THE TELOS ALLIANCE

Telos® Zephyr® iPort PLUS Multi-CODEC Gateway

16 Stereo Codecs in a Livewire® Gateway

OVERVIEW

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

Zephyr iPort PLUS is the workhorse of codecs, configurable as eight stereo bi-directional MPEG codecs, or for encode / decode of up to 16 uni-directional stereo streams. Zephyr iPort PLUS connects to Axia® IP-Audio networks using a single CAT-6 cable for all I/O. Don't have a Livewire network yet? Pair Zephyr iPort PLUS with Telos Alliance® xNode audio interfaces for use as a standalone multiple-stream codec.

Coding algorithms include AAC, AAC-LD, HE-AAC (plus v2), MP2, MP3, linear, and optional aptX® Enhanced*. Bit rates range from 24 to 320 kbps for MPEG codecs, plus standard fixed rates for aptX and linear to over 2 Mbps. In addition, iPort offers dual, parallel-path end-to-end streaming for ultra-reliability and redundancy. Up to 20 unidirectional GPIO contact closures per codec are available in several modes to allow considerable flexibility of control. End to end GPIO is supported for each codec. For network operators, a unique Content Delay feature allows independent local storage and scheduled delayed playout of any or all coded audio channels for up to six hours.

FEATURES

- Distributes multiple channels of coded audio between broadcast facilities over QoS-enabled IP links.
- Configurable as a CODEC with 8 bi-directional channels, each with GPIO and PAD — or, as a 16-channel stereo encoder or 16-channel stereo decoder.
- 8 PCM Stereo channels are available for use simultaneously alongside CODEC channels, (dependent upon available bandwidth).
- Can also deliver streaming audio channels for Internet transmission via SHOUTcast, Steamcast or compatible stream replication server.
- Wide choice of genuine Fraunhofer codecs, including Standard AAC, high-efficiency AAC-HE (aacPlus), AAC-HEv2, low-delay AAC-LD, and MP3, with a choice of bit rates from 24 kbps to 320 kbps, definable per stream.
- Optional aptX Enhanced audio coding may be ordered at time of purchase or added later, as desired.
- When used as part of a Livewire network, allows audio from remote facilities to be used as if they were local sources, with associated logic and control.
- Eight 5-input Virtual Mixer (VMIX) channels each allow combining and mixing of up to 5 networked Livewire audio streams on a single channel.
- Eight Virtual Mode (VMODE) channels allow audio to be split into left/right channels, summed L+R, and more, prior to encoding and transmission.
- Content Delay option enables delayed playout of any or all selected receive audio channels, along with time-synchronized ancillary data, for up to six hours. Each playback delay time is independently configurable on a per-channel basis, making Zephyr iPort PLUS ideal for network operators, program distribution networks, or delayed playout of received audio at network-affiliated stations.
- Remote control/configuration via any computer with a standard Web browser.
- Separate LAN and WAN ports help ensure network security.
- Fanless, convection-cooled DSP-powered platform with dual-redundant, auto-switching powersupplies for maximum uptime. Power supply modules are field-replaceable in minutes.
- Optional Time Zone Delay upgrade allows the iPort to delay the playout of material for one, two, or several hours, facilitating its use over different time zones.
- AES67 Support for increased interoperability available in vMode.

IN DEPTH

Powerful, advanced program distribution and facility connection.

If your facility is like most, rack space is a precious commodity. That's why Telos® engineers invented Zephyr iPort PLUS, a sophisticated multiple-CODEC device that saves you money and rack space by housing 16 broadcast-quality stereo codecs in one 2RU device.

A pair of Zephyr iPort PLUS on each end of a QoS-controlled IP link can send and receive 8 channels of bi-directional stereo MPEG audio. Or, use iPort as a one-way "push" link to encode and deliver 16 channels of broadcast-quality one-way audio to a remote destination. With its ability to send multiple MPEG channels over IP connections, Zephyr iPort PLUS is perfect for audio transmission over VPNs, satellite links, Ethernet radio systems, and Telco or ISP-provided QoS-controlled IP services such as T1, T3 or OC-3 links.

You can use iPort for studio-to-transmitter links, network distribution systems, multi-channel links to remote studios. Install a QoS-enabled IP link between two studios with Axia Livewire networks, put an iPort at each end, and you can pass audio and GPIO between locations as if they were just next door. Paired with an appropriate server, you can even use Zephyr iPort PLUS to generate multiple channels of MP3 or AAC coded audio for Internet streaming, broadcasting to mobile phones, and audio distribution systems.

Finally, Zephyr iPort PLUS' exclusive Content Delay option (available at extra cost) adds hardware and software that enables delayed playout of select received audio channels. Associated GPO and ancillary data is likewise delayed and synchronized with audio. Delay any or all coded audio channels up to six hours; each channel's delay time is independently configurable.

The Zephyr iPort PLUS rear panel is remarkably simple, thanks to the use of Livewire AoIP I/O. A single Ethernet cable is all that's needed for all inputs, outputs, GPIO and remote control. Uncompressed 24-bit/48kHz audio goes in from your network via Ethernet; compressed MPEG streams go out on the same cable — eliminating expensive, space-consuming converters and connectors. Or, use the separate WAN connection to send your audio over an outside network.

If you don't have an Axia network yet, that's no problem — just pair Zephyr iPort PLUS with Telos VX analog or digital audio interfaces, or Telos Alliance xNodes, to make a standalone high-density audio codec package.

Zephyr iPort PLUS streams sound fantastic, thanks to our long-standing relationship with Fraunhofer IIS, the inventor of MP3 and co-inventor of AAC. The encoding algorithms inside iPort are genuine FhG, not no-name knockoffs. A full range of state-of-the-art codec types and bitrates are supported; the highest-quality implementations possible, run by a powerful Intel floating-point processor. Choose AAC-LD for delay-sensitive applications, AAC-HE and AAC-HEv2 for low bitrate requirements, standard MPEG AAC for best quality and resilience to packet loss at higher bitrates, MP3 and MP2 for legacy applications.

You'd expect all this to cost a lot, but it doesn't: we built Zephyr iPort PLUS on a single industrial motherboard, rather than the usual "multiple DSP cards in a frame" approach. Together with the Livewire-only audio interface, Zephyr iPort PLUS delivers more power than a legacy cardframe design, at only a fraction of the cost.

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 dBor -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
- AES67 Support for increased interoperability available in vMode

Frequency Response

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

Network

- 1 LAN port, 1 WAN port; 100/1000BASE-T Ethernet interfaces.

Codecs

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

Power

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

Regulatory

North America: FCC and CE tested and compliant, power supply is UL approved.

Europe: Complies with the European Union Directive 2002/95/EC on the restriction of the use of certain hazardous substances in electrical and electronic equipment (RoHS), as amended by Commission Decisions 2005/618/EC, 2005/717/ EC, 2005/747/EC (RoHS Directive), and WEEE.

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