IQOYA X/LINK-ST

Product sheet

digigram

The 2 channel 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.

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

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

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



2 analog I/O's and 1 stereo AES/EBU I/O

2 internal redundant PSUs for secure operation



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

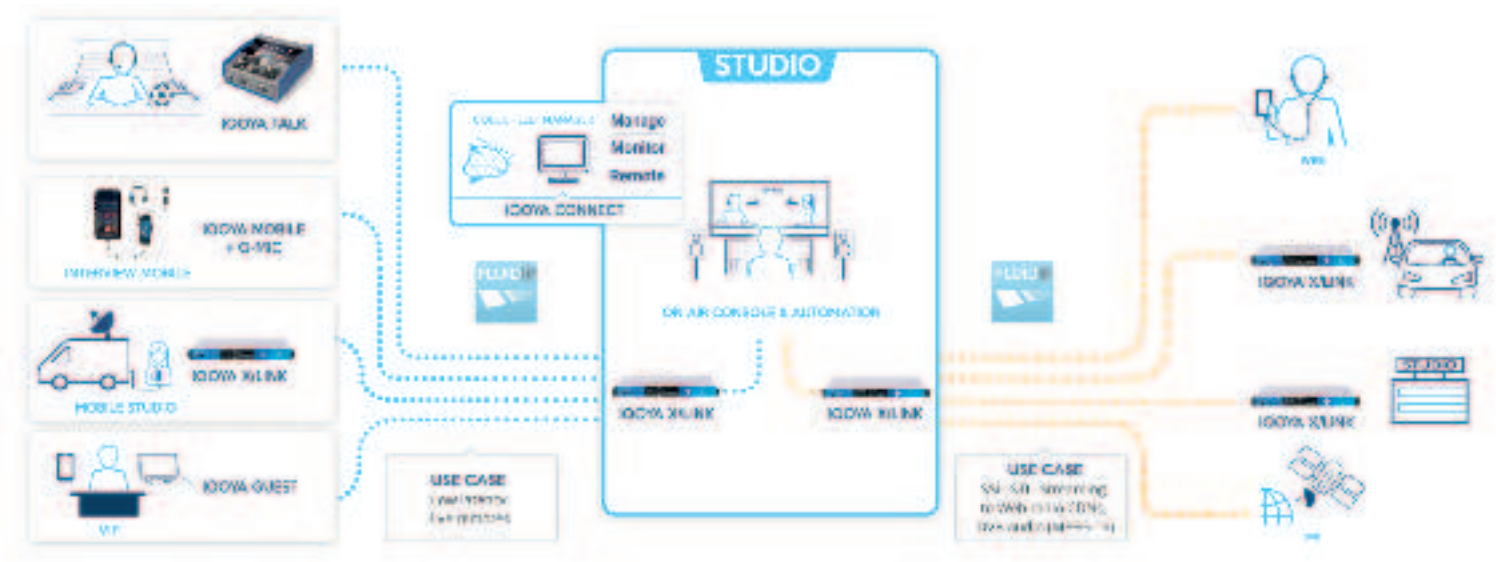
8 GPIO's for signalling of physical status

4 network ports for full separation of IP traffics:

- AES67, RAVENNA, Livewire (LAN)
- Redundant dual streaming (WAN)
- Remote management (LAN / WAN)

Key Points

- Adapted to legacy audio infrastructures and full-IP audio infrastructures
- Simultaneous delivery of 1 stereo (or 2 mono) radio program(s) in multiple encoding and transport formats to FM transmitter sites, Web radio CDNs, DVB multiplexers, and also to other studios
- 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	8 GPI/8 GPO on Sub-D25 connector
	1 RS232 port for serial data tunnelling
2 internal redundant PSUs 100-250VAC (Optional 100-250VAC / -48VDC) (Max 15W consumption)	4 network ports 1 x 100 Mbps + 3 x 1Gbps on RJ-45 connectors

NETWORKING

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

ENCODING, DECODING AND STREAMING 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) Pro-MPEG CoP #3 FEC for MPEG-TS streams
Dual port redundant streaming with spatial and time diversity (Fluid-IP)	Adaptive and resilient audio streaming (Fluid-IP)

FUNCTIONS

Simultaneous multi-format encoding and multi-protocol streaming of the audio sources allowing to address FM, satellite, and web distribution at the same time	Tunneling of physical or IP serial data and GPIs (including RDS-UECP)
Unicast, multi-unicast, multicast, multi-multicast addressing	Insertion of metadata to Icecast/Shoutcast streams (yellow pages and on-the-fly)
3 decoding priorities per output program with choice of the audio source on each priority: IP stream, playlists on SDHC card, or audio inputs. The switch between decoding priorities is automatic according to adjustable criteria	Network traffic separation (WAN, LAN, Management)
	Configuration and monitoring via intuitive Web GUI and via SNMP
Silence detection on the audio inputs and on the received IP audio streams	Hardware by-pass of the audio inputs to the audio outputs (analog and AES/EBU)
AES transparency without the need of an external GPS synchronized clock	
Available clock synchronization are internal, AES/EBU inputs, PTP, livewire clock	**Optional:** Audio output synchronization of decoders based on NTP or PTP
Place calls from an address book or receive calls for live broadcasts (SIP, Direct SIP, Symmetric RTP)	**Optional:** Transcoding (EBU/ACIP RTP, Icecast/Shoutcast, MPEG-TS)