DIGITAL AUDIO OVER IP-NETWORKS

- ✓ **Professional Audio over IP for STL**
- ✓ **Program Distribution and Live Studio Connectivity.**
- ✓ **Audio TX STL-IP offers 5ms delay and at up to 24bit / 96Khz audio!**

OVERVIEW

Audio TX STL-IP delivers live audio over IP networks with transmission grade audio quality & robustness and an extremely low delay of just 5ms!

- Audio TX STL-IP can transmit and receive audio using point-to-point UDP or TCP/IP, and point-to-multipoint UDP Multicast network protocols. A single system can transmit audio on up to six simultaneous connections, each using different audio coding and network protocols if required.
- Audio TX STL–IP works with linear (uncompressed) audio at up to 24 bits and 96kHz sample rate, professional grade MPEG Layer 2 and Layer 3 compression, AAC*, AAC Low Delay* and HE-AAC (SBR)*, J.41, G.722 and ADPCM.
- Optional Forward Error Correction (FEC) and network jitter compensation plus synchronous transmission of serial ancillary data and/or contact closures (GPIO) make Audio TX STL-IP an incredibly flexible and cost-effective solution.

APPLICATIONS

- Studio to Transmitter links (STL) over any IP Network: including Telco, Private/Dedicated, LAN/WAN, Wireless or Satellite, and the Internet.
- Distribution of live or shared programming for radio stations and groups.
- Permanent and ad-hoc broadcast quality audio connectivity between sites.
- Multicast audio distribution over Telco networks, Satellite, LAN/WAN and all multicast capable networks.
- Telco carrier-grade distribution of live audio signals (one or two way) via TCP/IP or UDP (including multicast) over IP networks.

FEATURES AND SPECIFICATIONS

- **System Features**
  Audio TX 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 system can receive audio from one location.
- **Audio Specifications & Protocols**
  Professional grade analogue balanced Stereo inputs and outputs plus AES/EBU digital in/out, external word-clock input. Audio in/out at up to 24bit, 96 kHz sample rate.
  Mono/Stereo audio transmit/receive using Linear (uncompressed) mode at up to 24 bit, 96 kHz sample rate, MPEG Layer 2 at up to 384 kbps,MPEG Layer 3 at up to 320kbps, AAC*, AAC-LD* and HE-AAC (SBR)*, J.41 at 384 kbps (mono) or 768 kbps (stereo), G.722 at 64kbps,ADPCM for 20 Hz – 22 kHz audio at 128 to 192 kbps (mono) or 256 to 384 kbps (stereo) Mono, Stereo, Joint-Stereo, Dual-Mono operation, optional MPEG TS (Transport Stream) mode. Built in Silence and overload Detectors
- **Network Capabilities & Specifications**
  Supports all IP networks including Telco, Private/Dedicated circuits, LAN/WAN, Satellite, Wireless (incl. WiFi), ATM, T1/E1 and Internet.
  Network modes: UDP, TCP/IP, UDP Multicast modes.
  Audio transmit/receive bitrates between 24 kbps and 4.6 MB/s.
  Optional transmission of ancillary serial data at up to 128 kbps, up to 4 in / 4 out GPIO (contact closures).
  Optional use of FEC (forward error correction) and/or network jitter compensation.
  Monitoring of hardware, network and connection status via SNMP or email alerts or logic level outputs.

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