THE 2 CHANNELS IP AUDIO CODEC WITH ALL THE ESSENTIALS AT AN ATTRACTIVE PRICE

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.

High processing power allowing low latency audio connections, multi-format encoding and multi-protocol streaming

SDHC card reader for backup playlists, and loading/saving the codec configuration

Easy and fast status monitoring: status LEDs, vu-meters, headphones

2 analog I/O's and 1 stereo AES3 I/O

2 internal redundant PSUs for secure operation

Switchable hardware by-pass of audio inputs to outputs in case of power supply failure

4 network ports for full separation of IP traffics:
- AES67, RAVENNA, Livewire (LAN)
- Redundant dual streaming (WAN)
- Remote management (LAN / WAN)

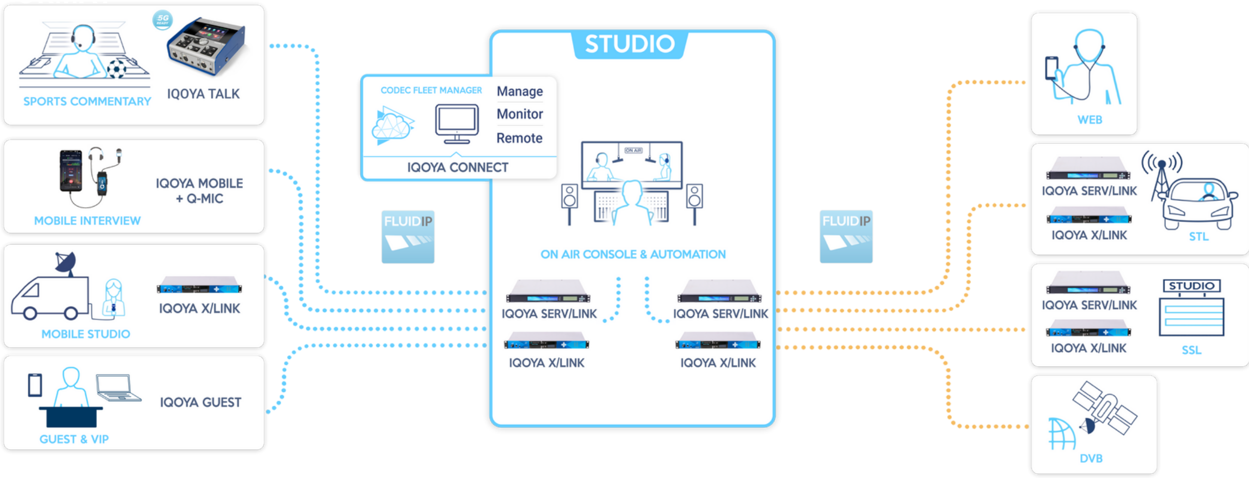
4 GPIO's for signalling of physical status

Handles 2 full-duplex mono remotes, or 1 full-duplex stereo remote

RS232 port for serial data signalling

KEY FEATURES

- 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
- Allows for 1 stereo, or 2 mono full-duplex remotes
- 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
- Control and configuration via SNMP and Web services for easy integration with codec and network management/monitoring systems.



1 I/Os AND POWER

- 2 input and output channel codec
- 2 balanced analog audio I/Os , 24 bit converters, +24 dBu max
- 1 stereo AES3 I/O with hardware sample rate converters
- 2 AES67 or Ravenna or Livewire I/O channels
- Adjustable analog input and output gains by steps of 0.5dB
- Adjustable digital input and output gains by steps of 0.1dB
- 4 Ethernet ports: 1x100Mbps + 3x1Gbps on RJ-45 connectors
- 1 RS232 port for serial data tunneling
- 8 GPI / 8 GPO on Sub-D25 connector for auxiliary data tunneling
- 2 internal redundant PSUs 100-250VAC (Max 15W consumption)

2 NETWORKING

- Streaming protocols:
  Encoding: RTP/UDP with or without MPEG-TS encapsulation,
  Icecast/Shoutcast, HLS multi-bitrate
  Decoding: RTP/UDP, Icecast/Shoutcast
- DHCP, IGMPv2 & v3, VLAN tagging, QoS (DSCP)
- Unicast, multicast, multi-unicast, multi-multicast
- PTP (AES67/Ravenna), NTP (date and time, audio synchronization)
- SIP, for remote broadcasting
- AES67/Ravenna: mDNS for streams discovery, RTSP for streams announcement
- Management and control: SNMP, HTTPS, WEB Service API

3 ENCODING, DECODING AND STREAMING

- Multi-format/bitrate encoding of each input
- Audio encoding formats: PCM linear 16/20/24 bits, ISO MPEG-1/2 Layer II and Layer III, MPEG-4 AAC-LC, AAC-LD, HE-AACv1, HE-AACv2, AAC-ELD, Opus
- Dual port redundant streaming with spatial and time diversity (up to 3 seconds)
- Selectable FECs for ACIP/RTP streams (from +10% to +100% IP bandwidth),
- Pro-MPEG CoP #3 FEC for MPEG-TS streams (SPTS, MPTS)
- Adaptive and resilient audio streaming (Fluid-IP)

- **Optional**: Multi-protocol streaming of each audio source: raw RTP/UDP, MPEG-TS SPTS/MPTS, Icecast/Shoutcast, HLS multi-bitrate

4 FUNCTIONS

Programs distribution

- Simultaneous multi-format/bitrate encoding of each audio source
- 3 decoding priorities per output program with a choice of the audio source on each priority: RTP or Icecast/Shoutcast stream, playlists on SDHC card, or audio inputs.
- Silence detection on the audio inputs and on the received IP audio streams
- AES transparent transport
- Tunneling of serial data (from a serial port or a UDP port) and triggers (GPIs or status bits in a UDP frame)
- Insertion of metadata to Icecast/Shoutcast streams (yellow pages and on-the-fly)
- Configuration and monitoring via intuitive Web GUI, SNMP, or WEB service API
- Hardware by-pass of the first stereo analog and AES3 audio inputs to the corresponding analog and AES3 outputs
- **Optional**:
  - Audio output synchronization of decoders based on NTP
  - Transcoding: RTP, Icecast/Shoutcast to RTP, Icecast/Shoutcast, HLS, MPEG-TS SPTS/MPTS)

Remote broadcasting

- Interoperable with EBU/ACIP compliant codecs
- Supports 2 mono remotes, or 1 stereo remote
- Address book and audio profile list
- Place calls from WEB GUI or from IQOYA Connect Saas application
- Auto-redial
- Manual or automatic call answering
- Connection modes: registered SIP, unregistered SIP (aka Direct SIP), Symmetric RTP
- Registration to main and backup SIP servers
- Redundant dual streaming between Digigram codecs
- Real-time network quality measurement
- Part of Digigram One IP remote broadcasting solution