○ Connect to _____ — Link partner IP address (preset #4)
- ○ Connect to _____ — Link partner IP address (preset #5)
- ○ Connect to _____ — Link partner IP address (preset #6)

Connection status

- #1 🟢 Stream active / RTSP OK
- #2 🟢 Stream active / RTSP OK
- #3 🔴 No audio stream / RTSP OK
- #4 🔴 No audio stream / RTSP OK

Custom/advanced settings will be reset to Unicast Link defaults

- Clock mode: STL Snake Slave
- Sources 1-4: Livewire Standard Mode
- Destinations 1-4: Connected to selected Unicast Link partner

Figure 26 - 2x stereo link: Active link partner

The bandwidth and latency of the link can be customized using Stream Mode setting in the Sources WEB page. By default Standard Stream audio format is used as it is the most bandwidth efficient option.

With Unicast Link configuration, the synchronization mode is to "STL Livewire+ Slave" which allows for a higher packet forwarding jitter. This setting can be changed on the QoS WEB page.

Unicast Status on xNode Front Panel

In Unicast mode, the front panel presents the link status information as little triangles below the meters.

- ♦ If no triangle is shown, the Source and Destination port is not configured for Unicast.

- ♦ If the triangle is blinking, the port is configured for Unicast, but there is no active connection.

- ♦ When a solid triangle displayed, there is a Unicast Link established to that port.

Figure 27 - xNode display

Advanced Options Pages

Sources Page

Sources page defines the audio inputs of the xNode. An input of the xNode is given a name and a unique channel number that is used to distribute through the AoIP network. The xNode provides an auto numbering process when you ID the xNode through the node ID process described in Chapter 3.

Figure 28 - Sources Page

Source Name

The Source Name column is used to identify the device that is connected to the xNode input (for documentation sake!!) and is also used in source advertisement through the network when operating in Livewire+ mode.

Channel Name

The channel column is the unique number for each source. This will auto populate based on the ID given to the xNode. It is critical that each source in the network has a unique number, not shared with any other source.

Stream Mode

Stream mode column defines the source as a stereo source, a mono source, or multichannel sources and how the packet is formed

♦ Live stereo (or Live mono) is the original Livewire+ stream intended for Microphone or other live audio content that needs a lower delay. A packet contains 12 samples or 250us of audio.

♦ Low Latency Stereo (or mono) is an option for support of Ravenna and AES67 standards. The packet contains 1ms of audio (48 samples).

♦ Standard Stereo (or mono) is the high efficiency stream that contains 240 sample (5ms of audio), maximizing the packet size at 1522 bytes. This option is intended for suitable sources such as playback machines that can use the higher efficiency with slightly more latency (not more than 100msec).

♦ Surround: Front L, R is available at source number 1 and 4. This setting creates an 8-channel multichannel group which includes the next 3 sources. The standard channel routing used is:

◊ Pair 1 = Front Left / Right

◊ Pair 2 = Center / LFE

◊ Pair 3 = Left Surround / Right Surround

◊ Pair 4 = Stereo Downmix（or Left Back / Right Back）

Note!

Live Streams provide low audio latency of a round trip delay of 2ms. Each packet takes 154 bytes on the wire where packets are sent at 4000 packets/s. Each Live stream requires 4.9MBits/s bandwidth.

Standard Streams provide higher network utilization efficiency. Typical audio delay with Standard Streams is 100ms (determined by a buffer size). Each packet takes 1522 bytes on the wire (maximum allowed Ethernet frame size), at 200 packets/s. Each stream requires 2.5MBits/s bandwidth.

When a source is enabled through the stream mode, an RTSP link will appear under the Source row. This link can be given to a media player that supports RTSP as well as 24-bit PCM streaming audio for direct auditioning of the audio stream. VLC 1.1.11, for instance, is a cross platform media player which supports RTSP.

Input Gain

Input gain column provides adjustment of +/-12dB.

Mode

The Mode column defines whether the input audio is synchronous to the Livewire+ network master clock or uses a different clock reference and Sample Rate Converters are inserted on the audio to prevent audible glitches in the signal.

Two options are provided, asynchronous（default）and synchronous. Asynchronous mode inserts Sample Rate Converters on the input to the Livewire+ network, and allows different reference clock domains between the SDI path and the Livewire+ network. This is the recommended mode when working with PCM audio signals.

In cases where coded audio（Dolby Digital, Dolby E, HE-AAC, etc.）is to be carried across the Livewire+ network, and/or the entire facility is being operated on a master house reference clock should Synchronous mode be used.

Note!

Just like any other audio devices you may be used to, if coded audio (Dolby Digital, Dolby E, HE-AAC, etc.) is input and desired to be sent across the Livewire+ network without decoding it back to PCM first, Synchronous Mode must be used and the Livewire+ network clock reference must be locked to the SDI input. Attempting to pass coded audio signals through sample rate converters destroys the data in the stream.

I/O

Selects the input signal from the SDI Input（s）to de-embed as the audio for the source. Any of the of the 8 audio pairs from within an SDI stream are available for de-embedding.

Destination Page

The Destination page defines the audio outputs of the xNode. These are the signals which will be taken from the AoIP network to be embedded into an SDI Output.
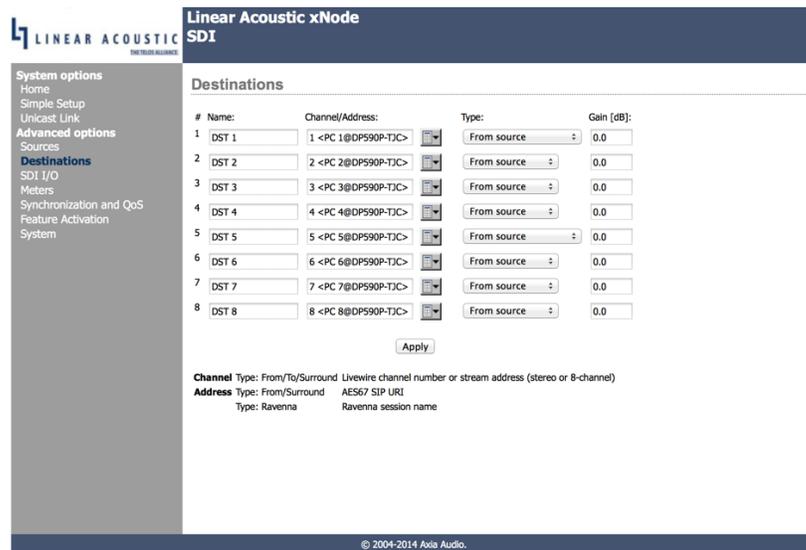

Figure 29 - Destinations page

Name

The first column is the name field which is used for documenting the output. This is helpful to recall what should be wired to the xNode or what source should be routed to the output.

Channel

Channel column is used to define the network source which is routed to the destination. You may type in the channel number or use the browse button to search the network for the desired source. When using the browse function, a popup will appear displaying all the sources in the network. Selecting a source will populate the channel information in the text field but the change will not take place until the Apply button is pressed. Selecting some "To:*" source from the network will automatically change the type field for you.

Type

The Type column defines if the output should behave as stereo or as a linked multichannel group, and if the output is the audio from the source defined in the channel field or a special dynamic return audio known as a backfeed. Backfeeds are created when one of the "To Source" options are chosen in the Type field.

Backfeeds turn the destination into a bidirectional link.   They are generally used with devices such as hybrids and telephone systems.

They exist in the network because of an Axia control surface which may have generated the source due to configuration. If you are comfortable with the concept of backfeeds, you can manually change the option. If not familiar, simply picking the desired source from the browse function should be all you need to do.

Figure 30 - Destination Type choices

The options for the Type column are:

♦ From Source: Selects a single direction signal path (default setting)

♦ To Source: Selects a bi-directional backfeed link.

♦ Surround: Front L, R: Selects an 8-channel multichannel output. This option is only available out at destination number 1 and 4, and links the following 3 destination pairs together into an 8-channel signal.

♦ Ravenna Stereo: Selects an output from a Ravenna session stream.

Gain

Gain column provides adjustment for the outputs from the range of +/-12dB.

SDI I/O Page

The SDI I/O page is where the SDI output streams are created.
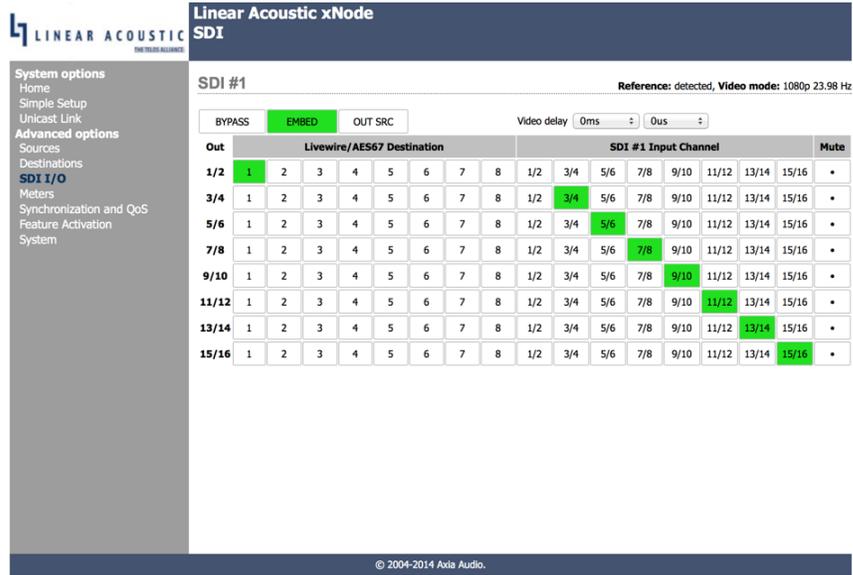
Figure 31 - SDI I/O page with single SDI output enabled

Bypass

This control engages the relay bypass, which directly connect the SDI Input and the SDI Output.

> **Note!**
>
> **When the SDI xNode is powered off, the relay bypass is automatically engaged in order to prevent breaks in the signal path, ensuring that the broadcast stays on-air.**

Embed

The Embed control page enables embedding audio into the SDI output stream. All 16-channels of the SDI output stream are available for embedding, pair shuffling, and pass-through.

Out
This column represents the audio channels of the output SDI stream. The signal source to be embedded into the SDI output will be selected from the same row. The audio being embedded into the SDI output can be selected from any Livewire+ destination or from the input SDI stream.

Livewire+/AES67 Destination
Livewire+ destinations 1 through 8 can be routed to, and embedded into the SDI output stream, on a pair by pair basis.

SDI Input Channel
Audio pairs from an input SDI stream can be routed to the SDI output. Pair shuffling from SDI Input to output may be performed.

Mute
These controls mute the individual channel pairs within the SDI Output stream.

Out SRC

16 channels of sample rate conversion are included on the outputs of the SDI xNode.   These sample rate converters may be used to either change the sample rate of the audio signal to 48 kHz, or to compensate for two different clock references.

This control bypasses all 16 channels of output sample rate conversion.   Individual control on a pair by pair basis is not available for the output sample rate converters.

Video Delay

The SDI xNode includes compensating video delay for each pair of SDI I/O. This delay is used to ensure synchronization between audio and video signals at the SDI output.

The delay may be adjusted in both milliseconds (ms) and in microseconds (us) increments.   Up to a full second of video delay for each SDI I/O is available.

Meter Page

The Meter page uses a Java script to provide real time meter representation of the audio at the inputs (source) and outputs (destination) of the Livewire+ network. The meter page shows a stereo channel or a mono channel based on the configuration of the xNode and the amount of channels enabled.

The meters are represented in dBFS.
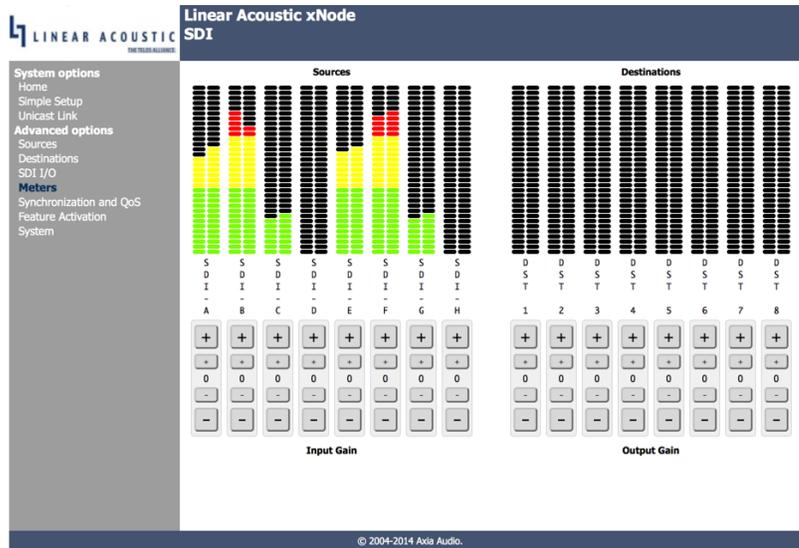
Figure 32 - Meters page

Gain Controls

Below the meters are gain adjustment buttons which allow for 1dB adjustment (Large button) and 0.1dB adjustment (Small button) of the audio signal.

Synchronization and QoS Page

The Synchronization and QoS page is where the Livewire+ network and SDI xNode reference clock is configured, along with network QoS configuration.

Livewire+'s clocking system is automatic and largely transparent to end users. If no clock is present on the network, Livewire+ devices negotiate a master by means of priority settings and the lowest MAC address. If a device comes online into a network that has a clock present, it will not generate clock unless it has priority setting of 7. The xNode does permit variance in determining the source of clock.

Figure 33 - Synchronization and QoS page

Clock mode

Livewire+ IP low rate (the default setting) is suggested for a typical Livewire+ network. Livewire+ IP "STL" is an option for an application where a node is at the remote end of an STL link (Live streams not supported while in this mode). Livewire+ IP high rate tasks the node to synchronize to the fast clock is not needed in most cases. PTP/IEEE 1588 ARB clock class 248 is the option to enable the xNode to participate in the execution of the best master clock algorithm for a PTP (Precision Time Protocol) network (enabling it to be a master PTP clock). PTP/IEEE 1588 slave enables the xNode to synchronize to a PTP network. NTP slave for special applications that synchronize from an NTP source (Live streams not supported under NTP mode).

Note!

If an xNode becomes PTP master and the Livewire+ clock master priority is set to a value greater than 0, the xNode will change its priority to 7. This allows for the xNode to also become the master source for Livewire+ clock, ensuring a single clock source in the network.

Livewire+ clock master priority

In a Livewire+ network, higher priority is given to nodes with a higher priority setting. A value of 0 will never generate clock. A value of 7 will always generate clock. Values between will generate clock if no clock is received from another device, and the system will elect a master source based on priority settings. 7 (Always master) "STL snake" – used for cases where the xNode is one end of a wireless link. This mode will cause the node to generate clock and will suppress the generation of fast clock. The default priority is three (3) which allows any device to become source of clock with equal priority.

PTP domain number

(0 – 127) A domain is one or more PTP devices communicating with each other. A domain association is used to define communication messages. A value of 0 is standard.

PTP delay mechanism

(E2E or P2P) PTP supports two methods of time correction which alter messaging between devices. All devices within the network should use the same mechanism. The recommended setting is to use End-to-End (E2E) since more networked devices prefer to support this option.

PTP clock priority1/priority2

Priority setting from 0-255 is used in the best master clock algorithm (BMC) to determine a Grandmaster. Lower numbers have higher priority.

PTP clock sync interval

Is the time for sending synchronization messages. A setting of -4 is 16 messages in 1 second. A setting of 1 is 1 message in 2 seconds. The default setting is 0, 1 message per second.

Master node clock timebase

Selects the reference clock source when the SDI xNode is the master node on the Livewire+ network. The options are the internal 48 KHz clock, SDI #1, and SDI #2.

System Page

The System page provides network settings, SNMP settings, Syslog settings, authentication password, and firmware control.

Figure 34 - System page

Host Name

Name of the device on the network. Up to 12 characters using only letters, numbers, or hyphens (no spaces or special characters). Host name is auto populated when you assign an ID to the xNode providing a unique name. You may customize this field if you desire.

Network Address

Network address: The IP address of the device, as configured either manually or via the Setup button or iProbe software. The address can be changed here, but once you apply the changes, you will lose connection and will need to connect to the device with the new IP address you just assigned.

You may also configure a management port. By default, NET2 address will be assigned to 0.0.0.0, which defines the second port as a redundant network port. If you'd rather forsake redundancy for a separate management port which does not pass Livewire+ audio, then set an IP address for the NET2 field, then define which port is the management port and which is the Livewire+ port.

Mask

The netmask is a filter to which helps to define the subnetwork. This is a topic covered in networking material widely available. For simplicity, a value of 255.0.0.0 is a wide open filter and a value of 255.255.255.254 is a very narrow filter. If you are not familiar with how subnetting works, please refer to your network administrator. A value of 255.255.255.0 is usually suitable for most cases.

Gateway

The gateway is a device that handles requests and responses from devices that are in different subnets. A gateway is what is needed when dealing with more than a single subnet. If creating a Livewire+ network on a dedicated local area network (LAN), then the value of the gateway can likely be ignored. If not familiar with the function of gateways, please ask your network administrator.

Livewire+ Enabled Ports

Default setting is for NET1 and NET2 to be Livewire+ enabled ports. This provides redundant network ports for audio and control to the same IP address. If you'd rather forsake redundancy to have a separate management port which is removed from the high bandwidth audio traffic, than apply a value to NET2, then select which port is to be Livewire+ enabled, and which is the management port.

NTP server

Specify the sync server if using NTP as the synchronization source. This mode of sync is only good for Standard streams and used for unique applications.

System Location

Define system location for SNMP

System Contact

Define system contact for SNMP

SNMP Community Name

Define the community for SNMP functionality

Syslog Server

IP address of the computer which is running a syslog server application.

Syslog Severity Level Filter

The 8 standard syslog levels are available for selection.

User Password

The authentication password can be changed be entering in the password twice in the two fields.

Firmware Version

The SDI xNode uses two storage banks to store software for the xNode. Switching to a new version of software is done by selecting the other radio button and pressing Apply button.

> **Note!**
>
> **Selecting a new version of software to operate under requires a reboot of the xNode. Rebooting will cause any audio streams to be interrupted until the reboot completes.**

If Bank 1 is empty, you will you use the Browse function to locate the desired software that has been downloaded to your computer. Once the file is selected, pressing the Apply button will upload the software into the bank. The process will not interrupt the operation of the xNode. Once complete, the System page will return and the two banks will show versions of software.

If operating with Bank 1, and you wish to upload a different version, you will select the "commit this version to Bank 0" and press Apply button. This will move the code to Bank 0 and empty Bank 1

5 AES67 Features

AES67

The AES67 standard defines an interoperability mode for transport of high-performance audio over networks based on the Internet Protocol. For the purposes of the standard, high-performance audio refers to audio with full bandwidth and low noise. These requirements imply linear PCM coding with a sampling frequency of 44.1 KHz and higher, and resolution of 16-bits and higher. High performance also implies a low-latency capability compatible with live sound applications. The standard considers latency performance of 10 milliseconds (ms) or less.

The xNode implementation of AES67 is compliant with the standard, supporting interoperability with third party devices. In particular, the following features are implemented:

◆ Multicast and Unicast Real Time Protocol (RTP) audio streaming

◆ Session Initiation Protocol (SIP) based connection management for Unicast streams

◆ Transmission of 1ms audio frames in linear PCM 24-bit format

◆ Ability to receive audio streams containing 1 through 8 channels

◆ Ability to receive 16-bit as well as 24-bit format

◆ Support of IEEE1588-2001 Precision Time Protocol standard

◆ Time-stamping of outgoing streams with IEEE1588 derived RTP timestamps

Network Synchronization Setup

The AES67 standard requires devices to use the IEEE1588 network time protocol standard. Synchronization related options are available on the QoS WEB page.

Synchronization:

Clock mode:	PTP/IEEE 1588 ARB clock class 248
PTP domain number:	0 default
PTP delay mechanism:	E2E – uses the end–to–end delay request–response (default)
PTP clock priority1/priority2:	128 / 128
PTP clock sync interval:	–3 (1/8) – AES67 Default

Figure 35 - AES67 Synchronization configuration

PTP/IEEE clock mode selects the protocol required by standard. The xNode offers a few options depending on other system requirements:

♦ **PTP/IEEE 1588 ARB clock class 248** mode allows the node to be a master when no better clock is available on the network. Since xNodes are not locked to GPS or any other high precision source, only Arbitrary (ARB) Clock class is provided.

♦ **PTP/IEEE 1588** slave only is recommended for networks with high precision grand masters present.

♦ **PTP Domain 0** should be selected unless there is a need for multiple clock domains on a network segment. Choosing a different domain a useful option for doing test without affecting other components present on the network.

♦ **PTP delay mechanism** specifies how the node determines network round trip latency. This setting MUST match the master clock setting. In practice, End-to-End (E2E) is always used by ordinary clocks and boundary clocks. Peer-to-Peer (P2P) mechanism was defined to limit amount of traffic reaching the master clock, but is rarely used.

♦ **PTP clock priority1 and priority2** numbers provide a control over master clock selection. Lower priority number makes the clock a preferred choice over devices with a higher number. The range is 0–255, and 128 is the default and middle value.

♦ **PTP clock sync interval** determines how often clock packets are sent. The faster those packets are sent, the faster and more accurate synchronization can be achieved. Some IEEE1588 devices do not support lower clock sync interval than 0.5s, so this setting may need to be adjusted depending on what other networking equipment is used. AES67 recommends an interval of 1/8 (8 packets per second). This setting affects behavior of the master clock and does nothing when the node is a slave.

AES67 Multicast Connection Setup

The xNode transmits a multicast stream as soon as Source is configured with a Livewire+ channel number or multicast IP address, and Stream Mode is set to a value other than "disabled". AES67 requires devices to support reception of at least 1ms frames – this corresponds to "Low Latency" Stream Mode setting. Other settings will work if the receiver supports different frame sizes in addition to required 1ms.

Figure 36 - Transmitting a 1ms AES67 stream to multicast group 239.194.0.1

Session Description (SDP) can be downloaded from an xNode by clicking on the 'AES67: Download stream description (SDP)" link shown above.

AES67 multicast stream reception is set up by entering the multicast address of the stream into the Channel/Address field of a Destination.

Figure 37 - AES67 receive destination setup

AES67 multicast reception requires the source to transmit in PCM 24-bit stereo or 8-channel format.

Surround 8-channel reception is set up by selecting Destination Type: Surround.

The Channel/Address field accepts an optional UDP port number which is entered following the IP address, separated with a ":". This is a useful option when the port number is different than RTP default 5004.

AES67 Unicast Setup

Each Source of an xNode can be configured as a multicast sender, or Unicast port supporting a single listener/receiver. Unicast configuration is achieved by clearing the Channel/Address field in a Source setup.

Figure 38 - AES67 Unicast Source

The media stream is accessed using a SIP URI (Session Announcement Protocol Uniform Resource Identifier). It contains "sip:" prefix, followed by Source/Destination port number, '@' sign and IP address of the node. xNodes provide the stream addresses on the Sources WEB page, so that you can conveniently copy them.

In Unicast mode, Source accepts SIP connections. There are a few possible scenarios:

♦ **Receive-only connection** (SIP Invitation with *recvonly* attribute specified in the SDP block). After answering, xNode will start sending a stream to the originator of the connection. Media format follows current Source configuration.

♦ **Send-only connection** (SIP Invitation with *sendonly* attribute specified in the SDP block). After answering, xNode will start receiving the stream, which can be 16- bit or 24-bit with channel count from 1 to 8. This type of a connection requires the corresponding Destination Channel/Address field to remain empty.

♦ **Bidirectional** (default SIP invitation). xNode will receive and send. The media format must be 24-bit. This type of a connection requires the corresponding Destination Channel/Address field to remain empty.

In order to setup AES67 Unicast stream receive, enter the SIP URI stream address in Channel/Address field of a destination.

Destinations

Figure 39 - AES67 stream reception

xNodes make a SIP call to the specified source. The call is receive-only (issued with *recvonly* SIP/SDP media attribute specified).

xNodes provide status of the Unicast connection on the front panel. Small triangle icons below the meters show the current status:

- ♦ If no triangle is shown, the Source and Destination port is not configured for Unicast.

- ♦ If the triangle is blinking, the port is configured for Unicast, but there is no active connection.

- ♦ When a solid triangle displayed, there is a Unicast Link established to that port.

The Channel/Address field accepts optional entry of a UDP port number in the URI, separated from the address with a ":". This is a useful option when the port number used is different than SIP default 5060.

6 Configuration Example

Introduction

This chapter explains and provides guidance on configuring the SDI xNode for usage. Chapter 3 described the initial configuration of the unit in order to control it. This chapter describes configuring inputs, outputs, and signal routing.
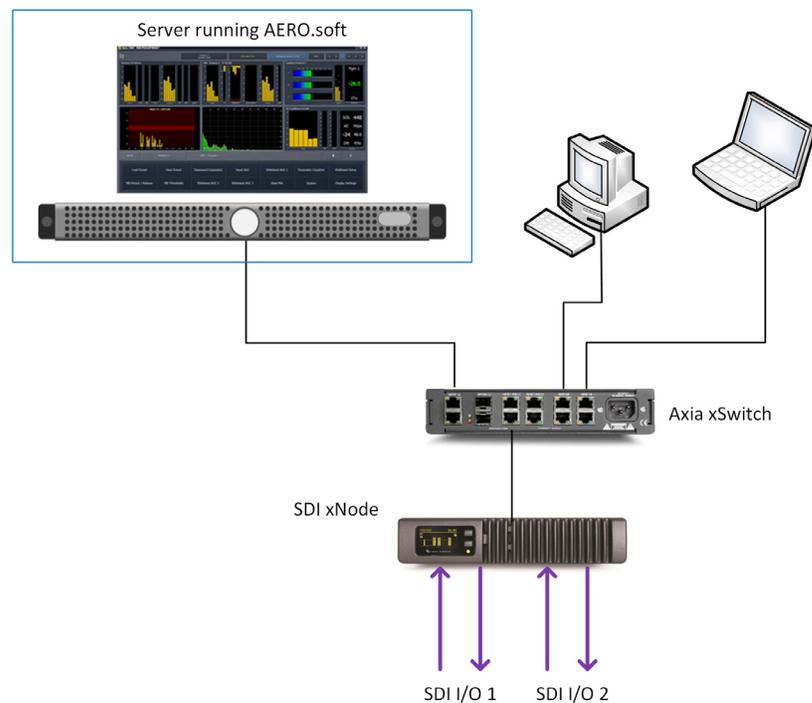

Figure 40 - Sample system block diagram

The use case example described in this chapter consists of a SDI stream on SDI Input 1 that contains three audio programs:

 ♦ a main stereo program in English

 ♦ a stereo program in a second language (SAP)

 ♦ a 5.1-channel program for the High Definition channel

The audio programs are sent via Livewire+ to a server running Linear Acoustic's AERO.soft Loudness Management Suite. AERO.soft is configured to process a 5.1-channel program and 2 stereo programs simultaneously.

The output streams from the AERO.soft server are returned to the SDI xNode via Livewire+, and are embedded into the output SDI stream in the same order they were input in.

SDI In to Livewire+ Source

This section describes how to configure the xNode to accept a source from SDI 1, de-embed the desired audio from it, and output it across the Livewire+ network.

We will use audio pairs 1 – 5 (channels 1/2, 3/4, 5/6, 7/8, and 9/10) from the SDI stream and send them out as Livewire+ Source numbers 11141 – 11143. The input SDI stream is constructed in this manner:

SDI Input Pair (Ch)	Signal	Signal Type
1 (1/2)	Stereo Program (English)	PCM
2 (3/4)	5.1-ch program - L / R	PCM
3 (5/6)	5.1-ch program - C / LFE	PCM
4 (7/8)	5.1-ch program - Ls / Rs	PCM
5 (9/10)	Stereo Program (SAP)	PCM
6 (11/12)	Not used	PCM
7 (13/14)	Not used	PCM
8 (15/16)	Not used	PCM

In the Sources page, we will map:

♦ SDI 1-Pair 1 to Livewire+ #11141, labeled with a Source Name of *Server-Eng*

♦ SDI 1-Pairs 2, 3, and 4 to Livewire+ #11143, labelled with a Source Name of *Server-Surround* *****

   * Livewire+ multichannel sources are configured in groups of 8. For this example, we will set channels 7 / 8 to a silent source.

♦ SDI 1-Pair 5 to Livewire+ #11142, labelled with a Source Name of *Server-SAP*

Figure 41 - Sources Configuration

Livewire+ Destinations

This section describes how to configure the SDI xNode for the proper Livewire+ destinations using the Destinations page.

In this example, we will configure Destination 1 as coming from AERO.soft-1-1 (the 5.1-ch processor), Destination 2 as coming from AERO.soft-1-2 (the first 2-channel processor), and Destination 3 as coming from AERO. soft-1-3 (the second 2-channel processor).

Figure 42 - Destinations configuration

SDI Output Embedding

This section illustrates configuring the SDI xNode to embed the Livewire+ destinations (outputs from AERO.soft) to the correct audio pairs in the SDI output stream. The desired output mapping of audio programs is the same as the input streams:

♦ The main stereo program (English) is embedded in SDI pair 1

♦ The stereo program (SAP) is embedded into SDI pair 5

♦ The 5.1-channel program is embedded into SDI pairs 2, 3, and 4

Of course, the audio signals presented at the SDI output would have been processed by AERO.soft, but are in the exact same channel order as the original, unprocessed, input signals.

In this configuration example, SDI output pairs 6, 7, and 8 are muted, but could easily carry audio signals through from the input SDI.

Linear Acoustic xNode
SDI

SDI #1   Reference: detected, **Video mode:** 1080p 23.98 Hz

BYPASS	EMBED	OUT SRC						Video delay	0ms ⇕	0us ⇕				

Out	Livewire/AES67 Destination								SDI #1 Input Channel							Mute
1/2	**1**	2	3	4	5	6	7	8	1/2	3/4	5/6	7/8	9/10	11/12	13/14	15/16	•
3/4	1	2	3	4	**5**	6	7	8	1/2	3/4	5/6	7/8	9/10	11/12	13/14	15/16	•
5/6	1	2	3	4	5	**6**	7	8	1/2	3/4	5/6	7/8	9/10	11/12	13/14	15/16	•
7/8	1	2	3	4	5	6	**7**	8	1/2	3/4	5/6	7/8	9/10	11/12	13/14	15/16	•
9/10	1	**2**	3	4	5	6	7	8	1/2	3/4	5/6	7/8	9/10	11/12	13/14	15/16	•
11/12	1	2	3	4	5	6	7	8	1/2	3/4	5/6	7/8	9/10	11/12	13/14	15/16	•
13/14	1	2	3	4	5	6	7	8	1/2	3/4	5/6	7/8	9/10	11/12	13/14	15/16	•
15/16	1	2	3	4	5	6	7	8	1/2	3/4	5/6	7/8	9/10	11/12	13/14	15/16	•

Figure 43 - SDI I/O page

7 Specifications

Physical

Mechanical	
Dimensions	1.72"H x 8.5"W x 11.75"D (4.4 x 21.6 x 29.9 cm)
	1 RU high x 1/2 rack width
Weight	5 lbs (2.3 kg)
Shipping Dimensions	12" x 7" x 16" (30.5 x 17.8 x 40.7 cm)
Shipping Weight	9 lbs (4.1 kg) approximate
Environmental	
Operating	0 – 50 degrees C
Non-operating	-20 – 70 degrees C
User Interface	
Display	dot matrix, single color OLED display
Front Panel Control	Two, multi-function, push-button switches.
Web Control	Internal http server. Enables full configuration, control, and monitoring via common web browser applications
Connectors	
Power	IEC universal power inlet DC power inlet for back-up power supply
Signal	Four (4) BNC-F connectors for SDI I/O
Ethernet	Two (2) RJ-45 connectors for Ethernet / Livewire+/AoIP

Electrical

Power	
Power Requirements	95-240vAC, 50/60 Hz. 18 Watts maximum. Universal IEC Power inlet. IEC cord matches country of delivery
Redundant Power	External 12 VDC power supply (optional)
Cooling	Passive
Digital I/O	
HD/SD-SDI	2 SDI input and 2 SDI output connectors. Connections are unbalanced, 75-Ω, with internal termination via BNC-F connectors. Auto-sensing 3GHz HD/SD-SDI with de-embedding of audio channels from each SDI input. De-embedded audio can be routed to Livewire+ output and/or re-embedded to its matching SDI output with SMPTE 292M (HD-SDI) and SMPTE 259M (SD-SDI) support 16-channels of audio (total) from SDI input can be routed to the Livewire+ network.
Ethernet	Two (2) 100 BASE-T connections via RJ-45 connector
Livewire+ and AES67	Full Livewire+ and AES67 compatibility
Reference	Livewire+ at audio input, SDI-derived audio clock at output

Audio

Audio	
Format	PCM, coded audio
Sample Rate	48 KHz
Bit depth	Up through 24 bits
Input / Output Formats	
HD-/SD-SDI	HD-SDI (SMPTE 292M), SD-SDI (SMPTE 259M), up to 3 Gb interface
Audio over IP	Livewire+, AES67, Ravenna

Other

Options	
	External redundant power supply
Warranty	
	Standard Telos Alliance 5-year warranty.
Regulatory	
North America	Designed to comply with the limits for a Class A digital device pursuant to Part 15 of the FCC rules (CFR).
Europe	Designed to comply with the limits of harmonised Low Voltage Directive 2006/95/EC and EMC Directive 2004/108/EC as indicated by the affixed CE marking; RoHS and WEEE compliant.

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Telos Alliance Limited Warranty

For the latest Telos Alliance warranty, visit: telosalliance.com/warranty.

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