Telos iPort High Density

iPort High Density

Telos iPort High Density is a networked multi-codec gateway that enables the transport of multiple channels of stereo audio across any QoS-enabled IP network, including T1 and T3 connections and private WANs with MPLS – perfect for large-scale distribution of audio to single or multiple locations.

iPort High Density is the workhorse of codecs, configurable with up to 64 bi-directional MPEG codecs. Optionally, you can add aptX® encoding to any codec or use Linear PCM audio. iPort HD connects to Axia® IP-Audio networks using a single ethernet cable for all I/O or can be used as a standalone multiple-stream codec.

Coding algorithms include AAC, AAC-LD, HE-AAC (plus v2), MP2, MP3, optional aptX® Enhanced, or Linear PCM. Bit rates range from 24 to 320 kbps for MPEG codecs, plus standard fixed rates for aptX, 24 bit 48k for Linear PCM. iPort High Density offers dual, parallel-path end-to-end streaming for ultra-reliability and redundancy. Up to 20 unidirectional GPIO contact closures per codec are available in several modes to allow considerable control flexibility. End-to-end GPIO is supported for each codec. An optional unique Content Dealy feature allows independent local storage and scheduled delayed playout of any coded audio channels.

> ⚠ **Important -** Content Delay requires additional installed hardware and a separate license for activation. The base model iPort High Density does NOT include this hardware.

Quick Start Guide

Getting Started

Registration

We know you're anxious to mount your new Telos iPort High Density and get it running, but please take a moment to register your product online to activate the warranty and ensure you will get quality support should you have questions regarding installation, operation, or troubleshooting. Visit http://telosalliance.com/product-registration/ to register.

Hardware Installation

The 1RU iPort High Density hardware mounts in a standard 19-inch rack with supplied rack rails. The rear of the unit must be supported. To prevent damage, if the supplied rack rails are not used, a supporting shelf is recommended. Install iPort High Density in a location where cooling fan noise is not an issue.

> ⚠ **Important -** Please see the User Warnings and Cautions page for important installation and electrical safety information.

Front Panel

The front panel gives basic status indications including power, SSD drive activity, and connectivity indicators for the NET 1 and NET 2 connections. There are no controls on the front panel. iPort High Density will power on automatically once connected to power.

Rear Panel Connections
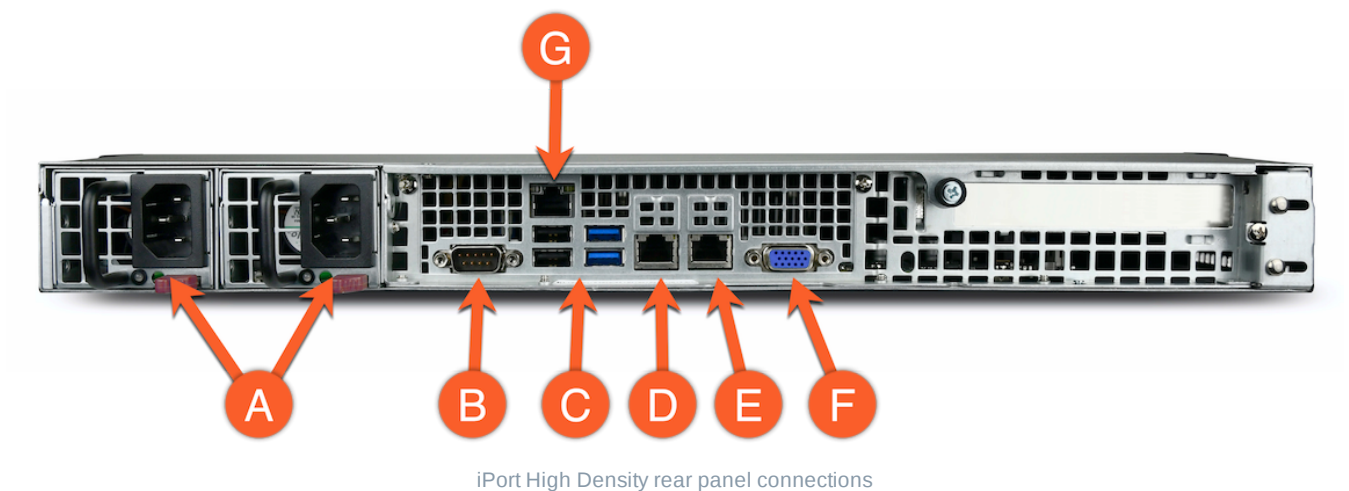
iPort High Density rear panel connections

A. Power
B. RS232 (Serial)
C. USB Ports
D. NET 1 (WAN)
E. NET 2 (Livewire - LAN)
F. VGA Connector
G. Intelligent Platform Management Interface (IPMI)

> ⓘ **Note -** USB and VGA port connections may be required for the initial IP address configuration. A

> connection is not required for normal operation or configuration.

Initial Configuration

Power

Power is supplied by dual, redundant, internal, auto-ranging power supplies, 100-264 VAC, 50/60Hz, 100W max. These supplies are auto-switching and easily field-replaceable for maximum uptime. A green LED indicates a normal status. A red LED (or no light at all) indicates a fault.

> ⓘ **Note -** Using both power supplies - ideally fed from two different sources - is recommended for redundancy, but iPort High Density can operate on a single power supply. If only one power supply is connected, release the latch and remove the second power supply to prevent the alarm from sounding.

Network

The Ethernet mode is set permanently to AUTO for both NICs, and so the switch or router to which they are connected must be as well. Problems can result if the router or switch is set to a fixed 10, 100, or 1000BT status.

CAT-5e cables may be used for the Livewire connection, though we recommend CAT-6 when possible for all 1000BT (Gigabit) connections.

> ⚠ **Important -** The NET 2 (the Livewire LAN) connection requires a 1000BT full-duplex connection.

For the initial setup, you can connect iPort High Density to your switch or temporarily connect it directly to a PC using a standard Ethernet cable (a cross-over cable is not required).

iPort High Density ships with network addresses pre-set as follows. If this fits your network scheme, you can continue with the configuration via a web browser.

WAN (NET 1)

- NET 1 has no configuration by default.

LAN (NET 2) Livewire or AoIP Network

- IP Address (static): 192.168.2.100
- Subnet Mask: 255.255.255.0

- Gateway: Not offered on LAN interface
- DNS: Not offered on LAN interface

If the pre-configured address scheme does not meet your needs, you will need to use a local monitor and keyboard and follow the instructions in the Alternate Method for IP Addressing section of the User Manual.

Connecting to the Web Interface

iPort High Density can be remotely accessed from a web browser on any PC on the network.

1. Connect the iPort High Density to your network using NET 2.

2. From a computer connected to the same network, point your browser to the iPort by entering the unit's IP address (http://192.168.2.100).

3. When prompted, enter the user name "user".

4. The default password is blank (no password).

After login, the Home screen is displayed:

iPort High Density Home screen

Completing Network Configuration

From the left side menu, choose **Network** under the **Configuration** heading. The current settings for the Livewire (LAN) and WAN interfaces are displayed.

Configuring the Livewire (LAN) Interface (1)

- Set the Network address
- Set the Netmask

Note that the Livewire interface does not require or support a Gateway or DNS.

Configuring the WAN Interface (2)

- Set Network Address A
- Set Netmask A
- Set Gateway A
- Check the "Enable Configuration Access from WAN" if desired (3)
- Click "Apply", then answer "Yes" to allow the system to restart

> (i) **Note -** Using the "B" network configuration is an advanced topic covered in detail in the User Manual.

Livewire (LAN) and WAN interface configuration

Adding Feature Licenses

iPort High Density comes with eight codecs pre-licensed. No additional steps are required if you did not purchase additional options.

To add additional licenses, please refer to the User Manual. Additional licenses may include additional codecs, aptX encoding options, or Content Delay features.

Congratulations! You have completed the initial setup. Please proceed to the User Manual for further instructions.

User Manual

Notices and Cautions

CAUTION:

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 auto-ranging line voltage input. Ensure the power voltage is within the specified range of 100-240v. 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: DOUBLE POLE/NEUTRAL FUSING

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 the fuse only with the 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 in the top 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.

WARNUNG:

Die Installations-und Serviceanleitung in diesem Handbuch ist für die Benutzung durch qualifiziertes Fachpersonal. Um Stromschläge zu vermeiden führen Sie keine andere Wartung durch als in dieser Betriebsanleitung aufgeführt, es sei denn Sie sind dafür qualifiziert. Überlassen Sie alle Reparaturarbeiten qualifiziertem Fachpersonal.

Dieses Gerät hat eine automatische Bereichseinstellung der Netzspannung.

Stellen sie sicher, dass die verwendete Netzspannung im Bereich von 100-240V liegt.

Das Symbol ~, falls verwendet, bezeichnet eine Wechselstromversorgung.

 Dieses Symbol, wo immer es auftaucht, macht Sie auf nicht isolierte, gefährliche elektrische Spannung (ausreichend um einen Stromschlag hervorzurufen) innerhalb des Gehäuses aufmerksam. Spannungen.

 Dieses Symbol, wo immer es auftaucht, weist Sie auf wichtige Bedienungs-und Wartungsanleitung hin. Lesen Sie die Bedienungsanleitung.

ACHTUNG: ZWEIPOLIGE ABSICHERUNG / NULLEITER ABSICHERUNG

Das Netzteil des Gerätes hat eine interne Sicherung eingebaut. Auch wenn die Sicherung durchgebrannt ist, können auf einigen primären Bauteilen noch gefährliche Spannungen vorhanden sein. Wenn ein Austausch der Sicherung erforderlich ist, ersetzen Sie die Sicherung nur mit gleicher Art und Wert für den kontinuierlichen Schutz gegen Feuer.

WARNUNG:

Das Gerätenetzkabel ist die Haupttrennvorrichtung. Die Steckdose sollte sich in der Nähe des Gerätes befinden und leicht zugänglich sein. Das Gerät sollte nicht so angeordnet sein, dass der Zugang zum Netzkabel beeinträchtigt ist. Wird das Gerät in ein Rack eingebaut, sollte eine leicht zugängliche Sicherheitstrennvorrichtung in den Rack-Aufbau mit einbezogen werden.

Um die Gefahr von Stromschlägen zu verringern, darf dieses Produkt nicht Regen oder Feuchtigkeit ausgesetzt werden. Dieses Gerät ist nur für die Benützung im Innenbereich. Dieses Gerät erfordert freie Luftzirkulation für eine ausreichende Kühlung. Blockieren sie nicht die Lüftungsschlitze auf der Geräteoberseite und den Seiten des Gerätes. Unzureichende Belüftung kann das Gerät beschädigen oder Brandgefahr verursachen. Platzieren Sie das Gerät nicht auf einem Teppich, Poster oder andere Materialien welche die Lüftungsöffnungen beeinträchtigen könnten.

Wird das Gerät anders als in der, vom Hersteller angegebenen Weise verwendet, kann der, durch das Gerät gegebene Schutz beeinträchtigt werden.

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.

Telos iPort High Density USER MANUAL

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NOTICE

All versions, claims of compatibility, trademarks, etc. of hardware and software products not made by Telos Alliance which are mentioned in this manual or accompanying material are informational only. Telos Alliance makes no endorsement of any particular product for any purpose, nor claims any responsibility for operation or accuracy. We reserve the right to make improvements or changes in the products described in this manual that may affect the product specifications or to revise the manual without notice.

WARRANTY

This product is covered by a two-year limited warranty, the full text of which can be viewed or downloaded from https://www.telosalliance.com/warranty-information.

UPDATES

The operation of iPort High Density is determined largely by software. We routinely release new versions to add features and fix bugs. Check the Telos Alliance website at www.TelosAlliance.com for the latest. We encourage you to sign-up for the email notification service offered on the site.

FEEDBACK

We welcome feedback on any aspect of iPort High Density, 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.

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 representatives to make arrangements for service.

WE SUPPORT YOU...

BY PHONE:

- You may reach our 24/7 Support team anytime around the clock by calling +1-216-622-0247.
- For billing questions or other non-emergency technical questions, call +1-216-241-7225 between 9:30 am to 5:00 PM, USA Eastern time, Monday through Friday.

BY E-MAIL:

- Technical support is available at support@TelosAlliance.com.
- All other questions, please email inquiry@TelosAllinace.com.

VIA WORLD WIDE WEB:

The Telos website has a variety of information that may be useful for product selection and support. The URL is www.TelosAlliance.com.

REGISTER YOUR PRODUCT

Did you know that all Telos Alliance products come with a 5-Year Warranty? Take a moment to activate your coverage online at http://telosalliance.com/product-registration/ .

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Introduction

Welcome to Telos iPort High Density

Welcome to your new Telos iPort High Density, a Multi-Codec Gateway designed on an IP backbone that is used by broadcasters worldwide, featuring built-in resilience and reliability even with challenging

connections. Worldwide networks use iPort for both Distribution and Contribution spanning multiple time

Now, the iPort legacy continues with the more powerful iPort High Density, which transports multiple channels of stereo, mono, and dual-mono audio across IP networks, including private WANs, IP-radio links, and over good quality Public Internet connections—perfect for large-scale distribution of audio to single or multiple locations.

iPort High Density is the workhorse of codecs and comes with eight bi-directional stereo codecs, configurable to run in MPEG or Linear PCM mode. You can license additional codecs up to a maximum of 64, as well as add Enhanced aptX™ encoding. iPort High Density connects to your existing Livewire® Network using a single ethernet cable (CAT-6 recommended) for all I/O. It can also pair with Telos Alliance xNodes via an adequately configured ethernet switch for use as a standalone multi-stream codec.

Coding algorithms include AAC, AAC-LD, HE-AAC (plus v2), MP2, MP3, linear PCM, and optional Enhanced aptX™. 24 to 320 kbps for MPEG codecs, standard fixed rates for Enhanced aptX™, and 24-bit PCM at 48kHz, mono, or stereo are supported. Dual, diverse-path, end-to-end connections are available for ultra-reliability and redundancy. Built-in streaming servers use SHOUTcast/ICEcast formatting at the output.

iPort High Density offers MPEG-standard ancillary data transport, up to three transparent control and metadata channels per codec and direction, and enhanced GPIO options with up to 20 end-to-end GPIO channels per codec and direction, all bundled and synchronized with the respective audio content. An optional Content Delay feature allows independent local storage and scheduled delayed playout of any or all coded audio channels.

Configuration

The first step is to configure the network parameters. Click on the Network link under the Configuration heading to open that page.

Network Configuration

Host Name

This is the Linux host name that is used for device identification, for example, when system logs are being sent to an external machine. These should be unique for each device in your network. If you have more than one iPort, you will want to change this from the default 'iPort' text., as we have done here. If this will be your only iPort, you can keep the default.

Livewire interface

This is the value for the Livewire Ethernet port. Normally, this is set to a value in the same address space as the Livewire network to which the iPort PLUS will connect. Note that the netmask value needs to correspond with that of your other Livewire AoIP equipment.

WAN interface

With iPort High Density, you may configure two IP addresses (A and B) on the same WAN port or each of the WAN and LAN ports for enhanced redundancy.

The WAN port may be set to any needed value for compatibility with the network to which it's connected. Here, you must enter a correct gateway IP address for the iPort to work correctly. The gateway address will typically be the router's IP address to access the Internet or other WAN.

You may use redundant WAN Paths by enabling the redundant path options, specifying its IP address, netmask, and gateway, and selecting the desired ethernet port.

By default, Web access via the WAN port is disabled. Click the box if you want to enable it.
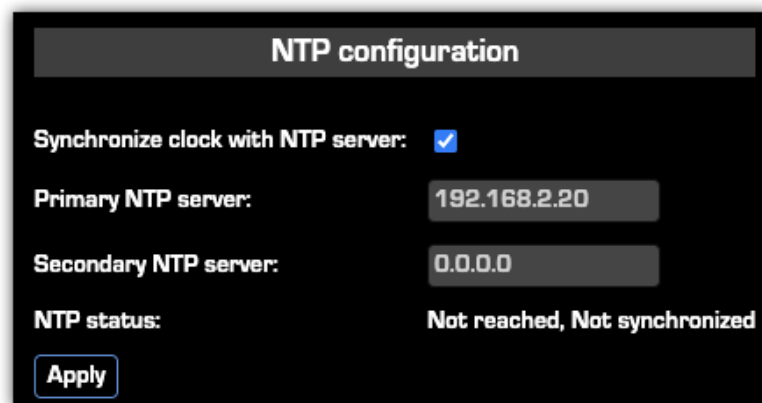
When you are finished making entries, remember to click the **Apply** button at the bottom of the page.

Setting the Time

iPort High Density maintains a very accurate clock by connecting to an NTP time server. Once the network connections are established, you can set them up.

Server Configuration

1. From the main menu, click on the AoIP and Options link under the Configuration heading.
2. Enter the IP address for the Primary and Secondary NTP servers
3. Click the Synchronize Clock with NTP Server box

After a few minutes, you should see the NTP status change to 'Reached, SYNC'.

Time Zone Configuration

On the AoIP and Options page, Under Time zone configuration, set your GMT offset and click Apply.
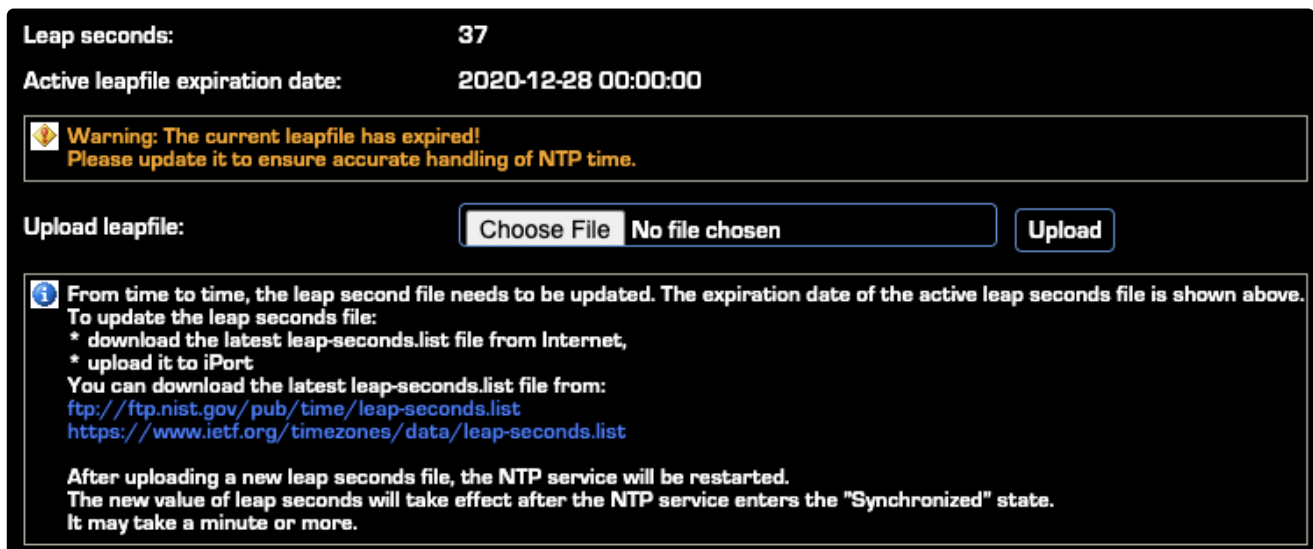
Adjusting Leap Seconds

From time to time, the leap seconds file needs to be updated. This is particularly important if your iPort uses the optional Content Delay feature where accurate playback time is needed.

 A leap second is a one-second adjustment that is occasionally applied to Coordinated Universal Time (UTC), to accommodate the difference between precise time (as measured by atomic clocks) and imprecise observed solar time. For more information on leap time, visit https://en.wikipedia.org/wiki/Leap_second.

To set leap seconds, upload a leap file to the NTP Configuration page. To obtain a leapfile, follow the instructions and links on the NTP Configuration webpage.

For web addresses and status information about time servers maintained in the United States by NIST, visit <http://tf.nist.gov/tf-cgi/servers.cgi>.

To find out more about NTP time servers globally, visit <http://www.pool.ntp.org/en/>.

Changing Passcodes

User passcodes may be changed from their default settings on the **Software** page under the Remote Access Password section.

The default for the Web interface is <blank>, meaning no password is needed

Backing-up and Restoring Configuration

Backup and Restore of the iPort High Density system configuration is done on the Backup & Restore webpage

- **Backup** - Saves your settings to an XML file on your computer.
- **Restore** - Click Choose File, browse for and locate the XML file backup, then click Restore.
- **Reset cfg** - Used to factory reset iPort High Density. You will be given the option to;
  - **Restore all**
  - **Restore all except netowrk configuration**
  - **Cancel**

> ⚠ Make sure that you have copies of any licenses before proceeding with a reset.

Restart

iPort High Denstity can be restarted from the Software page.
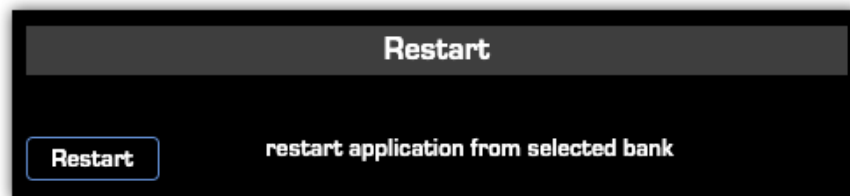
After answering yes to the Continue? prompt, iPort High Density will restart retaining all current configuration values. Answer NO to cancell the restart.

Firmware Upgrade

This is on the **Software** page.
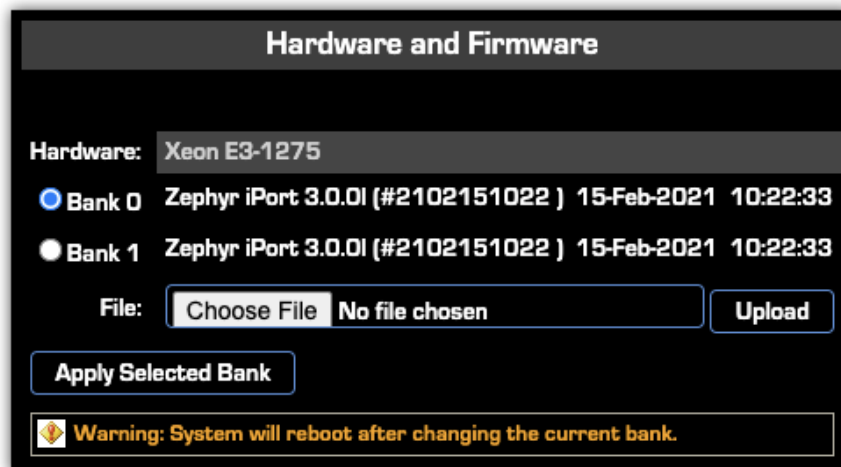
You might receive this via email from our support department, or you might find it on our website. To install new firmware, the first step is to download it to your PC. Then browse to locate it, then click the Upload button to install it into the iPort.

To ensure a smooth upgrade, this procedure must be strictly followed. New firmware can only be uploaded to Bank 1, after switching the operation to Bank 0.

If the upload is started but does not succeed, Bank 1 will become unusable, and you will continue on Bank 0 which is working. You can attempt to upload to Bank 1 any number of times, and you can still reboot from Bank 0.

In case booting starts from a corrupted bank and does not succeed after the third unsuccessful attempt the bootloader would automatically switch the system to using the other bank.

When the firmware has been successfully uploaded, click the Apply Selected Bank button to activate it.

The iPort High Density will be offline for 2 to 4 minutes while the firmware updates.

What's Next?

We've just covered the nuts and bolts of getting iPort High Density installed and connected to the network, and discussed how to update and maintain its firmware. You also learned how to set up your unit's configuration files.

Since the main function of iPort High Density is sending high-quality audio from point A to point B, we'll next discuss how to do that with the MPEG or optional aptX® Enhanced encoders.

Feature Licensing

Covers entry of additional feature licenses and activation for both on-line and off-line methods

Depending on what options you purchased for your iPort, you will have one to three different licenses that need to be entered.

- **Codec expansion licenses** - adds additional non-aptX codecs up to 64 total.
- **aptX license** - enables aptX encoding on licensed codecs
- **Time Zone delay** license (additional hardware required)

Additional licenses are provided to you as a PDF generally by email. If you have obtained a license, please proceed to the License Management page. If you do not have one yet, please contact a Telos dealer.

> ⚠ An internet connection **is required** from the computer completing the activation of any additional licenses.

Feature licenses are activated and deactivated on the License Management webpage.
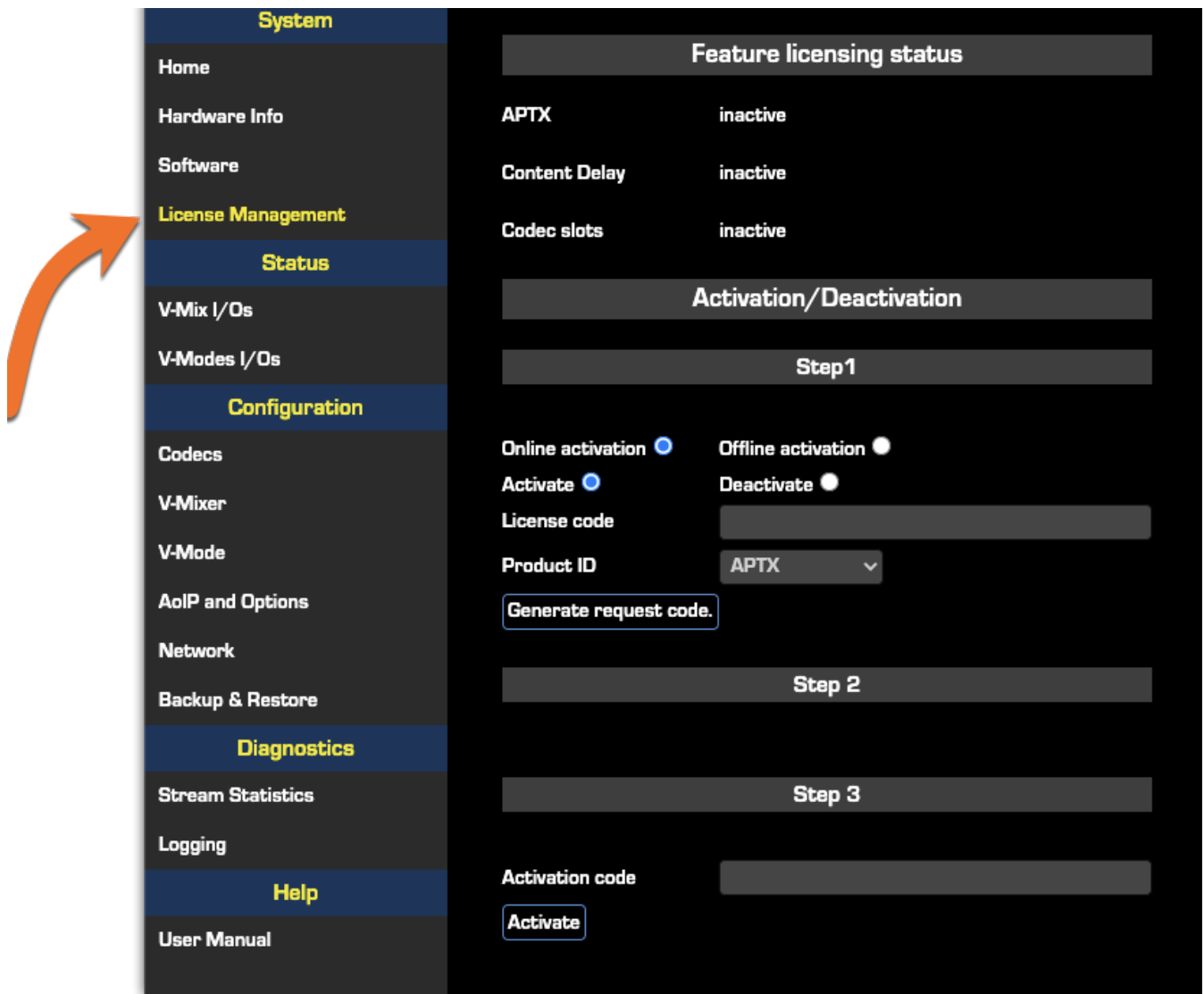
Audio Transport

Audio Transport via MPEG or Optional Enhanced aptX™

Livewire is a professional-grade IP Audio system, used over controlled local-area networks (LAN). It is a modern replacement for older analog or TDM digital systems. There are a wide range of products made for broadcast studio facilities that take advantage of Livewire: interface nodes for analog and AES3 audio, routers, mixing consoles, dynamics processors, ISDN and POTS telephone interfaces, delay units, satellite encoders and receivers, PC-based delivery systems, and – of course – codecs. Facilities span the gamut from a single interface xNode to multi-studio installations with dozens or even hundreds of connected devices.

If you already have a Livewire-based installation, the iPort High Density is a simple and low-cost way to extend it over a wide-area IP network. Connect the local network to one of the iPort jacks and the WAN to the other, make some configuration choices, and you are ready to go. This application nicely illustrates the advantages of IP Audio – were you to do the same thing with traditional equipment, you'd have a rack full of codec boxes, expensive router interface cards, and a plethora of cables.

If you don't already have a Livewire installation, no problem. You can use an xNode audio interface or two (available from The Telos Alliance), and still benefit from the iPort High Density's simplicity and low cost.
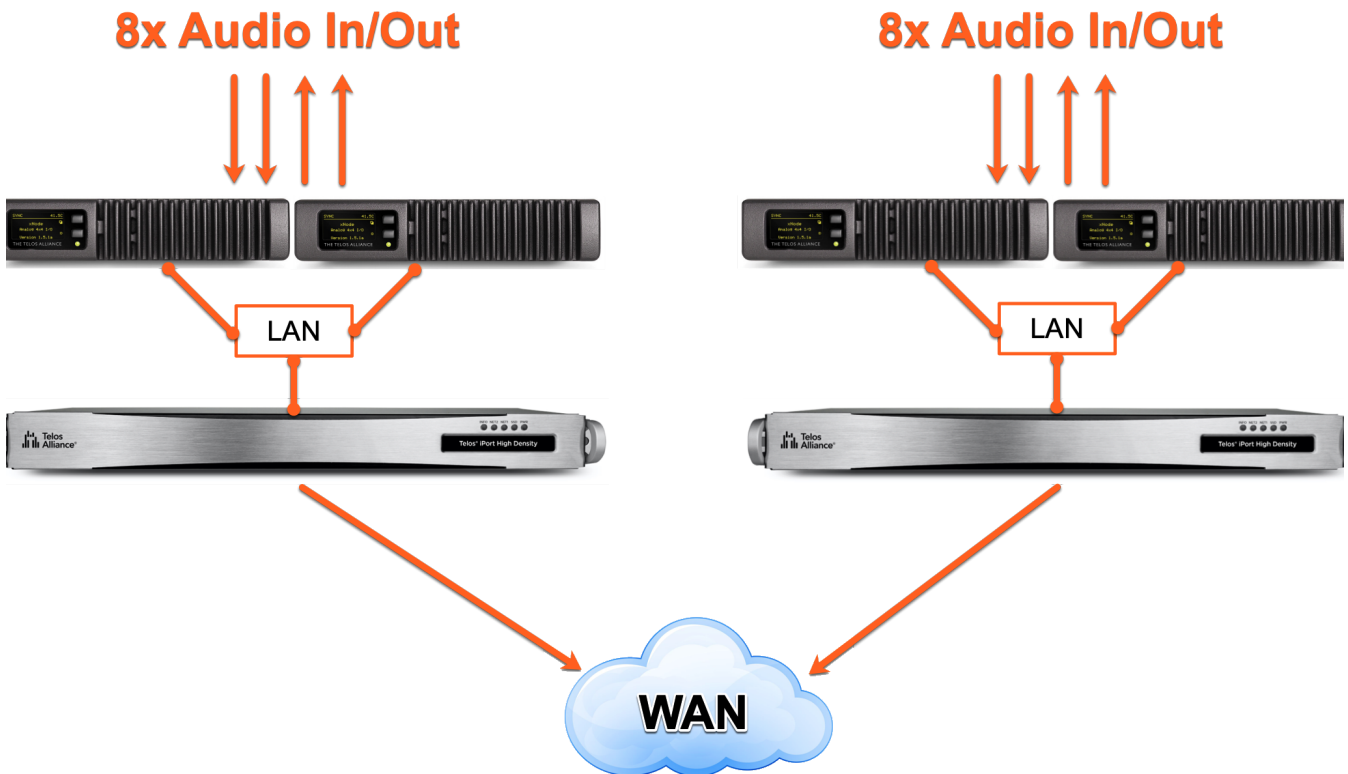
Because Livewire is uncompressed, it requires a lot of bandwidth - up to 5 Mbps for a stereo channel. And because Livewire needs to be very low delay, it cannot tolerate a network with too much latency or jitter. These conditions are easily fulfilled with a switched Ethernet LAN, but they don't generally exist for wide area networks.

iPort High Density bridges the two environments. It reduces the needed bandwidth and accommodates the delay and jitter caused by WANs. Remember that 5 Mbps data rate? After MPEG AAC compression, a typical rate might be 256 kbps – about 20 times less. For most ears, a 256 kbps AAC compression rate is unimpeachable, meaning near full quality. Most of the reduction comes from the compression process, but some reduction comes from iPorts ability to use larger packets with comparatively less header overhead. iPort High Density uses state-of-the-art MPEG codec technology to conserve network bandwidth while preserving excellent audio quality.

Codec Types

The following tables list the codec types in iPort High Density, along with their related bitrates, sample rates, and input samples per channel:

MP3 Codecs

Codec/Mode	Bitrate	Sample Rate	Input Samples per Channel
MP3 Stereo	24 Kbps	16 kHz	3458 (3456)
	32 Kbps	16 kHz	3458 (3456)

	40 Kbps	24 kHz	2304
MP2 Codecs	48 Kbps	24 kHz	2304
	56 Kbps	24 kHz	2304
	64.Kbps	24 kHz	2304
	80 Kbps	24 kHz	2304
	96 Kbps	32 kHz	3458 (3456)
	112 Kbps	48 kHz	2304
	128 Kbps	48 kHz	2304
	160 Kbps	48 kHz	2304
	192.Kbps	48 kHz	2304
	256 Kbps	48 kHz	2304
	320 Kbps	48 kHz	2304
MP3 Mono	24 kbps	24 kHz	1152
	32 kbps	24 kHz	1152
	40 kbps	24 kHz	1152
	48 kbps	24 kHz	1152
	56 kbps	48 kHz	1152

MP2 Codecs

Codec/Mode	Bitrate	Sample Rate	Input Samples per Channel
MP2 Stereo/Dual mono	64 kbps	48 kHz	2304
	96 kbps	48 kHz	2304
	112 kbps	48 kHz	2304
	128 kbps	48 kHz	2304
	160 kbps	48 kHz	2304
	192 kbps	48 kHz	2304
	256 kbps	48 kHz	2304
	320 kbps	48 kHz	2304
	384 kbps	48 kHz	2304
MP2 Mono	32 kbps	48 kHz	1152
	48 kbps	48 kHz	1152
	56 kbps	48 kHz	1152
	64 kbps	48 kHz	1152
	80 kbps	48 kHz	1152
	96 kbps	48 kHz	1152
	112 kbps	48 kHz	1152
	128 kbps	48 kHz	1152
	160 kbps	48 kHz	1152
	192 kbps	48 kHz	1152

AAC-LC Codecs (LC = Low Complexity)

Codec/Mode	Bitrate	Sample Rate	Input Samples per Channel
AAC-LC Stereo/Dual Mono (MP2-AAC)	32 kbps	24 kHz	4096
	40 kbps	32 kHz	3072
	48 kbps	32 kHz	3072

	56 kbps	32 kHz	3072
	64 kbps	32 kHz	3072
	80 kbps	48 kHz	2048
	96 kbps	48 kHz	2048
	112 kbps	48 kHz	2048
	128 kbps	48 kHz	2048
	160 kbps	48 kHz	2048
	192 kbps	48 kHz	2048
	256 kbps	48 kHz	2048
	320 kbps	48 kHz	2048
AAC-LC Mono (MP2-AAC)	24 kbps	24 kHz	2048
	32 kbps	32 kHz	1536
	40 kbps	32 kHz	1536
	48 kbps	32 kHz	1536
	56 kbps	48 kHz	1024
	64 kbps	48 kHz	1024

AAC-LD Codecs (LD = Low Delay)

Codec/Mode	Bitrate	Sample Rate	Input Samples per Channel
AAC-LD Stereo/Dual Mono	64 kbps	32 kHz	1442
	80 kbps	32 kHz	1442
	96 kbps	32 kHz	1442

	112 kbps	48 kHz	960
	128 kbps	48 kHz	960
	160 kbps	48 kHz	960
	192 kbps	48 kHz	960
	256 kbps	48 kHz	960
	320 kbps	48 kHz	960
AAC-LD Mono	32 kbps	32 kHz	721
	40 kbps	32 kHz	721
	48 kbps	32 kHz	721
	56 kbps	48 kHz	721
	64 kbps	48 kHz	480
	80 kbps	48 kHz	480
	96 kbps	48 kHz	480
	112 kbps	48 kHz	480
	128 kbps	48 kHz	480
	160 kbps	48 kHz	480

AAC-HE Codecs (HE = High Efficiency)

Codec/Mode	Bitrate	Sample Rate	Input Samples per Channel
AAC-HE Stereo/Dual Mono	24 kbps	48 kHz	2048
	28 kbps	48 kHz	2048
	32 kbps	48 kHz	2048
	40 kbps	48 kHz	2048
	48 kbps	48 kHz	2048
	56 kbps	48 kHz	2048
	64 kbps	48 kHz	2048

	80 kbps	48 kHz	2048
	96 kbps	48 kHz	2048
AAC-HE Mono	24 kbps	48 kHz	2048
	28 kbps	48 kHz	2048
	32 kbps	48 kHz	2048
	40 kbps	48 kHz	2048
	48 kbps	48 kHz	2048
	56 kbps	48 kHz	2048

AAC-HE-V2 Codecs

Codec/Mode	Bitrate	Sample Rate	Input Samples per Channel
AAC-HE-V2	24 kbps	48 kHz	2048
	28 kbps	48 kHz	2048
	32 kbps	48 kHz	2048
	40 kbps	48 kHz	2048
	48 kbps	48 kHz	2048
	56 kbps	48 kHz	2048

Enhanced aptX™ Stereo/Mono Codec

Codec/Mode	Bitrate	Sample Rate	Input Samples per Channel
Enhanced aptX Stereo (16 bit)	256 kbps	32 kHz	516
	384 kbps	48 kHz	512
Enhanced aptX Stereo (24 bit)	384 kbps	32 kHz	516
	576 kbps	48 kHz	512
Enhanced aptX Mono (16 bit)	128 kbps	32 kHz	516
	192 kbps	48 kHz	512

Enhanced aptX Mono (24 bit)	192 kbps	32 kHz	516
	288 kbps	48 kHz	512

Stereo / Mono Uncompressed Audio

Codec/Mode	Bitrate	Sample Rate	Input Samples per Channel
Uncompressed Stereo Audio (24 bit)	2304 kbps	48 kHz	216
Uncompressed Mono Audio (24 bit)	1152 kbps	48 kHz	216

Getting Started
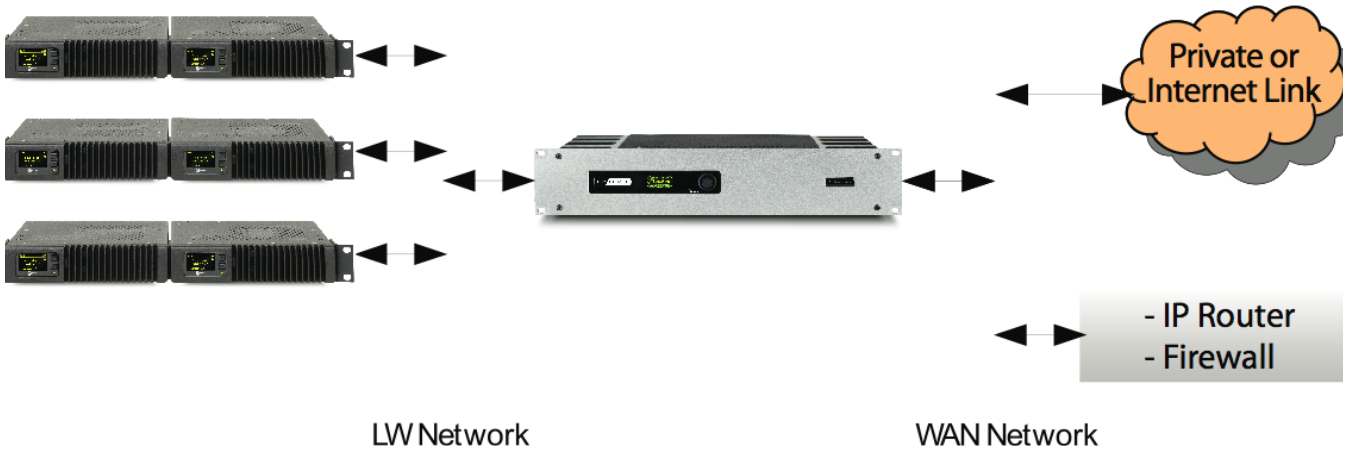
A simple installation might look like the one in the block diagram above. In this case, a Livewire xNode is providing the audio interface. Both analog and AES3 xNodes are available. Each xNode provides 4 inputs and 4 outputs (more if Mono is used). These can be expanded up to the 64 channel capacity of the iPort High Density. The xNodes must be connected with a qualified Ethernet switch. You may then connect a PC for configuration to a spare switch port or use an Ethernet cross-cable for a direct connection to the iPort PLUS.

If you will be using your iPort High Density within a facility that is already Livewire-equipped, the xNode will not be required. Just connect the Livewire port of iPort High Density to an open Ethernet port in the Livewire
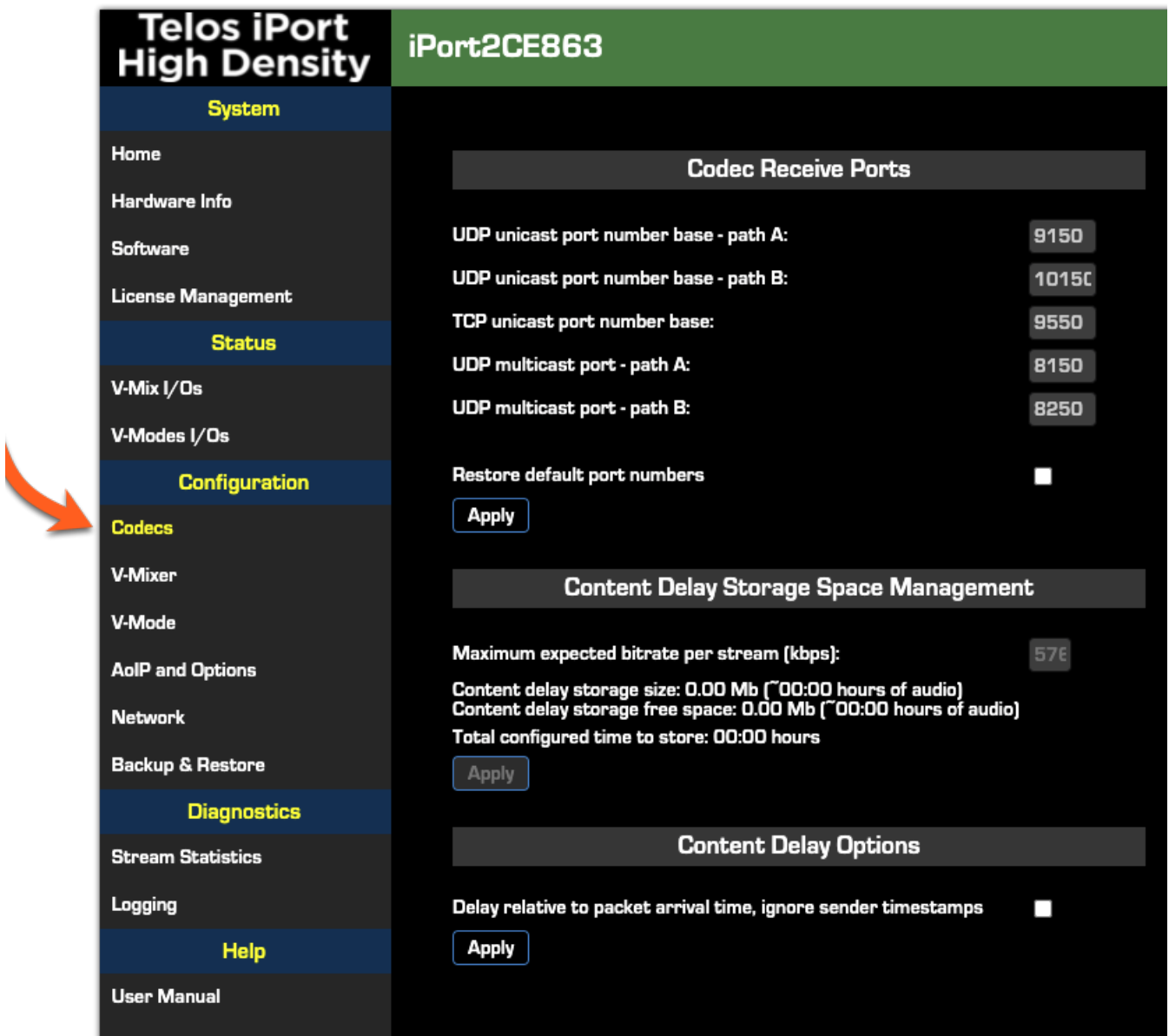
system (1 Gig port required). Livewire sources can be from any Axia product including xNodes, consoles, or PC software drivers. Remember, however, there must be at least one hardware Livewire xNode in the system to provide the required clock signal to the network. If you already have a Livewire network, you like already have this covered.

Most Livewire installations will be like that found in the diagram above, with separate networks for the Livewire audio and the general network, which has the WAN interface. The iPort is ready for this case, with its two Ethernet interface ports. It keeps the two networks isolated.

Codec Configuration

Global Codec Options

From the Codec Configuration page, click the Global Options button at the right top to set the parameters that are common for all codec channels. Having done so, you will see this page:

Codec Receive Ports

The base port numbers for both UDP and TCP determine the receive port number assigned to the corresponding function of the first codec channel. Port numbers for other codec channels are incremented by one per channel automatically. Note that if multiple WAN paths are used the A and B paths MUST have different base ports defined. The defaults are shown.

As an example – using the default configuration – Codec #1 would use the receive port 9150. Codec #2 uses 9151. Codec #64 (on a fully licensed iPort) would use port 9214. These are the ports that the SENDING iPort will send to. Feel free to adjust these ports to fit your needs. Most of the time there is no reason to change these from their defaults.

USD Multicast ports for Path A and B settings determine a single receive port number for all multicast traffic arriving over the path.

The "Restore default port numbers" option will reset all port numbers to the factory default values as shown in the screenshot above.

The lower two sections of this page are used to configure the Content Delay function - see the Optional Content Delay section of this manual.

Codec and Channel Settings

From the Codec Configuration page, click the Options button for the channel you are configuring. There is an Options button for EACH codec.

The Options Page

Transmit

Set the desired Encoder type, Bitrate, and Channel Config (stereo/mono/dual mono) for this channel.

Use the Encoder Type dropdown to select the method of MPEG encoding you wish to use. Along with several options for AAC, MP3 and MP2 encoding, you can also choose linear (uncompressed) audio or GPIO only. If the optional Enhanced aptX is licensed, you can choose options for Enhanced 16-bit and 24-bit encoding.

With AAC, up to around 10% random packet loss can be effectively concealed. This is one of the strengths of AAC. Some other codecs also have concealment, but it will not be as effective.

> ⓘ The Enhanced aptX encoding options are shown even thouh your unit may not be licensed to use them.

Network links with guaranteed Quality of Service are a plus but can lead to a false sense of security. The main weakness of UDP is that it does not recover lost packets, whereas TCP does. QoS only reduces the variance of network latency for the higher-priority traffic, thus allowing the use of less buffering at the receiving end. QoS does not prevent packet loss in general.

DSCP Class of Service refers to Differentiated Services Code Point, which sets the 8-bit Differentiated Services Field (more commonly, the DS field) in the IP header for packet classification purposes. This is where you can fine-tune the management of network traffic and determine the Quality of Service. Settings range from low-latency for streaming media to best-effort for non-critical services.

Unless you are very familiar with DSCP, we recommend you keep this set for 46 Expedited Forwarding.

The Output Configuration section allows you to specify the IP of the main output, as well as up to 3 more replicated streams.

Output configuration

	Protocol type:	IP:	Port:
Main output (path A):	UDP ⌄		9150
Main output (path B):	UDP ⌄		
Enable SHOUTcast protocol:	☐		
ⓘ Non-SHOUTcast devices will stop receiving when this option is selected.			
External stream server password:	telos1		
	Protocol type:	IP:	Port:
Additional output (path A) 1:	UDP ⌄		
Additional output (path B) 1:	UDP ⌄		
Additional output (path A) 2:	UDP ⌄		
Additional output (path B) 2:	UDP ⌄		
Additional output (path A) 3:	UDP ⌄		

Select the Protocol Type and specify an IP Address and Port for the receiver. Note the port used on the receiver is determined by the Global Options from the previous section.

The Protocol type may be UDP, TCP, or multicast. There are tradeoffs in this choice that are only fully understood with knowledge of this aspect of IP networks. Here is a simple overview: UDP is the usual choice because it offers a lower delay than TCP and is not impaired by TCP's flow/rate control. However, with UDP, there is no network recovery of lost packets as there is with TCP, so the codec's concealment is used to reduce the audibility of these events.

Checking the box for the option to **Stop encoded stream in the absence of LW input stream** will disable all encoding (send) if the Livewire input stream fails, is disabled, or removed. Also, note that this option only disables the audio component and if GPIO is configured it will continue to be sent.

The Enable SHOUTcast protocol option is used for sending a SHOUTcast compliant stream to an external SHOUTcast server.

If you do select to use the SHOUTcast stream, the next section is where you set it up. For more information, see the Streaming section of this manual.

Receive

Scrolling further down the codec configuration page, we come to the Receive section.

The Buffering drop-down box lets you choose how much buffer is applied to the received streams. A lower value gives a lower delay but depends upon the network to have low jitter. A range of values allows experimentation to find the optimum for your network conditions. One procedure is to try successively lower values until you hear audio interruptions. Then back off to a higher value allowing for margin. A good rule of thumb is to set your buffer to three times the highest jitter.

> ✓ A note about buffering. Increasing the buffer in an iPort receiver can only account for "jitter" in the receive stream. It can not account for or correct packets that are acually missing. More on this in the Diagnostics and Stream Statistics section.

The Protocol type must correspond to the value set at the encode side unit, be it UDP, TCP, or multicast.

When all values are entered, click the Apply button and return to the Codec page. Here are sample configurations for the first two channels:

GPIO

Each codec channel has a robust set of options for both send and receive GPIO. They are described in the Using GPIO and Data section of this manual.

FAQs

Do all the channels need to go to a single unit at the other end?

No. Each of the Zephyr iPort PLUS channels are independent and may be used individually. Simply enter the IP numbers/ports for the unit you want to use at one or more other ends. Indeed, one codec instance may send an audio stream to up to 8 different IP addresses and port numbers. Note that each iPort codec may receive from one other codec only.

Can I use the Zephyr iPort PLUS with Telos Z/IP ONE™ or Zephyr Xstream® codecs at the other end?

Yes, with some care. Only a limited subset of all iPort functionality is interoperable. For example, there is no support for user data channels or the advanced GPIO operations on other devices.

Can I use Zephyr iPort PLUS with codecs from other manufacturers?

Zephyr iPort PLUS creates and consumes standard MPEG streams with standard RTP/UDP/IP packet

formatting - nothing proprietary or special. Theoretically, it should work with other manufacturers codecs, but we cannot make any guarantees. We recommend you do some experimentation on your own before committing to an equipment purchase or broadcast of a major event.

Does Zephyr iPort PLUS conform to the ITU N/ACIP specification?

No, it does not. Zephyr iPort PLUS is intended for a different class of applications. The N/ACIP standard envisions VoIP call-like operation with SIP control, whereas the iPort is generally used in a 'nailed-up' way.

The Telos Z/IP ONE codec does conform to N/ACIP.

What about firewalls?

You will need to open the appropriate ports in your firewall to accept incoming IP-audio streams. The IP and port numbers are easily set/determined from the Zephyr iPort PLUS's Web pages, so you know which have to be opened. Port translation in the router is allowed if that technique is valuable to your network operation. This follows from the usual nailed-up applications for which Zephyr iPort PLUS is intended.

Telos Z/IP ONE codecs have sophisticated technology for automatically traversing most kinds of firewalls. To do so, it uses a special ZIP Server that resides outside the firewall. (You can use the one we operate as a complimentary service to Z/IP ONE users.) The Zephyr iPort PLUS has no way to use such a server because it does not use SIP for call set-up.

Which codec type should I use?

There are tradeoffs among those available in the Zephyr iPort PLUS, with each having advantages and disadvantages. That's why we give you the choice. Here are some guidelines:

AAC is the best all-round codec for bitrates of 96kbps and above (stereo). It has excellent packet-loss concealment.

AAC-HE (AAC+) should be used at rates under 96kbps. It has good audio quality at 64kbps and is pretty good even down to 48kbps. It also has good packet-loss resilience, but not as good as AAC.

AAC-HEv2 is the most efficient codec for stereo. It has a new "parametric stereo" function that kicks-in at low bitrates. Rather than sending the left/right channels discretely, it sends a core mono signal together with steering control. This makes reasonable quality stereo possible down to 32kbps, and useful stereo even to 24kbps.

AAC-LD has the lowest delay of the psychoacoustic codecs and is the best choice when inter-activity is important, such as for on-air interactions with remote guests. It has about 30% less efficiency than AAC, which means that for equal quality, you would need to use 30% higher bitrate. Its packet loss concealment is good, but not as good as AAC.

MP3 (MPEG layer 3) is not as efficient as AAC and has the worst packet-loss concealment. It is included mostly for compatibility with codecs and software players that only support MP3.

Where can I learn more about TCP, UDP and Multicasting?

Any good network engineering book would explain these in detail. One of our favorites is Computer Networking by Kurose and Ross. There is a section in our Introduction to Livewire that introduces networking concepts to audio engineers, including a discussion of TCP and UDP. If a copy was not included

I need to calculate the actual network bitrate. There will be packet overhead, right?

Yes, the network rate is higher than the codec rate, owing to the headers for the IP packets taking some additional bandwidth. MPEG streams are very efficient in this regard, however. The overhead varies with the specific codec but is typically under 15%.

Will Zephyr iPort PLUS work over the public Internet?

That depends. There are no guarantees of any kind on most Internet connections. This is certainly true when multiple ISPs are involved, since nobody can take full responsibility for the entire link. When you choose AAC as your codec, Zephyr iPort PLUS provides quite good packet-loss concealment up to 10% random loss. That's pretty good and would probably allow many Internet links to work reasonably well. Higher buffer time helps, of course, but at the expense of delay.

If you can take even more delay, you can use the TCP protocol option. In this case, lost packets are recovered by re-transmission, making bad links more usable. This is why streaming audio over the Internet works fairly well. The streaming servers use TCP to connect to players. Delay is not an issue. Indeed, multiple seconds of buffering is the norm.

As we mentioned before, the Telos Z/IP ONE is intended for such applications. It has a suite of adaptive technologies to accommodate bad and variable network conditions.

What's Next?

In this chapter, we showed you how to configure the codecs in Zephyr iPort PLUS to send and receive high-quality audio. In many Zephyr iPort PLUS installations, there's a fair amount of remote control, tally and status overhead that needs to accompany each of the codec channels. The GPIO capabilities of Zephyr iPort PLUS are very robust, so we've devoted an entire chapter to explaining how to set it all up. That's coming up next.

Using GPIO and Data

Using GPIO & Data with iPort High Density

As multiple audio signals are encoded in iPort High Density and distributed over a WAN or other network, their associated GPIO information needs to travel with them. Zephyr iPort PLUS makes this happen by embedding GPIO events in the coded audio stream.

Zephyr iPort PLUS has a GPIO capacity of 20 contact closures per codec channel, and per transmit direction. GPIO events always travel from the encoder to the decoder. To establish a two-way exchange of GPIO events, a duplex codec link is required.

The GPIO functionality of Zephyr iPort PLUS represents a significant upgrade over that of the original iPort. It is not possible to describe every drop down menu and permutation that is available. Instead, we will describe the four operation modes for GPIO. And, with a bit of experimentation, you should be able to

develop a solution that works for you. If you're still having difficulties, remember that Telos Alliance 24/7 tech support is just a phone call away.

Audio and control functions aren't the only things running around the data stream. There are "associated data channels," too. We'll cover these options near the end of this chapter.

GPIO Operation Modes

Zephyr iPort PLUS supports four operation modes, which are selected individually on the dropdown menus, per GPIO ports and transmission direction.

On the Livewire interface side, the Virtual Endpoint and Transit Point modes are TCP/unicast-centric. Zephyr iPort PLUS provides individual point-to-point event transportation channels, which repeat the configuration model and behavior of the "GPIO snake" as implemented in Axia GPIO devices.

In the "emulation" modes, Zephyr iPort PLUS communicates with other devices over the GPIO multicast channel and repeats the behavior of the controller (e.g. Axia console) or I/O device (GPIO xNode) respectively. It allows you to create a "virtual presence" for devices physically located on separate networks.

The four operation modes are:

Virtual Endpoint

Transit Point

I/O Emulation

Controller Emulation

First, a Few Definitions

As with most IT technologies, Livewire uses acronyms. You don't need to know these in order to set up the GPIO functions, but there are a few terms that will be used frequently in the following explanations:

LWRP – LiveWire Routing Protocol

GPIO - General Purpose Input-Output

GPI – General Purpose Input

GPO – General Purpose Output

LWCH – LiveWire CHannel number

Virtual Endpoint

The virtual endpoint mode serves for forwarding the GPIO events between clients (PC applications, Axia consoles) connected on the local side of the Zephyr iPort PLUS, and the codec link. In the endpoint mode, the decoder terminates the snake, therefore in the decoder the GPI events received from the codec link are translated to virtual GPO states. This is functionally equivalent to the snake mode of the Axia IP driver.
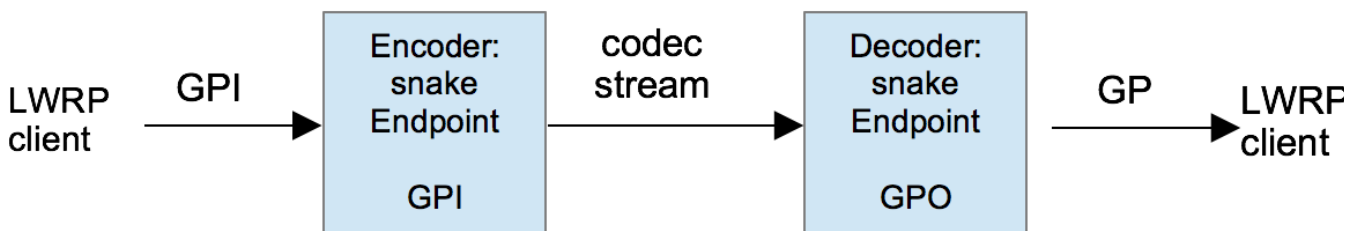
Snake between virtual endpoints in Zephyr iPort PLUS

There is a set of virtual GPI pins associated with each encoder and a set of virtual GPO pins associated with each decoder. When two iPorts are connected via a codec link, GPO states in the decoder automatically follow the corresponding GPI states of the connected encoder.

LWRP (LiveWire Routing Protocol) clients can set and read the states of both GPI pins in the encoder and GPO pins in the decoder, as well as receive change notifications in the standard way. Axia consoles can set the states of GPI pins in the encoder and receive state change notifications about GPO pins in the decoder. GPI states set in the encoder using any of the two methods above would be forwarded over the codec link to the decoder. GPO states set directly in the decoder and states received over the codec link would override each other on the last-arrived basis.

To enable interfacing with an Axia console, a Livewire channel number must be assigned to the corresponding GPIO port of an encoder or decoder, by means of the configuration UI. The iPort's encoders monitor the common GPIO multicast transport channel, to receive events from consoles. The events are reported using multicast command messages and addressed to a specific Livewire channel.

Similarly, the iPort's decoders use multicast event reports to report to consoles about events that were received from the codec link.

Both encoders and decoders monitor the GPIO multicast transport channel to receive GPIO state read requests, and they use multicast messages to report the states requested.

If no Livewire channel number is assigned, the console commands will not be received by the iPort, and notifications will not be generated, while the LWRP interface operations will not be affected.

Transit Point

The Transit Point mode serves for forwarding GPIO events between a GPIO node connected on the local side of the Zephyr iPort PLUS, and the codec link. The transit point mode differs from the endpoint mode by the following:

Zephyr iPort PLUS - encoder: On the hardware GPIO node connection it acts as a client, not server

Zephyr iPort PLUS - decoder: The snake is not terminated in iPort, therefore GPI events received from the codec link are forwarded to client connections

It functions as GPI, rather than being translated to GPO.
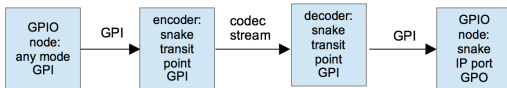
Snake between GPIO nodes

There is a set of virtual GPI pins associated both with each encoder, and with each decoder. When two iPorts are connected via a codec link, GPI states in the decoder automatically follow the corresponding GPI states of the connected encoder.

This mode requires specifying an address of a GPIO event source to be followed - port of a GPIO node connected on the local side of the encoder.

The address is entered into encoder's configuration interface in URL format:

<ip address>[:<udp port number>]/<GPI port number>

This configuration determines the association between the source GPIO port number n in the node and the GPIO port number m in the iPort- encoder. The iPort-encoder (client) would set up a snake connection ("CFG GPO" command) with the specified GPIO node/port (server) that it has to follow. Further, port numbers found in GPI events received from the GPIO node will be automatically translated, and events will be accordingly directed to the linked GPIO ports of the encoder.

LWRP clients can inject GPI events directly into the encoder instance m, bypassing the GPIO port number translation that is applied to server events. Further these logic states would be memorized and forwarded over the codec link identically with those received from a node.

Similarly, LWRP clients can inject GPI events directly into the decoder, overriding the states received via the codec link from the connected encoder. These GPI events would be memorized in the decoder and indicated to other LWRP clients identically with those received from the codec link.

LWRP clients can read the logic states from encoders or decoders using the GPI command, or they can subscribe (ADD GPI) to receive automatic GPI change notifications in the standard way. The following exceptions apply:

GPI state changes in the encoder, initiated by the locally connected GPIO node that is acting as a server (snake origin), do not trigger event reports back to the server.

GPI state changes in the decoder, initiated by the codec link, do not trigger event reports back to the codec link.

I/O Emulation

In this mode, iPort emulates local presence of a physically remote I/O device, e.g. GPIO node.

In brief, this mode is "symmetric" to the controller emulation mode in the following way:

Use of console messages for command and response are swapped - encoder is listening to command and decoder is issuing response.

Virtual GPI's and GPO's are swapped – the encoder has GPO and decoder has GPI.

Console: LWCH →	CMsg WRNI	encoder: emulate I/O ← LWCH GPO	Codec stream	decoder: emulate ctrl. LWCH → GPO	CMsg WRNI	GPIO node: multicast ← LWCH GPO
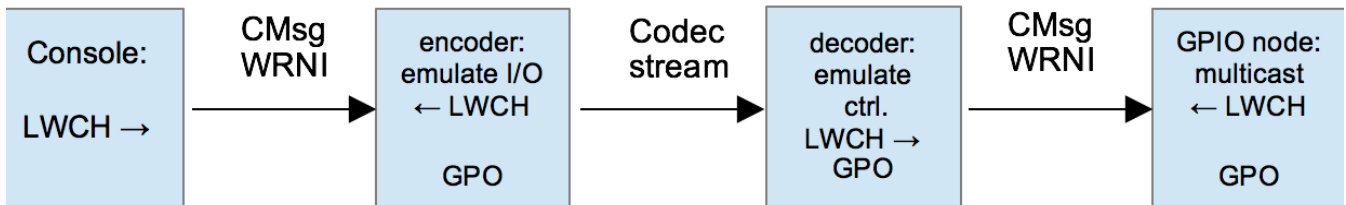
Console to GPIO node, emulation

Controller Emulation

In this mode, iPort emulates local presence of a physically remote controller device, such as an. Element or iQ console.

The Zephyr iPort PLUS encoder is listening to console event reports on the Livewire side. The logic states from the received console messages are applied to virtual GPI and over the codec link forwarded to the decoder.

iPort decoder receives logic states from the codec link, applies them to virtual GPO, and forwards to Livewire side in command messages.

Alternatively, LWRP clients can set and read the states of both GPI pins in the encoder and GPO pins in the decoder, as well as receive change notifications, in the standard way. GPI states set in the encoder are forwarded to the decoder the same way as those received with event reports.

When GPI or GPO states are modified via different interfaces (LWRP, console or codec link) concurrently, the state arriving later in time overrides the former, regardless of the interface of origin.

GPIO node: multicast LWCH → GPI	CMsg INDI	encoder: emul. ctrl ← LWCH GPI	codec stream	decoder: emulate I/O LWCH → GPI	Cmsg INDI	Element: ← LWCH
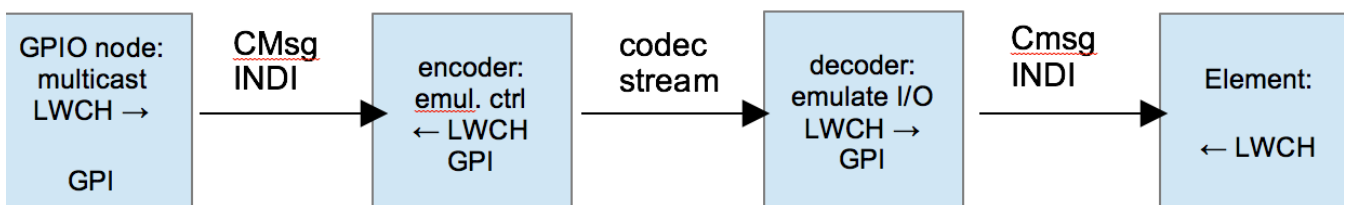
GPIO node to Element, emulation

Configuring the Data Channels

Zephyr iPort PLUS provides three unidirectional user data channels per each audio direction. These are provided typically to transport metadata (artist, song title, album and related info) associated with each audio stream.

To configure these channels, go to the Codec Configuration page, click Options for the channel being set up. Scroll to the bottom of that page and click Data Channels.

Set the Protocol dropdown for TCP Server, TCP Client or UDP.

Enter the IP address along with the correct port on your computer where the metadata connects from your automation system or other source.

Do the same for the other two channels if needed, and don't forget to click Apply when you're done. If everything is working correctly, you should see the status change from Not Established to Established.

What's Next?

This chapter has introduced you to the fundamentals of Zephyr iPort PLUS GPIO. The next chapter will cover configuration and usage of the Zephyr iPort PLUS Content Delay option.

Optional Content Delay

Using Optional Content Delay

If your requirements include recording content for playback and distribution to different sites in different time zones, you may want to order iPort HD with the Content Delay option. Its features include:

- NTP synchronization for the content delay on absolute time.
- Built-in SSD with dynamic storage space allocation, configurable per codec.
- Delay of GPIO and user data channels synchronized with audio.

Content Delay is not a field-installable option. The factory must install it at the time you place your order.

To verify that your iPort HD does indeed contain Content Delay, go to the License Management page. Feature licensing status is shown at the top of this page. Make sure that Content Delay is marked Active (not Inactive as shown in this example).
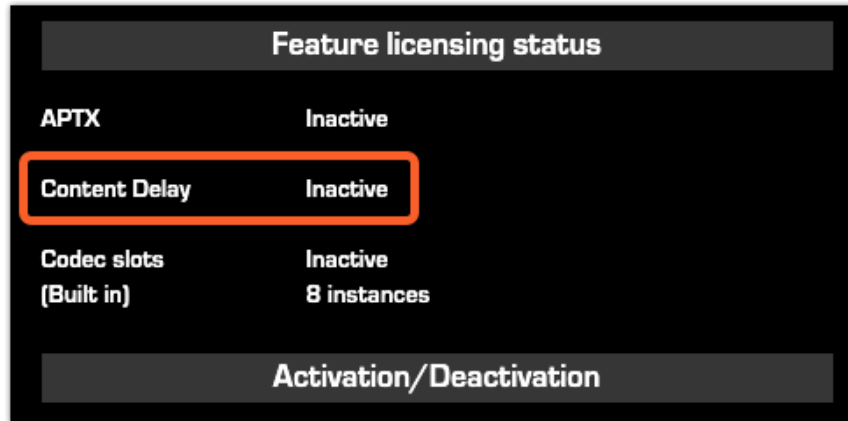
Contnet Delay showing INACTIVE status

Content Delay Configuration

Setting the Storage Space Management Options

The total amount of content that you can store on the iPort HD is based on several factors.

- The size of the installed storage drive
- The number of codecs using the Content Delay
- The bitrate of the codecs using the Content Dealy

From the **Codec Configuration** page, click the **Global Options** button at the top right. Set the following two parameters (bottom two sections of the page):
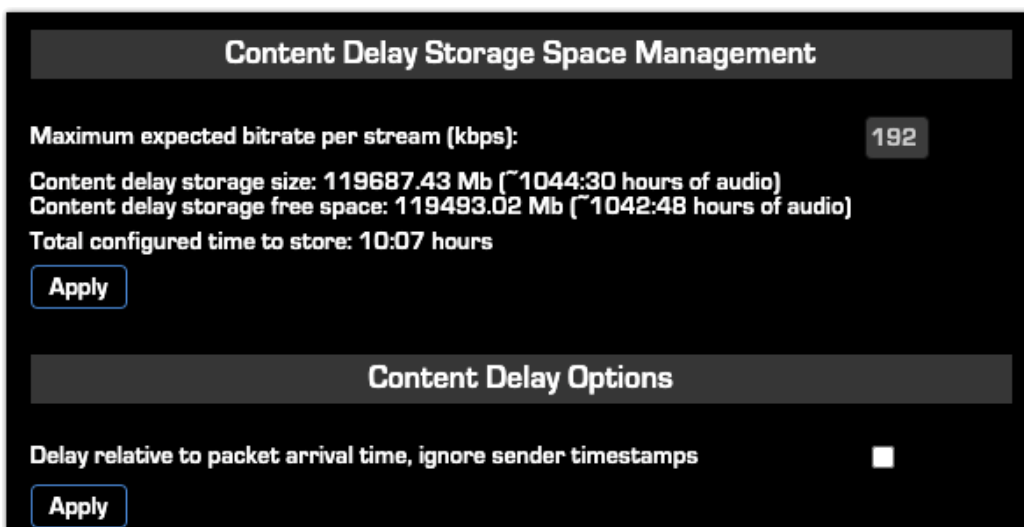
Maximum expected bitrate per stream (kbps)

The maximum expected bitrate per stream setting is used to calculate the total program storage time of the Content Delay feature based on the installed physical storage space. It shows maximum storage space, current free storage space, and currently configured storage time from all codecs.

In this example, based on a stream rate of 192 kbps, there is a total storage time of 1044 hours, 30 minutes. Only 10 hours, 7 minutes of storage are currently configured.

Delay relative to packet arrival time, ignore sender timestamps

This option determines whether the content will be delayed by the set time relative to its creation time at the sender (option unchecked) or relative to the receiver's arrival time (option checked).

Individual Codec Storage Options

The configuration is done individually from each Codec's Options page.

1. Click the Options button for the Codec you want to configure.
2. Scroll to the bottom of the page.
3. Check the box to **Enable content delay:** then click **Apply.**
4. Using 24-hour time, enter the **Storage duration**.
5. Enter the **Playback** offset.
6. Click **Apply.**

 A good analogy is to think of Content Delay as a pipe. The **Duration to Store** determines the pipe's length, while the **Playback offset** controls the point at which you tap into the pipe.
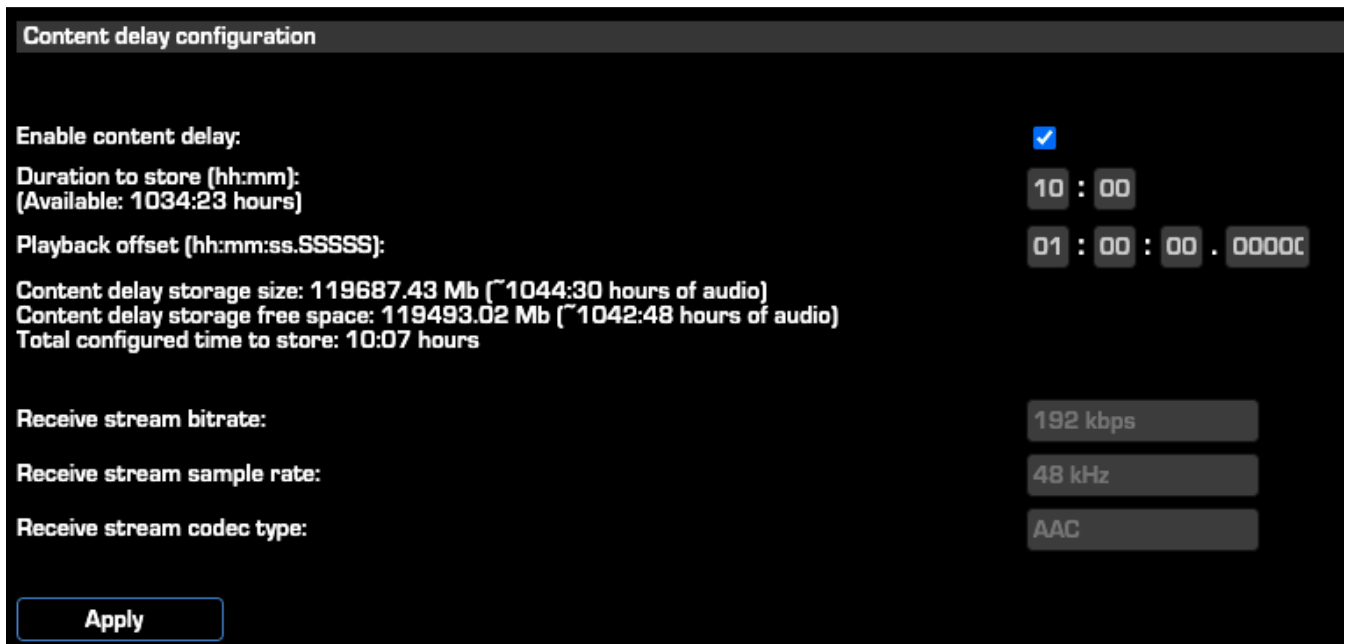
This scheme allows you to instantly switch to any point of the stored program, back and forth, as long as the fragment is still in the pipe.

Note that 'playback offset' is the actual delay time as measured from the instant of content creation. All calculations under the hood are based on UTC, regardless of the local time-zone setting.

> ⓘ Note that the resolution for playback offset goes to milliseconds, allowing you to correct delays in digital transmission links.

The display also tells you the total amount of storage space, how much is still available, and the total time available for this particular codec. Each codec instance is limited to 24 hours. Within that range, free storage

space is available to any codec.
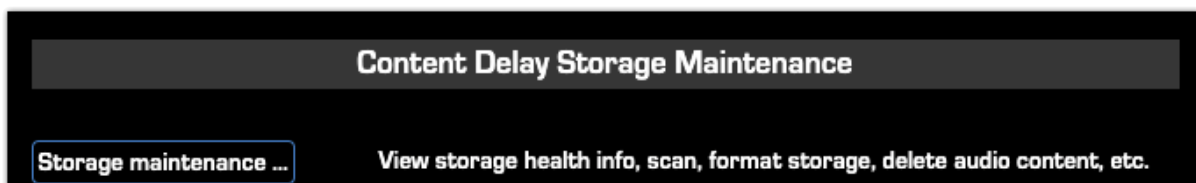
SSD Housekeeping

If you want to check up on the Solid State Drive's health, make repairs, or format the drive, go to the bottom of the Logging page and click on the Storage Maintenance button.

Streaming

When used with an appropriate SHOUTcast distribution server, Zephyr iPort PLUS can be used as a simple and reliable encoder for Internet or internal audio distribution. Because the iPort generates standards-based MPEG streams, a wide variety of internet and mobile devices can be used for listening.

For delivering audio over a LAN or private network, the encoder and server can be together in the same rack. For public streaming over the Internet, the encoder typically runs in the place where the audio is generated (e.g. a studio) and the server is in a place where a lot of bandwidth is available such as an Internet co-location site.

Streams that are served by SHOUTcast protocol servers can be heard on Winamp, Apple iTunes, XMMS, VLC, Foobar, Microsoft Windows Media Player, and many other PC software players.
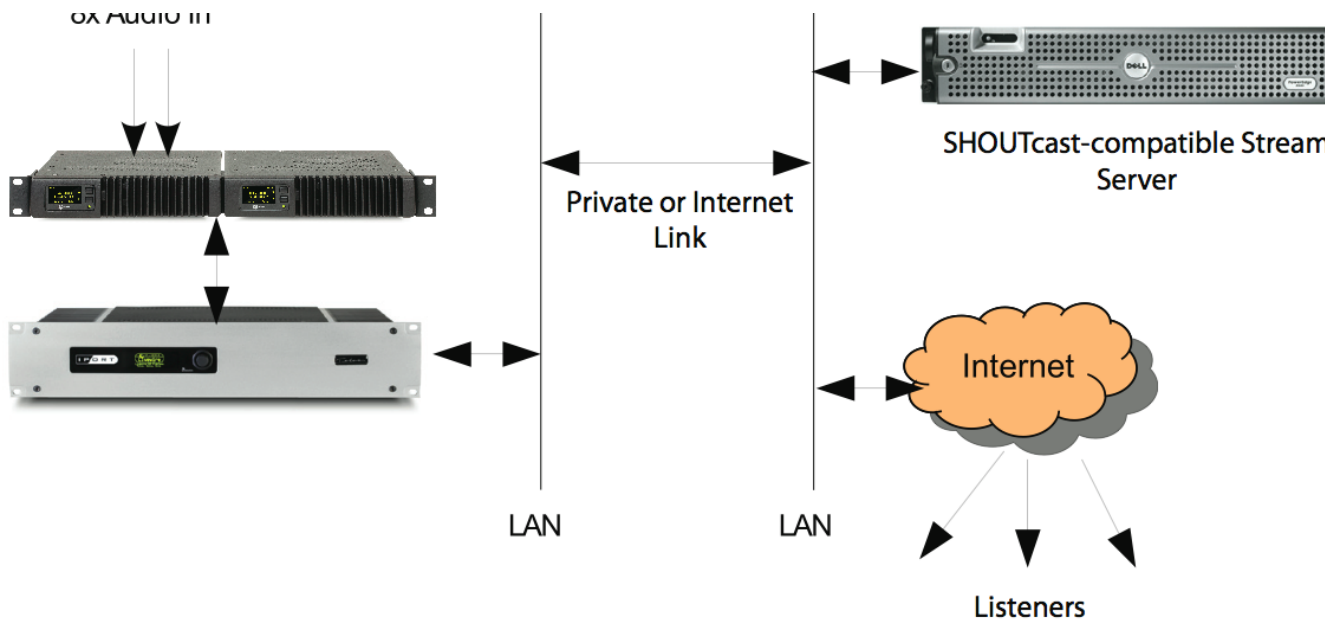
The diagram above shows a simple set-up for streaming. A Livewire xNode AoIP interface provides live audio input, which connects directly to the Livewire Ethernet jack on the iPort. The iPort's WAN jack connects to a network (typically the Internet via your router) that leads to a remotely located streaming server. That streaming server provides streams to listeners over the Internet.

Many variants are possible. For example, a Livewire PC driver could provide the source audio rather than xNode hardware (remember, there must be an xNode somewhere on the network to provide clock synchronization). An installation which already has Livewire equipment would not need another xNode dedicated to the iPort - you would just select channels you want to stream directly from the Livewire network.

If you are planning to make really large scale public broadcasts via the Internet, you may want to host your broadcast through a Content Delivery Network (CDN).

This type of installation would be similar to the one diagrammed above. In this case, one LAN ties together the xNode, Zephyr iPort PLUS, as well as an Omnia® audio processor. Another network links to the WAN (presumably with a firewall in the picture). This offers good security since the Livewire network is isolated from the WAN. It would also be possible to use a single network on the studio side. In this case, the firewall would be responsible for protecting the Livewire network.

Configuring Zephyr iPort PLUS for Streaming

Go to the Codec Configuration page. For each stream:

Enter the Livewire channel number you want to use as your source.

Enter the Livewire text name.

Enter the stream server IP number and port (the port must correspond to that set in the server's configuration).

1.

Enable:	Channel (1..32766):	Type:	Source name (up to 16 chars):	Status:	Audio:	Status:	IP:Port		IP:Port		

Global Option

dec 1	☐	10804	From source ⬍	iPod				multipoint ⬍	➝	multipoint ⬍	Options
	☐	5801	Standard Stereo ⬍	iPort .58 Codec			192.168.0.58 : 9150		⬅	0.0.0.0 : 0	Apply

Encoder: AAC, 128 kbps, sampling 48 kHz, Stereo, DSCP:46 Decoder: buffering 500 ms, , sampling , , , UDP, UDP, DSCP: ,TZD: OFF

Press the Options button to access the codec detail configuration.

Encoder type:	MP3 (mpeg1 layer 3) ⬍
Bitrate:	128 kbps ⬍
Sample rate:	48 kHz
Channel config:	Stereo ⬍
DSCP Class of Service:	46 Expedited Forwarding (recommended) ⬍

Output configuration

	Protocol type:	IP:	Port:
Main output (path A):	TCP ⬍	192.168.1.107	9150
Main output (path B):	UDP ⬍		9850
Enable SHOUTcast protocol: (Non-SHOUTcast devices will stop receiving when this option is selected)	☑		
External stream server password:	************		
	Protocol type:	IP:	Port:
Additional output (path A) 1:	TCP ⬍	192.168.1.8	10098
Additional output (path B) 1:	UDP ⬍		
Additional output (path A) 2:	TCP ⬍	192.168.1.8	10163
Additional output (path B) 2:	UDP ⬍		
Additional output (path A) 3:	UDP ⬍		
Additional output (path B) 3:	UDP ⬍		

 Enter the encoder type, bitrate, and other parameters.

Select 'TCP' as the protocol (used for the iPort to server link).

Click the box Enable SHOUTcast protocol, and enter a password that will be used by the server to authenticate the stream.

Install, configure, and start the server.

Using the Integrated SHOUTcast Stream Server

Click the Enable stream server checkbox, then complete the following fields:

Integrated stream server

Enable stream server:	☑
Stream on LW interface:	
http://10.0.0.58:8000/live1	Listen
Stream on WAN interface:	
http://192.168.0.58:8000/live1	Listen

SHOUTcast stream information

Source name:	stream1
Homepage for the stream:	http://positiveenergy.com

Genre:	disco
IRC:	70s channel
AIM:	N/A
ICQ:	70s channel2

Source name **–** Used for the stream directory listings.

Homepage for the stream **-** Enter the stream's web address.

Genre – Enter the format of your stream or station here, for example rock, country or urban. SHOUTcast will display this information in the metadata field.

IRC – Internet Relay Chat - If you have an IRC channel, enter it here and it will be displayed on your SHOUTcast server stats.

AIM – AOL Instant Messenger - If you have an AIM handle, enter it here and it will be displayed on your SHOUTcast server stats.

ICQ – I See You - If you have an ICQ handle, enter it here and it will be displayed on your SHOUTcast server stats.

MPEG License & Music Copyright

The MPEG encoders in Zephyr iPort PLUS are properly licensed for professional, non-public applications such as broadcast remotes, private distribution systems, and the like. We've paid the fees as part of the cost of making the Zephyr iPort PLUS (in fact, it is a significant part of our expense), and the use license is passed to you when you buy the equipment. For this kind of application, you have no need to pay any additional fees.

However, public streaming applications may require additional license payments. It's your responsibility to investigate whether you have this obligation.

Musical compositions and sound recordings are creative works that are protected by the copyright laws of the United States (title 17, U.S. Code) and other countries. Under U.S. law, the owner of a copyright has the exclusive right to (and to authorize others to) reproduce the work, use parts of the work in a new creation, distribute the work in whole or in part, and to publicly display or perform the work (including on web pages and through webcasting). With few exceptions, it is illegal to reproduce, distribute or broadcast a sound recording without the permission of the copyright owner. It is your responsibility to comply with the copyright laws of the United States and other countries in which you broadcast and to pay all applicable royalties to the copyright owners when you become a webcaster.

There have been recent amendments to the copyright law regarding webcasting of sound recordings. These new provisions allow webcasting under the terms of a statutory license, as a way to help webcasters get permission without having to go to each sound recording's owner. The statutory license, however, has strict requirements that you must follow. Some of these requirements include the payment of license fees, limitations on the number of songs from the same album or artist that may be played in a three hour period (called the sound recording performance complement); a prohibition on publishing advance playlists; and a requirement to identify the song, artist and album on the website. There are other requirements as well. The

Recording Industry Association of America provides quite a bit of information on copyright law as it applies to webcasting, and both ASCAP and BMI have created license agreements that they are willing to grant to webcasters that they believe conform to the provisions of the new copyright rules for webcasting. For additional information on the statutory license and other aspects of webcasting, please visit the following sites:

The U.S. Copyright Office

http://www.copyright.gov

The Recording Industry Association of America

http://www.riaa.com/issues/music/webcasting

ASCAP

http://www.ascap.com/weblicense/webintro.html

BMI

http://www.bmi.com/iama/webcaster/index.asp

VMix and VMode

If you're familiar with Axia mixing consoles, then you already know about VMix and VMode, the virtual mixing and stream manipulation utilities that are built into these products.

That same functionality is available with Zephyr iPort PLUS. VMix (short for Virtual Mixer) provides up to 8 channels of "virtual mixing", which can be used to pre-mix up to 5 audio sources each for presentation as a single. VMix works completely independently of the rest of your iPort. In addition to static control of VMix through its web pages, Axia's Pathfinder routing control tools can also be used to dynamically control VMix and create mixing functions based on a variety of system-wide parameters.

VMode (short for Virtual Mode) can be used to convert channel count, channel order, and/or channel content between audio streams at their input and output. For example, passing some channels while cutting others, summing stereo to mono, upmixing or downmixing between stereo and multi-channel, combining channels from two inputs into one output. Select audio as input, define the "Mode" (conversion type, or in other words, the routing within an internal matrix), and define the output as a network source into the AoIP network. There are 8 VMode instances in an iPort engine. Normally, these would be used to manipulate Livewire audio channels before or after the iPort's inputs or outputs.

Virtual Mixing with VMix

What's it all about?

VMix is a 40-input "virtual" mixer accessed using the Zephyr iPort PLUS web browser. This mixer consists of 40 stereo input channels, a direct output for each channel, 8 submixer outputs, and one master out. The 40 channels are divided equally among the 8 subgroups, providing 5 stereo channels in each subgroup mixer.

Using these, you can combine multiple Livewire audio channels into a single summed channel, which can then be sent to a receiving Zephyr iPort PLUS.

The various VMix outputs described above are sources that can feed your Livewire network and they can be manipulated in the same manner as any other audio source. A VMix source can be applied to a console fader, assigned to an audio node destination, or monitored by Pathfinder.

To understand this concept, think of VMix as a standalone piece of hardware. If you visualize wiring an external line mixer to your network, Livewire audio sources would be connected to the mixer inputs; your VMix outputs then become mixdowns that you can use anywhere else on the network, just like any other audio source.

To access VMix setup, open a browser on a computer connected to your Axia network and browse to the IP Address assigned to Zephyr iPort PLUS or, if you have Axia iProbe, open the UI page by right-clicking on the device's icon in the left pane. Then, choose the V-Mixer and V-Mode item from the navigation bar.

The following image is an excerpt of the VMode setup screen. It shows the VMix Main and Sub-Mixer 1 controls. These sub-mix controls are duplicated for all 8 VMix submixers.

Note that adjustments made to VMix take effect as soon as you apply them — so changes saved "on the fly" will affect your output streams immediately.
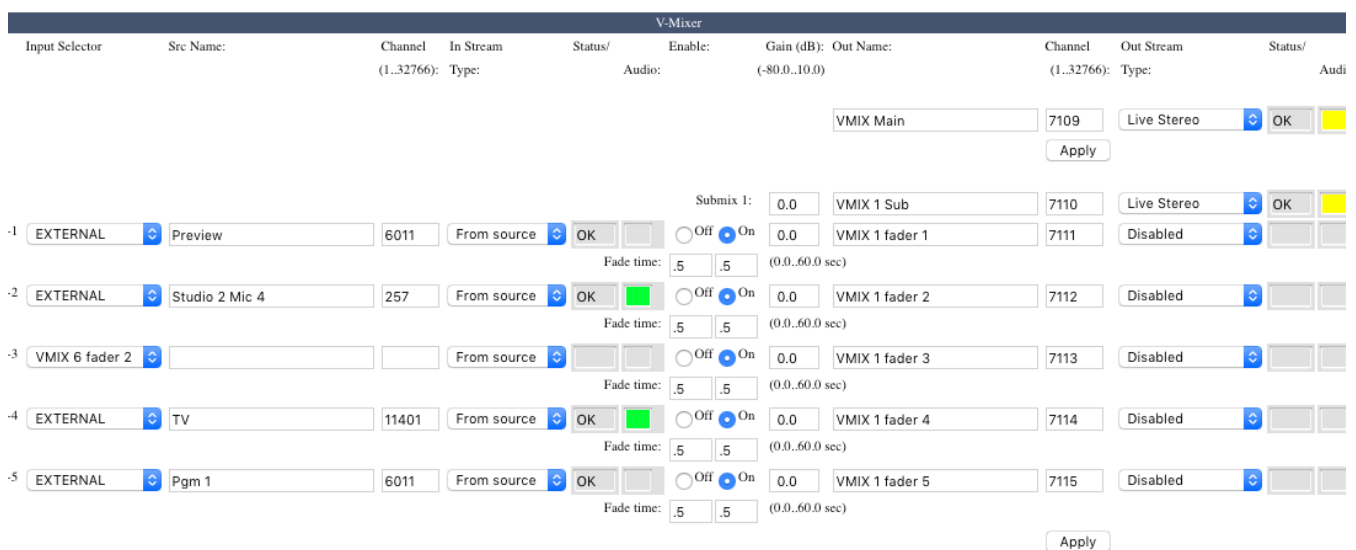
VMix Main Controls

The VMix Main output provides a summed mix of all 8 submixes. Unless you need a single output that combines the audio from of all the submixes, you may leave this disabled — it doesn't need to be enabled for the submixers to work. There are only a few settings for this control:

Out Name **-** You can enter a friendly name for the VMix Main output, which will be displayed as a source name on your Axia network.

Out Stream Type **-** Choose from Live Stereo or Standard Stereo, or leave it Disabled if no Main stream is desired.

The Status window will normally display "OK" when the stream is enabled.

The Audio box will be green when audio is present.

VMix Submixer Controls

In most cases, the VMix Subs are the only channels you will need to enable, since each Sub has its own direct output. Only enable Submixes you intend to actively use; active submixes without any activity clutter up your network with empty streams.

First, you'll want to enable the Submixer you're working with. At the top of the section, you'll see (in this case) Submix 1 displayed on the screen. Next to that are controls for:

Gain. Set this at whatever output level you want the submix output stream volume to be.

Channel. All Livewire audio streams are assigned a Channel number; put a unique value in this box.

Out Stream Type. Choose from Live Stereo or Standard Stereo. Disabled turns off the submixer.

Each VMix Sub submixer input includes an on/off setting, a gain setting, and automatic fade-up/fade-down time parameters. Using one is easy; let's walk through the steps:

In the Src Name box, enter the name of the source you'll be assigning to the input.

In the Channel box, enter the unique Livewire channel number of your audio source.

In Stream Type will normally be set to "From Source", meaning that the source itself (a mic, CD player, etc.) will be providing the audio. However, you can select "To Source" to use the source's automatically generated Backfeed (mix-minus audio) as in input.

Example: Selecting "From Source" when a phone hybrid is assigned as a VMix Sub input would use the caller audio; selecting "To Source" would instead use the Mix Minus sent to the hybrid.

Select the Enable box next to the input to turn the Submixer input on.

Leave the Fade Time boxes set at their default.

The Fade Time function won't be used in normal operation, but can be used to create cross fades between sources when Pathfinder is dynamically making changes to the VMix.

If, say, 1.0 is entered in the first box, the submixer channel's audio will rise from -∞ to the Gain value set in the next field in 1.0 seconds. If the field is set to 0, the audio will simply turn on at the gain value specified.

The second box works the same way, but controls the ON to OFF fade time.

In the Gain box, enter the setting, in dB, that you want for the Submixer input. Each input has its own individual Gain setting.

That's all you need to do to configure a VMix Submixer input stream. You can configure up to 5 input streams per VMix Submixer.

At the bottom of each of the submix sections is an Apply button. Any changes you make will be saved when this button is pressed. Be sure to save the changes for each submixer as they are configured.

VMIx Submixer Advanced Options

There are some additional options provided for advanced users.

In most cases only the Submix output itself needs a unique channel number, but if you so desire, each VMix input can also be sent back to the network as a unique source, post the ON/OFF and gain stage of the VMix. Some users find this useful for constructing "cascaded" mix channels to suit unique situations. To do so, enter values for these controls:

Out Name: The name you give the post-on/off submix channel to send back to the Axia network.

Channel: the channel number you assign to the post-on/off submix channel to send back to the Axia network.

Out Stream Type. Choose from Live Stereo or Standard Stereo to enable the direct output stream for this VMix "fader". Disabled means no stream will be generated.

Be sure to click Apply after you've finished making any options changes.

Some VMix Examples

Now that you know how to enable and set up VMix submixers, what might you do with it? The possibilities are limited only by your imagination. To get your creative juices flowing, here are some examples.

A Mix of Sources to Monitor

Some facilities may need to monitor one or more sources in addition to program audio, like the "squawk channel" some satellite feed providers use to relay announcements.

If you wanted to monitor this "squawk" audio on your Preview speaker without taking up a fader assignment, you could create a mix of the "squawk" source and the Preview mix from the engine. The VMix Sub Channel would be the audio source that you would route to the Preview speaker.

Virtual VMix Control
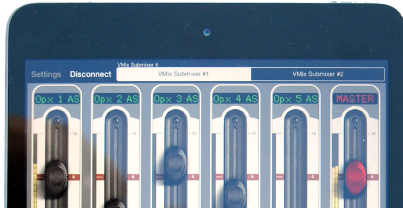
GPIO control of your VMIX with Pathfinder

Imagine that you have a night jock that should monitor all four radio stations in your cluster. To help make sure this actually happens, you could send all four off-air signals as sources into a VMix submixer, and take the output of that submix to a monitor. A Fusion accessory panel or external button wired to a GPIO port could then provide a "press and hold" function to allow the jock to monitor the sources momentarily. (This example is only possible with Pathfinder control of VMix.)

Manipulating Streams With VMode

VMode Application

Each of the 8 VMode instances supports audio streams with up to 8 channels at input and up to 8 channels at output. Between the input and output, there is a matrix, where the conversion is performed, according to the selected audio routing option ("Mode"). Livewire 8-channel streams carry 5.1 content in channels 1-6, and stereo content in channels 7 and 8, and the matrix is partitioned the same way, to fit them natively:

- Channels 1 to 6 of the matrix form a multi-channel part
- Channels 7 and 8 of the matrix form a stereo part

Routing option ("Mode") names refer to the parts of the matrix and conversions performed between them. They do not refer to channel maps of the input/output streams.

For receiving and transmitting, two stream classes are distinguished:

- Class "mono/stereo": streams with 1 or 2 channels
- Class "multi-channel": streams with 3 to 8 channels

Input streams put their content into, and output streams take their content from, the stereo or multi-channel parts of the matrix, according to their mono/stereo or multi-channel classes. Multi-channel streams with 7 and 8 channels span across the both parts of the matrix, starting from channel 1, in the natural order. Livewire 8-channel "Surround" streams fit the matrix mapping without reordering.

The VMode view provides status of the current configuration.

Input Selector	Src Name:	Address	Status/Audio	Audio Mode:	Out Name:	Address	Status/Audio	
None				Pass stereo	V-Mode 1	Disabled		Configure...
None				Pass stereo	V-Mode 2	Disabled		Configure...
None				Pass stereo	V-Mode 3	Disabled		Configure...
None				Pass stereo	V-Mode 4	Disabled		Configure...
None				Pass stereo	V-Mode 5	Disabled		Configure...
None				Pass stereo	V-Mode 6	Disabled		Configure...
None				Pass stereo	V-Mode 7	Disabled		Configure...
None				Pass stereo	V-Mode 8	Disabled		Configure...

To change the configuration, select the far right "Configure…" button.

SDP file: If the input stream to the VMode is an AES67 stream, an SDP file may be supplied and uploaded using the "Choose File" and "Import" buttons.

SDP document: Alternatively, the textual content of an SDP file may be copied and pasted in to this box and applied by clicking "Import"

Input Selector: used to select any source that is native to the iPort engine or to select an "EXTERNAL" source from the network.

Source Name: Text field used to describe the input

Address: This is used to manually define an EXTERNAL source. Note that this supports Livewire, AES67, Unicast and AES67 SIP. Usage is shown in this screen shot and on the Web UI.
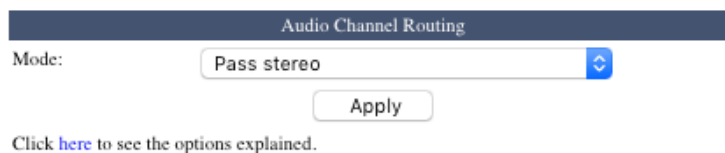
Notes regarding SIP ports used:
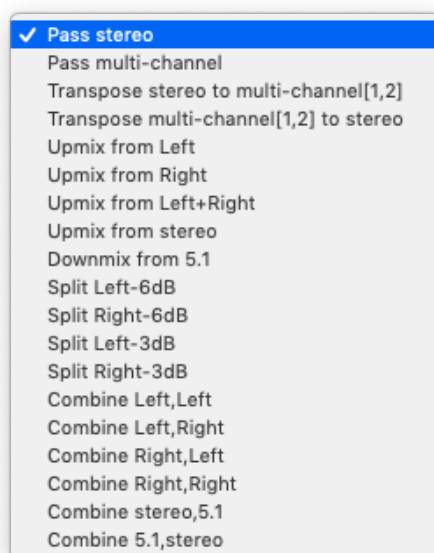
Default start port for SIP audio is 16384

Valid range for RTP stream protocol is from 16384 to 65535

SIP URI protocol supports ports from 1 to 65534 (CPSIP Stack does not allow port 65535)

Default SIP port is 5060

Audio Channel Routing

Mode: Pass stereo

Apply

Click here to see the options explained.

Mode: The manipulation within the matrix is referred to as rouging and is configured by selecting a Mode from the list.

- ✓ Pass stereo
- Pass multi-channel
- Transpose stereo to multi-channel[1,2]
- Transpose multi-channel[1,2] to stereo
- Upmix from Left
- Upmix from Right
- Upmix from Left+Right
- Upmix from stereo
- Downmix from 5.1
- Split Left-6dB
- Split Right-6dB
- Split Left-3dB
- Split Right-3dB
- Combine Left,Left
- Combine Left,Right
- Combine Right,Left
- Combine Right,Right
- Combine stereo,5.1
- Combine 5.1,stereo

Pass stereo: passes the stereo part of the matrix transparently, and blocks the multi-channel part.

Pass multi-channel: passes all 8 matrix inputs to the corresponding outputs.

Transpose options: These options move two channels between the stereo part of the matrix and channels 1 and 2 of the multi-channel part, as indicated in the name of the option.

Upmix and Downmix options: These options mix, copy, and/or apply gains to input channels, to obtain output channels. Only the stereo part of the matrix is involved in transformations, where both inputs and outputs are of the mono/stereo class. Both parts of the matrix are involved in transformations between the mono/stereo and multi-channel classes. All "Upmix …" options output silence in the LFE (sub woofer) channel.

Split Options: These options create a mono sum from a stereo input, apply the specified attenuation, and send it to either Left or Right output channel as indicated in the name of the option, while sending no signal to the other. For example, "Split L –6dB" sums Left and Right, reduces by 6dB, and places the mix on the Left channel.

Combine Options: These options are special in that they work with a pair of inputs. All 8 Vmode inputs are

arranged in 4 fixed pairs – [1,2], [3,4], etc. When a "combine" option is selected on a Vmode instance, it takes input channels from the two inputs (designated A and B) of its pair, as indicated in the name of the option. For example, "Combine Left,Left" takes the Left channel from the A input and passes it to the Left channel of the output, and the Left channel from the B input and passes it to the Right channel of the output.

Additional details, with examples, can be obtained by following the "click here" link from the iPort while selecting your Mode.

SDP File: An SDP file may be supplied and uploaded using the "Choose File" and "Import" buttons.

SDP document: Alternatively, the textual content of an SDP file may be copied and pasted in to this box and applied by clicking "Import"

Source Name: Text field used to describe the input

Address: This is used to manually define the Livewire source channel or AES67 address. Note that this supports Livewire, AES67, Unicast and AES67 SIP. Usage is shown in this screen shot and on the Web UI.

Stream Type: This defines the type of stream to be generated from the matrix output. Similar to the input type field, there are many options available. Common AES67 and Livewire types are listed. The most common is L24 2ch (Livewire From source).

Packet time: This defines the latency and bandwidth consumption of the stream. Livewire has historically given this field a user-friendly name such as Live or Standard stereo. Packet times are now listed with their more common names referenced in parenthesis.

What's Next?

VMix and VMode are two of the features that make Zephyr iPort PLUS an even more versatile product. We've explained the fundamentals of each, and given you some examples of what to do with them. Hopefully, we've also brought out your inner hacker, and you'll develop something that solves an issue that is unique to your installation. If so, be sure to tell us about it.

Diagnostics and Stream Statistics

Diagnostics

The Diagnostics page offer information and functions useful for checking Zephyr iPort PLUS for proper operation, and for troubleshooting should problems arise. There are three ways that you can be alerted to problems and obtain diagnostics.

Syslog – Problems can be logged to an external standard Linux-style syslog device. Syslog is mostly intended for our engineers. Should you have a problem that might be diagnosable via these logs, our support people will ask you to download and email us the files.

In the Diagnostics page, you can set the Syslog severity level filter to determine how much detail you want. On the low end of the scale, 'Emergency: system is unusable', will only bother you when the system has crashed. At the other end of the spectrum is 'Notice: normal but significant condition', for a more proactive information flow. Syslog error events and thresholds are set up individually for each codec under the Options section.

SNMP – Zephyr iPort PLUS uses SNMP (Simple Network Management Protocol) to provide diagnostic information about software version and product ID, device status, Livewire interface and codecs.

Internal Log - Zephyr iPort PLUS can log problems with various levels of detail to either an internal or external file that can be downloaded and inspected. The internal file is the easiest to use. Simply download it with a web browser. Note however that this file is really a loop in RAM, meaning that when it is full, it starts to overwrite the old records. About 10K lines of text can be recorded. Note also that this log is in volatile memory, so it will be lost after a reboot.

On the other hand, the external file requires a simple client to be run on a PC, but the amount of data is not limited, and the log will not be lost when Zephyr iPort PLUS is rebooted.

All three of these options are configured from the Diagnostics page.

Host name:	iPort8085F3
Local time:	Mon Feb 9 15:00:48 2015

Syslog Configuration	
Syslog server (IP address):	0.0.0.0
Syslog severity level filter:	Emergency: system is unusable
	Apply

SNMP Configuration	
Target network manager IP:	0.0.0.0
Target network manager port:	0

Additionally, if you purchased the Content Delay option, there is a Storage Maintenance section where you can check up on the health of the solid state drive, scan, format storage, and delete unwanted audio content.

SNMP Configuration

SNMP (Simple Network Management Protocol), enables a management workstation or another SNMP client application on the same LAN to communicate directly with Zephyr iPort PLUS, and for the iPort to initiate an alarm on the network. Although the S in SNMP stands for Simple, the reality can be anything but. SNMP is a complex utility, and the ground rules really depend on whatever SNMP Manager you are using. Be sure to read and understand the instructions for your SNMP Manager. Having done all of that, it may also be a good idea to have a current networking reference textbook close at hand.

Zephyr iPort PLUS will interface directly with an SNMP Manager. It communicates through an MIB (Management Information Base) The MIB is a definition file that presents the management data in a more human-friendly manner. Let's look at an example.

The raw data is ordered in a tree structure, where every item has a tree-based identifier, such as "1.3.6.1.4.1.42463.1.11", for example. SNMP protocol itself can only deliver these raw numeric id's. The MIB file associates meaning with the position in the tree. Now, if the same MIB is used at the both ends, you can better understand what's going on. For example, if your SNMP client does not have MIB, it will display:

1.3.6.1.4.1.42463.1.11 = "2.0.0a"

On the other hand, if the Zephyr iPort PLUS MIB is uploaded to the SNMP client, it will display:

active software version = "2.0.0a"

Note that with the current firmware version, Zephyr iPort PLUS does not make its MIB file available for downloading.

The following text summarizes the contents of the MIB file in Zephyr iPort PLUS:

internet

 mgmt

 mib-2

 private

 enterprises

 42463- PEN: Telos Corp.

 -1- system

 -10- product id: string

 -11- active software version: string

 -2- device status

 -2- traps

 -10- operational state change (var: operational state)

 -11- CPU temp normal (var: CPU temp)

 -12- CPU temp high (var: CPU temp)

 -13- CPU load normal (var: CPU load)

 -14- CPU load high (var: CPU load)

 -15- CPU load critical (var: CPU load)

 -10- operational state: Integer {ok(1), general error(2)}

 -11- CPU temp: Integer

 -12- CPU load: Integer

 -3- Livewire interface

 -2- traps

 -10- link state change: (var link state)

-11- sync state change: (var sync state)

-10- link state: Integer {fdx(1), unplugged(2), hdx(3)}

-11- sync state: Integer {locked(1), not locked(2), no sync signal(3)}

-12- link speed: Integer {100Mbs(1), 1Gbs(2)}

-13- link load in: Integer

-14- link load out: Integer

-10- codecs

-11- cdNumber: Integer

-12- cdTable

-1- cdEntry

-1- index: Integer

-2- traps

-10- Encoder LW receive OK (var: index)

-11- Encoder LW receive lost (var: index)

-12- Encoder LW audio OK (var: index)

-13- Encoder LW audio lost (var: index)

-14- Encoder LW buffer error (var: index)

-15- Encoder general alarm (var: index)

-30- Decoder WAN receive OK (var: index)

-31- Decoder WAN receive lost (var: index)

-32- Decoder LW audio OK (var: index)

-33- Decoder LW audio lost (var: index)

-34- Decoder WAN jitter alarm (var: index)

-35- Decoder WAN sequence alarm (var: index)

-37- Decoder general alarm (var: index)

-10- encoder

-10- operational state: Integer {on(1), off(2)}

-11- LW receive state: Integer {ok(1), lost(2)}

-12- LW audio state: Integer {ok(1), lost(2)}

-11- decoder

-10- operational state: Integer {on(1), off(2)}

-11- WAN receive state: Integer {ok(1), lost(2)}

-12- LW audio state: Integer {ok(1), lost(2)}

The Zephyr iPort PLUS SNMP extension agent requires Windows SNMP feature to be installed. To install this feature, navigate to Start, Control Panel, Programs and Features and select Turn Windows features on or off. Select "Simple Network Management Protocol (SNMP)" and press OK to install it.

Once the Windows SNMP feature is installed, it needs to be configured. Open Windows Services control panel by navigating to Start, Control Panel, Administrative Tools and with administrative privileges open Services item:

Locate the "SNMP Service" item, right-click and select Properties.

Enter Contact and Location information in the Agent tab and select the SNMP Service type.

Configure the access information in the Security tab. Use community "public" and READ_ONLY rights to set up a public access.

If needed, set up an IP address which will receive the generated traps in the Traps tab.

Press Apply. You may also need to restart the SNMP Service for these changes to take effect.

SNMP Security

SNMP 'Communities' serve as passwords for this function. Go into the Read: and Write: boxes to enter text that is specified for the read/write strings on the SNMP Manager monitoring device. As with similar text-entry screens, execute a Save after entering.

SNMP Traps

When Zephyr iPort PLUS initiates an alarm as opposed to being queried or polled for one, the alarm is known as a 'trap.'

SNMP Ports

The default SNMP Ports, General Port 161 and Trap Port 162, are the default for most SNMP operations. These may be changed as needed in the usual way.

Stream Statistics

The Stream Statistics page gives you a summary of key metrics for: encode and decode streams, Livewire receive streams, Virtual mixer inputs and Vmode External inputs. It will keep a running tally until you click

the Clear Counters button. If you would like a hard copy you can save the page as an HTML document, or

Host name: iPort8085F3 [Clear counters]

Codec Receive Streams

	Session Count:	Decoder Changes:	Bitrate Changes:	Sample Rate Changes:	Packets Received(A/B) (%):	Packets Received A (%):	Packets Received B (%):	Packets Expected:	Small Packets:	Lost Packets:	Overflow Packets	Underrun Events	Lost Samples:	Samples Decoded
Decoder 1	0	0	0	0	0 (0.00)	0 (0.00)	0 (0.00)	0	0	0	0	0	0	0
Decoder 2	0	0	0	0	0 (0.00)	0 (0.00)	0 (0.00)	0	0	0	0	0	0	0
Decoder 3	0	0	0	0	0 (0.00)	0 (0.00)	0 (0.00)	0	0	0	0	0	0	0
Decoder 4	0	0	0	0	0 (0.00)	0 (0.00)	0 (0.00)	0	0	0	0	0	0	0
Decoder 5	0	0	0	0	0 (0.00)	0 (0.00)	0 (0.00)	0	0	0	0	0	0	0
Decoder 6	0	0	0	0	0 (0.00)	0 (0.00)	0 (0.00)	0	0	0	0	0	0	0
Decoder 7	0	0	0	0	0 (0.00)	0 (0.00)	0 (0.00)	0	0	0	0	0	0	0
Decoder 8	0	0	0	0	0 (0.00)	0 (0.00)	0 (0.00)	0	0	0	0	0	0	0
Decoder 9	0	0	0	0	0 (0.00)	0 (0.00)	0 (0.00)	0	0	0	0	0	0	0
Decoder 10	0	0	0	0	0 (0.00)	0 (0.00)	0 (0.00)	0	0	0	0	0	0	0
Decoder 11	0	0	0	0	0 (0.00)	0 (0.00)	0 (0.00)	0	0	0	0	0	0	0
Decoder 12	0	0	0	0	0 (0.00)	0 (0.00)	0 (0.00)	0	0	0	0	0	0	0
Decoder 13	0	0	0	0	0 (0.00)	0 (0.00)	0 (0.00)	0	0	0	0	0	0	0
Decoder 14	0	0	0	0	0 (0.00)	0 (0.00)	0 (0.00)	0	0	0	0	0	0	0
Decoder 15	0	0	0	0	0 (0.00)	0 (0.00)	0 (0.00)	0	0	0	0	0	0	0
Decoder 16	0	0	0	0	0 (0.00)	0 (0.00)	0 (0.00)	0	0	0	0	0	0	0

Codec Transmit Streams

	Block Count:	Byte Count:	Overflow Events:	Overflow Bytes:	Bytes Written:
Encoder 1-1 (path A)	0	0	0	0	0
Encoder 1-2 (path A)	0	0	0	0	0
Encoder 1-3 (path A)	0	0	0	0	0
Encoder 1-4 (path A)	0	0	0	0	0
Encoder 2-1 (path A)	0	0	0	0	0
Encoder 2-2 (path A)	0	0	0	0	0
Encoder 2-3 (path A)	0	0	0	0	0
Encoder 2-4 (path A)	0	0	0	0	0
Encoder 3-1 (path A)	0	0	0	0	0
Encoder 3-2 (path A)	0	0	0	0	0

What's Next?

If you've read this manual straight through from Chapter 1, congratulations, you made it to the end. You have learned just about everything there is to know about this subject, and are hereby awarded a Ph.D. in Zephyr iPort PLUS!
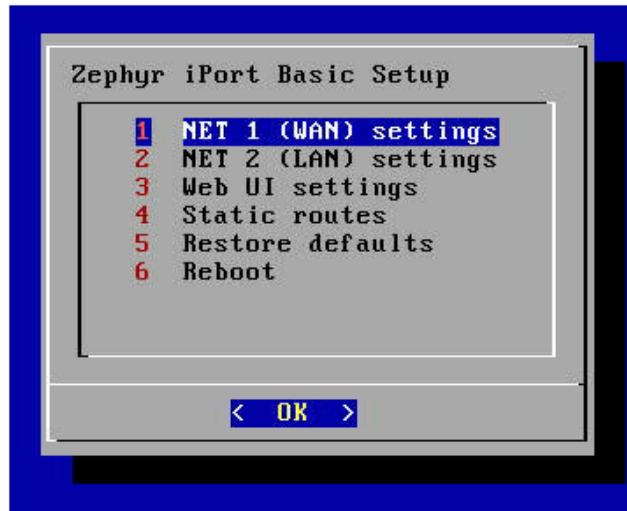
Alternate Method for Initial IP Addressing

This configuration method requires using a local VGA monitor and USB Keyboard for setting the initial IP addresses.
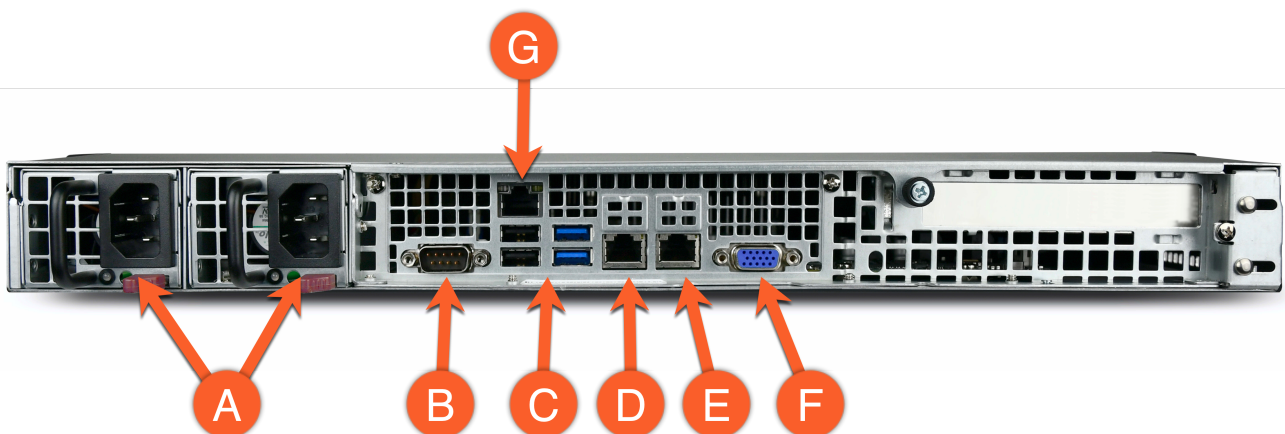
1. Connect a VGA monitor to the VGA port on the rear of the iPort HD (F)
2. Connect a USB keyboard to any USB port (C)

The Basic Setup console is displayed on the monitor.

Set or change the NET 1 (WAN) address

> ⓘ Use the **UP** and **DOWN** arrow keys to move between fields. Use the **BACKSPACE** or **DELETE** keys to remove any old IP addresses. Use the **TAB** key to change menu options.
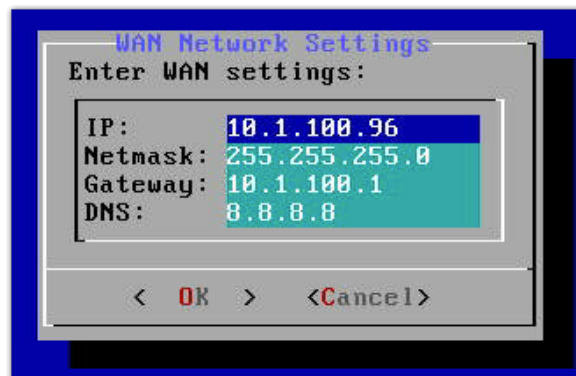
1. Using the keyboard, select item 1, `NET 1 (WAN) settings` from the menu and press `Enter.`
2. Select item 2, `Configure network settings` and press `Enter.`
3. Using the keyboard, set the IP, Netmask, Gateway, and DNS.
4. Select `< OK>` when finished.
5. Press `Enter` to accept the "WAN settings are updated" notice.
6. Select `<Done>` on the WAN Network Menu

Set or change the NET 1 (LAN) address
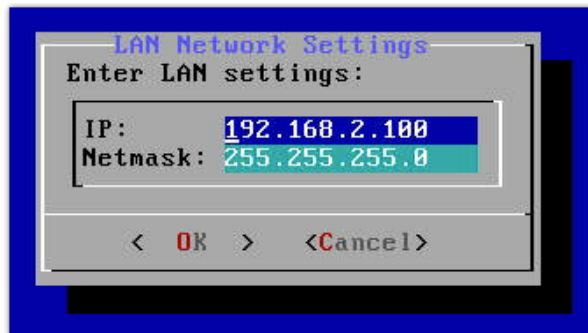
1. Using the keyboard, select item 2, `NET 2 (LAN) settings` from the menu and press `Enter.`

2. Select item 2, `Configure network settings` and press `Enter.`

3. Using the keyboard, set the IP and Netmask. (Gateway and DNS not supported on LAN.)

4. Select `< OK>` when finished.

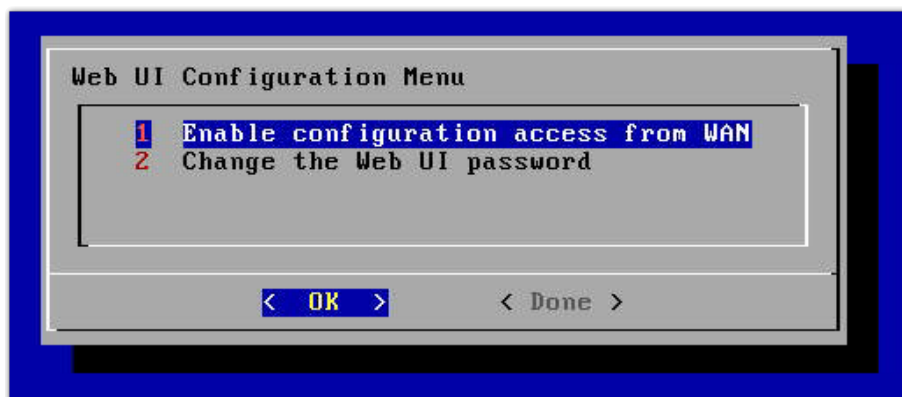5. Press `Enter` to accept the "LAN settings are updated" notice.

6. Select `<Done>` on the LAN Network Menu

(Optional) Web UI Configuration

Configuration must be enabled to allow configuration from the WAN interface. The Web UI configuration menu allows you to enable or disable this option and change the Web UI password. If configuration from the WAN is not required, you may skip this step. Access can be allowed later via the Web interface.

1. Using the keyboard, select item 3, `Web UI settings` from the menu and press `Enter.`

2. Select item 1, `Enable configuration access from WAN` and press `Enter.`

3. Select <Yes> or <No> to enable or disable access.

4. Select `< OK>` when finished.

5. Select `<Done>` on the Web UI Configuration Menu.

Similarly, use this menu to set (or Reset) the web UI Password.

If you got here from the Quick Start Guide, click here to go back to .

Specifications and Warranty

Specifications

Audio

Zephyr iPort PLUS has no native audio I/O, operating on streams provided by attached Livewire audio devices. All audio specifications below are representative of Axia Livewire audio interfaces.

Analog Line Inputs

- Input Impedance: >40 k ohms, balanced
- Nominal Input Range: Selectable, +4 dB 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 dB(-20 dB FSD)

- Impedance: 110 Ohm, balanced (XLR)
- Signal Format: AES3 (AES/EBU)
- AES3 Input Compliance: 24-bit with selectable sample rate conversion, 32 kHz to 96 kHz input sample rate capable.
- AES3 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

Frequency Response

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

Network

- 1 LAN port, 1 WAN port; 100/1000-BaseT Ethernet interfaces

Codecs

- Standard AAC, high-efficiency AAC-HE (aacPlus), AAC-HEv2, low-delay AAC-LD, MP3, MP2.
- Optional: aptX from CSR.

Power

- Dual-redundant internal auto-ranging power supplies, 90 – 132 / 187 – 264 VAC, 50Hz/60Hz.
- Power consumption: 100 Watts.

Warranty

Telos Alliance 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 or on behalf of TLS Corp. and its affiliated companies, collectively doing business as The Telos Alliance (hereinafter "Telos").

With the exception of software-only items, the Products are warranted to be free from defects in material and workmanship for a period of two (2) years from the date of receipt of such Product by the end-user (such date of receipt the "Receipt Date"). Software-only items are warranted to be free from defects in material and workmanship for a period of 90 days from the Receipt Date. Telos will repair or replace (in its discretion) defective Products returned to Telos within the warranty period, subject to the provisions and limitations set forth herein.

This warranty will be void if the Product: (i) has been subjected, directly or indirectly, to Acts of God, including (without limitation) lightning strikes or resultant power surges; (ii) has been improperly installed or misused, including (without limitation) the failure to use telephone and power line surge protection devices; (iii) has been damaged by accident or neglect. As with all sensitive electronic equipment, to help prevent damage and or loss of data, we strongly recommend the use of an uninterruptible power supply (UPS) with

all of our Products. Telos products are to be used with registered protective interface devices which satisfy regulatory requirements in their country of use.

This Warranty is void if the associated equipment was purchased or otherwise obtained through sales channels not authorized by Telos.

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

In no event will Telos, its directors, officers, employees, agents, owners, consultants or advisors (its "Affiliates"), or authorized dealers or their respective Affiliates, 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, the Product must be registered via Telos' website (found at: http://telosalliance.com/legal/warranty) at time of receipt by end-user and notice of a warranty claim must be received by Telos within the above stated warranty period and warranty coverage must be authorized by Telos. Contact may be made via email: support@telosalliance.com or via telephone: (+1) 216-241-7225. If Telos authorizes the performance of warranty service, the defective Product must be delivered to: Telos, 1241 Superior Avenue, Cleveland, Ohio 44114 or other company repair center as may be specified by Telos at the time of claim.

Shipping Costs and Warranty Service:

If the date the customer's notice of warranty claim is received by Telos (such date the "Warranty Claim Notice Date") is within the first 90 days following the Receipt Date, Telos will pay the costs of shipping such warranted Product to and from the end user's location, and the cost of repair or replacement of such warranted Product.

If the Warranty Claim Notice Date occurs after the first 90 days following the Receipt Date and before the end of the second (2nd) year, the customer will pay the freight to return the warranted Product to Telos. Telos will then, at its sole discretion, repair or replace the warranted Product and return it to the end user at Telos' expense.

If the Warranty Claim Notice Date occurs between the end of the second (2nd) year following the Receipt Date and the completion of the fifth (5th) year, the customer will pay the costs of shipping such warranted Product to and from the end user's location. Telos will then, in its sole discretion, repair or replace the warranted Product at Telos' expense. Telos also reserves the right, if it is not economically justifiable to repair the warranted Product, to offer a replacement product of comparable performance and condition direct to the customer at a discounted price, accepting the failed warranted Product as a trade-in.

The end user will in all cases be responsible for all duties and taxes associated with the shipment, return and servicing of the warranted Product.

No distributor, dealer, or reseller of Telos products is authorized under any circumstances to extend, expand or otherwise modify in any way the warranty provided by Telos, and any attempt to do so is null and void and shall not be effective as against Telos or its Affiliates.

Out of warranty units returned to the factory for repair may be subject to a $500 evaluation fee, which fee must be prepaid prior to shipping the unit to Telos. If no repairs are required, the $500 fee will be retained by

Telos as an evaluation charge. If repairs are required, the $500 fee will be applied to the total cost of the repair.

To activate your product warranty, visit http://www.telosalliance.com/product-registration .