8x8 Analog Node
8x8 AES Node
Installation & User's Guide

AXIA
A TELOS COMPANY

IMPORTANT NOTE:

Axia nodes are intended for use with an Ethernet Switch that supports multicast and QOS (Quality of Service). On a non-switched Ethernet hub, or a switch that is not enabled for multicast, this will result in network congestion that could disrupt other network activity. Should you wish to connect a node to a non-Livewire network for access to the web configuration interface, etc, you must first confirm that streaming is disabled as described in Section 2 page 6 "Programming the Node's Streaming Mode".

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 only 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 present appareil numerique n'emet pas de bruits radioelectriques depassant les limites applicables aux appareils numeriques (de les Class A) prescrites dans le Reglement sur le brouillage radioelectrique edicte par le ministere des Communications du Canada."

Important Safety Information

To reduce the risk of electrical shock, do not expose this product to rain or moisture. Keep liquids away from the ventilation openings in the top and rear of the unit. Do not shower or bathe with the unit.

Caution

The installation and servicing instructions in the 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.

Electrical Warning

To prevent risk of electric shock: Disconnect power cord before servicing.

This equipment is designed to be operated from a power source that includes a third "grounding" connection in addition to the power leads. Do not defeat this safety feature. In addition to creating a potentially hazardous situation, defeating this safety ground will prevent the internal line noise filter from functioning.

Ventilation Warning

The Axia 8x8 node uses convection cooling. Do not block the ventilation openings in the side of the unit. Failure to allow proper ventilation could damage the unit or create a fire hazard. Do not place the unit on a carpet, bedding, or other materials that could interfere with the rear and top panel ventilation openings.

Customer Service

We support you...

By Phone/Fax in the USA.

- Customer service is available from 9:30 AM to 6:00 PM USA Eastern Time, Monday through Friday at +1 216.241.7225. Fax: +1 216.241.4103.

By Phone/Fax in Europe.

- Service is available from Axia Europe in Germany at +49 81 61 42 467. Fax: +49 81 61 42 402.

By E-Mail.

- The address is **Support@AxiaAudio.com**.

Via World Wide Web.

- The Axia Web site has a variety of information which may be useful for product selection and support. The URL is **http://www.AxiaAudio.com.**

Feedback

We welcome feedback on any aspect of Axia 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.

Updates

The operation of the Axia node is determined largely by software. Periodic updates may become available - to determine if this is the case check our web site. Contact us to determine if a newer release is more suitable to your needs.

Our electronic newsletter has announcements of major software updates for existing products, as well as keeping you up to date on the latest Axia, Telos, and Omnia product releases. To subscribe go to:

http://www.axiaaudio.com/news/eNews.htm

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Notice

All versions, claims of compatibility, trademarks, etc. of hardware and software products not made by Axia mentioned in this manual or accompanying material are informational only. Axia makes no endorsement of any particular product for any purpose, nor claims any responsibility for operation or accuracy.

Warranty

This product is covered by a one year limited warranty, the full text of which is included in the rear section of this manual.

Service

You must contact Axia before returning any equipment for factory service. Axia 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 at Axia. Be sure to adequately insure your shipment for its replacement value. Packages without proper authorization may be refused. US customers please contact Axia technical support at +1 (216) 241-7225. All other customers should contact their local representative to arrange for service.

We strongly recommend being near the unit when you call, so our Support Engineers can verify information about your configuration and the conditions under which the problem occurs. If the unit must return to Axia, we will need your serial number, located on the rear panel.

About This Manual

This manual covers the details of the Axia 8x8 Analog and 8x8 AES nodes. However it is assumed in this document that you are familiar with Livewire's basic concepts, as outlined in the companion Introduction to Livewire: System Design Reference & Primer manual.

If you have not done so, please review that material first. In it 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. After reading Introduction to Livewire you will know what's up when you are speaking with gear vendors and the network guys that are often hanging around radio stations these days.

As always, we welcome your suggestions for improvement. Contact Axia Audio with your comments:

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A Note From The Founder/CEO of Telos

It's been a tradition since Telos' very first product, the Telos 10 digital phone system, that I share a few words with you at the beginning of each manual. So here goes.

In radio broadcast studios we're still picking up the pieces that have fallen out from the digital audio revolution. We're not using cart machines anymore because PCs are so clearly a better way to store and play audio. We're replacing our analog mixing consoles with digital ones and routing audio digitally. But we're still using decades-old analog or primitive digital methods to connect our gear. Livewire has been developed by Telos to provide a modern PC and computer network-oriented way to connect and distribute professional audio around a broadcast studio facility.

Your question may be, "Why Telos? Don't you guys make phone stuff?" Yes, we certainly do. But we've always been attracted to new and better ways to make things happen in radio facilities. And we've always looked for opportunities to make networks of all kinds work for broadcasters. When DSP was first possible, we used it to fix the ages-old phone hybrid problem. It was the first use of DSP in radio broadcasting. When ISDN and MP3 first happened, we saw the possibility to make a truly useful codec. We were the first to license and use MP3 and the first to incorporate ISDN into a codec. We were active in the early days of internet audio, and the first to use MP3 on the internet. Inventing and adapting new technologies for broadcast is what we've always been about. And we've always been marrying audio with networks. It's been our passion right from the start. In our genes, if you will. As a pioneer in broadcast digital audio and DSP, we've grown an R&D team with a lot of creative guys who are open-eyed to new ideas. So it's actually quite natural that we would be playing marriage broker to computer networks and studio audio.

What you get from this is nearly as hot as a couple on their wedding night: On one RJ-45, two-way multiple audio channels, sophisticated control and data capability, and built-in computer compatibility. You can use Livewire as a simple sound card replacement – an audio interface connecting to a PC with an RJ-45 cable. But add an Ethernet switch and more interfaces to build a system with as many inputs and outputs as you want.

Audio may be routed directly from interface to interface or to other PCs, so you now have an audio routing system that does everything a traditional "mainframe" audio router does – but at a lot lower cost and with a lot more capability. Add real-time mixing/processing engines and control surfaces and you have a modern studio facility with many advantages over the old ways of doing things. OK, maybe this is not as thrilling as a wedding night – perhaps kissing your first lover is a better analogy. (By the way, and way off-topic, did you know that the person you were kissing was 72.8% water?)

While we're on the subject of history… you've probably been soldering XLRs for a long time, so you feel a bit, shall we say, "attached" to them. We understand. But no problem – you'll be needing them for microphones for a long while, so your withdrawal symptoms won't be serious. But your facility already has plenty of Ethernet and plenty of computers, so you probably already know your way around an RJ-45 as well. It's really not that strange to imagine live audio flowing over computer networks, and there's little question that you are going to be seeing a lot of it in the coming years.

The 20th century was remarkable for its tremendous innovation in machines of all kinds: power generators, heating and air conditioning, cars, airplanes, factory automation, radio, TV, computers. At the dawn of the 21st, it's clear that the ongoing digitization and networking of text, audio, and images will be a main technology story for decades to come, and an exciting ride for those of us fortunate to be in the thick of it.

Speaking of years, it has been a lot of them since I wrote the Zephyr manual intro, and even more since the Telos 10 – 20 years now. Amazing thing is, with all the change around us, I'm still here and Telos is still growing in new ways. As, no doubt, are you and your stations.

Steve Church

A Note From The President of Axia

20 years ago, I designed my first broadcast console for PR&E. I look back on that time with great fondness; we were building bullet-proof boards for the world's most prestigious broadcasters, making each new console design bigger and fancier to accommodate a wider variety of source equipment and programming styles. The console was the core of the studio; all other equipment was on the periphery.

Then things changed: the PC found its way into broadcast audio delivery and production. At first, PC audio applications were simple, used only by budget stations to reduce operating expenses. But soon the applications evolved and were embraced by larger stations. Slowly, the PC was taking center stage in the radio studio.

Like many, I was captivated by the PC. Stations retired carts, phonographs, open-reel decks, cassettes — even more modern digital equipment such as DAT and CD players, replacing all with PC apps. Client/server systems emerged and entire facilities began using PCs to provide most – or all – of their recorded audio. Yet consoles continued to treat PCs as nothing more than audio peripherals. I knew that we console designers were going to have to rethink our designs to deal with computer-centric studios.

During this time, traditional broadcast console companies began producing digital versions. But early digital consoles were nearly identical in form and function to their analog predecessors. It took a fresh look from a European company outside broadcasting to merge two products – audio routing switchers and broadcast consoles – into a central processing engine and attached control surface. Eventually nearly every console and routing switcher company followed suit, and a wide variety of digital "engines" and control surfaces flooded the market.

But, advanced as these integrated systems were, they still handled computer-based audio sources like their analog ancestors. Sure, the router and console engine were now integrated, but the most important studio element – the PC – was stuck in the past, interfaced with 100-year-old analog technology. The PC and console couldn't communicate in a meaningful way – strange, considering that PCs everywhere were being networked, fast becoming the world's most popular and powerful communication tool.

Then a group of Telos engineers developed a method of using Ethernet to network real-time audio devices, allowing computers and consoles, controllers and peripherals to interact smoothly and intelligently. Powerful, flexible networks had finally come to our studios. As with the transition from carts to computers, the benefits are many and impressive. A few networked components can replace routing switchers, consoles, processing peripherals, sound cards, distribution amps, selector switches and myriad related devices.

This deceptively simple networked system costs a fraction of other approaches, yet has capabilities surpassing anything else. The system is modular and can be used to perform discrete functions in a traditional environment. Concurrently, it easily scales to serve both the humblest and the very largest of facilities. Console, router, and computer work in harmony.

So, equipped with this new technology and countless ideas, we launch *Axia*, the newest division of Telos. *Axia* is all about delivering innovative networked audio products to future-minded broadcasters. On behalf of our entire team, I welcome you as a charter client. Axia is the culmination of nearly 40 man-years of some of the most ambitious R&D ever applied to the radio industry. And this is only the beginning. We have more products, innovations, and partnerships in the pipeline.

You already know your Axia system is unlike anything else. So it shouldn't be surprising that your new system is loaded with new thinking, new approaches, and new ideas in virtually every conceivable area. Some concepts will challenge your traditional ideas of studio audio systems, but we're certain that once you have experienced the pleasures of the networked studio, you'll never want to go back. And now, for something completely different...

Michael "Catfish" Dosch

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Chapter One:

Introducing the 8x8 Analog and 8x8 AES Nodes

This section will allow you to get to know the 8x8 Node and describes the unit's features, display, and connectors.

Description

The Axia Livewire 8x8 Audio Node has eight stereo inputs, and eight stereo outputs. 8x8 Analog nodes have balanced analog inputs and outputs whereas the 8x8 AES nodes have AES3 inputs and outputs. Therefore, each 8x8 Node can create 8 stereo Livewire streams, each of which becomes available to other devices on the Livewire network.

Each stereo output can be assigned to output a stereo Livewire stream acquired from the network.

Basic point-to-point (e.g. "Livewire Snake") applications require only two Livewire nodes and a category 6 "Crossover Ethernet cable". More sophisticated multipoint networks can be used by connecting multiple Livewire nodes to an appropriate Ethernet Switch.

NOTE: Only an approved, and properly programmed Ethernet switch, incorporating the proper Multicast and QoS standards may be used.

Front Panel Controls and Indicators

The Livewire 8x8 Audio Node incorporates a number of front panel indicators to allow the operator to verify proper operation quickly and confidently.

Status LED indicators

Four LEDs indicate the status of the Livewire and Ethernet connections, as well as system synchronisation as follows:

LINK

When illuminated continuously, this LED represents the presence of a live Ethernet link to another Ethernet 100 Base-T device. This LED indicates that a connection is present and some device is connected. It does not indicate the quality of the connection however. If no Ethernet link is present, this will flash slowly.

LIVEWIRE

This LED indicates that the connected Ethernet segment has Livewire traffic present. If the link LED is illuminated, and the LIVEWIRE LED fails to illuminate, there are either no other Livewire devices connected, or the Ethernet switch has not been programmed to pass such traffic to the port to which this node is connected.

SYNC & MASTER

Only one of these two LEDs should be illuminated, with the one exception noted below. If neither LED illuminates, something is not correct. The SYNC LED indicates the receipt of clock information from another (Master) Livewire Node. The MASTER LED indicates that this node is acting as the master clock source for the Livewire network. More specifically:

SYNC – If Sync packets are being received by the Livewire node, this LED will begin to flash. The LED will continue to flash until the Livewire node has locked

its local clock to the network master. Once the local node's PLL is locked, the LED will illuminate solidly.

MASTER – The Livewire system employs a sophisticated master/slave clocking system over the Ethernet network. By default any device may become the clock master, however this can be changed if desired. The system has the ability to automatically change to a different clock master should the current master become disconnected, or otherwise inoperable. This happens transparently without audio glitches. This LED indicates that this node is currently acting as MASTER.

SYNC & MASTER with the 8x8 AES Node

When the AES Node is used in the default configuration the SYNC and MASTER LEDs operate as described above. However, if the AES sync input as Livewire master timebase is set to YES from the QOS web pages (see section 3, Advanced Programming) these lights will operate differently, as described here:

SYNC – If a valid AES signal received by the AES input designated as AES sync source in the QOS web page, this LED will begin to flash. The LED will continue to flash until the Livewire node has locked its local clock to the AES input signal. Once the local node's PLL is locked, the LED will illuminate solidly.

MASTER – This LED indicates that this node is currently acting as MASTER clock source for the Livewire network.

Bargraph LED Meters

The Livewire 8x8 nodes have a bargraph meter for each input and each output.

INPUT (Transmit to network) METERS

The left eight meter-pairs represent the left and right

channels of each input. The meters are continuously active, and indicate that audio is present at the associated input. The lowest LED segment will illuminate at a signal level of -42 dBfs. The top-most LED segment represents a level of 0 dBfs – this segment should not illuminate except in cases of extreme input overload.

OUTPUT (Receive from network) METERS

The rightmost 8 LED meter pairs represent the left and right channels of each audio stream being received. Each meter is associated with the audio on the corresponding output. The meter calibration is the same as for the input meters.

The lowest Output LED segment has a special meaning. These LEDs will be illuminated whenever the stream designated for that channel is present, and the Livewire node is locked to that stream. Therefore, the lowest LED segment will be illuminated during silence, to indicate that the associated Livewire stream is correctly being received.

Rear Panel

The rear panel of both the 8x8 Analog Node and the 8x8 AES Node are essentially identical and are pictured below:

AC (Mains) Power

The AC receptacle connects mains power to the unit with a standard IEC (International Electrotechnical Committee) power cord. The power supply has a "universal" AC input, accepting a range from 85 to 265 VAC, 47-63 Hz. A fuse is located inside on the power supply circuit board

IMPORTANT! As with any piece of modern electronic gear, it is advisable that precautions be taken to prevent damage caused by power
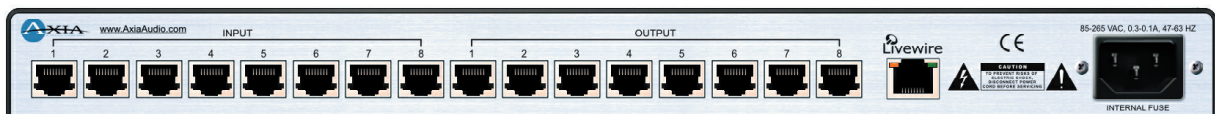
surges. Standard line surge protectors can be used to offer some degree of protection. It is the user's responsibility to ensure protection adequate for their conditions is provided. This equipment is designed to be operated from a power source, which includes a third "grounding" connection in addition to the power leads. Do not defeat this safety feature. In addition to creating a potentially hazardous situation, defeating this safety ground will prevent the internal line noise filter from functioning. If fuse replacement is required, please note: For continued protection against fire, replace fuse only with same type and value. See the DETAILED TECHNICAL INFORMATION section for information and cautions.

Livewire (100 Base-T) Connector

This connector is for connection to another Livewire node, or an approved Ethernet switch. It has two integral LEDs. The green "Link" LED indicates the presence of a live signal (same as the front panel "Link" LED). The "Activity" LED indicates that Ethernet packets are being sent or received over the link.

IMPORTANT NOTE: Axia nodes are intended for use with an Ethernet Switch that supports multicast and QOS (Quality of Service). On a non-switched Ethernet hub, or a switch that is not enabled for multicast, this will result in network congestion that could disrupt other network activity. Should you wish to connect a node to a non-Livewire network for access to the web configuration interface, etc, you must first confirm that streaming is disabled as described in Section 2 page 6 "Programming the Node's Streaming Mode".

Input Connectors

Each pair of audio inputs share a 8-position / 8-pin miniature modular jack (e.g. RJ45 style). The connector pin functions are the same for both the AES and Analog nodes and are as follows:
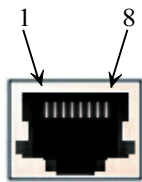
INPUT CONNECTORS	
Pin	**Function: Analog/AES**
1	**Left Channel Input + /AES +**
2	**Left Channel Input - /AES -**
3	**Right Channel Input +**
4	**Not Connected**
5	**Not Connected**
6	**Right Channel Input -**
7	**Not Connected**
8	**Not Connected**

AES Input Characteristics AES3/EBU Compliant

IMPORTANT NOTE: Axia recommends using balanced audio connections for analog audio connections. If unbalanced sources are to be connected to these inputs, we strongly recommend using a balanced to unbalanced buffer amplifier (or transformer) at the source device. Such devices are readily available, for example the Studio Hub "Match Jack – Output".

Analog Input Characteristics

Level: +4 dBu nom (+24 dBu clip point)

Impedance : >/= 10 K_

For additional technical information please see the Specifications section.

Output Connectors

Each pair of audio outputs share a 8-position / 8 pin miniature modular jack (e.g. RJ45 style). The connector pin functions are the same for both the AES and Analog nodes (see above).

IMPORTANT NOTE: Axia recommends using balanced audio connections for analog audio connections. If unbalanced devices are to be connected to these outputs, we strongly recommend using a balanced to unbalanced buffer amplifier (or transformer) at the destination device. Such devices are readily available, for example the Studio Hub "Match Jack – Input".

Analog Output Characteristics

- Level – +4 dBu (+24 dBu clip point)
- Impedance – < 50 _

AES Input Characteristics

- Balanced 110 _ (XLR)
- AES3/EBU Compliant

Chapter Two:

Operation via the 8x8 node's front panel

Here we cover everything you need to know to create a simple Livewire snake using only the unit's front panel.

Configuration & Testing

Basic configuration of the Livewire node can be performed from the front panel. In addition, the system can be configured over the network by using a web browser accessing the node's built in web server, see section 3 Advanced Programming.

Powering up

When the Livewire node is powered on, you should observe the following: The 4 LED indicators should illuminate briefly, and the LED meter display will display a screen test. Next, you should see the node's name displayed (default name is "LiveIO"). After a brief period of time, you will see the normal front panel bargraph meter display (see section 1, above).

Restoring Defaults

To restore the Livewire node to the default settings follow the steps below:

1. Power the node OFF.
2. Depress and hold the SELECT and ID buttons.
3. Power ON the unit while continuing to hold the above buttons.
4. After about 8 seconds will see the word "RESET 3 S" displayed. If you release the buttons within 3 seconds no changes will occur. If you continue to hold the buttons after 3 seconds the default settings will be set and "REBOOT" will be displayed. At this time release the SELECT and ID buttons.

Bench Testing

Two Livewire nodes may be connected together in "Point to point" mode (e.g. Ethernet snake mode) to verify operation of the units. When connected in this way the audio fed to "input 1" of "node A" will be output on "output 1" of "node B" whereas the audio on "input 1" of "node B" will be output on "output 1" of "node A". Likewise, the other inputs for the two nodes will be mapped correspondingly. To connect two units in "point to point" fashion follows these steps:

1. Restore default settings on both of the Livewire nodes to be connected. See "Restore Defaults", above.
2. Enable streaming on each unit as follows: Press the <SELECT> button repeatedly until MODE is displayed. Press and hold the <ID> button until a cursor appears under the current setting (e.g. Off). Press <SELECT> repeatedly until SNAKE is displayed. Press <ID> to confirm your entry.
3. Connect the two units using a "Crossover 10/100 Base-T" Category 6 cable 100 meters maximum (see Introduction to Livewire; System Design Reference & Primer).
4. The LINK and LIVEWIRE LEDs should illuminate on both nodes. The MASTER LED should illuminate on one unit and the SYNC LED should illuminate on the other unit.
5. The eight receive (output) meters on each of the two units should show the lowest segment illuminated to indicate streams are being received.
6. Audio may now be fed into each input and will be recieved on the corresponding output of the other unit.

Basic Programming - via the front panel

Basic programming of the Livewire node can be accomplished using the front panel display and the SELECT and ID buttons. The node's name and other information can also be checked from the front panel.

Programming the unit's IP address

Each Livewire node must have a unique IP address. The only exception is when two nodes are connected in the point-to-point (snake) configuration.

To program the node's IP address follow these steps:

1. Starting from the metering screen, press the <SELECT> button once. The default IP address is "0.0.0.0", so unless the unit has previously been programmed, the screen will show "000.000.000.000".

2. Press and hold the <ID> button for 4 seconds. A blinking cursor will appear below the first digit. Use <SELECT> to change the digit indicated by the cursor (each press of this number will increment the displayed digit by one).

3. Press the <ID> button to jump to the next digit. Use <SELECT> to change the digit indicated by the cursor. Continue until all digits of the IP address have been entered.

4. Once the changes are complete, press the <ID> button repeatedly until no cursor is shown then press <SELECT> to exit.

5. If you do not wish to save your changes do not press <SELECT> after reaching the last digit. After approximately 10 seconds the display will return to the meter screen and the previous settings will be restored.

Checking the Livewire Node Name

With the meter screen displayed, press SELECT twice. The name of this Livewire node will be displayed. The default is LiveIO. To change the name of a node you must use the Browser interface, see section 3 Advanced Programming.

Checking the Livewire Node Software Version

With the meter screen displayed, press SELECT three times. The software version will be will be displayed.

Programming the Node's Streaming Mode

The streaming mode can be selected from the front panel. The default mode is to have streaming disabled (e.g. OFF), thereby allowing safe connection to a computer or LAN for programming. See the Introduction to Livewire; System Design Reference & Primer manual for more on streaming. The possible settings for streaming mode are shown here:

- OFF – This disables both streaming of Livewire audio and Livewire clock packets. This setting is useful when you wish to connect the unit to a non-Livewire-LAN or directly to a computer to configure the IP address and other settings.
- FAST – This enables Fast Stream generation and also permits clock packet generation. This is the typical setting for most Livewire audio network applications.
- SLOW – This enables Slow Stream generation as well as Livewire clock packets. This is the usually
- setting for a Axia node used in conjunction with a computer or other device where delay is unimportant.
- SNAKE – The mode is appropriate for when two Axia nodes are directly connected with a null Ethernet cable. This enables Livestreams and Livewire clock packets only. SNAKE mode differs from fast by disallowing local loopback capability in the node.
- CUSTOM – This indicates that the Streaming Mode parameters have been configured using the unit's web page interface.

To change the Streaming Mode from the front panel follow these steps:

1. Press the <SELECT> button repeatedly until MODE is displayed.
2. Press and hold the <ID> button until a cursor appears under the current setting.
3. Press <SELECT> repeatedly until SNAKE is displayed.
4. Press <ID> to confirm your entry.

Programming the unit's Transmit Base Channel (TxBCH)

The Transmit Base Channel is the number assigned to the first of the eight streams to be transmitted by this unit. If the Transmit Base Channel is set to 00101 then the system's eight streams would be on channel numbers 101 through 108. Non-contiguous numbering is possible, however in that case they must be assigned using the node's browser user interface, see section 3 Advanced Programming.

If you have read the Introduction to Livewire; System Design Reference & Primer manual, you will know that no each stream must have a unique channel number, so don't forget that now.

To program the node's Transmit Base Channel follow these steps:

1. Starting from the metering screen, press the <SELECT> button 5 times. The default TxBCH is "00001", so unless the unit has previously been programmed, the display will show that entry.
2. Press and hold the <ID> button for 4 seconds. A blinking cursor will appear below the first digit. Use <SELECT> to change the digit indicated by the cursor (each press of this number will increment the displayed digit by one).
3. Press the <ID> button to jump to the next digit. Use <SELECT> to change the digit indicated by the cursor. Continue until all digits of the TxBCH have been entered.
4. Once the changes are complete, press the <ID> button repeatedly until no cursor is shown then press <SELECT> to exit.
5. If you do not wish to save your changes do not press <SELECT> after reaching the last digit. After approximately 10 seconds the display will return to the meter screen and the old settings will be restored.

Programming the unit's Receive Base Channel (RxBCH)

The Receive Base Channel is the number assigned to the first of the eight streams to be received by this node and output from its audio connectors. If the Receive Base Channel is set to 00201 then the system will search the Livewire network for eight streams on channel numbers 00201 through 00208. If a designated stream(s) is(are) present the node will indicate this by illuminating the lowest LED segment on the corresponding output meter. Of course in many scenarios you will want to use non-contiguous numbering for receive channels; these must be assigned using the node's web interface, see section 3 Advanced Programming.

To program the node's Receive Base Channel follow these steps:

1. Starting from the metering screen, press the <SELECT> button 6 times. The default RxBCH is "00001", so unless the unit has previously been programmed, the screen will show that entry.
2. Press and hold the <ID> button for 8 seconds. A blinking cursor will appear below the first digit. Use <SELECT> to change the digit indicated by the cursor (each press of this number will increment the displayed digit by one).
3. Press the <ID> button to jump to the next digit. Use <SELECT> to change the digit indicated by the cursor. Continue until all digits of the TxBCH have been entered.
4. Once the changes are complete, press the <ID> button repeatedly until no cursor is shown then press <SELECT> to exit.
5. If you do not wish to save your changes do not press <SELECT> after reaching the last digit. After approximately 10 seconds the display will return to the meter screen and the old settings will be restored.

Chapter Three:

Advanced Programming

Using the node's built in web pages to configure advanced features is fast and easy.

Assigning an IP Address Remotely

If you have not assigned an IP address you must do so. You can do this from the front panel as described in section 2, or the node's IP address can be remotely assigned over the network using a program included with the your node called BootP. To do so follow these steps:

1. Start bootps.exe program on any Windows 2000/XP

2. Hit ID button on GPIO front panel. You will be prompted for new IP address entry:

3. Enter new IP address and press ENTER:

IMPORTANT NOTE: Axia nodes are intended for use with an Ethernet Switch that supports multicast and QOS (Quality of Service). On a non-switched Ethernet hub, or a switch that is not enabled for multicast, this will result in network congestion that could disrupt other network activity. Should you wish to connect a node to a non-Livewire network for access to the web configuration interface, etc, you must first confirm that streaming is disabled as described in Section 2 page 6 "Programming the Node's Streaming Mode".

Make note of the IP address you have entered, so that you can access the Node using a Web browser, see section 3. You can now continue to assign additional Node IP addresses, or shut down the Bootp program.

Accessing the Node's Web Pages

IMPORTANT NOTE: Axia nodes are intended for use with an Ethernet Switch that supports multicast and QOS (Quality of Service). On a non-switched Ethernet hub, or a switch that is not enabled for multicast, this will result in network congestion that could disrupt other network activity. Should you wish to connect a node to a non-Livewire network for access to the web configuration interface, etc, you must first confirm that streaming is disabled as described in Section 2 page 6 "Programming the Node's Streaming Mode".

A number of parameters may be configured using the Livewire nodes built in web pages. To access the built in web server from a computer, the computer and node must be connected to the same LAN (or the computer and node can be connected using a "crossover 10/100 Base-T" Ethernet cable). To connect enter the following in your browser:

http://123.456.789.101

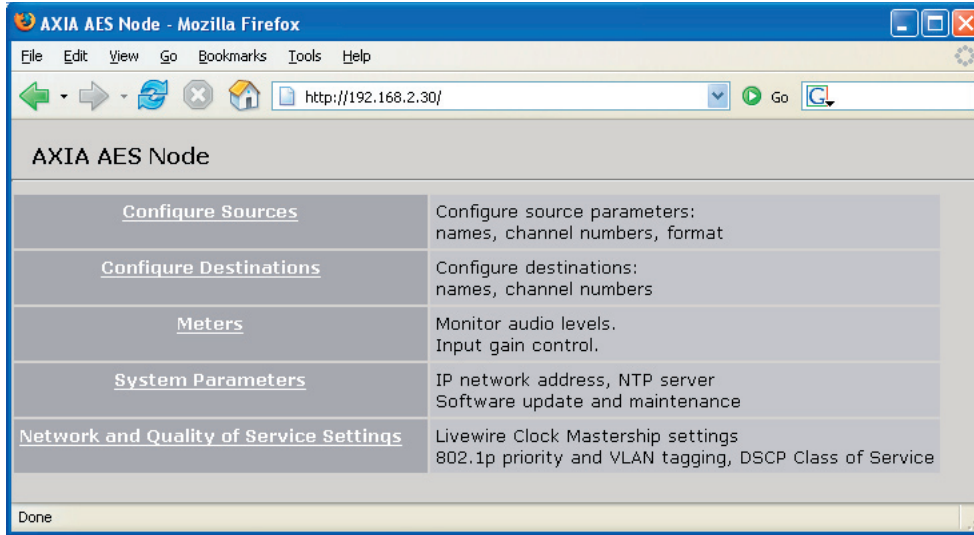where "123.456.789.101 is the IP address if the node to be configured.

Your browser should now display the login window to allow you to access the node:

Enter a valid user name and password and click on "OK" to log in.
- The default user name for all Axia nodes is: "user"
- The default password for all Axia nodes is: <enter>

NOTES:
- The IP range (e.g. the first three numbers of the four numbers of the IP address of the computer and the node must match, or additional configuration and router will be required.
- Microsoft Internet Explorer version 5 and later have been tested with the Livewire 8x8 Nodes. Other browsers may work, however they have not been tested.
- Your browser must have Java enabled and must allow "pop up" windows.
- Once you have logged in you will see the Axia 8x8 node home page as shown here:

The Sources screen for the 8x8 Analog and 8x8 AES nodes are slightly different. The Analog 8x8 Node Sources screen is shown:

Source Name and Channel

As described in the Introduction to Livewire; System Design Reference & Primer manual, each Livewire stream must be assigned a unique channel number. The channel number must be a number between 1 and 32767. Optionally a text name can be entered.

Livewire names may contain any printable character and can be up to 24 characters long (when entering names excess characters will be truncated to 24 characters). Note however, that the displays on some Livewire nodes can display only 10 or 16 characters. In this case the leftmost characters will be displayed, so keep this in mind.
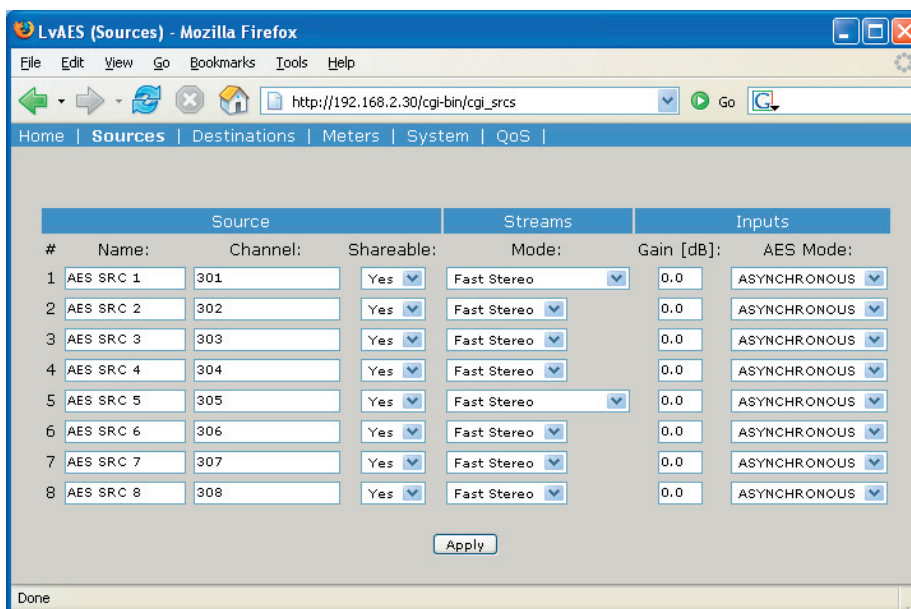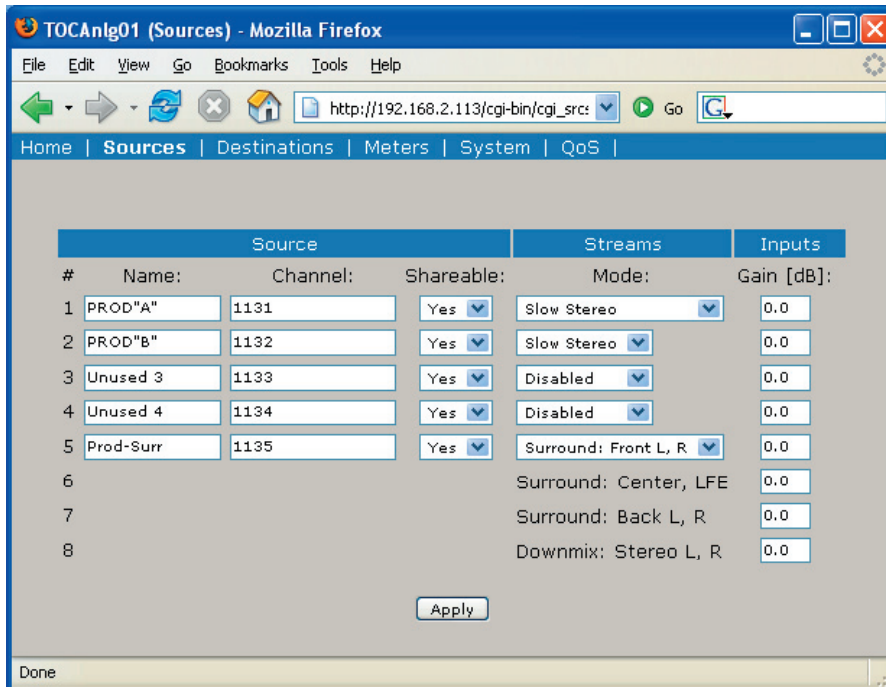
The 8x8 Node Home Page

The home page simply acts as a homepage, to allow access to each of the configuration pages. We describe each of those pages in detail.

The Configure Sources (local inputs) page

This is where you configure the local inputs to this node, and assign Livewire channels and parameters to each. Once configuration is complete (or at any time in the configuration process) click on Apply to save your changes to the node.

You will want to develop a logical naming plan for your facility. For example you may wish to include the studio or rack name as part of your names to make life simpler when identifying sources in the future. We give some examples in the Introduction to Livewire manual.

Choices:
• Disabled – Audio source is disabled. It is not advertised. No stream is generated.
• Fast Stereo – Enables low delay stream for this source (used for Microphones and other live sources).
• Slow Stereo – Enables

TOCAnlg01 (Sources) - Mozilla Firefox

File Edit View Go Bookmarks Tools Help

http://192.168.2.113/cgi-bin/cgi_src: Go

Home | **Sources** | Destinations | Meters | System | QoS |

	Source			Streams	Inputs
#	Name:	Channel:	Shareable:	Mode:	Gain [dB]:
1	PROD"A"	1131	Yes	Slow Stereo	0.0
2	PROD"B"	1132	Yes	Slow Stereo	0.0
3	Unused 3	1133	Yes	Disabled	0.0
4	Unused 4	1134	Yes	Disabled	0.0
5	Prod-Surr	1135	Yes	Surround: Front L, R	0.0
6				Surround: Center, LFE	0.0
7				Surround: Back L, R	0.0
8				Downmix: Stereo L, R	0.0

Apply

Done

els at nominal levels of +4 dBu with a clip point of 24 dBu (e.g. 20 dB headroom). When feeding the node's inputs from a "+4 nominal" device that clips at some lower level (for example +18 dBu, e.g. 14 dB headroom), you can increase the gain (by 6 dB in our example). to match this device's clip point to the node's 0 dBfs point.

This adjustment can also be used in cases where a low-level signal source must be used. Again you should use the rated clip point of the source device to determine the closest setting. Simply add gain to bring this rated clip point up to +24 dBu.

streams, which can be used for all common sources, like CD players.

Tip: Slow streams conserve network bandwidth and are a better choice for delivering audio to computers for recording and playback.

• • Surround – Enables "5.1+Stereo" surround mode. This choice is only available on ports 1 and 5. Selecting surround mode creates a bundle of 4 ports: 1-4 or 5-8.

Gain (dB)

This allows you to change the input gain in the digital domain. Care must be taken to ensure peak levels do not exceed 0 dBfs to prevent clipping. Up to 18 dB of Gain may be selected in steps of 1 dB.

The analog inputs clip point remains at +24 dBu so this level must not be exceeded. The rare device with a clip point in excess of +24 dBu will require an external pad.

Analog nodes – The Gain setting may be use to adjust for differing peak output levels between different "+4 nominal" equipment, or it can be used to accommodate analog sources that are below +4 nominal levels.

• The default setting of 0dB accommodates input lev-

AES nodes – in the case of AES nodes, this adjustment can be used to adjust system headroom. This should be done with care and deliberation.

While we don't recommend setting levels "by eye," if you choose to do so, you can view the Source levels and adjust the Gain setting from the Meters page, see below.

AES Mode

This option is only available on the Sources page for the AES node. There are two possible options for the setting:

• Asynchronous – this is the usual setting and enables sample rate conversion. Any valid AES source can be used in this mode without concerns about dropouts due to mismatched clocks.

• Synchronous – this setting can be used if the device transmitting the AES signal is synchronised to the Livewire network. For example, if the device synchronised its outputs to its input, and the input were fed we from an Axia AES node. Or, if the device had at Sync input fed from a Axia AES node. Enabling

the Synchronous mode turns off sample rate conversion thereby reducing the delay at the input by approximately 3 msec. Primarily useful for use with digital microphones or for purist applications.

The Configure Destinations (Local Outputs) Page

The page permits entering information related to this node's local outputs. For example these can be named, and for each you can assign a Livewire stream from the network to be output.

The Sources screen for the 8x8 Analog and 8x8 AES nodes are slightly different. The Analog 8x8 Node Destinations screen is shown below:

Destination Name

This is the name used to identify this destination (local output) within the Livewire network. Destination names are primarily used by Router software such as Axia Pathfinder PC.

Destination Channel

These are Livewire channels to be output from each local output. If the channel to be output is not yet available on the network, you can enter the channel number here. In the usual case you can click on the choose channel button to the right of this field, and the following Select Source screen will be displayed, see below.

You can now click on the name or channel number of the desired source to assign it to this Destination (local output). If a Livestream (e.g. a low delay stream) is available this will be used, otherwise a Standard Stream will be used.
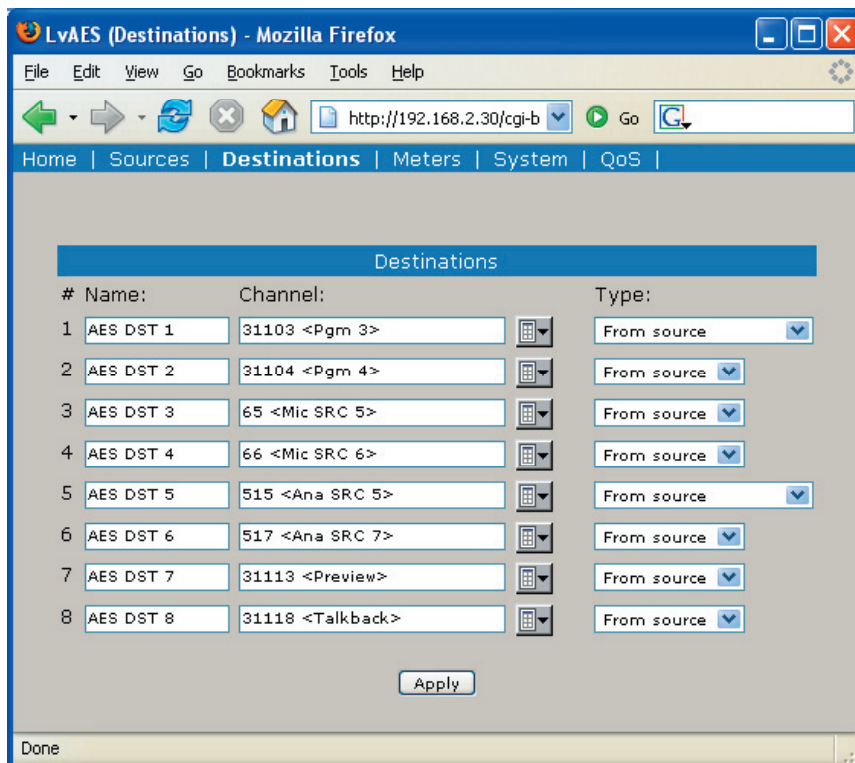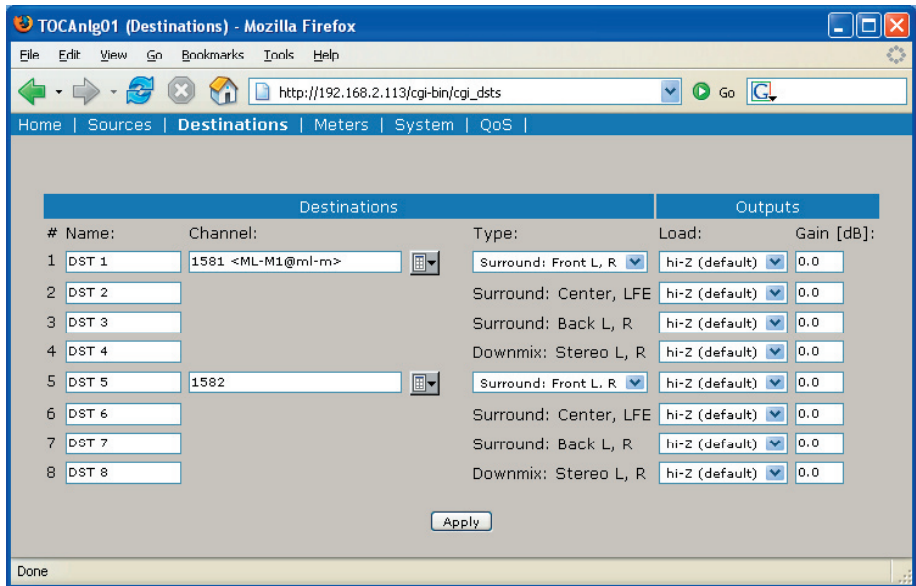
Destination Type

There are four choices for this setting. Usually you will use From Source (Livestream) for delay sensitive audio (such as microphones or headphone feeds) or From Source (Standard Stream) for non-delay sensitive cases.

You will recall from the Introduction to Livewire; System Design Reference & Primer manual that Livewire permits special bi-directional streams for use with cases where a source and destination are associated, such as a codec or hybrid. The return feed to such devices is usually a mix-minus (clear feed) generated by a mixer or mix engine fed back to the device that is the primary audio source (and usually the name of the stream in question). In effect you have thus created a bi-directional Livewire channel with a single channel number.
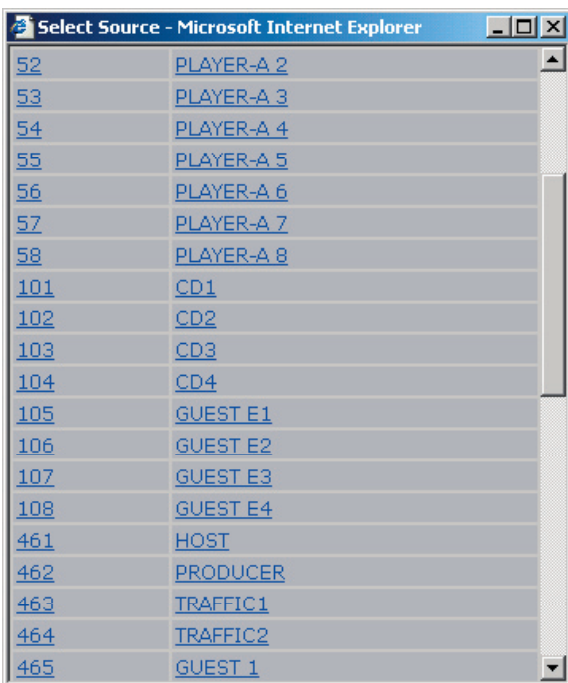
What does this all mean in practice? If the destination is a codec or hybrid you'll set the Destination Type to To Source and use the same Channel number as the stream representing the Codec or Hybrids output (the caller or far end codec audio).

Ohm equipment to ensure unity gain. The clip point remains at 24 dB.

Listen Links

For each destination stream where a Standard stream is available, the Destinations page will show a clickable link "Listen". Clicking on this link will launch the local PC application currently registered as the default players for ".sdp" files. For example, the Real Audio player.

NOTE: while the Apple QuickTime Player supports .sdp files, it does not support multicast streams and will not work.

The Meters Page

The Meters screen, shown on the below, is simply a metering screen that shows the audio level of all local sources (local inputs) and destinations (local outputs) for the node. The screen is divided into two sections, with inputs on the left and outputs on the right. Each section has 8 pairs of meters, with a left and right meter for each input or output. Note that the levels shown are in the digital domain, and are therefore calibrated in dBfs. Therefore it is worth noting that the color-coding of the meters is somewhat arbitrary; the meters turn red 9 dB before the clip point (e.g. 9 dB below digital full scale) but this does not represent an overload condition.

Meters Inputs

While we recommend setting the gain setting for inputs based on the peak output clip point of the source equipment (see Source Page, above), you can use the meter screen to tweak input Gain settings "by eye" if desired.
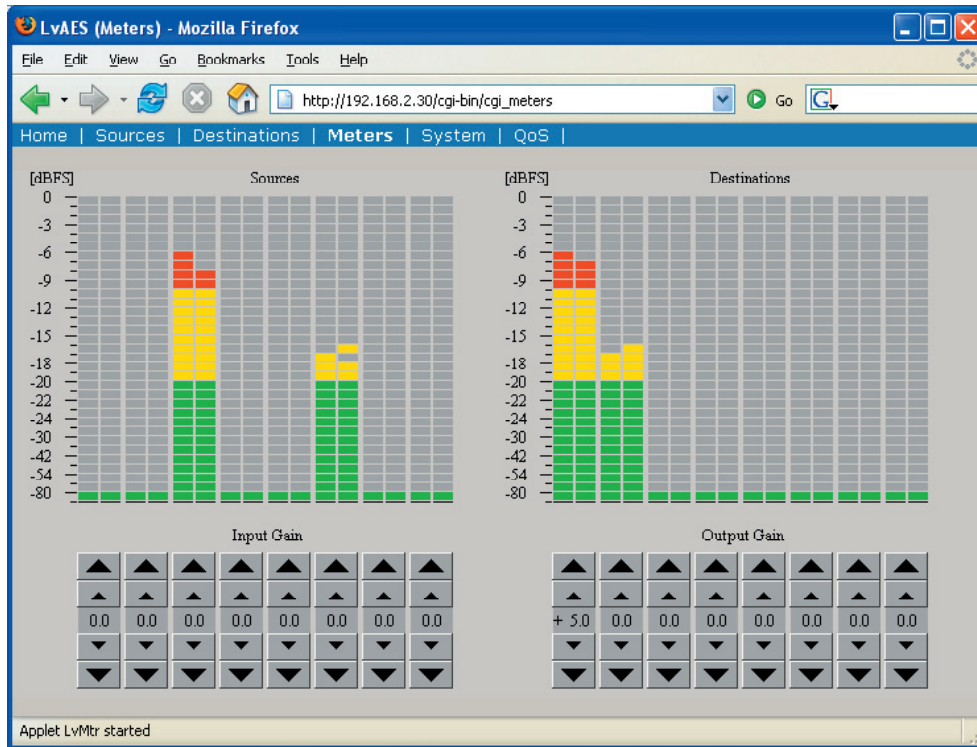
Meters Outputs

These eight pairs of meters represent the Livewire streams being output from this node. These are primar-

Output Load

This option is only available on the 8x8 Analog node's Destinations screen. This setting has two options. The usual selection is Hi-Z and is used when the node's outputs are fed to High impedance destination devices. When the node is feeding 600 Ohm inputs the 600 ohm option should be selected. This boosts the node's output level by ~1 dB to maintain true +4 levels into 600

IP address. For more detail on these see Introduction to Livewire.

Host name

An alphanumeric name for this node. This is used to identify the node on the network. You may wish to include the location of the node (studio and rack) in the name for future reference.

Network address (IP Address)

The IP address of the local unit. Each Livewire node must have a unique IP address. The only exception is when two nodes are connected in the point-to-point (snake) configuration. Normally this would be set using the front panel or using the BootP program, but it can be checked or changed here, if needed.

NOTE: If you change this entry you will lose your browser connection when you click Apply, and will need to reconnect using the new IP address.
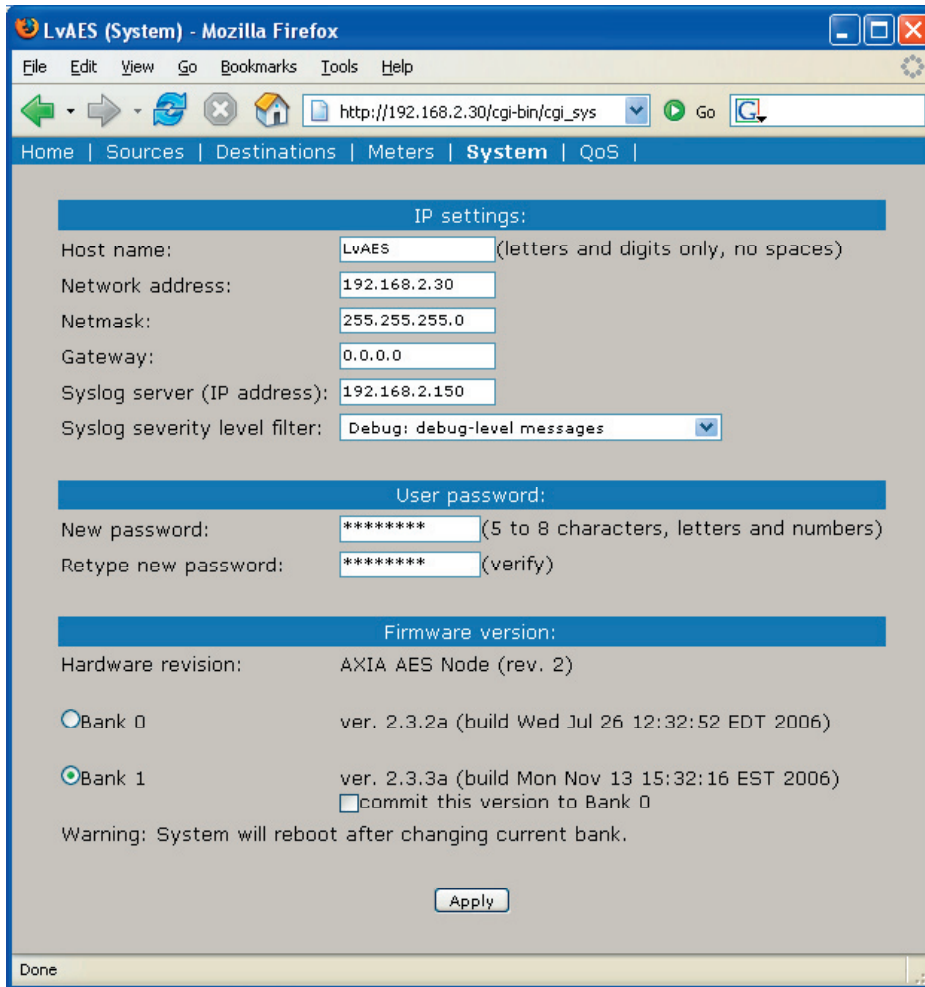

Netmask (Subnet mask)

This is the IP subnet mask of the local unit. Normally this would be set using the front panel or using the BootP program, but it can be changed here, if needed. The typical setting in most cases is 255.255.255.0 .

NOTE: If you change this entry you will lose your browser connection when you click Apply, and will need to reconnect using the new IP address.

Gateway (Router)

This is the IP address of the IP Router connecting the local IP network with some other IP network. This is needed if communication with other IP equipment that do not share the same IP address range is required.

ily for confidence monitoring. As with the front panel, the lower-most segment indicates that the designated Livewire stream is present, even if no audio is currently playing.

The System Parameters Page

The System Parameters page, shown on the next page, allows configuring the node's IP address and related settings. It also permits choosing between a primary and secondary bank of software and to download new software into the secondary bank. The currently running software version is displayed here as well. You must click the Apply button for changes to take place.

IP Settings

These are the usual IP-related settings (see Introduction to Livewire; System Design Reference & Primer for an overview and some good references to additional information). Your network administrator should be able to provide the needed values. Each unit must have a unique

- Informational: informational messages
- Debug: debug-level message
- Only messages with a severity higher than that specified by the filter will be forwarded to the remote logger.

User password

This is the password required to connect to the unit. It must be at least 5 characters long and may be as long as 8 characters. Only alphanumeric characters are permitted. To change the password you must enter the new and old passwords and then click Apply. NOTE: If you changed the IP or Firmware settings the unit will reboot. If you have only entered a new password the unit will not reboot.

When logging into the node any of the following "user names" may be used: user, USER, axia, Axia, AXIA. The password is the same for any of the above users.

IMPORTANT! If the unit was upgraded from an earlier version, only the user name "user" will be active unless the Restore Defaults process described in Section 2 has been performed
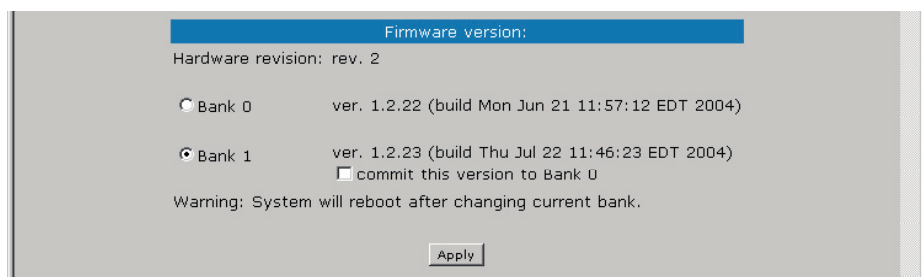
Syslog Server (IP address)

Various services generate syslog (RFC 3164) messages, which can be forwarded to a remote daemon. The remote syslog daemon IP address can be entered on the System WEB page.

Syslog severity level filter

You can customize logging by choosing log detail level:

- Emergency: system is unusable
- Alert: action must be taken immediately
- Critical: critical condition
- Error: error conditions
- Warning: warning conditions
- Notice: normal but significant condition

Firmware version

An Axia node has two internal memory "banks". Each bank contains room for a complete version of operating software. This approach allows a software update download to be completed and checked without danger of making the unit inoperable if the download were to be incomplete or corrupted. It also provides and easy way to try a new software version and still return to the old version.

The software version in each bank is displayed here. To change banks simply click in the "radio button" for the desired bank and then click on Apply.

IMPORTANT! The node will reboot after you click Apply if you change the software version. This will result in loss of audio locally, and at any unit using the local sources.

Saving Bank 1 Software

Software is always downloaded to bank 1 (the secondary bank). Downloading software (see below) will overwrite the software currently in this bank, if any. If you wish to save the software currently residing in bank 1, you can save it by moving it to bank 0 as follows:

1. Click on commit this version to Bank 0 box (see above).

2. Click on Apply. The node will now reboot.

IMPORTANT! The node will reboot after you click Apply if you change the software version. This will result in loss of audio locally, and at any unit using the local sources. In addition you will reset your browser connection and will need to reconnect.

Downloading new software

A new software version can be downloaded into bank 1 as follows:

1. Go to the Axia web site www.axiaaudio.com/downloads/ and download the desired software update for your node to your computer (this should be the computer that you will use to access the node's web page). Your local computer operating system should

display a prompt to permit you to choose where you wish to locate the downloaded file. You can choose any convenient location, just be sure to note the drive and location where the file is to be saved.

2. Open a web browser and connect to the node to be updated. Enter the complete path and file name for the software file (e.g. the file downloaded from the Axia site), or click on the Browse button to locate the file. Once the proper path and filename are displayed, click on Apply to download the file.

3. A successful download will be indicated by the new version being displayed in the Bank 1 field. If the download is unsuccessful the field for Bank 1 would be blank.

4. To run the new software click on Bank 1 and then click on Apply to reboot the node.

IMPORTANT! The node will reboot after you click Apply when changing between software versions. This will result in loss of audio locally, and at any unit using the local sources. In addition you will reset your browser connection and will need to reconnect.

The Network & QOS Page

This screen is slightly different for the Analog vs. AES nodes. The Analog 8x8 node screen is pictured below:

The settings on this screen are advanced settings, and generally the default settings should be used. For more on these settings see Introduction to Livewire; System Design Reference & Primer. Specific application notes or Axia support may direct you to change some of these items in which case use the settings directed.

One setting that is commonly adjusted is *Livewire*

Clock Master as described below.

Livewire Clock Master

Livewire's clocking system (see Introduction to Livewire for details) is automatic and largely transparent to end users. By default the Axia hardware node with the lowest Ethernet MAC address will be the clock "master". The system will automatically and transparently switch to a new unit as clock master if need be. We do however permit you to force clock mastership to a particular node (not recommended) or set certain nodes as preferred for clock mastership while maintaining automatic operation. For example you may prefer to have nodes that are on backup power be preferred clock masters. Note that in the automatic modes clock mastership is determined only when the current master becomes unavailable (adding a new node will not change clock mastership regardless of the new node's setting). The only exception is the 7 (Always Master) setting. You have the following choices for this setting:

- 0 (always slave) – This unit will never be used as clock master.
- 3 (default) – The usual setting.
- 4 (Secondary Master) – Nodes with this setting will be used as clock masters before those set to 3.
- 5 (Primary Master) – Nodes set to this setting will be used as clock masters before those set to 4.
- 7 (Always Master) – This forces a particular node to be clock master, even if another node is currently clock master. If this mode becomes available then the usual prioritization is used.

IMPORTANT! Only a single node on a Livewire network should ever be set to 7 (Always Master). For this reason we do not recommend using that selection.

Livewire Clock Mode
Provided for compatibility with older revisions:
- IP low rate (default) – recommended setting
- Ethernet – compatible with 1.x firmware
- IP High rate – compatible with 2.1.x master

Receive Buffer Size

Determines the amount of buffering in the receiver. Buffering is needed to compensate for jitter in network packet delivery. Usually the biggest source of the jitter is the source PC. Real-time performance varies a lot from one system to another; some computers can provide very low timing irregularities and allow the receive buffer to be reduced to achieve lower audio delay. Default setting is 100 ms.

801.1p tagging, 802.1p VLAN ID, 802.1q Priority, & DSCP Class of Service

802.1p tagging is necessary within the Livewire network to mark high-priority audio packets . This information is used by the Ethernet switches in the packet scheduling and queuing mechanism. It provides low-jitter packet forwarding of Livewire clock and low-latency audio streams.

On the other hand, Standard streams don't need tagging, because they are not low-latency. By default, standard streams are marked with Type of Service (DSCP code points) information in the IP header which can be used by L3 switches to provide better service to our audio streams than to best effort IP traffic.

There is an option to enable L2 802.1p tagging on standard streams, and this may be used with switches which do not use the DSCP information included in the TOS field of the IP header. We do not enable this tagging by default, because it wouldn't work in cross-over Ethernet connection to PCs; most network cards do not accept 802.1p frames by default.

You should not need to change these default settings unless you are building a system which is not based on our recommendations.

The "VLAN ID" setting is related to the VLAN ID in Ethernet switches. For instance, in HP switches, VLAN starts at "0", but in the web page, "0" is the default. So, how do you dertermine the correct value?

VLAN ID=0 means that VLAN information is not

set in 802.1p tag. The Ethernet switch, upon reception of such frame, uses the port default VLAN ID. Only 802.1Q priority information is used.

"DSCP Class of Service" is a standard describing the tagging of IP frames with service information. Network equipment can be set up to provide different forwarding delay and drop precedence depending on the service information. Our defaults are compatible with most Ethernet equipment defaults for class of service Livewire requires; you should not need to change them unless instructed by Axia Support.

AES Synchronization and Clock

These setting determine two factors. The Livewire Clock Master Priority setting determines the clock mastership options as described above for the Analog Node. The AES node also permits additional synchronization options given that it can be locked to an AES source, as discussed below.

AES Sync Source & AES Master Timebase

If the AES sync input as Livewire master timebase box is checked, then this node will use the AES source fed to the input to chosen in AES sync source as the clocking source for this node if and when it becomes the Livewire master clock source. If this node becomes the Livewire clock master (see above), then the entire Livewire network will be synchronized to the AES signal fed to the selected input.

Of course Livewire clock mastership can change (as described above) so the careful user will feed house AES sync to multiple 8x8 AES nodes. In this case, those nodes would all be set to priority 4 or 5 (see above) to ensure that these nodes will be the source for the Livewire master clock whenever one of this group is available.

If AES sync source is set to Livewire 48 kHz then the system will simply use the unit's internal clock source if and when it becomes clock master.

AES Output Sync

This sets the output sample rate and synchronisation

for this node's AES outputs. It as two options:

- Livewire 48kHz.
  This would cause the AES outputs be synced directly to the Livewire system clock and no sample rate conversion will be performed. However the receiving unit would need to have sample rate conversion or to be synchronous to the Livewire system clock or dropouts due to buffer over/under run will occur.
- AES Sync in
  In this case, the Livewire stream will be sample rate converted to the clock stripped of the designated AES input. This permits operation at rates other than 48 kHz, but only if an external source at that rate is used.

Appendices:

Here are a few useful bits of information we think might prove useful. Don't forget that the manual Introduction to Livewire; System Design Reference & Primer should be your companion and has many useful tips.

Unbalanced Connections

We've told you, both earlier in this manual, and in Introduction to Livewire; System Design Reference & Primer, that Axia recommends balanced audio connections when connecting analog source and destination gear to the inputs and outputs, respectively, of Axia nodes. Not only do we recommend this for the usual reasons, but because inter-channel crosstalk between the left and right channels of unbalanced signals sharing the same Cat. 5 cable is a possibility. As we've mentioned before, we recommend converting between balanced and unbalanced at the unbalanced device and then using the standard Cat. 5 connection from there to the Axia node. There are a number of active balanced-to-unbalanced and unbalanced-to-balanced adaptors commercially available at a reasonable cost (see www.studiohub.com for a pair of units that will easily plug and play with our gear). We'll suggest one more time that this approach is the way to go, and that using unbalanced cable runs is a good way to compromise the performance of your state of the art Axia audio network. However, if you are in a bind, or otherwise determined to do so, here is how we recommend connecting Axia nodes to unbalanced equipment:

Unbalanced Destinations

To feed audio to an unbalanced destination from the 8x8 Analog node you must use a separate cable for the left and right signals, and you will need a shielded RJ-45 plug so you can terminate the shield of the audio cables. Pin 1 will feed the Left signal with the signal common (e.g. cable shield) connected to the RJ-45 shield. Pin 3 will feed the Right signal with the signal common (e.g.

cable shield) connected to the RJ-45 shield.
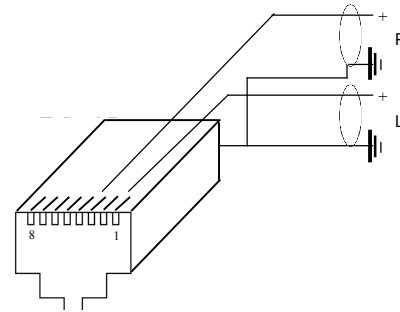
Feeding an unbalanced destination inputs from 8x8 node's analog outputs

An external pad may be required if the destination equipment's inputs cannot accept signals with peak levels of +24 dBu.

Generally the unused output pin should not be tied to the shield. Doing so will not harm the node, however doing so will activate a feature that will increase the output level by 6 dB, which is generally not desirable.
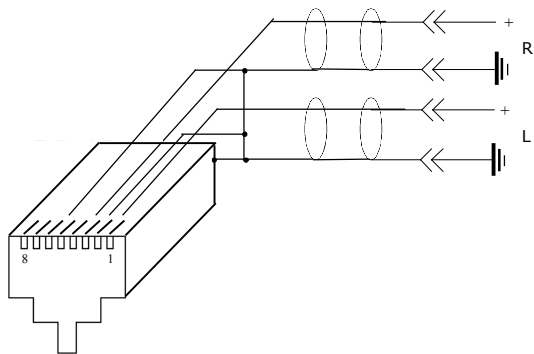
Sources

To feed an unbalanced signal from a source into the inputs of the analog 8x8 node you must use a separate cable for the left and right signals. We generally prefer the method where the unbalanced signal is presented across the differential balanced inputs of the node. The handling of the shield will depend on the equipment and grounding practices used.

If both pieces of equipment are grounded to a facility grounding system then the shield may be left open at one end (or both ends), as follows.
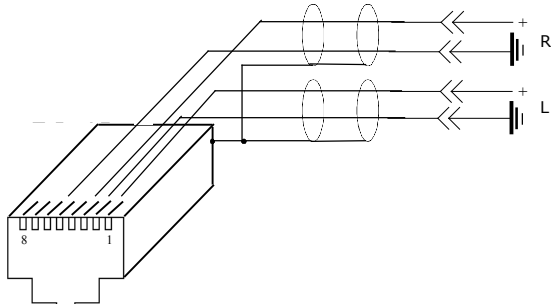
Axia node's analog inputs fed from an unbalanced source where both pieces of equipment are tied to a facility ground.

Alternatively, if both pieces of equipment are not both tied to a common facility ground, both sides of the shield must be connected. In this case the "-" side of the nodes inputs are tied to the shield of the RJ-45 plug as follows:

Axia node's inputs fed from a floating source, with no facility ground in common with the Axia node.

Using Axia Nodes with Ethernet Radios

Applies to Node Software v2.3.2a and higher

There are several changes and additions to Axia Audio Nodes software beginning with v2.3.2a designed to simplify the operation of STLs and audio snakes using Axia nodes in conjunction with Ethernet Radios.

These additions include:

STL Slave and STL Snake modes on Clock Master Priority options

IP Low Rate is now set as the default receive Clock mode in the LW Clock Mode options field. (Note that this setting only defines the RECEIVE type of stream. It does not change the clock stream type when the node is acting as the Master Clock. We recommend that you do not change this setting.)

A Standard Stream Buffering option, which is set to 100ms as default. (Note that this setting should not be adjusted unless advised otherwise by Axia Technical Support.)

A "Master/Sync" confidence tally is added to the Router Selector Node display.

Using Axia Audio Nodes as snakes or STLs

Note that the setup options described below require that Node Software v2.3.2a or higher must be installed to work correctly.

OPTION 1 - Using two nodes back to back without an Ethernet switch

In this scenario, the clock sync mode will set the Clock Rate to a Low Rate sync packet regardless of the Livewire Clock Mode setting. This enables a more stable SYNC mode, eliminating the need for an Ethernet switch between the nodes handling QOS of the clock sync signal.

Navigate to the "QOS" web pages of the Audio Nodes you'll be using. Determine which one will be the master and which the slave, and set the new "STL Snake" and "STL Slave" clock priority modes to the appropriate values.

Typically, you will set the Clock Master Priority option on the Node located in the studio to "7 (Always Master) STL Snake". The Node on the remote end of the link should be set to "0 - (Always Slave) STL".

All stream types must be set to Standard streams.

Leave Standard Stream buffering at 100ms (the default setting).

OPTION 2 - Connecting a "remote" Audio Node to an existing Axia network using Ethernet Radio

If you are using Studio Engines and/or existing nodes connected to an Ethernet switch, then these instructions assume that you have a current Livewire Network and are adding a node at a remote location connected via an IP radio. You must maintain the high rate Master clock sync packets for these devices in order for all nodes to sync properly. This is especially important for the well-being of the Axia Studio Mix Engines.

In this case, you will need to have at least ONE Audio Node on the main Livewire network designated as the MASTER CLOCK and running version 2.3.2a software. It should be set to a higher priority than all other nodes running earlier software versions. We recommend choosing "7 - (Always Master)". Do not select "7 – (Always Master) STL Snake" for this application.

The remote node at the receive end of the Ethernet radio should likewise be running v2.3.2a software. Its clock setting should be "0 - (Always Slave) STL".

Deep Tech: A node running version 2.2.0, when operating as the current Clock Master, will generate two clock streams: a High rate and a Low rate clock sync. Nodes running version 2.1.x and earlier do not have this dual clock feature and require the High rate sync to operate as well.

Streams sent to the "remote" node should all be STANDARD streams. Leave Standard Stream buffering at 100ms (the default setting) on the receiving node.

IP Radio Settings and Recommendations:

Settings on your Ethernet radios will have to be tweaked as needed. Unfortunately, due to the large number of Ethernet radios on the market, we are unable to make specific recommendations on which radio to choose, or their exact optimal settings.

Some Quality of Service options may assist or hinder the operation of the radio for multicast UPD data packets. This may involve turning ON or OFF some or all the "smarts" within the radio. User experience will differ from model to model. We suggest that you contact your radio's manufacturer for additional support on the operation of the radios in this mode. For the purposes of passing Livewire streams reliably, we desire that the IP radios behave as much as possible like a simple piece of CAT6 cable, with minimal latency.

Questions on the operation of the Axia Audio Nodes can be emailed to Axia support at support@axiaaudio. com.

Specifications and Warranty

Axia System Specifications

Microphone Preamplifiers

- Source Impedance: 150 ohms
- Input Impedance: 4 k ohms minimum, balanced
- Nominal Level Range: Adjustable, -75 dBu to -20 dBu
- Input Headroom: >20 dB above nominal input
- Output Level: +4 dBu, nominal

Analog Line Inputs

- Input Impedance: >40 k ohms, balanced
- Nominal Level Range: Selectable, +4 dBu or -10dBv
- Input Headroom: 20 dB above nominal input

Analog Line Outputs

- Output Source Impedance: <50 ohms balanced
- Output Load Impedance: 600 ohms, minimum
- Nominal Output Level: +4 dBu
- Maximum Output Level: +24 dBu

Digital Audio Inputs and Outputs

- Reference Level: +4 dBu (-20 dB FSD)
- Impedance: 110 Ohm, balanced (XLR)
- Signal Format: AES-3 (AES/EBU)
- AES-3 Input Compliance: 24-bit with selectable sample rate conversion, 32 kHz to 96kHz input sample rate capable.
- AES-3 Output Compliance: 24-bit
- Digital Reference: Internal (network timebase) or external reference 48 kHz, +/- 2 ppm
- Internal Sampling Rate: 48 kHz
- Output Sample Rate: 44.1 kHz or 48 kHz
- A/D Conversions: 24-bit, Delta-Sigma, 256x oversampling
- D/A Conversions: 24-bit, Delta-Sigma, 256x oversampling
- Latency <3 ms, mic in to monitor out, including network and processor loop

Frequency Response

- Any input to any output: +0.5 / -0.5 dB, 20 Hz to 20 kHz

Dynamic Range

- Analog Input to Analog Output: 102 dB referenced to 0 dBFS, 105 dB "A" weighted to 0 dBFS
- Analog Input to Digital Output: 105 dB referenced to 0 dBFS
- Digital Input to Analog Output: 103 dB referenced to 0 dBFS, 106 dB "A" weighted

- Digital Input to Digital Output: 138 dB

Equivalent Input Noise

- Microphone Preamp: -128 dBu, 150 ohm source, reference -50 dBu input level

Total Harmonic Distortion + Noise

- Mic Pre Input to Analog Line Output: <0.005%, 1 kHz, -38 dBu input, +18 dBu output
- Analog Input to Analog Output: <0.008%, 1 kHz, +18 dBu input, +18 dBu output
- Digital Input to Digital Output: <0.0003%, 1 kHz, -20 dBFS
- Digital Input to Analog Output: <0.005%, 1 kHz, -6 dBFS input, +18 dBu output

Crosstalk Isolation and Stereo Separation and CMRR

- Analog Line channel to channel isolation: 90 dB isolation minimum, 20 Hz to 20 kHz
- Microphone channel to channel isolation: 80 dB isolation minimum, 20 Hz to 20 kHz
- Analog Line Stereo separation: 85 dB isolation minimum, 20Hz to 20 kHz
- Analog Line Input CMRR: >60 dB, 20 Hz to 20 kHz
- Microphone Input CMRR: >55 dB, 20 Hz to 20 kHz

Power Supply AC Input

- Auto-sensing supply, 90VAC to 240VAC, 50 Hz to 60 Hz, IEC receptacle, internal fuse
- Power consumption: 35 Watts

Operating Temperatures

- -10 degree C to +50 degree C, <90% humidity, no condensation

Dimensions and Weight

- Microphone node: 1.75 inches x 17 inches x 10 inches, 6 pounds
- Analog Line node: 1.75 inches x 17 inches x 10 inches, 6 pounds
- AES/EBU node: 1.75 inches x 17 inches x 10 inches, 6 pounds
- Router Selector node: 1.75 inches x 17 inches x 10 inches, 6 pounds
- GPIO node: 1.75 inches x 17 inches x 13 inches, 8 pounds
- Studio Mix Engine 3.5 inches x 17 inches x 15 inches, 10 pounds

Axia Node Limited Warranty

This Warranty covers "the Products," which are defined as the various audio equipment, parts, software and accessories manufactured, sold and/or distributed by TLS Corp., d/b/a Axia Audio (hereinafter "Axia Audio").

With the exception of software-only items, the Products are warranted to be free from defects in material and workmanship for a period of one year from the date of receipt by the end-user. Software-only items are warranted to be free from defects in material and workmanship for a period of 90 days from the date of receipt by the end-user.

This warranty is void if the Product is subject to Acts of God, including (without limitation) lightning; improper installation or misuse, including (without limitation) the failure to use telephone and power line surge protection devices; accident; neglect or damage.

EXCEPT FOR THE ABOVE-STATED WARRANTY, AXIA AUDIO MAKES NO WARRANTIES, EXPRESS OR IMPLIED (INCLUDING IMPLIED WARRANTIES OF MERCHANTABILITY AND FITNESS FOR A PARTICULAR PURPOSE).

In no event will Axia Audio, its employees, agents or authorized dealers be liable for incidental or consequential damages, or for loss, damage, or expense directly or indirectly arising from the use of any Product or the inability to use any Product either separately or in combination with other equipment or materials, or from any other cause.

In order to invoke this Warranty, notice of a warranty claim must be received by Axia Audio within the above-stated warranty period and warranty coverage must be authorized by Axia Audio. If Axia Audio authorizes the performance of warranty service, the defective Product must be delivered, shipping prepaid, to: Axia Audio, 2101 Superior Avenue, Cleveland, Ohio 44114.

Axia Audio at its option will either repair or replace the Product and such action shall be the full extent of Axia Audio's obligation under this Warranty. After the Product is repaired or replaced, Axia Audio will return it to the party that sent the Product and Axia Audio will pay for the cost of shipping.

Axia Audio's authorized dealers are not authorized to assume for Axia Audio any additional obligations or liabilities in connection with the dealers' sale of the Products.

Axia Audio's products are to be used with registered protective interface devices which satisfy regulatory requirements in their country of use.

rev 12/08/04 v 1.0 RKT
rev 12/28/04 v 1.0b RKT
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Analog memories fade.

The future beckons!