Telos® iPort

Telos® iPort High Density Multi-Codec Gateway gives you more than ever before. License up to 64 codecs for distribution and contribution, in half the rack space of the previous iPort model.

Telos Alliance®

Telos® iPort High Density

Multi-Codec Gateway

Control with Pathfinder

AES67 Livewire+™

More Codecs, More Storage, Less Rack Space

The Multi-Codec Gateway that Gives You More

Telos® iPort is a Multi-Codec Gateway designed on an IP backbone that is used by broadcasters worldwide, featuring built-in resilience and reliability even with challenging connections. Worldwide networks use iPort for both Distribution and Contribution spanning multiple time zones.

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

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

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

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

Resilient, Reliable, Powerful

Features

- Transports multiple channels of stereo, mono, and dual-mono audio across IP networks
- Comes with eight bi-directional stereo codecs
- Codecs are configurable to run in multiple MPEG or linear PCM modes
- Up to four redundant IP stream destinations per encoder
- Unicast UDP and TCP, or UDP Multicast stream types, independently configurable per WAN stream
- Each codec independently configurable
- Additional codecs can be licensed up to a maximum of 64
- Enhanced aptX™ encoding optional
- Connects to existing Livewire® networks, allowing audio from remote facilities to be used as if they were local sources, with associated logic and control
- Pairs with Telos Alliance xNodes and an adequately configured ethernet switch for use as a standalone multi-stream codec
- Coding algorithms include:
  - AAC
  - AAC-LD
  - HE-AAC (plus v2)
  - MP2 (MPEG 1 layer 2)
  - MP3 (MPEG 1 layer 3)
  - Linear PCM
  - Optional Enhanced aptX™ (24 and 16 bit)
- Supported bit rates include:
  - 24 to 320 kbps for MPEG
  - Standard fixed rates for Enhanced aptX
  - 24-bit PCM at 48kHz, mono, or stereo
- Dual, diverse paths for ultra-reliability and redundancy
- SHOUTcast/Icecast formatting is offered by built-in streaming servers

License Up to 64 Codecs!

- Associated data transport:
  - MPEG-standard embedded ancillary data
  - Three transparent control and metadata channels per codec and direction
  - Up to 20 end-to-end GPIO channels per codec and direction
  - All data bundled and synchronized with the respective audio content
- Sixteen five-input Virtual Mixer (VMIX) channels each allow mixing of up to five networked Livewire audio streams on a single channel
- Sixteen Virtual Mode (VMODE) channels allow audio to be split into left/right channels, summed L+R, and more, and allow conversion of stream formats between AES67 and Livewire. VMODE channels can be used independently, or they can feed encoders and process decoder outputs as well
- Optional Content Delay with SSD-based dynamic storage space allocation, configurable per codec, with synchronized delay of GPIO and user data channels
- NTP synchronization for Content Delay on absolute time
- Remote control and configuration via any computer with a standard Web browser
- Remote status signaling and control using virtual GPIO pins
- SNMP monitoring with traps and attribute read supported
- Extended remote control and monitoring support via LWRP
- Separate LAN and WAN ports help ensure network security
- 1RU form factor chassis including rack rails
- Dual-redundant, auto-switching power supplies for maximum uptime. Power supply modules are field-replaceable
- Gigabit Livewire and WAN Ethernet interfaces

For iPort High Density + Content Delay

- Capabilities identical to iPort High Density
- Adds hardware and exclusive software to enable delayed playout of selected received audio channels
- Associated GPIO and ancillary data are likewise delayed and synchronized with audio
- Each playback delay time is independently configurable on a per-channel basis
- The total amount of hours, bitrate, and the number of streams depends on storage

Control and Configure via Any Web Browser

Powerful, advanced program distribution and facility connection.

If your facility is like most, rack space is a precious commodity. iPort High Density is a third-generation, sophisticated, multiple-CODEC device that saves you money and rack space by housing up to 64 broadcast-quality stereo codecs in one 1RU device.

A pair of iPort High Density connected via a QoS-controlled IP link can send and receive up to 64 channels of bi-directional stereo MPEG audio. Or, use iPorts as a one-way “push” link to encode and deliver up to 64 channels of broadcast-quality one-way audio to remote destinations. With its ability to send multiple MPEG channels over IP connections, iPort High Density is perfect for audio transmission over VPNs, satellite links, Ethernet radio systems, and Telco or ISP-provided IP services such SD-WAN, MPLS, or more traditional (legacy) data links such as T1.

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

Finally, iPort High Density’s exclusive Content Delay option (available at extra cost) adds hardware and software that enables delayed playout of select received audio channels. Associated GPIO and ancillary data are likewise delayed and synchronized with audio. Delay any or all coded audio channels; each channel’s delay time is independently configurable. Total storage time varies by configuration.

Uncompressed 24-bit/48kHz audio is ingested from your network via Ethernet; compressed MPEG (and/or optional Enhanced aptX) streams are output 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.

Available with Content Delay Option for Delayed Playout

iPort High Density's 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. 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 bit rates, MP3 and MP2 for legacy applications.

Specifications

Audio

iPort High Density has no native audio I/O, operating on streams provided by attached Livewire+ AES67 audio devices. For full audio specifications, see the Telos Alliance xNode page.

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: Enhanced aptX™.

Power

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

Regulatory

Click here to view the current regulatory compliance.

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