THE 2 CHANNELS AUDIO CODEC FOR ROCK-SOLID AUDIO TRANSMISSION OVER IP

IQOYA X/LINK-ST is a 1U rack 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 links, DVB audio, WEB radio, and also for full-duplex live remote broadcasts. It can be used in legacy analog or AES/EBU audio infrastructures, as well as in full-IP audio infrastructures (AES67, Ravenna, Livewire), making it a perfect investment for the migration to IP audio. Like all the IQOYA products, X/LINK-ST 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-ST 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, LCD display 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)

8 GPIO's for tunneling of physical status

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

RS232 port for serial data tunneling

KEY FEATURES

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

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

Efficient all-in-one 2-channel codec in a 1U rack. Simultaneous multi-bitrate and multi-protocol streaming for programs delivery for STL, SSL, WEB radios, DVB audio.

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

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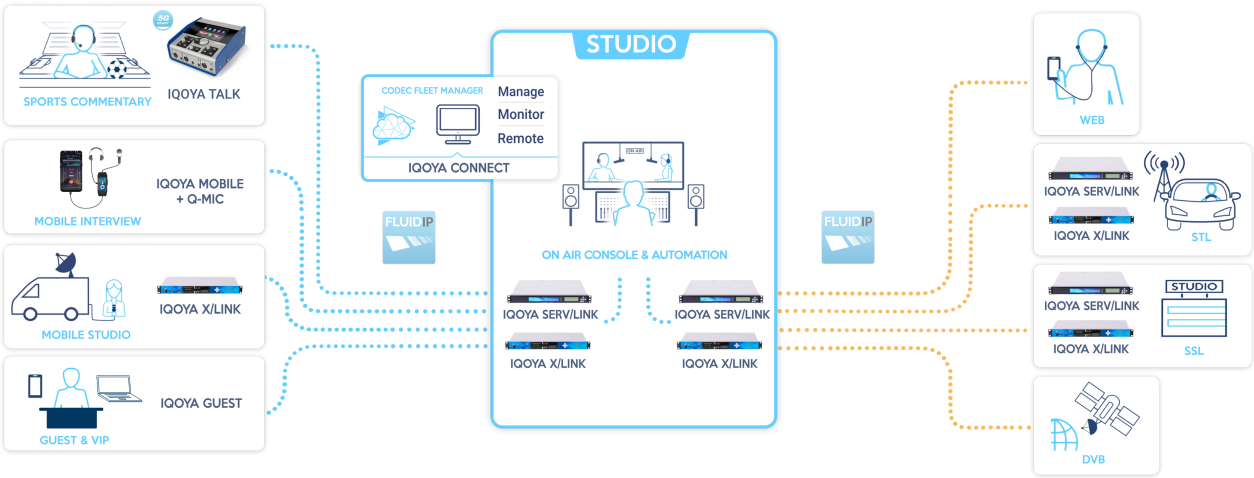

1 I/Os AND POWER

- 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
- 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
- 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 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