

ZEPHYR® IPORT PLUS

Multi-CODEC Gateway



USER'S MANUAL

Manual Version 3.0 for software version 1.6.1g and later

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iPort version 2.2.1i

Telos®

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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 autoranging 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 fuse only with same type and value for continued protection against fire.

WARNING:

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

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

This equipment requires the free flow of air for adequate cooling. Do not block the ventilation openings 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 durchgeführt, 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 Benutzung 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.

Zephyr iPort PLUS USER MANUAL

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WARRANTY

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

UPDATES

The operation of Fusion is determined largely by software. We routinely release new versions to add features and fix bugs. Check the Telos Alliance web site 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 Zephyr iPort PLUS, 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 The Telos Alliance before returning any equipment for factory service. We will need your unit's serial number, located on the back of the unit. We will issue a return authorization number, which must be written on the exterior of your shipping container. Please do not include cables or accessories unless specifically requested by the Technical Support Engineer. Be sure to adequately insure your shipment for its replacement value. Packages without proper authorization may be refused. US customers, please contact Telos Alliance Technical Support at +1-216-622-0247. All other customers should contact local representative to make arrangements for service.

WE SUPPORT YOU...

BY PHONE / FAX:

- ◆ 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 6:00 PM, USA Eastern time, Monday through Friday.
- ◆ Our Fax number is +1-216-241-4103.

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 web site has a variety of information which may be useful for product selection and support. The URL is www.TelosAlliance.com.

REGISTER YOUR PRODUCT

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

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Creating the Most Exciting and Engaging Audio Experiences Imaginable

Congratulations on your new Telos Alliance product!

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

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

The Telos Alliance

1 Introduction

Welcome to Zephyr iPort PLUS

Welcome to Zephyr iPort PLUS, a Livewire®-to-MPEG/-aptX® Enhanced/-linear PCM gateway that enables the transport of multiple channels of stereo, mono, and dual mono audio across any QoS-enabled IP network, including T1 and T3 connections, as well as private WANs with MPLS (Multi-Protocol Label Switching). It is the perfect solution for large-scale distribution of audio to single or multiple locations.

Zephyr iPort PLUS represents the natural evolution of the original Zephyr iPort, with several additions and enhancements, including:

- ◆ Sixteen bi-directional IP-audio codecs, each capable of compressed or linear audio transport at bit rates ranging from 24 kbps to 2.4 Mbps.
- ◆ Up to eight IP stream destinations per encoder.
- ◆ UDP, TCP and Multicast stream types independently configurable per WAN stream.
- ◆ SHOUTcast interface support.
- ◆ Associated data transmission via three uni-directional (RS-232 style) user data channels per each audio direction per codec.
- ◆ Optional SSD (Solid State Drive) Content Delay with dynamic storage space allocation, configurable per codec with synchronized delay of GPIO and user data channels.
- ◆ NTP synchronization for content delay on absolute time.
- ◆ Increased GPIO capacity to 20 independent uni-directional contact closures per codec per transmit direction.
 - ◇ Virtual endpoints, with all the same logic as attached to hardware circuits.
 - ◇ Cut-in logic for snake between hardware GPIO nodes.
 - ◇ Cut-in logic for Element to hardware GPIO link.
 - ◇ Element console emulation logic.
 - ◇ GPIO node emulation logic.
 - ◇ Remote status signaling and control using virtual GPIO pins.

Quick Start

Register and Sign Up

We know. You're anxious to mount your new Zephyr iPort PLUS in the rack and get it running. But first, please take a moment to register your product online. Registration activates the warranty, and is the best way to assure you will get quality support should you have questions regarding installation, operation, or troubleshooting. Visit <http://telosalliance.com/product-registration/> to register.

A copy of The Telos Alliance 5-Year Limited Warranty is included at the back of this manual.

While visiting our web site, you may also want to sign up for our Direct Current e-mail, so you'll receive notification of new firmware upgrades available for your Zephyr iPort PLUS.

Installation

Zephyr iPort PLUS mounts in a 2 RU high space in a standard 19-inch rack. The unit will operate in any environment where the stirred air temperature around the unit is between 0 and 40 degrees Celsius (that's 32 to 104 degrees Fahrenheit) with a relative humidity of 0 to 98 percent (non-condensing).

Power is supplied by fanless, dual-redundant internal auto-ranging power supplies, 90-132 / 187-264 VAC, 50-60 Hz. These supplies are auto-switching and easily field-replaceable for maximum uptime. Power consumption is 100 Watts. Normally, the PWR and FUSE LEDs on each of the power supplies will light up green. They will glow red if the power supply has failed or the fuse is open.

Connections



The rear panel of Zephyr iPort PLUS is simple, thanks to Livewire AoIP I/O. A single Ethernet cable is all that's needed for all inputs, outputs, GPIO, and remote control. Uncompressed 24-bit/48 kHz audio goes in from your network via Ethernet, and compressed MPEG, aptX® Enhanced (optional), or linear PCM streams go out on the same cable – no need for converters and connectors. Alternately, you can use the separate WAN connection to send your audio over an outside network.

For both Ethernet jacks, the Ethernet mode is set permanently to AUTO, so the other end must be, as well. (Problems can result if the other end is set to a fixed 10, 100, or 1000BT.)

Usually, these jacks are connected to an Ethernet switch or an IP router. A PC is connected to the same logical network to support the Web control interface. An Ethernet cross-cable can be used for a temporary direct connection to a PC, allowing configuration.

CAT-5e cables may be used for the Livewire connection, though we recommend CAT-6 when possible. The 100/1000 connection may be CAT-5e if used with a 100BT port at the other end, but must be CAT-6 when the other end is 1000BT (gigabyte).

You'll also notice that there are two USB ports and a video out connector (labeled DVI). These are available for upcoming features that can be added with firmware upgrades. Also, you may be asked by a tech support engineer to plug in a keyboard and monitor if there are issues with the unit's Linux operating system.

Front-Panel OLED

The front-panel OLED display gives basic status indication. Together with the adjacent knob, it lets you enter IP values during initial setup.

Initial Start-Up

From the moment you plug Zephyr iPort PLUS in until it boots up takes just a minute or so. Once it is booted, you will see a screen that tells you the status of the power supplies, ambient temperature, and status of Livewire and Ethernet connections.

Configuring Zephyr iPort PLUS

Initial Network Configuration

Make your initial settings of the IP address using the front-panel OLED display and jog/select wheel. This front panel configuration controls the Ethernet port marked “Livewire” on the rear panel. Set this to an IP address compatible with your Livewire network. Also set the proper Netmask address. (typically 255.255.255.0) . Be sure to highlight and click the checkmark at the right-hand side of each IP address field.

You may complete the rest of the setup and configuration using a web browser, as we discuss below.

After you have made your initial settings via the front-panel OLED, connect a PC to the iPort, either using a cross-connect cable, or through an Ethernet switch.

Web Interface

Zephyr iPort PLUS contains a Web server to support a PC with Web browser for configuration and control. We've fully tested it with Firefox, and to a lesser extent, Internet Explorer and Chrome. Other browsers may work as well, but we can't guarantee it. Java must be installed in your browser so that dynamic functions such as sensitive pieces of the web cfg pages work properly.

Point your browser to the iPort by entering the unit's IP address, like this: <http://192.168.1.7>

You will be asked for a username and password. The defaults are:

- ◆ User name: user
- ◆ Password: <blank>

Next, you will see a top-level menu with the following links:

- ◆ Codec configuration - Enable / disable codec instances, assign Livewire channels and link addresses, set options, display status.
- ◆ V-Mixer and V-mode - Display virtual mixer and V-Mode module settings.
- ◆ Stream Statistics - Display incoming stream performance statistics.
- ◆ Network - Set host name, IP address, subnet mask and gateway IP address.
- ◆ System - Display system and network status. Configuration backup / restore / reset to factory defaults. Firmware setup. Restart.
- ◆ Diagnostics - View and capture system diagnostics info.

Configuration

The first step is to configure the network parameters, so let's click on the Network link to open that page.

Host Name	
Host name:	<input type="text" value="iPort400021"/>
Domain name syntax - series of labels concatenated with dots. Labels may contain only letters, digits, and hyphens, and must start with a letter or digit.	
Livewire interface	
Network address:	<input type="text" value="192.168.2.150"/>
Netmask:	<input type="text" value="255.255.255.0"/>
MAC address:	<input type="text" value="58:50:ab:40:00:21"/>
WAN interface	
Network address A:	<input type="text" value="10.1.100.99"/>
Netmask A:	<input type="text" value="255.255.255.0"/>
Gateway A:	<input type="text" value="10.1.100.1"/>
VLAN number for path A: (Leave blank to disable tagging)	<input type="text" value=""/> (Untagged)
Use Ethernet port:	<input type="text" value="WAN (88:88:88:88:87:88)"/> ↕
Enable redundant WAN path:	<input type="checkbox"/>
Network address B:	<input type="text" value="0.0.0.0"/>
Netmask B:	<input type="text" value="255.0.0.0"/>
Gateway B:	<input type="text" value="0.0.0.0"/>
VLAN number for path B: (Leave blank to disable tagging)	<input type="text" value="0"/>
Use Ethernet port:	<input type="text" value="WAN (88:88:88:88:87:88)"/> ↕
Enable configuration access from WAN:	<input checked="" type="checkbox"/>
DNS Server:	<input type="text" value="75.75.75.75"/>
Warning: all changes except host name take effect after restart. Attempts to set network address and netmask to 0.0.0.0 will not be accepted.	
<input type="button" value="Apply"/>	

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 'ZEPHYR' 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 Zephyr iPort PLUS, you may configure two IP addresses (A and B) on the same WAN port or on each 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. This will normally be the IP address of the router used to access the Internet or other WAN.

Redundant WAN Paths may be used 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

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

From the main menu, go to the **Options** page. Under **NTP Synchronization**, enter the IP address for primary and secondary NTP servers. Click the Synchronize clock with NTP server box and click **Apply**. You should see the NTP status change to 'Reached, SYNC'.

Under **Time zone configuration**, set your GMT offset and click **Apply**.

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 **System** page. You can do this for both the front panel OLED/knob and Web access.

Default for the OLED/knob is <blank>, meaning no password is needed.

Default for the Web interface is also <blank>.

Backing-up and Restoring Configuration

Backup and Restore of the iPort PLUS system configuration is done on the **System** page.

It can be time-consuming to enter all the needed Zephyr iPort PLUS configuration values. So we give you a way to save them all in a file on your local PC. This can also be used to "clone" additional Zephyr iPort PLUS units after you have configured the first in a multiple Zephyr iPort PLUS installation. (Don't forget to change the Host name, IP numbers, etc. before connecting to your network...).

You can also reset Zephyr iPort PLUS to the factory defaults by using the **Reset cfg** button.

Restart

This is on the System page.



Clicking the Restart button causes Zephyr iPort PLUS to restart, while retaining all the current configuration values.

Firmware Upgrade

This is on the System page.



Warning: System will reboot after changing the current bank.

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 Zephyr iPort PLUS will be offline for 2 to 4 minutes while the firmware updates.

To Learn More

The TelosAlliance.com website has a wealth of information on Livewire technology, the wide range of Live-wire-equipped gear and partners, as well as information on MPEG codecs, networks, IP streaming, etc.

Our support team is standing-by 24/7 to answer your specific questions. In the USA, contact +1 216 241.7225. Worldwide, email support@TelosAlliance.com.

What's Next?

We've just covered the nuts and bolts of getting Zephyr iPort PLUS 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.

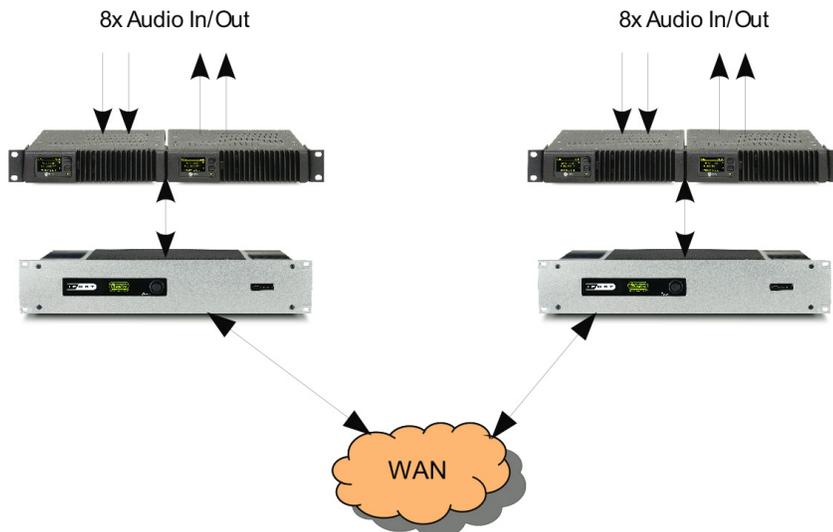
Since the main function of Zephyr iPort PLUS is sending high-quality audio from point A to point B, we'll next discuss how to do that with the MPEG or optional aptX® Enhanced encoders.

2 Audio Transport via MPEG or Optional aptX® Enhanced

Livewire is a professional-grade IP Audio system, used over controlled local area networks. 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 Zephyr iPort PLUS 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 Zephyr iPort PLUS'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.

Zephyr iPort PLUS 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. Most of the reduction comes from the compression process, but some comes from using larger packets with comparatively less header overhead. Zephyr iPort PLUS uses state-of-the-art MPEG codec technology to conserve network bandwidth, while preserving excellent audio quality.

The following tables list the codec types in Zephyr iPort PLUS, 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
	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
32 kbps		24 kHz	1152
40 kbps		24 kHz	1152
48 kbps		24 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
256 kbps		48 kHz	1152
320 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

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	
80 kbps		48 kHz	1024	
96 kbps		48 kHz	1024	
112 kbps		48 kHz	1024	
128 kbps		48 kHz	1024	
		160 kbps	48 kHz	1024

AAC-LD Codecs

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
	192 kbps	48 kHz	480

AAC-HE Codecs

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

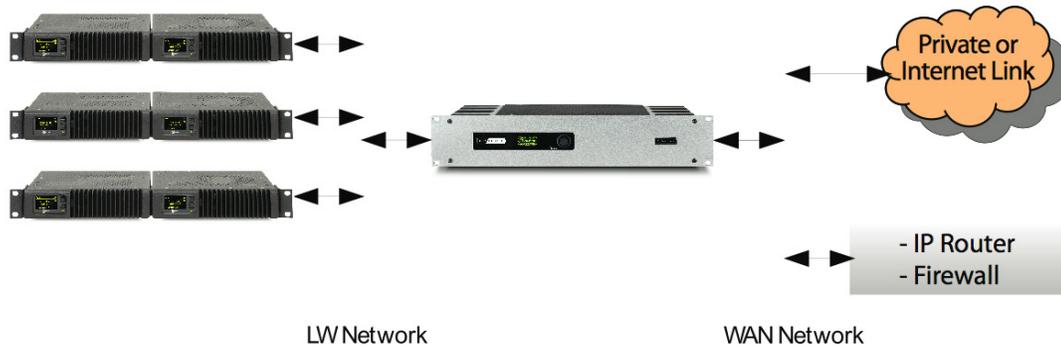
aptX® Enhanced Stereo/Mono Codec

Codec/Mode	Bitrate	Sample Rate	Input Samples per Channel
aptX® Enhanced Stereo (16 bit)	256 kbps	32 kHz	516
	384 kbps	48 kHz	512
aptX® Enhanced Stereo (24 bit)	384 kbps	32 kHz	516
	576 kbps	48 kHz	512
<hr/>			
aptX® Enhanced Mono (16 bit)	128 kbps	32 kHz	516
	192 kbps	48 kHz	512
aptX® Enhanced 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
<hr/>			
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. So, with 4 xNodes, you have a perfect correspondence with the channel capacity of Zephyr iPort PLUS. The xNode may 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 Zephyr iPort PLUS within a facility that is already Livewire-equipped, the xNode will not be required. Just connect the Livewire port of Zephyr iPort PLUS to an open Ethernet port in the Livewire system. Livewire sources may be from/to either xNodes 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.

Most Livewire installations will be like that 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. However, some Livewire installations will have the AoIP and WAN-accessible networks combined.

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:

Unit Configuration	
16 encoder / decoder channels:	<input checked="" type="radio"/>
16 encoder channels:	<input type="radio"/>
16 decoder channels:	<input type="radio"/>
<input type="button" value="Apply"/>	

Codec Receive Ports	
UDP unicast port number base - path A:	<input type="text" value="9150"/>
UDP unicast port number base - path B:	<input type="text" value="10150"/>
TCP unicast port number base:	<input type="text" value="9550"/>
UDP multicast port - path A:	<input type="text" value="8150"/>
UDP multicast port - path B:	<input type="text" value="8250"/>
Restore default port numbers	<input type="checkbox"/>
<input type="button" value="Apply"/>	

Content Delay Storage Space Management	
Maximum expected bitrate per stream (kbps):	<input type="text" value="576"/>
Content delay storage size: 0.00 Mb (~00:00 hours of audio)	
Content delay storage free space: 0.00 Mb (~00:00 hours of audio)	
Total configured time to store: 00:00 hours	
<input type="button" value="Apply"/>	

Content Delay Options	
Delay relative to packet arrival time, ignore sender timestamps	<input type="checkbox"/>
<input type="button" value="Apply"/>	

“Unit Configuration” options determine whether iPort will operate with 16 bidirectional codecs, or 16 encoders only, or 16 decoders only.

In the “Codec Receive Ports” section:

- ◆ “Port number base” determines 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. Note that if multiple WAN paths are used the A and B paths MUST have different base ports defined. The defaults are shown.
- ◆ “Mcast port” settings determine a single receive port number for all multicast traffic arriving over the path A or path B accordingly
- ◆ The “Restore default port numbers” option will reset all port numbers to the factory default values

The lower two sections of this page are used to configure the Content Delay function - see section 4 of this document.

Codec and Channel Settings

From the Codec Configuration page, click the Options button for the channel you are configuring to set the details of the codec and protocol parameters. Having done so, you will see this page:

Codec:	1		
Transmit			
Encoder type:	AAC		
Bitrate:	128 kbps		
Sample rate:	48 kHz		
Channel config:	Stereo		
DSCP Class of Service:	46 Expedited Forwarding (recommended)		
Output configuration			
Main output (path A):	Protocol type:	IP:	Port:
	UDP	10.1.100.99	9152
Main output (path B):	UDP	0.0.0.0	
Enable SHOUTcast protocol: (Non-SHOUTcast devices will stop receiving when this option is selected)	<input type="checkbox"/>		
External stream server password:	telos1		
Additional output (path A) 1:	Protocol type:	IP:	Port:
	mcast	239.192.100.101	5004
Additional output (path B) 1:	UDP		
Additional output (path A) 2:	UDP		
Additional output (path B) 2:	UDP		
Additional output (path A) 3:	UDP		
Additional output (path B) 3:	UDP		
Stop encoded stream in absence of LW input stream:	<input type="checkbox"/>		
Integrated stream server			
Enable stream server:	<input type="checkbox"/>		
Stream on LW interface:	http://192.168.2.150:8000/live1 Listen		
Stream on WAN interface:	http://10.1.100.99:8000/live1 Listen		

Transmit

Set the desired codec type, bitrate, and stereo/mono/dual mono mode 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 aptX® Enhanced encoder is licensed, you will also see the options for Enhanced 16-bit and 24-bit encoding.

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 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.

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.

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**.

Checking the box for the option to Stop encoded stream in the absence of LW 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 Output Configuration section allows you to specify the IP of the main output, as well as up to 3 more replicated streams.

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, read Chapter 6: Streaming.

SHOUTcast stream information

Source name:	<input type="text" value="stream1"/>
Homepage for the stream:	<input type="text" value="<empty_name>"/>
Genre:	<input type="text" value="<empty_name>"/>
IRC:	<input type="text" value="N/A"/>
AIM:	<input type="text" value="N/A"/>
ICQ:	<input type="text" value="N/A"/>

GPIO Each codec channel has a robust set of options for GPIO. They are described in Chapter 3: GPIO.

Receive

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

Receive

Buffering: 500 ms

Bitrate:

Sample rate:

Decoder type:

Protocol type (path A): UDP

Remote MCAST Address (path A):

Protocol type (path B): UDP

Remote MCAST Address (path B):

Syslog event configuration

Frame Error Rate (FER) threshold: FER events OFF

Frame Order Error Rate (FOER) threshold: FOER events OFF

Frame Receiving Jitter threshold: Jitter events OFF

The **Buffering** drop-down box lets you choose how much buffer is applied to the received streams. A lower value gives 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.

The **Protocol type** is set to 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. This both accepts your changes and takes you back to the main Codec page. Here are sample configurations for the first two channels:

Host name: iPort400021

Codecs Channels	Local Livewire	Local	Remote	
Enable:	Channel Type:	Source name Status: Audio:	Status: IP:Port	IP:Port
	(1...32766):	Source name (up to 16 chars):		
Codecs 1	<input checked="" type="checkbox"/> 11401 From source	Source OK ■ OK	multicast → multicast	Options
	<input type="checkbox"/> 15001 Live Stereo	ThisSource ■ ■ ■	0.0.0.0 : 8150 ← 0.0.0.0 : 0	Apply
Encoder: AAC, 128 kbps, sampling 48 kHz, Stereo, DSCP-46 Decoder: buffering 500 ms., sampling . . . mcast, UDP, DSCP:				
Codecs 2	<input type="checkbox"/> 11401 From source	<empty_name> ■ ■ ■	multicast → multicast	Options
	<input checked="" type="checkbox"/> 15002 Live Stereo	dec-src-name-2 OK ■ ■	10.1.100.99 : 9151 ← 0.0.0.0 : 0	Apply
Encoder: AAC, 128 kbps, sampling 48 kHz, Stereo, DSCP-46 Decoder: buffering 500 ms., sampling . . . UDP, UDP, DSCP:				

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 with your iPort PLUS, you can download one from our website. Indeed, the TelosAlliance.com site has a number of papers and other resources that could be useful to you.

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.

3 Using GPIO & Data with Zephyr iPort PLUS

As multiple audio signals are encoded in Zephyr iPort PLUS 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.

Warning! Encoder is OFF! GPIO Ports are working only on local side!

GPIO

Port	Protocol	Operation mode	Livewire channel or IP/Port
Port 1 (GPI 1)	iPort enhanced	Virtual Endpoint	
Port 2 (GPI 2)	Disabled	Virtual Endpoint	
Port 3 (GPI 3)	Disabled	Virtual Endpoint	
Port 4 (GPI 4)	Disabled	Virtual Endpoint	

Missing Plug-in

Status and control (Port 4)

	GPI 4	GPO 4
Pin 1:	(user mode)	(user mode)
Pin 2:	(user mode)	(user mode)
Pin 3:	(user mode)	(user mode)
Pin 4:	(user mode)	(user mode)
Pin 5:	(user mode)	(user mode)

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

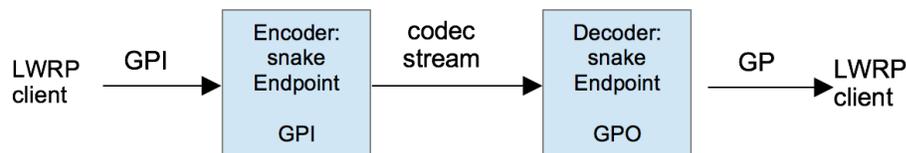
GPI – General Purpose Input

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.

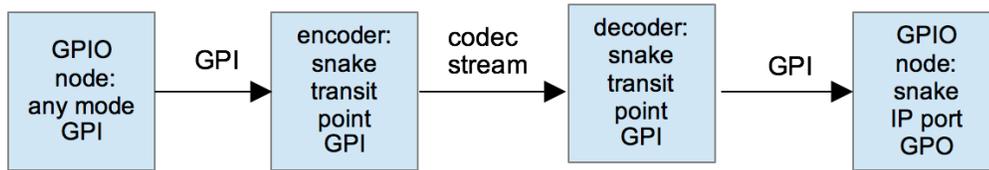
Both encoders and decoders monitor the GPIO multicast transport channel to receive GPIO state read requests, and they use multicast messages to report the states requested.

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.

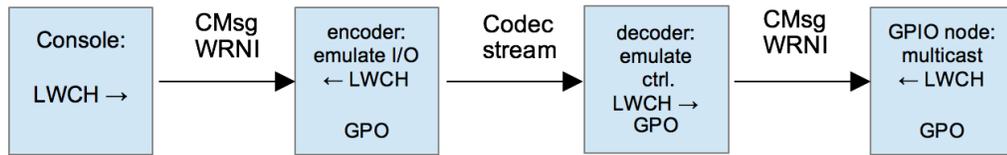
I/O Emulation

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 to GPIO node, emulation

Controller Emulation

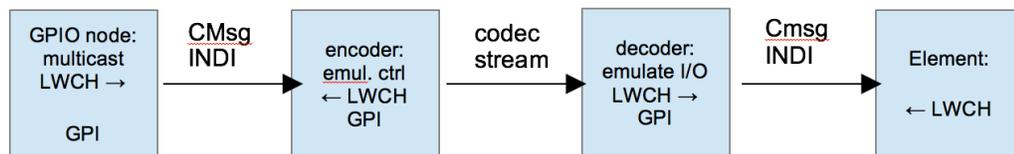
In this mode, iPort emulates local presence of a physically remote controller device, such as an. Element or iQ console.

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 to Element, emulation

Using the Associated Data Channels

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**.

Codec:

User data channel configuration

	Encoder	Decoder
Channel #1		
Status:	Not Established	Not Established
Enabled:	<input type="checkbox"/>	<input type="checkbox"/>
Protocol:	TCP Server ▾	TCP Server ▾
IP Address:	<input type="text"/>	<input type="text"/>
Port:	<input type="text" value="0"/>	<input type="text" value="0"/>
Bidirectional:	<input type="checkbox"/>	

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.

4 Using Optional Content Delay

If your requirements include recording content for playback and distribution to different sites at different times, you may want to order Zephyr iPort PLUS with the Content Delay option. Its features include:

- ◆ NTP synchronization for 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. It must be installed by the factory at the time you place your order.

To verify that your Zephyr iPort PLUS does indeed contain Content Delay, go to the **Options** page, and look at the bottom under **Feature Licensing**. Make sure that **Time-zone delay** is marked **Active**.

Content Delay Configuration

From the Codec Configuration page, click the Global Options button at the right top to set the following common parameters (bottom two sections of the page):

- ◆ “Maximum expected bitrate per stream” is used to convert the installed physical storage space to the total program time that the content delay feature will be able to store (summed per all codec channels).
- ◆ “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 arrival time at the receiver (option checked).

Configuration is done individually for each codec from the **Options** page. Scroll down to the bottom of the page, and use a 24 hour time format to enter the duration to store. A good analogy is to think of the Zephyr iPort PLUS as a pipe. The program comes in one end and goes out the other.

The ‘Duration to Store’ determines the total time length of the pipe that the program is traveling through, while the recording and playback process can continue indefinitely. The ‘playback offset’ determines 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 content creation instant. All calculations under the hood are based on UTC, regardless of local time-zone setting. Note that the resolution for playback offset goes to milliseconds, allowing you to correct for delays in digital transmission links.

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

Next, enter the receive stream bitrate, sample rate and codec type. When you’ve got everything the way you want it, remember to click the **Apply** button.

Content delay configuration

Enable content delay:

Duration to store (hh:mm): :
(Available: 184:49 hours)

Playback offset (hh:mm:ss.SSSSS): : : .

Content delay storage size: 55839.65 Mb (~197:04 hours of audio)
 Content delay storage free space: 54641.81 Mb (~192:51 hours of audio)
 Total configured time to store: 12:15 hours

Receive stream bitrate:	<input type="text"/>
Receive stream sample rate:	<input type="text"/>
Receive stream codec type:	<input type="text"/>

SSD Housekeeping

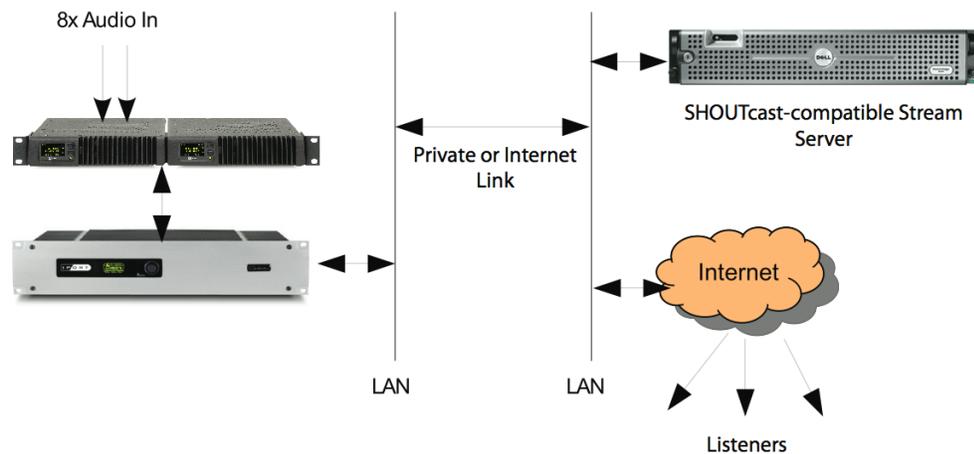
If you want to check up on the health of the Solid State Drive, make repairs, or format the drive, go to the **Diagnostics** page and click on the **Storage Maintenance** button.

5 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.

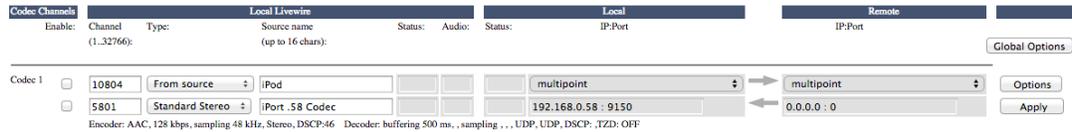
If you are planning to make really large scale public broadcasts via the Internet, you may want to host your broadcast through a Content Delivery Network (CDN).

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:

1. Enter the Livewire channel number you want to use as your source.
2. Enter the Livewire text name.
3. Enter the stream server IP number and port (the port must correspond to that set in the server's configuration).



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

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: *****

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

4. Enter the encoder type, bitrate, and other parameters.
5. Select 'TCP' as the protocol (used for the iPort to server link).
6. Click the box **Enable SHOUTcast protocol**, and enter a password that will be used by the server to authenticate the stream.
7. 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:	<input checked="" type="checkbox"/>
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:	<input type="text" value="stream1"/>
Homepage for the stream:	<input type="text" value="http://positiveenergy.com"/>
Genre:	<input type="text" value="disco"/>
IRC:	<input type="text" value="70s channel"/>
AIM:	<input type="text" value="N/A"/>
ICQ:	<input type="text" value="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>

6 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 in to 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 its 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 a 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.

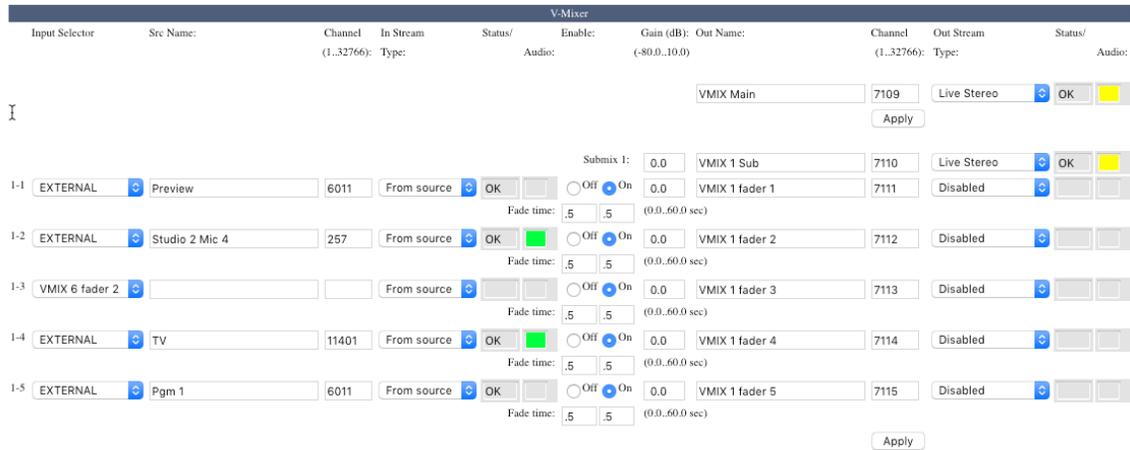
The various VMix outputs described above are sources that can feed your Livewire network and they can be manipulated in the same manner as any other audio source. A VMix source can be applied to a console fader, assigned to an audio node destination, or monitored by Pathfinder.

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 $-\infty$ 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

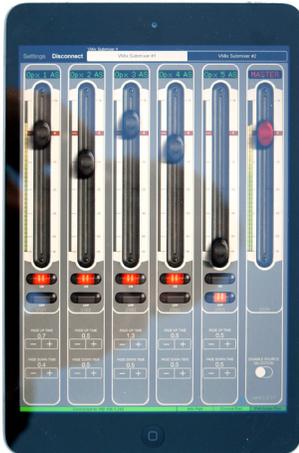
Now that you know how to enable and set up VMix submixers, what might you do with it? The possibilities are limited only by your imagination. To get your creative juices flowing, here are some examples.

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



Using third-party products like *Vmix for Axia* from BSI, you can remotely control the VMix channels in Zephyr iPort PLUS via your iPad. This gives studio talent or producers direct control of iPort’s VMix, without the need for a physical console. This app has been tested and verified by our engineers. There is one caveat, however. The wireless connection you are using **MUST** be on the Livewire network. It is not possible to “route” it from a different network as you can with an Axia Studio Engine or Power Station.

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.

Home | Codec configuration | V-Mixer | **V-Mode** | Stream statistics | Options | Network | System | Diagnostics

Host name: iPort400021

V-Mode							
Input Selector	Src Name	Address	Status/Audio	Audio Mode	Out Name	Address	Status/Audio
None				Pass stereo	V-Mode 1	Disabled	
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.

Input (Destination)

SDP file: No file chosen

SDP document:

Input selector: ▾

Source name:

Address:

Stream type: ▾

Using the "Address" field:

- Blank: Disconnected.
- Integer number: Channel number of Livewire multicast stream, range 1 to 32766. Standard RTP port number 5004 will be used.
- <Multicast IP>: Address of AES67 multicast stream, range 239.0.0.0 to 239.255.255.255. Standard RTP port number 5004 will be used.
- [0.0.0.0]:<port>: Unicast stream receive port number. A stream from any sender will be accepted.
- <Unicast IP>:<port>: Unicast stream sender, and the receive port number. Only a stream from the indicated IP address will be accepted.
- <AES67 SIP URI>: Initiate SIP connection. (Maximum length 299 symbols)

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

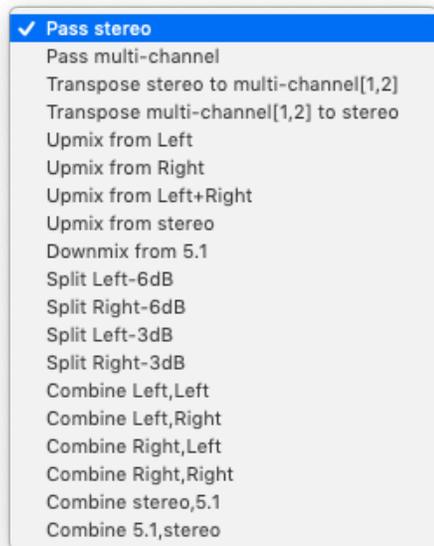
Valid range for RTP stream protocol is from 16384 to 65535

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

Default SIP port is 5060



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



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.

Output (Source)

SDP file: No file chosen

SDP document:

Source name:

Address:
AES67: sip:1@192.168.2.150

Stream type: Disabled (SIP connections possible) ▾

Note: Stream must be disabled to make SIP connections possible.

Packet time (ptime): 0.25ms (Livewire Live) ▾

Using the "Address" field:

Integer number	Channel number of Livewire multicast stream, range 1 to 32766. Standard RTP port number 5004 will be used.
<Multicast IP>	Address of AES67 multicast stream, range 239.0.0.0 to 239.255.255.255. Standard RTP port number 5004 will be used.
<Unicast IP>:<port>	Unicast stream destination address and port number.

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.

7

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:

Local time:

Syslog Configuration

Syslog server (IP address):

Syslog severity level filter:

SNMP Configuration

Target network manager IP:

Target network manager port:

Internal Log

disable select diagnostics trace detail level
 simple (default)
 medium
 detailed

Warning: after reset trace will be automatically set to default level.

 download diagnostics trace

 empty trace buffer

 open and view startup log file

Content Delay Storage Maintenance

 View storage health info, scan, format storage, delete audio content, etc.

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.

Storage health info

Content delay storage size: 55839.65 Mb (~197:04 hours of audio)
 Content delay storage free space: 54605.59 Mb (~192:43 hours of audio)
 Total configured time to store: 12:15 hours

Storage reconditioning

Check and repair content delay storage filesystem:
 During this operation all active content delay processes will be stopped.

Delete stored audio content

Data partition formatting

Format content delay storage:
 Warning! All stored content will be lost!

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)}
lost(2)
-11- LW receive state: Integer {ok(1),
-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 print it as a PDF file.

Host name: iPort8085F3		Codec Receive Streams														Clear counters
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	0	
Decoder 2	0	0	0	0	0 (0.00)	0 (0.00)	0 (0.00)	0	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	0	
Decoder 4	0	0	0	0	0 (0.00)	0 (0.00)	0 (0.00)	0	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	0	
Decoder 6	0	0	0	0	0 (0.00)	0 (0.00)	0 (0.00)	0	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	0	
Decoder 8	0	0	0	0	0 (0.00)	0 (0.00)	0 (0.00)	0	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	0	
Decoder 10	0	0	0	0	0 (0.00)	0 (0.00)	0 (0.00)	0	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	0	
Decoder 12	0	0	0	0	0 (0.00)	0 (0.00)	0 (0.00)	0	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	0	
Decoder 14	0	0	0	0	0 (0.00)	0 (0.00)	0 (0.00)	0	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	0	
Decoder 16	0	0	0	0	0 (0.00)	0 (0.00)	0 (0.00)	0	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!

A Specifications

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.

B Telos Alliance Warranty

Telos Alliance Limited Warranty



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

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