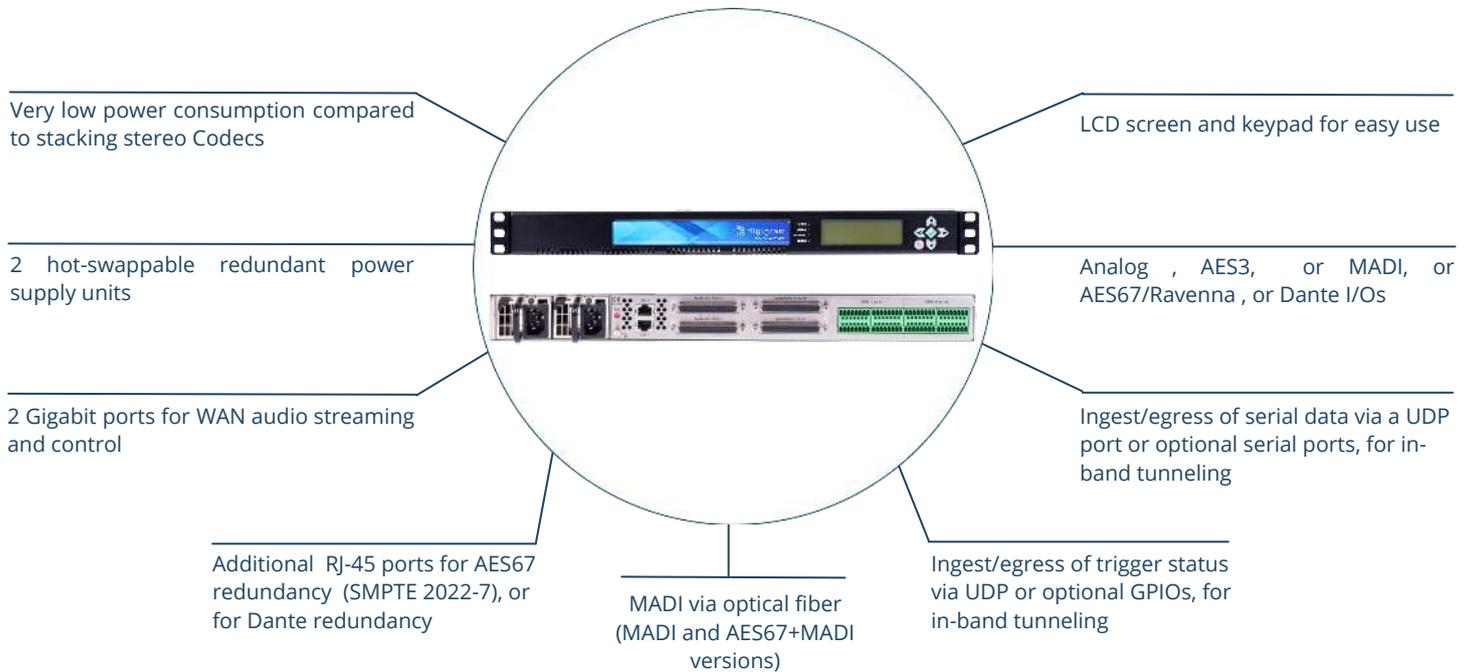


OUTSTANDING POSSIBILITIES IN ONLY 1U RACK

IQOYA SERV/LINK is the high density 1U rack multi-channel IP audio codec designed for live remote broadcasts or program distribution over IP networks.

It supports from 4 to 64 stereo (8 to 128 mono) input and output channels with the possibility to simultaneously encode, decode and transcode IP audio streams.

Only 1U to host up to 64 stereo IP audio codecs



ANALOG

AES/EBU

MADI

AES67

Dante

RAVENNA

KEY FEATURES



High audio channels density in a 1U rack



Simultaneous encoding and delivery of multiple audio programs to transmitter sites, WEB radio CDNs, DVB multiplexers, or studios



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



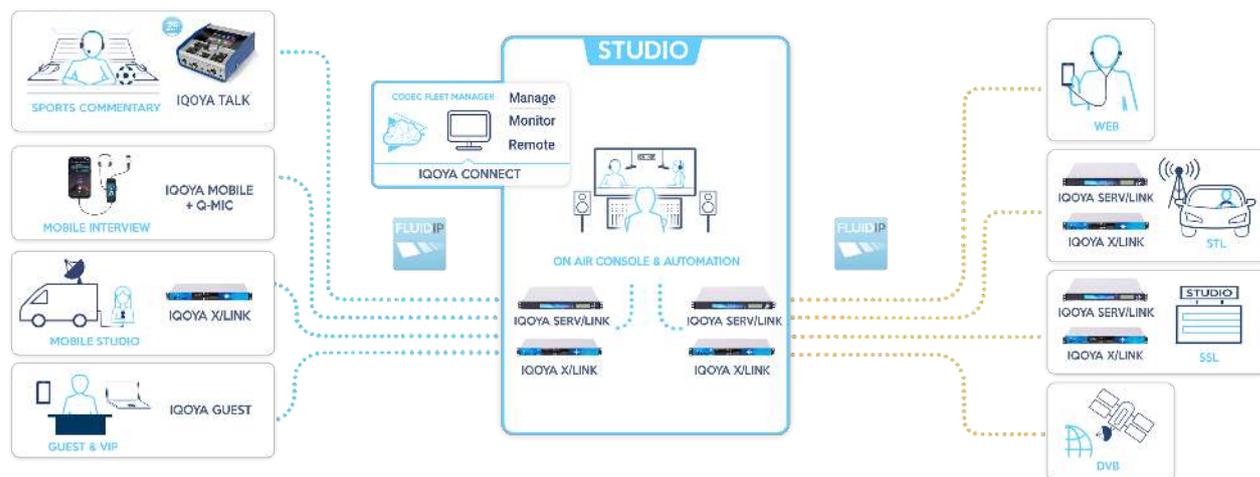
Isolation between the internet world and the LAN audio (AES67 or DANTE) to preserve the security of the internal IP-audio



Multiple levels of redundancy for audio service continuity and failsafe operation: 2 hot-swappable power supply units, 2 Gbit network ports with stream redundancy, audio failovers, and 1+1 hot device redundancy



Features Fluid-IP, the Digigram technology for resilient audio transmission over unmanaged IP networks.



1 I/Os AND POWER

- 1U Rack
- 2 redundant 90 - 264VAC hot-swappable power supply units

2 CONNECTIVITY

- Models from 4 to 64 stereo audio channels (128 mono)
- 2 Gbit Ethernet ports for AoIP streaming over WAN
- 2 or 4 Gbit Eth ports for AES67/Ravenna connectivity. 2 Gbit Eth ports for Dante connectivity
- According to variants, 8 or 16 GPIs (TTL) and GPOs (relays)
- According to variants, 8 or 16 RS232 ports for serial data tunneling

3 ENCODING, DECODING, TRANSCODING AND STREAMING

- Audio encoding formats: G.711/722, 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, Qualcomm aptX (option)
- Support of multichannel audio 5.1 (PCM, AAC-LC) and 7.1 (PCM)
- Streaming protocols: RTP/UDP with or without MPEG-TS encapsulation (SPTS, MPTS), Icecast/Shoutcast, HLS (PUSH and PULL modes, multi-bitrate)
- Transcoding of IP streams: UDP/RTP/Icecast/Shoutcast to UDP/RTP/Icecast/Shoutcast/HLS multi-bitrate/MPEG-TS, with or without audio format change.
- Full EBU/ACIP compliance (tech3326 and 3368)
- Support of VLAN tagging and DiffServ QoS (DSCP)
- 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 (up to 3 seconds)
- Adaptive and resilient audio streaming (Fluid-IP)
- Support of unicast, multi-unicast, multicast, multi-multicast addressing (IGMPv2 and v3)

4 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
- 3 decoding priorities per output program with a 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
- Fully compatible with any SIP infrastructure
- Silence detection on the audio inputs and on the received IP audio streams
- **Optional:** Transcoding (EBU/ACIP RTP, Icecast/Shoutcast, MPEG-TS)
- Tunneling of serial data (from 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 and via SNMP
- Possibility to deploy a pair of IQOYA SERV/LINK in High Availability mode (1+1 hot redundancy)