Gateway NeXus

High Density IP Audio Codec

Delivering 32 Channels of Deterministic, Low Latency IP Audio

AoIP Protocols

AES67 Livewire+ · AES67 · SMPTE ST 2110-30 ST 2022-7 · NMOS AMWA · RAVENNA · Dante · EMBER+

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Tieline®
The Codec Company

Gateway Nexus Multichannel IP Codec (TLR6232)

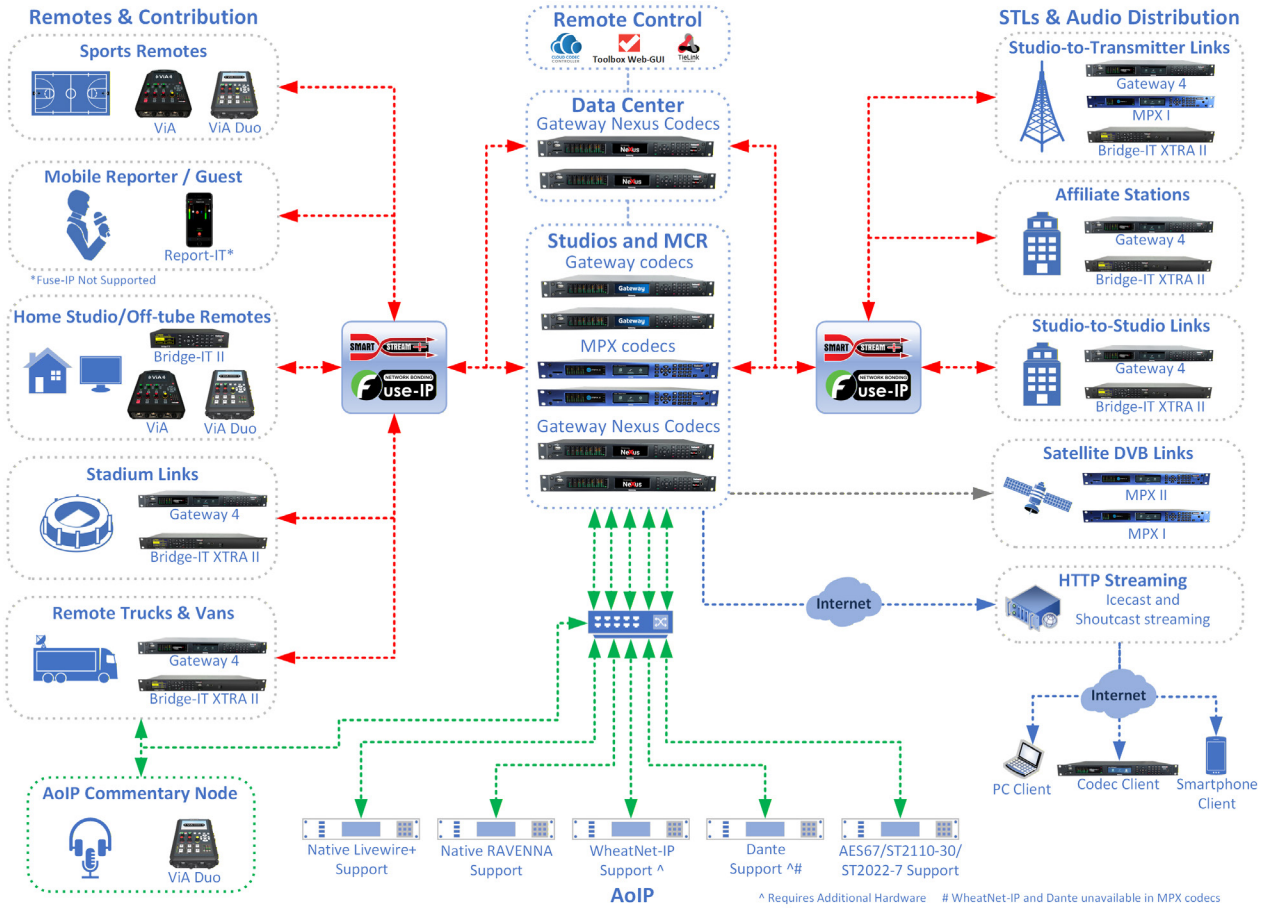
The Gateway Nexus is a powerful and flexible 32-channel, high-density AoIP codec delivering deterministic, low-latency multichannel audio transport using dedicated and reliable hardware.

The Gateway Nexus is ideal for a wide range of deployments and stands out as a compact, powerful and cost-effective alternative to virtualized codec and server-based systems. It can be installed at the broadcast plant, or in hosted data center environments. It is engineered to natively support AES67, ST2110-30, ST2022-7, RAVENNA, Livewire+, NMOS IS-04, IS-05, IS-07 and Ember+. Additionally, a Dante card can be installed at purchase. Compliance with these standards ensures seamless integration into IP-based audio with vendors supporting these protocols.

Applications:

Gateway Nexus connects over any IP network and is ideal for:

- Hosted or data center environments requiring high channel density and low latency connections for live programming and communications applications
- Audio contribution and managing multiple remotes
- Network syndication of programming
- Multiple Studio-to-Transmitter Links
- Multi-channel links to remote studios (inter-studio links)



Gateway Nexus integrated into network broadcast solutions

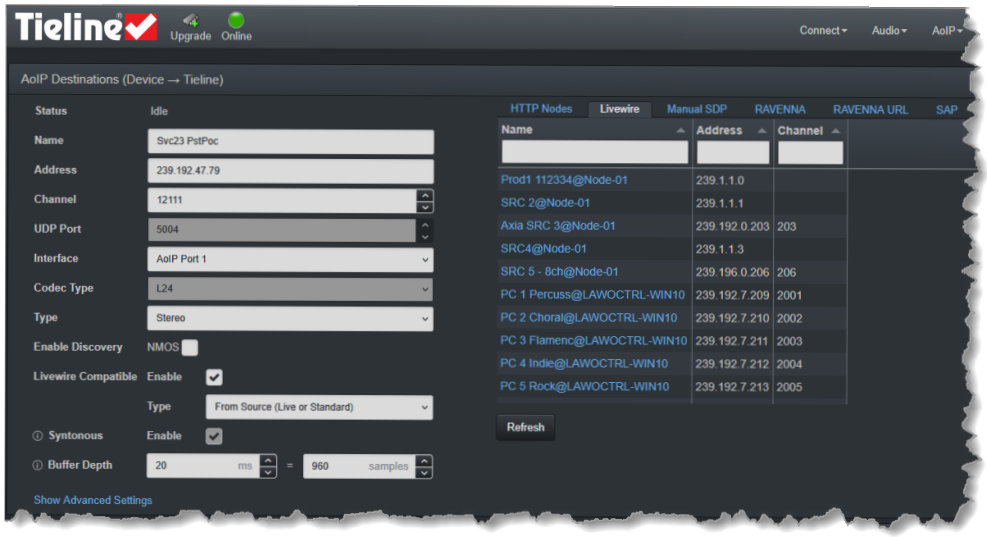
High Density Deployments with Deterministic Performance

As the broadcast industry shifts towards centralized and remote production models for live broadcast applications, the codec's DSP-based platform allows broadcasters to upscale infrastructure efficiently. The Nexus is purpose-built for hosted data center deployments, delivering 32 channels of full-duplex high-density AoIP audio. Unlike virtual codec solutions, it provides fully deterministic performance that is predictable, low-latency, and fully hardware-based. The Gateway Nexus will expand your encoding capabilities by providing a centralized hub for all your AoIP streams.

Interoperability

Advanced features and support for a wide range of protocols combine to deliver consistent latency, predictable performance and operational simplicity, making it ideal for state-of-the-art broadcast networks.

The Gateway Nexus is interoperable with all Tieline IP codecs and it is also compatible over SIP with all EBU N/ACIP Tech 3326, 3347 and 3368 compliant codecs and devices. The codec is also fully AES67, ST 2110-30, ST 2022-7, Livewire+ and RAVENNA compliant for AoIP streaming. It also supports AES3 and analog I/O as standard. The optional Dante card delivers seamless operation using Dante Controller. It is also AMWA NMOS IS-04, IS-05 and IS-07 compliant for AoIP discovery, registration, connection management, and notifications, simplifying setup and configuration of networked AoIP devices and systems. The Ember+ control protocol is also supported.



Key Features

- High channel density with 32 bidirectional mono or 16 bidirectional stereo IP streams
- 16 hardware GPIOs, plus 3 virtual inputs, 48 logical outputs, 64 WheatNet or Livewire Logic Inputs/Outputs and 64 Ember+ GPIOs
- 10 band EQ, compressor, noise gate and expander, plus IGC Limiting on all analog inputs
- Ideal for data centers, studio headend distribution, managing multiple remotes and contribution
- Stream IP audio over WANs using dual Ethernet ports
- Natively AES67, ST 2110-30, ST 2022-7, Livewire+ and RAVENNA compliant for AoIP streaming
- Optional Dante card at purchase supports Dante interoperability
- Interoperable with Dante (if no card installed) and WheatNet-IP in AES67 compatibility mode
- Support for Ember+ and NMOS IS-04, IS-05 and IS-07
- Supports multicasting and multi-unicasting, multi-multicasting & multiple multi-unicasts
- Supports HTTP Icecast or Shoutcast streaming.
- Dual DSP based platform with dual internal power supplies, dual Ethernet ports, dual AoIP ports, and dual USB ports.
- Supports uncompressed audio and a huge range of encoding options
- SD card slot for firmware upgrades and memory options
- Fully SIP EBU N/ACIP 3326, 3368 and 3347 compliant to operate with 3rd party codecs
- Module slot for future technologies and hardware upgrades

Redundancy and Remote Control

Maintain rock solid connections with multiple layers of redundancy and backup including:

- Dual internal power supplies, dual Ethernet ports and dual AoIP ports
- SmartStream PLUS hitless packet switching
- Fuse-IP data aggregation
- Alternative connection backup
- Audio file backup
- Icecast streaming client
- FEC and RIST

Gateway Nexus is fully configurable through an embedded HTML5 Toolbox Web-GUI interface and is also fully controllable using Tieline's Cloud Codec Controller. It also supports monitoring and alerts using SNMP and automated alarm notifications using Toolbox software.

Configuring SmartStream PLUS hitless packet switching

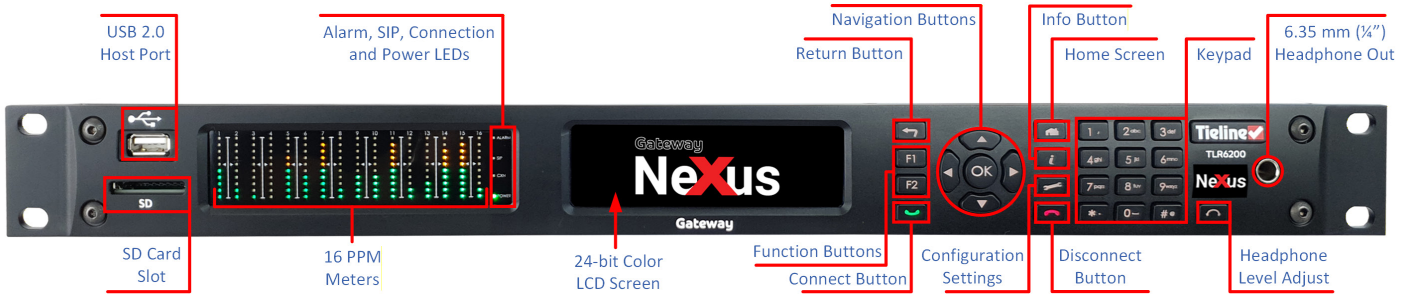
Flexible Audio Distribution

The Gateway supports 32 mono or 16 stereo full-duplex audio streams. Gateway also supports flexible multi-unicasting and multicasting configurations for audio distribution.

- Stream up to 32 different multi-unicast audio streams to up to 80 endpoints in total.
- Stream up to 32 different multicast audio streams to unlimited endpoints over compatible networks

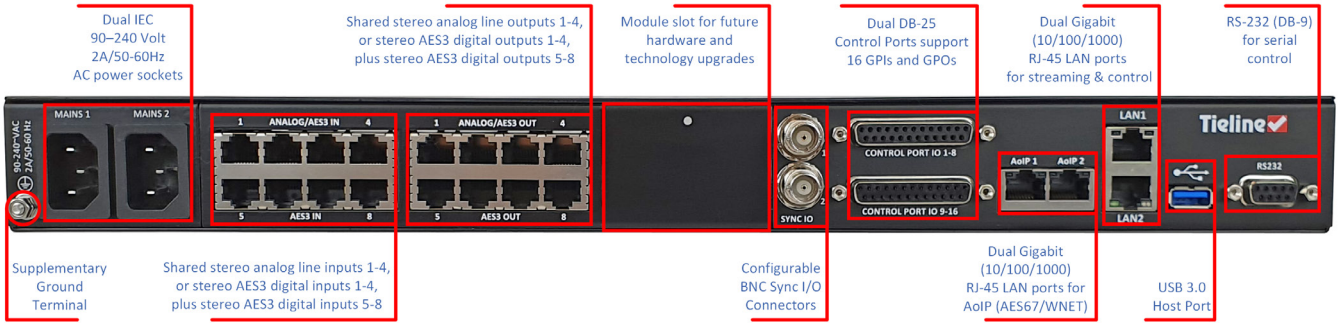
Front Panel Interfaces

The Gateway Nexus front panel features a high-quality color LCD screen for at-a-glance monitoring and configuration. Silicon menu, navigation and keypad buttons deliver simple configuration, control and audio monitoring. An SD card slot allows simple access for firmware upgrades.



Rear Panel Interfaces

The Gateway rear panel facilitates maximum flexibility with support for analog, AES3 and AES67 / ST 2110-30 / ST 2022-7 I/O. Tieline also offers optional Dante compatibility. The codec supports 16 GPIOs as well as software logic I/Os. The module slot enables future hardware innovation upgrades to maximize codec capability over time.



Specifications

Input and Outputs

RJ45 Analog Inputs 1-4/AES3 Inputs 1-4	Shared analog stereo line inputs 1-4, or stereo AES3 digital inputs 1-4
RJ45 Analog Outputs 1-4/AES3 Outputs 1-4	Shared analog stereo line outputs 1-4, or stereo AES3 digital outputs 1-4
RJ45 AES3 Inputs 5-8	Stereo AES3 digital inputs 5-8
RJ45 AES3 Outputs 5-8	Stereo AES3 digital outputs 5-8
Gigabit LAN Ports	2 x Gbit Ethernet ports for IP streaming over WANs (32 bidirectional mono or 16 bidirectional stereo)
Gigabit AoIP Streaming Ports	2 x Gbit Ethernet ports for AES67/ ST 2110-30/ ST 2022-7; Optional Dante streaming via AoIP1
Front Panel USB 2.0 Host Port	Supports USB to LAN or USB Wi-Fi for control only
Rear Panel USB 3.0 Host Port	Supports USB to LAN for Control only
Front Panel Headphone Output	1 x 6.35mm (1/4") headphone Jack
Front Panel SD Card Slot	Full size push-pull SD card slot for firmware upgrades [1]
BNC Sync Input/Output	2 configurable input/output BNC sync connectors (Wordclock, AES3 Input, AES11id, PTP Sample Clock, and Fixed)
Control Port In/Out	Dual DB-25 Control Ports support 16 CMOS control inputs and 16 opto-isolated solid-state relay outputs
Analog Input Impedance	> 10k ohm
Analog Output Impedance	< 30 ohm balanced
Clipping Level	Nominal output level +4dBu with maximum output + 24 dBu
A/D & D/A Converters	24 bit
AES3 (AES/EBU)	24 bit inputs with support for sample rates from 32kHz to 192 kHz[1]; 24 bit outputs
Frequency Response	20Hz to 22kHz
Total Harmonic Distortion	<0.002% at +22dBu unweighted
Signal to Noise Ratio	>100dB at +24dBu, unweighted, 20Hz – 20kHz
Crosstalk	< -95dB between adjacent channels

Data and Control

Configuration and connectivity	HTML5 Toolbox Web-GUI, Cloud Codec Controller (CCC)
Serial (DB-9)	RS232 up to 115kbps with or without CTS/RTS flow control can be used as a proprietary data channel
Software Logic I/Os (SLIO)	16 hardware GPIOs, plus 3 virtual inputs, 48 logical outputs, and 64 WheatNet or Livewire Logic Inputs/Outputs, 64 Ember + GPIOs

Encoding and IP Streaming

Encoding Formats	Tieline Music, Tieline MusicPLUS, Opus, G.711, G.722 (SIP only), MPEG Layer 2, MPEG Layer-3 LC-AAC, MPEG audio technologies licensed by Fraunhofer IIS (http://www.iis.fraunhofer.de/audio)
Uncompressed IP	Linear PCM16/24 bit 32kHz, 48kHz
IP Sample Frequencies	8kHz, 16kHz, 32kHz, 48kHz
Phase-locked Surround Sound Channels	Stream 4, 6 and 8 digital channels in phase-locked mode to deliver live content in both stereo and surround sound formats (LC-AAC only)
Asymmetric encoding	Supports asymmetric multi-format encoding
Protocols	RTP, DHCP, SNMP, DNS, HTTP, IGMP, IPv4/IPv6, RTCP, STUN, SSL Security Certificate, RIST, RTSP, RFC5109, RFC5888, RFC4756, RFC3388, RFC2733, RFC3190
SmartStream PLUS	Primary plus a redundant stream supported for each connection
Multi-unicasting	Support for Multi-unicast and 1, 2, 3, 4, 6, 8 , 16 or 32 Multiple multi-unicast streams (80 connections in total)
Multicasting	Support for Multicast (8-16) streams over compatible IP networks
Fuse-IP	Bond IP interfaces to aggregate data

AoIP Standards & Specifications

EBU N/ACIP 3326	Audio Contribution Over IP Compliant
I3P EBU Tech 3347	Intercom Over IP Compliant
EBU N/ACIP 3368	SIP Profiles Compliant
AES67 Compliant	44.1.kHz, 48kHz, 96kHz Sample rates; 16 & 24 Bit, 1-8 Channels, SDP, SAP
ST 2110-30 Compliant	Class A, Ax, B, Bx Sender and Receiver Compliant
ST 2022-7 Seamless Protection Switching	Compliant with requirements for sending multiple redundant streams of RTP packets to enable seamless protection switching
RAVENNA Compliant	RAVENNA Stream Discovery and Advertisement
Livewire+ Compliant	Livewire+ Stream Discovery and Advertisement
NMOS Compliant	NMOS IS-04 & IS-05 Discovery, Registration and Connection Management, NMOS IS-07 Event Notifications
Ember+	Supports Ember+ control protocol
Dante (Optional)	Install an optional Dante card at purchase for Dante AoIP streaming support.
Supported Audio Frames	125us, 250us, 333µs, 1ms, 4ms
Clock Modes Supported	Primary Leader (Master), Follower (Slave), Follower Only (Slave Only)

Advanced Networking

VLAN Tagging	IEEE 802.1Q,802.1p
Quality of Service (QoS)	Support for DiffServ (DSCP)
Synchronization	IEEE 1588-2008(PTPv2)
Multicasting	IGMP v2 and v3 and IGMP v3 SSM
SAP	SAP v2 (Session Announcement Protocol) as defined in RFC 2974

General

Display	24-bit Color LCD Screen (480 x 128 pixels)
Keypad	26 button silicon keypad
Navigation	5 navigation and selection buttons
Size	1U x 19" Rackmount
Dimensions	19" x 1 3/4" x 11 13/16" [482mm (W) x 44.45mm (H) x 300mm (D)] excluding rear connectors
Weight	4 Kg /8 lb 13.100 oz
AC Power	Dual AC 90-240V IEC power inlets; 2A 50-60 Hz
Nominal Power	30W
Maximum Power	45W
Inrush Current (cold start)	60A 230 VAC (per PSU) ; 30A/115VAC (per PSU)
Operating Temperature	0°C to 45°C (32°F to 113°F)
Humidity Operating Range	10% ≥ RH ≤ 90% (0 to 45°C/32°F to 113°F), non-condensing

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