THE MULTICHANNEL AUDIO CODEC FOR ROCK-SOLID AUDIO TRANSMISSION OVER IP

IQOYA X/LINK-AES67 is a 1U rack IP audio codec designed for live remote broadcasting and program distribution over IP. It is a perfect fit for delivering stereo and/or mono audio sources over IP networks, for STL, SSL, DVB audio, WEB radio applications and also for full-duplex live remote broadcasts. It is dedicated to full-IP audio infrastructures based on AES67, Ravenna, or Livewire (standard mode) technologies. Like all the IQOYA products, X/LINK-AES67 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 a low consumption, fanless and powerful hardware platform, IQOYA X/LINK-AES67 is designed for 24/7/365 use.

High processing power allowing low latency audio connections, simultaneous encodings in different formats and simultaneous streaming in different protocols

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

4 GPIO's and 1 RS232 port for data tunneling

Handles from 2 to 16 full-duplex mono remotes, or 1 to 8 full-duplex stereo remotes, or any combination of mono and stereo remotes

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

2 internal redundant PSUs for secure operation

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

KEY FEATURES

Designed for full IP audio environments (supports AES67, RAVENNA, and Livewire standard mode technologies)

Scalable number of supported AES67/Ravenna input and output channels, via software options

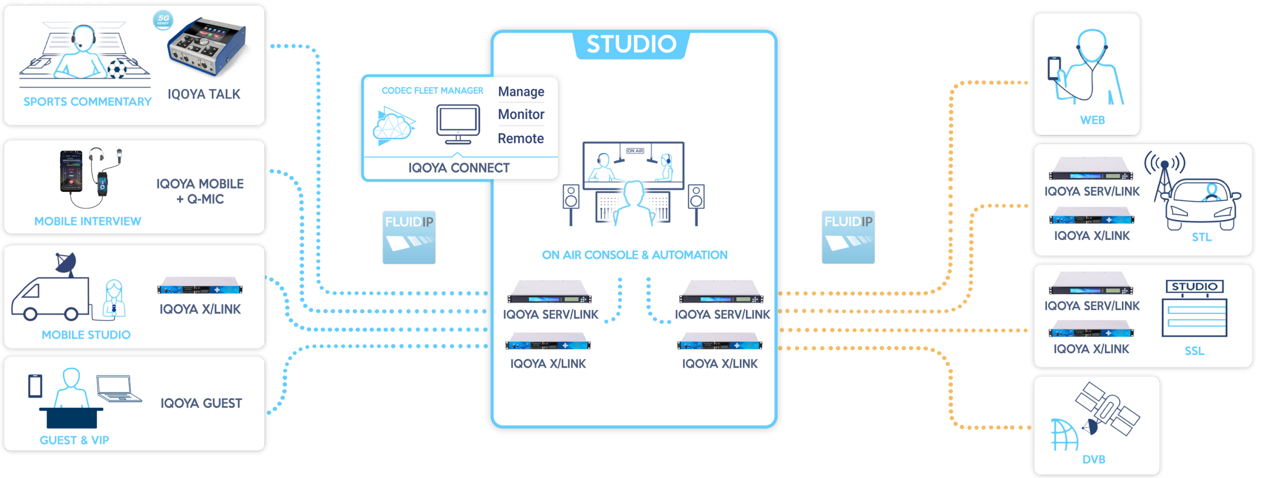
EBU/ACIP compliance for interoperability with third-party codecs and any SIP infrastructure

Allows for up to 8 stereo, or 16 mono, or any combinations of stereo and mono full-duplex remotes

Simultaneous delivery of up to 8 stereo (or 16 mono) programs in multiple formats and transport protocols to FM transmitter sites, Web radio CDNs, DVB multiplexers, and also to other studios

Multiple levels of redundancy for audio service continuity and failsafe operation: 2 power supply units, 4 network ports with stream redundancy, audio failovers

Control & configuration via SNMP and Web services for easy integration with codec and network monitoring systems.



1 I/Os AND POWER

- 2 to 16 AES67 / Ravenna or Livewire audio channels.
- The number of supported channels is upgradeable via software licenses
- Adjustable digital input and output gains by steps of 0.1dB
- 4 ethernet ports: 1x100Mbps + 3x1Gbps on RJ-45 connectors
- 8 GPI / 8 GPO on Sub-D25 connector
- 1 RS232 port for serial data tunneling
- 2 internal redundant PSUs 100-250VA (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
- Multi-protocol streaming of each audio source: raw RTP/UDP, MPEG-TS SPTS/MPTS, Icecast/Shoutcast, HLS multi-bitrate
- 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)

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
- 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 up to 16 mono remotes, 8 stereo remotes or combinations of mono and stereo remotes
- Address book and audio profile list
- Place calls from WEB GUI or front panel, 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