AEQ

Audio, Video and Communications for Broadcasters

COMPACT SIMPLE EFFICIENT

Broadcast Communications for your Station or Network of Stations

PHOENIX MERCURY

Full Duplex IP Audiocodec

MAIN FEATURES

IP AudioCodec, stereo, bidirectional. Allows for stereo dual or mono connections. PC User interface provides local or remote control for one or an array of codecs.

Compatible with the most audiocodecs in the market. Supports SIP and the most popular encoding algorithms and is 100% compatible with N/ACIP EBU Tech3326 technical recommendations.

IP Advantages: Adaptive buffer to absorb network jitter. Automatic IP parameter configuration through DHCP. FEC.

AEQ SIP Server. AEQ SIP Server - To simplify IP connections, AEQ offers its own SIP server – and at no cost to you.

Continuous data channel. Allows you to transport embedded data within the audio stream, as an example, for equipment remote control.

Includes encoding algorithms for any purpose. Includes the compulsory algorithms according to N/ACIP EBU Tech3326 technical recommendation plus the AEQ LD low delay algorithms. AAC algorithms can be supplied optionally.

Professional Audio. The unit has balanced analogue inputs and outputs on XLR connectors. Optionally, the unit can be equipped with AES/EBU Digital I/O's.

Reduced foot-print. Two Phoenix Mercury can be allocated together in 1RU 19" rack mount.

APPLICATIONS

STL Link (Studio-Transmitter Link). Through IP connections on private VLAN, IP Radio-links, WiMAx, WiFi, ADSL, Cable MODEM, etc., can be used to send the program signal from the production centre to the radio transmission sites.

Radio station networks. The unit can be used to interconnect the audio of the different radio stations in the network through IP. Since the unit is bidirectional the audio can be distributed to the different radio stations in the network and at the same time audio contribution can be established in the opposite direction.

Outside broadcasts and contributions. The unit is able to establish connections with other codecs and smartphone codecs, Phoenix (Studio, Mobile, Mercury, PC, Pocket o Lite) or from other manufacturers that are N/ACIP compliant.

Digital Hybrid for IP telephony. Can communicate through SIP with an IP switchboard (Asterisk or similar) that accepts calls in G711 and HD voice calls in G722 from IP telephony smartphone codecs, IP trunking and through gateways, calls from ISDN and POTS (PSTN). This type of communications is converted into 4W audio and made readily available for On-Air broadcasting.

Technical Specifications and Features

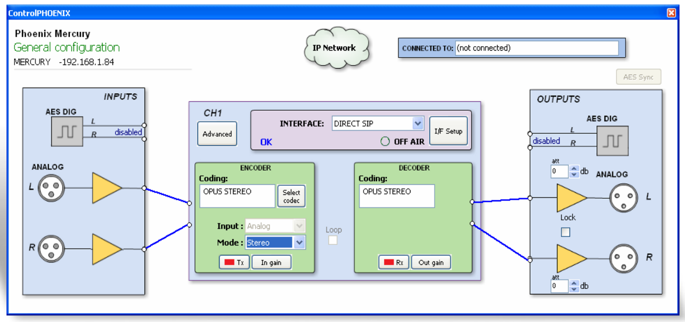
Broadcast Communications for your Station or Network of Stations

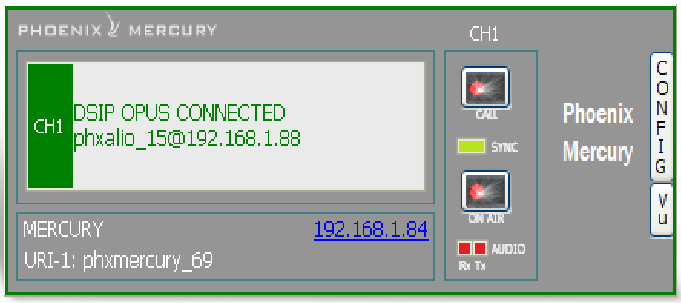
The control application allows the user to discover all the available Phoenix Mercury, Phoenix Venus and Phoenix Studio units within a local area network and to control them coordinately using a PC or a group of workstations. Further, remote units outside the LAN can be controlled, enabling integral management of an IP communications network.

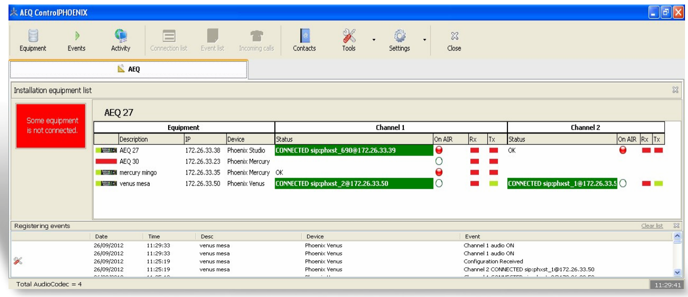
Each codec will show as a separate window for operation that will enable us to operate as if we were in front of the equipment's front panel, establishing, answering and cancelling calls.

This window also gives access to a configuration panel where we can choose the encoding algorithms, connection modes, etc. All the operation and configuration options are available in an intuitive and handy way.

Also, there is a status window that is displaying the codec(s) status at all times. We can verify the status of all codecs in our system at a glance. The application also provides a convenient way of managing the Codec's internal phone book or a common system connections database that can be transferred to each codec in the network.







Analog audio inputs
- 2 x XLR female. 9Kohm. Electronically balanced.
- Professional line level.

Analog audio outputs
- 2 x XLR male. Output impedance < 100 ohm.

Electronically balanced. Professional line level
- Digital audio I/O.
- DB15 connector. AES/EBU interface. SRC on input.
- Audio specifications.
- Nominal input level. 0dBu.
- Maximum input level. +20dBu.
- Nominal output level. 0dBu.
- Maximum output level. 20dB above nominal level.
- Max. distortion, linear audio <0.03%.
- THD + N in SRC @1KHz: -117 dB.
- Dynamic range for linear audio.>105dB.
- Crosstalk <-70dB.
- Frequency response (+/- 0.2dB).
- 50Hz – 15KHz in MPEG 1 L II.
- 20Hz – 20KHz in MPEG 4 AAC*, and for linear audio PCM.
- 50Hz – 7KHz in G722.
- 50Hz - 3KHz in G711.
- Analog I/O: A/D and D/A 24 bit Sigma-Delta converters, 48 kHz max.
- Modes: Mono, Dual Mono, Stereo.

IP communications interface
- Ethernet port. LAN 10/100 base T.
- Connector RJ45.
- Fully compliant N/ACIP EBU Tech 3326.
- SIP: Compliant with EBU-Tech 3326 recommendation.

AUX DATA
- DB 9. 9,6 Kbps to 38,4 Kbps flow embedded in the RTP stream.

USB
- Type A. Master or Slave operational mode.

Encoding algorithms
- **OPUS** with Fs= 48KHz, mono, stereo, with 3 mono and 4 stereo presets. Bit rates between 12 and 256Kbps. Audio bandwidth between 6 and 20 kHz.
- **G711** A law, µ law (64kbps, low delay, 3,5 KHz audio bandwidth)
- **G722** (64 Kbps, low delay, 7 KHz audio bandwidth).
- **AEQ-LD** Fs=16, 32 or 48KHz, mono or stereo. Bit-rates between 64 and 384Kbps, audio bandwidths between 7 and 20KHz.
- **MPEG1 y 2-LII**, Fs between 16 and 48 KHz, mono, stereo, dual channel and joint stereo. Bit-rates between 64 and 384 Kbps. Audio bandwidths between 10,5 and 16,5 KHz.
- **AAC-LC*** high quality, with Fs=24, 32 and 48 KHz, mono, stereo, MsStereo, bit-rates between 32 and 256Kbps., audio bandwidths between 9 and 20 KHz.
- **AAC-LD*** high quality and low delay, Fs=48 KHz, mono, stereo and MsStereo. Bit-rates between 32 and 256Kbps, audio bandwidths between 8 and 20KHz.
- **PCM (linear)** very low delay and transparent quality. Fs=48 KHz or 32 KHz @ 12, 16, 20 or 24 bits/sample, mono or stereo (bit-rates between 576 and 2304 Kbps), audio bandwidths between 15 and 20 KHz.
- **Smart RTP** call-initiation protocol that simplifies connection to compatible codecs.

General Features
- Range of operational temperature: -10 to + 45 º C (14 to 114 º F)
- Dimensions: 1/2 Rack - 1RU, 211 x 44 x 200 mm (8.30" x 1.75" x 7.88").
- Weight: 1 kg (2,2 lbs).
- Power supply 12 V DC. External power supply included.
- Input Power: 110-240V AC, 50-60Hz. 8.5 w. Auto-range.
- Ventilation. Natural convection – totally silent.
- Apt for in-Studio use.

* Specifications subject to change without prior notice.
* The AAC encoding algorithms are optional for the Phoenix AudioCodecs.

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