

Applications

- ▶ Professional Audio Networks
- ▶ DVB compliant encoding of audio signals
- ▶ Satellite feeds
- ▶ Distribution networks for radio stations
- ▶ Feeding digital cable networks
- ▶ Studio Transmitter Link
- ▶ Point-to-point connections
- ▶ Point-to-multipoint connections (up to 16 destinations)

Features

- ▶ Up to 16 digital stereo channels
- ▶ Up to 8 analog stereo channels
- ▶ MPEG TS over IP outputs (unicast or multicast)
- ▶ Output of elementary streams over IP
- ▶ Digital MPX transport
- ▶ Transmission of ancillary data (over IP or serial interface)
- ▶ 8 GPIO inputs
- ▶ Remote controllable over IP via SNMP
- ▶ RAVENNA / AES67, Axia® Lifewire+™, SMPTE ST 2110-30, SMPTE ST 2110-31, Dante®

Compression algorithms for TS

- ▶ MPEG 1/2 Layer 2
- ▶ AAC
- ▶ Enhanced aptX

Compression algorithms for ES

- ▶ Fraunhofer xHE-AAC®
- ▶ AAC
- ▶ Linear PCM
- ▶ MPEG-1/2 Layer 2/3
- ▶ Enhanced aptX
- ▶ OPUS
- ▶ G.711, G.722

Support Options

We are convinced of the high quality of our products. Hence, we are granting 2 years warranty without making compromises.

For the time after that, we offer affordable subsequent contracts.

For optimal support and for software updates and upgrades we offer budget friendly support contracts.

- 2 years warranty
 - Hardware warranty extension up to 10 years
 - Service Contract Basic (Updates, Email support) (mandatory)
 - Service Contract Advanced (Updates, Email- and phone support, replacement devices etc.)
- Legend: ▶ ● Default ▶ ○ Optional

Multi-Channel Audio Encoder

The Q9X-E IP Audio Encoder features MPEG-4 Advanced Audio Coding (MPEG-4 AAC) and MPEG-1 Layer 2 encoding capabilities. All of the common bit rates and sample rates are offered to enhance the IP delivery of Audio.

The Qbit Q9X-E IP Audio Encoder sets standards for high quality audio encoding.

Based on the proven Qbit platform, it provides highest signal quality, best build quality and service without compromise. Customers around the world trust our market leading IP Audio Encoder.

Either up to 16 digital or up to 8 analog stereo audio channels can be encoded simultaneously. They can then be output as MPEG-2-compliant DVB transport streams or elementary streams via the IP interface. Several configurations are available to flexibly adapt the audio input configuration to the network requirements. Each channel can be configured individually.

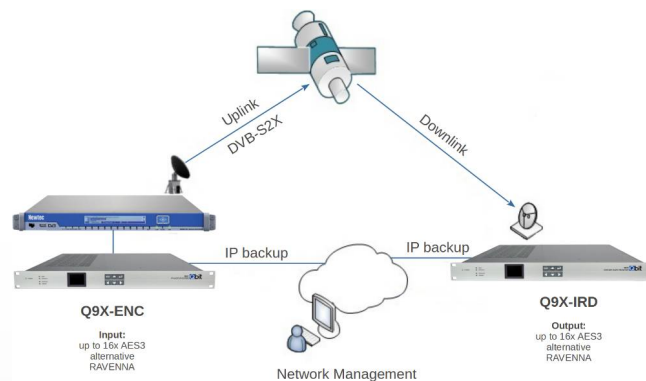
The Q9X-E IP Audio Encoder can be conveniently managed via the integrated web interface with all common web browsers. The device can also be monitored and managed via SNMP or REST-API.

The unit is built in a compact 19" 1 U housing. The transmission of ancillary data and switching contact information (GPIO) is possible with the default interfaces. Optionally, a second data interface can be used as a backup function.

The Q9X-E is CE and RoHS compliant to meet the demand of users worldwide.

The basic unit is licensed for one audio encoder channel and the functions RDS and GPIO. Further channels and options are possible.

If you need more than 16 stereo channels, our Q8V appliance is the best choice.



Specifications



Audio Input

Digital (max. 16 stereo) ▶ Sub-D 25 connectors for digital audio signals AES3 (TASCAM pin assignment)
▶ Digital MPX transport

Analog (max. 8 stereo) ▶ Sub-D 25 connectors for analog audio signals (TASCAM pin assignment)
▶ Audio Frequency Range: 20 Hz to 20 kHz (± 0.3 dB)
▶ THD+N (1 kHz at max. level): $< 0,01$ % at 1 kHz
▶ Crosstalk attenuation at 1 kHz: > 100 dB
▶ S/N ratio (weighted): > 80 dB

Audio over IP ▶ Axia® Lifewire+™ (Legacy Lifewire supported)
▶ RAVENNA / AES67: Input of up to 16 stereo channels
Formats: L16, L24, L32
Sampling rate: 32 kHz, 48 kHz
▶ Dante®
▶ AES67
▶ SMPTE ST 2110-30
▶ SMPTE ST 2110-31

Audio Compression

Algorithms ▶ ISO/IEC 1172-3, 13818-3 MPEG-1/2 Layer 2/3
▶ ISO/IEC 13818-7 MPEG-2 AAC-LC
▶ ISO/IEC 14496-3 MPEG-4 AAC-LC, HE-AAC V1/2, AAC-LD, AAC-ELD
▶ Fraunhofer xHE-AAC®
▶ Enhanced aptX
▶ Linear PCM
▶ OPUS
▶ G.711, G.722

Encoding bit rate ▶ All bit rates are supported according to the standards of the respective algorithms
▶ Sampling rate: 32 kHz, 48 kHz

Modes Stereo, Dual, Mono channel

Ancillary data ▶ Private stream via UECP within the MPEG-2 transport stream or embedded in MPEG audio data
▶ RS-232 interface
▶ Breakout cable (optional): 4 or 8 connectors, conversion from Sub-D 25 to Sub-D 9

Transport Protocols

Over IP ▶ Output of DVB MPEG-2 transport streams including service information according to ETSI EN 300 468, compliant to „Pro-MPEG Code of Practice #3 release 2“
▶ Output of elementary streams
▶ FEC compliant to Pro-MPEG Code of Practice #3 release 2 / SMPTE ST 2022-1 (optional)
▶ Seamless Protection Switching according to SMPTE ST 2022-7 (optional)
▶ SRT (Secure Reliable Transport)

Network Interfaces

3 separate Ethernet interfaces (IEEE 802.3, RJ45, 100/1000 Mbit/s)
▶ 2x Data (elementary / transport streams via IP)
▶ 1x Control (Web interface, SNMP and Ancillary Data)

VLAN Management

System Configuration, Control and Monitoring

Via Ethernet by accessing the on-system HTTP web server with any Internet browser

REST API

Silence detection (optional)

Via Ethernet with SNMP Traps in case of triggered alarms

Via the front panel keyboard and display

Power Requirements

Supply voltage ▶ Integrated switching power supply, input voltage 100 to 240 V AC ± 10 %, 50 to 60 Hz
▶ -48 V DC (optional)

Redundant power supply (optional) The optionally available redundant power supply protects the operation of the device and comes with the following functions:
▶ Measurement of the power supply voltages, values are provided via web GUI or SNMP
▶ SNMP trap generation on power supply fail
▶ Activation of switching contacts on power supply fail
▶ Automatic switch-over in case of power supply fail

Power consumption < 20 W

Physical Parameters

Chassis 19" rack mount cabinet, 1 U

Size ▶ Width: 483 mm
▶ Depth: 360 mm
▶ Height: 44 mm

Weight 4,5 kg

Environmental Conditions

Operating temperature 0 °C to 45 °C

Storage temperature -20 °C to 70 °C

Humidity 20 % to 90 %, non-condensing