PHOENIX STRATOS

Dual Multi-Format Full Duplex Audiocodec

AEQ

Dual Stationary Audiocodec with IP, ISDN and X21/V35 connectivity

Two stereo, bidirectional audiocodecs for communications over IP, ISDN and X21/V35 lines. Allows the sending of two independent stereo/dual or four mono signals to different receivers.

Unique advantages from design, such as fully independent dual channels for program and coordination or backup, each one with its respective returns, and advanced user interface.

Physical user interface in the front panel and control software running on a PC or set of PCs, that allows local or remote, control of a single unit or pool of units.

Compatible with most codecs from other manufacturers on IP and ISDN interfaces: Supports SIP and the most popular encoding algorithms and is 100% compliant with N/ACIP EBU Tech 3326 technical recommendations.

IP Advantages: Adaptive buffer to absorb network jitter. Automatic IP parameter configuration through DHCP. Automatic adjustment of the reference clock to synchronize both ends. FEC. It sends a same stream to multiple destinations with multiple-unicast.

Remote monitoring: includes SNMP server that allows the user to monitor its status, alarms, etc. together with other pieces of equipment in the system, even from different manufacturers, by simply using any standard SNMP client.

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

Ancillary data channel. Offers transport of embedded auxiliary data via ISDN or RTP stream over IP, in order to control equipment at the remote end.

GPIs and GPOs. It incorporates 4 general purpose inputs and outputs for signalling and control.

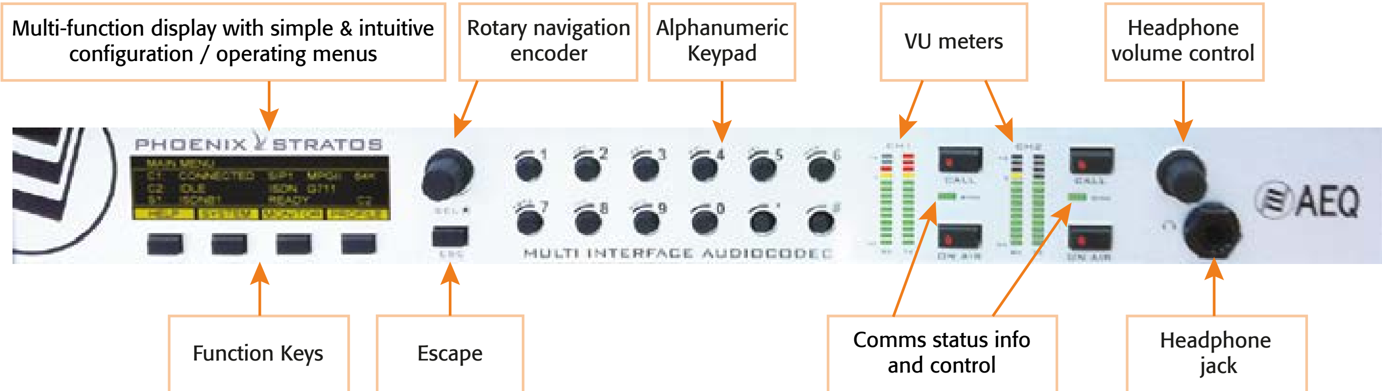
Includes encoding algorithms for any purpose. Includes the compulsory algorithms according to N/ACIP EBU Tech3326 technical recommendation plus the AEQ LD low delay algorithms. AAC algorithms can be supplied optionally.

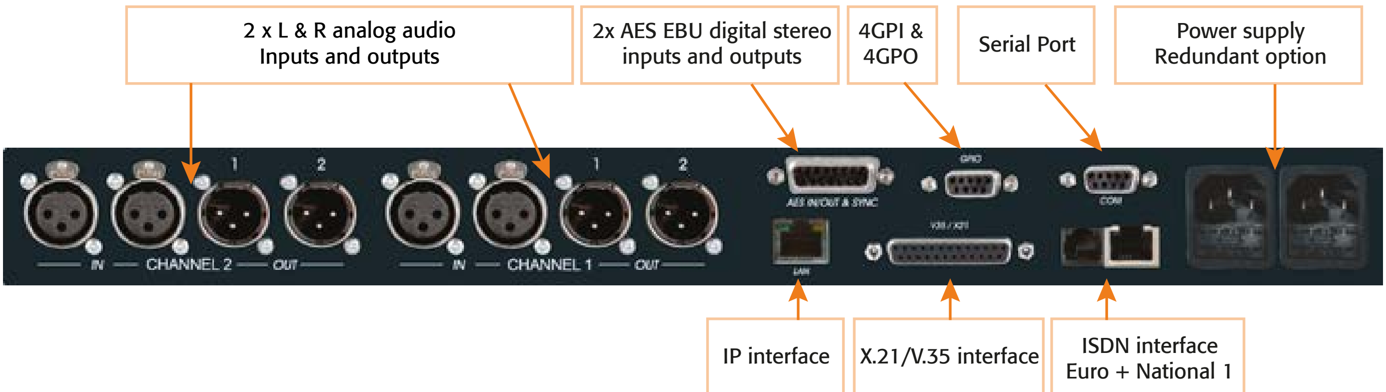
Analogue and Digital Professional Audio. The unit includes two pairs of line level XLR balanced analogue inputs and outputs, and AES/EBU digital I/O’s.

Silent operation. This device is cooled by natural convection in order to be suitable for Studio installation. No fans are installed.

Professional, comprehensive, redundant, simple and controllable. For your station or network communications.

Product details.





Applications

STL Link (Studio-Transmitter Link). Through IP connections on private VLAN, IP Radio-links, WiMAx, WiFi, ADSL, Cable MODEM, etc., it is possible to send up to two stereo or four mono programs to the radio transmission sites (one or two destinations) and to remote control and supervise from the production centre.

STL links can also be built using point-to-point V35/X21 synchronous links, switched ISDN, or IP connections, with optional backup on a synchronous switched ISDN network.

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 provides bidirectional audio, a signal 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.

IP Multiple unicast mode allows a single STRATOS codec to send up to two different programs, each one to a group of correspondents, while receiving feedback from one in each group. This way, the number of required audiocodecs at the network headquarters can be reduced.

Radio station networks can also be built using synchronous point-to-point V35/X21 links or switched ISDN networks.

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

Audio contribution can be performed on assorted IP networks, such as private VLANs, IP radio links, WIMAx, WiFi, ADSL, cable Modem, Inmarsat or similar IP satellite links, etc.

It can be connected via ISDN with most audiocodecs on the market including, of course, AEQ Phoenix Stratos, Mobile, Studio, Eagle, Course ISDN, SWING, M-PAC and TLE02. The integrated ISDN module features both S and U interfaces on RJ45 and RJ11 connectors, and internally supports Euro ISDN and National 1 protocols for worldwide usage.

IP APPLICATION NOTES USING AEQ AUDIOCODECS

A collection of application notes elaborated by AEQ is available, showing more than 10 connection examples for installed and mobile uses, suggesting particular network access devices and describing the way to configure them and of course the codecs, in order to ease system implementation.

http://www.aeq.eu/products/phoenix-app-note or **http://www.aeqbroadcast.com/products/phoenix-application-notes**

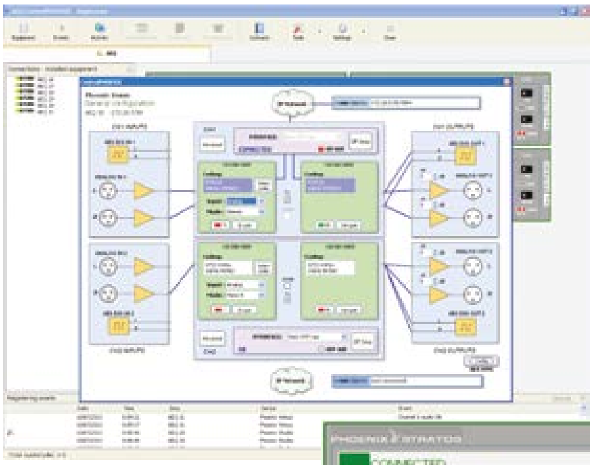
Control software

Each STRATOS unit includes all the necessary controls for its operation and configuration on the front panel, so a PC is not required to setup or use it. But audiocodecs are not readily accessible to an operator very often, but located in a central control room, links room or at any remote location that we need to control in order to connect to our local codec.

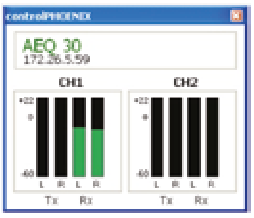
Phoenix audio codec system at London 2012 for Eurovision. 24 codecs pool working together with another 48 units distributed in different countries.

That's why each unit is supplied with an individual control and configuration PC application, including friendly, well differentiated setup (Fig 1) and operation (Fig 2) windows. The application allows the user to work with the unit just as being located in front of its front panel controls, selecting encoding modes, connection methods and establishing, answering and ending calls. All configuration and operational functions are presented in a very intuitive way.

(Fig 1)



(Fig 2)



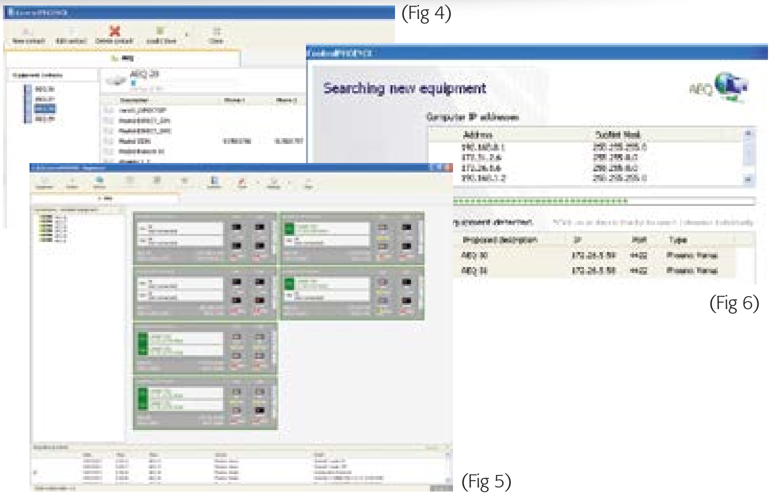
(Fig 3)

The software includes audio presence indicators as well as remote real-time VU-meters allowing the monitoring of incoming and outgoing audio levels for each device, no matter where physically located. (Fig 3)

It also provides a call book management application with copy functions that allows the user to generate a central contacts list and individual subsets for each codec in the network. (Fig 4)

If desired, it is possible to open as many instances of the application as required in order to control all the audiocodecs in your network from a single PC. But as the quantity of units increases, purchasing the full license for multi-codec control becomes more convenient, enabling well organized exploit of the codecs pool from a single program instance. (Fig 5)

Multi-codec control software allows you to automatically discover all the Phoenix Stratos, Mercury, Venus and Studio equipment in the same local network in order to coordinately control them from a PC or group of PCs. Remote units can also be controlled over Internet, thus enabling an integral management of the communications network. (Fig 6)

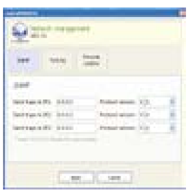


(Fig 4)

(Fig 6)

(Fig 5)

(Fig 7)



(Fig 8)

A summary list of the main status of all codecs in the system is also available. (Fig 7)

Furthermore, if your station has a SNMP management system installed, you will be able to incorporate your STRATOS codec units so they can also send the "traps" with issues of interest to your SNMP agent. (Fig 8)

phoenix stratos
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Technical Specifications

Inputs and Outputs

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

Analog audio outputs
4 x XLR male. Output impedance < 100 ohm. Electronically balanced. Professional line level.

Digital audio I/O
DB15 connector. 2 AES/EBU interfaces. SRC on inputs.

Headphone Output
1 x ¼" Stereo Jack, with front panel volume control.

Synchronization: AES/EBU output can synchronized with the inputs sampling frequency.

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 <0.003%.
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.

Communications interfaces

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.

ANCILLIARY DATA in IP: DB 9. 9,6 Kbps., 19,2 kbps. or 38,4 Kbps. flow embedded in the RTP stream.

X.21/V.35 communications interface: DB25, binary rates of 64/128/256Kbps.

ISDN / RDSI communications interface: Supports use of Euro ISDN and National-1 communications module. With up to two B channels supported per module. "S" interface (2B+D) Euro ISDN compliant (ETS 300 012, ETS 300 125, ETS300 102), RJ-45 connector. "U" interface (2B1Q) ANSI compliant (ANSI T1.601-1992, T1.602-1996,T1.607 - 1998), RJ-11 connector.

ANCILLIARY DATA in ISDN: DB 9. 9,6 Kbps flow embedded into the audio data stream.

GPIO: DB15 connector with 4 optocoupled GPIs and 4 open collector GPOs.

Other features:

- Automatic backup, selectable between IP, V35 and ISDN.
- Multicast IP: transmission and reception.
- Multiple-unicast in RTP-raw mode: allows the unit to send a same stream to up to 10 different IPs.
- SIP: according to EBU-Tech 3326 recommendation. Possibility of operation with or without SIP Proxy server.

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 384 Kbps, audio bandwidths between 7 and 20 KHz.
MPEG1 & 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 256 Kbps, 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 256 Kbps, 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).

Front panel control:
- 1 x 12 key, alphanumeric keypad.
- 1 x OLED display.
- 4 x function keys (used with display menus).
- 1 x rotary encoder and escape key (used with display menus)
- 4 x 14 segment LED VU meters.
- 4 x comm status LED indicators.
- Configuration Wizard (internal menu).

Dimensions and weight: 1RU, 482 x 44 x 280 mm ; 19" x 1.75" x 11").Weight: 3,5 kg (7,7 lbs)

Input power: 110 240 V AC, 50 60Hz. 12 w. autoranging. 3 PIN IEC connector*.
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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