IP codec live audio over networks Internet LAN WAN

AudioTX STL-IP Features & Specifications...

SYSTEM SUMMARY

AudioTX STL-IP IP codec provides for live audio transmission over IP networks with transmission grade audio quality & robustness and extremely low delays - as low as 5ms!

- IP Codec for live audio: Transmit and receive audio using point-to-point UDP or TCP/IP, and point-to-multipoint UDP Multicast network protocols over ANY IP Network - including private networks (LAN/WAN), Satellite, Wireless networks, T1/E1, ATM or the Internet.
- A single STL-IP system can transmit audio on up to six simultaneous connections, each using different audio coding and network protocols if required. Using Multicast, audio can be sent to an unlimited number of destination units. Audio can be received from one remote location, independently from transmission.
- AudioTX STL –IP works with linear (uncompressed) audio at up to 24 bits and 96kHz sample rate, or compressed audio via built-in Professional Grade MPEG Layer 2 or MPEG Layer 3 coding/decoding, J.41, ADPCM, G.722 or our extra Low-Bitrate speech codec. **Plus** MPEG4 AAC, AAC Low-Delay and HE-AAC v2 with the optional AAC Coding Pack for Stereo audio from just 14kbps!
- Optional APTx Coding (Enhanced APTx, 16 and 24 bits) with the APTx Codec Pack.
- Optional Forward Error Correction (FEC) and network jitter compensation where required.
- Synchronous transmission of serial ancillary data and/or contact closures (TTL GPIO).
- Built-in silence and audio overload detectors.
- Monitor and control via web-browser control interface, SNMP traps and queries, E-mail alerts, Telnet style IP remote control interface (using simple text commands and responses), included software and logic level status outputs.
- Incredibly flexible and cost-effective solution.

AudioTX STL-IP can send live audio using up to 6 simultaneous connections -

including any combination of UDP, TCP/IP or UDP Multicast (to an unlimited number of destinations for each UDP Multicast connection). The IP Codec system can receive audio from one location.

AUDIO SPECIFICATIONS AND PROTOCOLS

Summary: Professional grade analogue balanced Stereo audio inputs and outputs plus AES/EBU digital audio in/out, external wordclock input. Audio in/out at up to 24 bit, 96 kHz sample rate	
Mono/Stereo audio transmit/receive using Linear (uncompressed) audio, Broadcast Quality MPEG Layer 2, MPEG Layer 3, J.41, Mono, Stereo, Joint-Stereo, Dual-Mono operation. MPEG4 AAC, AAC Low-Delay and HE-AAC v2 with the optional AAC Coding Pack. Enhanced APTx coding with the APTx Codec Pack.	
Detailed specification:	

Linear (uncompressed) audio:	Uncompressed audio at 8kHz to 96kHz sample rate, 16 or 24 bit. Mono or Stereo modes.	Full-bandwidth linear audio with a 5ms delay.
MPEG4 AAC:	Professional grade AAC coded audio at between 16 and 48 kHz sample rate, 16 bit, Mono, Stereo, Joint-Stereo and dual-mono modes. 24-320kbps.	Provides full-bandwidth broadcast quality stereo audio at bitrates between 64kbps and 320kbps. Near transparent Stereo audio from 64kbps. 150ms delay.
MPEG4 AAC-Low Delay:	Professional grade Low Delay AAC at 16 to 48 kHz sample rate, 16 bit, Mono, Stereo, Joint-Stereo and dual-mono modes. Low Delay version. 24-320kbps.	Provides full-bandwidth broadcast quality stereo audio at bitrates between 80kbps and 320kbps with just 40ms delay.
MPEG4 HE-AAC and HE-AAC v2: (High Efficiency AAC, AACPlus):	Offering expectional audio quality at very low bitrates. HE-AAC coded audio at between 32 and 48 kHz sample rate, 16 bit, Mono, Stereo, and Parametric Stereo. 14 to 96 kbps.	Provides full-bandwidth excellent quality stereo audio at bitrates between 14kbps and 96kbps. 260ms delay.
MPEG Layer 2 coded audio:	Professional MPEG Layer 2 coded audio at between 16 and 48 kHz sample rate, 16 bit, Mono, Stereo, Joint-Stereo and dual-mono modes.	Provides full-bandwidth broadcast quality stereo audio at bitrates between 128kbps and 384kbps. Mono audio from 64kbps. 45ms delay.

MPEG Layer 3 coded audio:	Professional MPEG Layer 3 coded audio at between 16 and 48 kHz sample rate, 16 bit, Mono, Stereo, Joint-Stereo and dual-mono modes.	Provides full-bandwidth broadcast quality stereo audio at bitrates between 128kbps and 384kbps. Mono audio from 64kbps. 125ms delay.
J.41:	High-grade audio compression for near transparent professional audio, 32kHz sample rate, Mono/Stereo.	Mono audio at 384kbps, Stereo at 768kbps. 5ms delay.
ADPCM:	Professional quality compression, 32kHz or 48kHz sample rate, Mono or Stereo mode.	Mono audio at 128kbps or 192kbps, Stereo at 256kbps or 384kbps. 5ms delay.
G.722:	Good quality algorithm for speech/voice with a 7.5kHz audio bandwidth. Runs at 16kHz sample rate.	Mono audio at 64kbps. 5ms delay.
LB-1:	Extra Low-Bitrate speech codec offering 7.5kHz audio bandwidth. Runs at 16kHz or 24kHz sample rate and a range of bitrates determine quality.	Mono audio from 12kbps. 40ms delay.
APTx:	Enhanced APTx Coding - low delay, high quality compressed audio, choice of 16 or 24 bits.	Bitrates range from 64kbps to 576kbps. 9ms delay.
Source audio selection:	User-selectable channel inputs to audio transmission modules - Left channel, Right Channel, Stereo or MonoMix (L+R).	

NETWORK CAPABILITIES AND SPECIFICATIONS	MONITORING/CONTROL
Supports all IP networks including Telco, MPLS, Private/Dedicated circuits, LAN/WAN, Satellite, Wireless (incl. WiFi), ATM, T1/E1 and The Internet for IP codec operation. Network modes: UDP, TCP/IP, UDP Multicast modes Audio transmit/receive bitrates between 24 kB/s and 4.6 mB/s Optional transmission of ancillary serial data at up to 57600 bps, up to 4 in / 4 out GPIO (contact closures) Optional use of FEC (forward error correction) and/or network jitter compensation/safety buffer configurable in 1ms increments from zero to 5 seconds	**Monitoring and control via:** - Web-browser control interface - SNMP traps and queries - E-mail alerts - Telnet style IP remote control interface (using simple text commands and responses) - Included software - Logic level (TTL) status outputs Built-in silence and audio overload detectors.

AUDIO, NETWORK & DATA CONNECTIONS

Analog audio inputs:	Stereo balanced inputs, 2x XLR (F)	-18db nominal signal level. +18db at analog inputs = 0dbFS (digital full scale).
Analog audio outputs:	Balanced Stereo outputs, 2x XLR (M)	-18db nominal signal level. 0dbFS (digital full scale) = +18db at analog inputs.
Digital audio input:	AES/EBU digital input, XLR (F)	Input accepts both AES/EBU and SPDIF type of signals.
Digital audio output:	AES/EBU digital output, XLR (M)	
Clock input:	Wordclock input, BNC.	System clock-source is user-selectable: internal clock, wordclock input or use clock from incoming AES/EBU source.
GPIO:	TTL level inputs (4) and outputs (4) plus an additional 4 status output signals, D-Sub 25 pin connector.	GPIO TTL inputs & outputs provide end-to-end transmission of signals from transmitting to receiving units.
Ancillary Data:	RS-232 serial connection for ancillary data in and out, D-Sub 9 pin connector.	Serial data can be transmitted/received alongside audio at up to 57600 bps.
Network Connection:	Neutrik Ethercon RJ45 connector, accepts standard RJ45 connector or locking Ethercon cable.	10/100 Ethernet connection for TX, RX audio and web-management interface.

POWER

AC Power:	96-264 VAC, 50-60Hz autosensing for worldwide operation.	

Contact us: sales@stl-ip.com tel. +44 (0)121 256 0200 (GMT)

quicklink: home product info features/specifications applications sales info/contact
quicklink: stl-ip-16 and stl-ip-8 models features examples