IQOYA X/LINK-LE is a 1U rack streamlined IP audio codec designed for live remote broadcasting and program distribution over IP. It is a perfect fit for delivering a stereo source (or two mono sources) over IP networks for STL & SSL link, DVB audio, WEB radio and also for full-duplex live remote broadcasts.

IQOYA X/LINK-LE benefits from all the major features of X/LINK but comes at a lower price point. It can be used in legacy analog or AES/EBU audio environments, as well as in full-IP audio infrastructures (AES67, Ravenna, Livewire), making it a good investment for the migration to IP audio. Like all the IQOYA products, X/LINK-LE is based on Fluid IP, the Digigram technology for reliable and resilient audio transmission over all types of IP networks including inexpensive unmanaged IP networks. Based on low consumption and fanless powerful hardware platform, IQOYA X/LINK-LE is designed for 24/7/365 use.

Key Points

- Cost-effective solution with essential features, and no compromises on reliability
- Adapted to legacy audio infrastructures and full-IP audio infrastructures
- EBU/ACIP compliance for interoperability with third-party codecs and any SIP infrastructure
- Multiple levels of redundancy for audio service continuity and failsafe operation: 2 power supply units, 4 network ports with stream redundancy, audio failovers, audio hardware by-pass, and 1+1 hot device redundancy
- Control and configuration via SNMP and Web services for easy integration with codec and network management/monitoring systems.
I/Os and POWER

- 2 balanced analog audio I/Os with 24 bit converters, and 1 stereo AES/EBU I/O with hardware sample rate converter, and 2 AES67 or Ravenna or Livewire I/Os
- 4 ethernet ports: 1x100Mbps + 3x1Gbps on RJ-45 connectors
- 1 RS232 port for serial data tunneling
- 8 GPI / 8 GPO on Sub-D25 connector
- 2 internal redundant PSUs 100-250VAC (Optional 100-250VAC / -48VDC) (Max 15W consumption)

NETWORKING

- RTP/UDP, Icecast/Shoutcast, HLS, SIP/SDP, STUN
- DHCP, IGMPv2 and v3, QoS (VLAN tagging, DSCP)
- SNMP, HTTP
- PTP, NTP

ENCODING, DECODING AND STREAM PROTECTION

- Audio encoding formats: PCM linear 16/20/24 bits, ISO MPEG-½ Layer II and Layer III, MPEG-4 AAC-LC, AAC-LD, HE-AACv1, HE-AACv2, AAC-ELD, Opus
- Selectable FECs for ACIP/RTP streams (from +10% to +100% IP bandwidth)
- Dual port redundant streaming with spatial and time diversity
- Pro-MPEG CoP #3 FEC for MPEG-TS streams
- Adaptive and resilient audio streaming (Fluid-IP)

FUNCTIONS

- Multi-format encoding of the audio sources
- Unicast, multi-unicast, multicast, multi-multicast addressing
- Place calls from an address book or receive calls for live broadcasts (SIP, Direct SIP, Symmetric RTP)
- Auxiliary data tunneling of physical or IP serial data and GPIs (including RDS-UECP)
- Silence detection on the audio inputs and on the received IP audio streams (SIP, Direct SIP, Symmetric RTP)
- Network traffic separation (WAN, LAN, Management)
- Configuration and monitoring via intuitive Web GUI and via SNMP
- Hardware by-pass of the audio inputs to the audio outputs (analog and AES/EBU)
- Available clock synchronization are internal, AES/EBU inputs, PTP, livewire clock
- Optional: Simultaneous multi-protocol streaming
- Optional: Audio output synchronization of decoders based on NTP or PTP
- Optional: AES transparency without the need of an external GPS synchronized clock