PHOENIX VENUS 4 AND 4+

Dual channel, full duplex stereo ip audiocodec with local analogue, digital and dante aoip connectivity

CARRIER GRADE AUDIOCODEC

FOR YOUR REMOTE COMMUNICATIONS

General Description and Equipment Features

Dual Channel, Rack-mounted Stereo Audiocodec with IP Connectivity. Allows for two stereo/dual or four mono connections (to one or two destinations).

Unique possibilities from design, such as fully independent, dual channels for program and coordination, each one with its respective returns, or to send two mono or stereo programs to the same or different destinations.

Exclusive Connectivity Tools. With AEQ audiocodecs you benefit from the Smart RTP communications protocol that simplifies the connection, even through Dynamic Domain Name Service – DDNS. This avoids the need to know the destination IP, since this may change over time.

Compatible with third-party audiocodecs on the market. Supports SIP and the most popular encoding algorithms, and is 100% compatible with N/ACIP EBU Tech-3326 technical recommendations.

AEQ provides a SIP server as a courtesy, free service, to customers with AEQ Codecs. Optional RTP Relay service that transports the audio through the SIP server, simplifying connectivity for difficult scenarios.

IP connectivity tools. Adaptive buffer to absorb network jitter. DHCP -automatic configuration of the connection parameters-, automatic adjustment of the reference clock to synchronize both ends, error correction (FEC). Sending the same audio to several receivers is possible using multiple unicast.

3 network ports, Allows for the separation of control traffic, encoded audio (WAN) and local audio over IP (AES67/Dante).

Remote monitoring, includes SNMP server that allows the user to remotely monitor its status, alarms, etc. by simply using any standard SNMP client.

Double continuous data channel. Allowing for the transport of an independent data flow associated to each communication channel.

GPIs and GPOs. Equipment incorporates 4 general purpose inputs and 6 outputs, for signaling and control.

Includes encoding algorithms for any purpose. In addition to the recommended OPUS algorithm, VENUS 4 incorporates the encoding algorithms specified as Mandatory in the N/ACIP EBU Tech3326 recommendation, plus the low-delay AEQ LD. For other options, please contact us.

Analogue, digital and IP professional audio. The equipment has, as standard, two pairs of balanced analogue audio inputs and outputs with professional line level and XLR connectors, duplicated with AES/EBU digital audio inputs and outputs. Local audio over IP (AES67 / Dante) is available as an option. The input mode is selectable (Analogue, AES/EBU or Dante), and the outputs are distributable (Analog + AES/EBU or Analog + Dante), independently for each of the two channels.

Silent. The equipment is cooled by natural convection, making it suitable for studios, as it doesn't have fans.

Reliable. Possibility of including double AC power supply and/or one or two 48 V DC power inputs.

FRONT PANEL

Phoenix Venus 4 with basic indicators on the front panel, for control through the Control Phoenix application.

Status indicators for networks and communications, AoIP connection and power.

The Phoenix Venus 4+ + includes a user interface on its front panel. This simplifies operation and maintenance tasks, especially when making calls and activating Presets. Further, it also allows access to numerous configuration options for the device and also displays real-time stereo vu-meters for inputs and outputs of both channels.

In addition to the network and communication status indicators, mains switch and mains power LED, it includes a full-colour TFT display, rotary encoder with push button and, below it, ESC cancel/return key. 12-key numeric keypad, 2 call/hang-up keys and a key to load PRESETs.

REAR PANEL

Two analogue audio inputs and outputs through XLR connectors, dual AES3 input and output, two serial ports for ancillary data, 4 GPI, 6 GPO, two network ports (NET1 and NET2) and a third AoIP Dante network port, power supply (up to two AC and two DC-48V inputs).

CONTROL PHOENIX APP

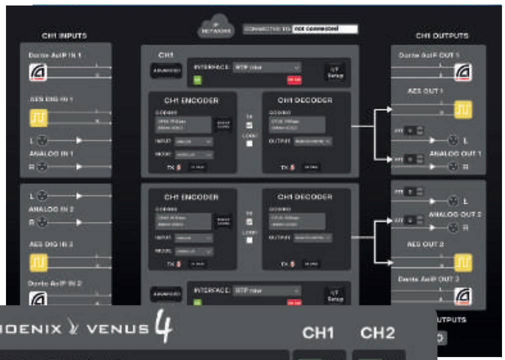


Fig.1

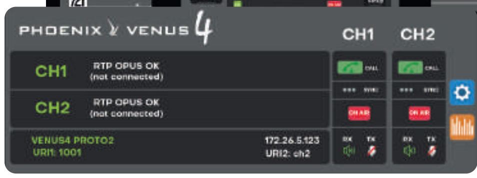


Fig.2

There is a very User-friendly graphical interface for configuration (Fig.1) and operation (Fig. 2). This allows the user to work with the unit just as if being located in front of its controls, selecting encoding modes, connection methods and making, answering and ending calls.

All configuration and operational functions are presented in a very intuitive way.

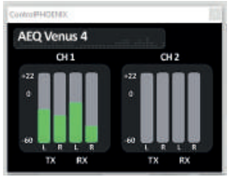


Fig.3

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

The application also provides a phone-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). You can control all the local or remote codecs in your network from a single PC.

If desired, it is possible to open as many instances of the application as required in order to control audiocodec pairs 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).

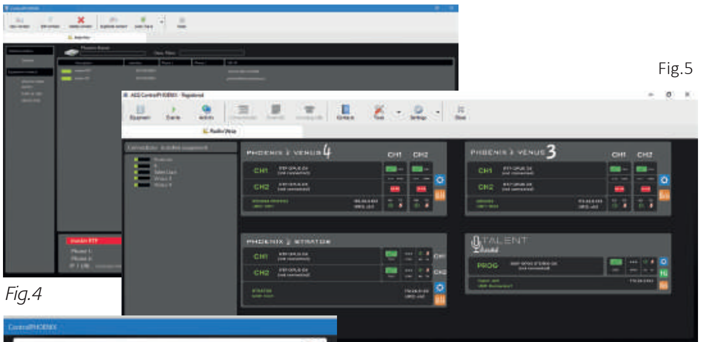


Fig.4

Fig.5

Multi-codec control software allows you to automatically discover all the AEQ IP Audiocodecs on 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). You will be able to view the summarized general status of all the codecs in the system from a single window.

It is also possible to manage a common phone-book or connection database (Fig. 7). Furthermore, if your station has a SNMP management system installed, you will be able to incorporate your VENUS 4 codec units so they can also send the “traps” with issues of interest to your SNMP agent. (Fig. 8).

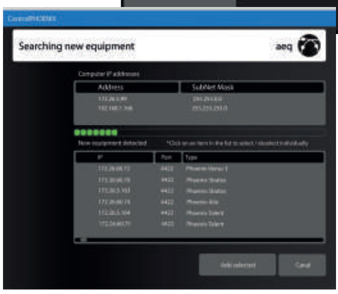


Fig.6

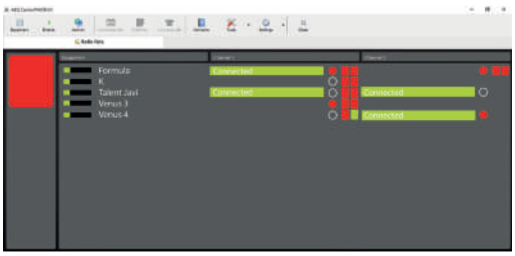


Fig.7

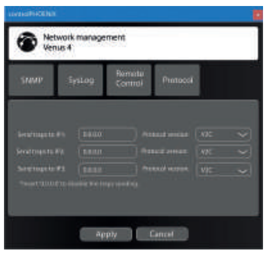
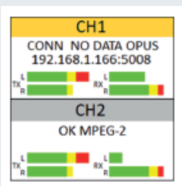


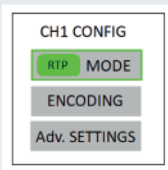
Fig.8

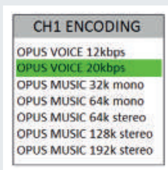
FRONT PANEL INTERFACE OF PHOENIX VENUS 4+

The user menu allows for basic operation of the equipment. More specifically, the front panel allows the user to perform the following operations:



Show the general status of each channel of the equipment: idle, bidirectionally connected, connected but without data in reception, registration error..., (represented in different colors for a quick visualization), selected encoding family and the complete vu-meters for both channels.





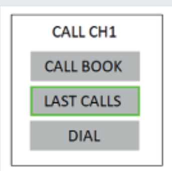
Access to a settings menu, in which you can select the mode for each channel (RTP / Direct SIP / Proxy SIP), change the encoding mode or profile, input mode (analogue / digital AES / Dante), choose a SIP account and modify the type and size of the receive buffer. Checking and modifying the IP configuration of the NET1 and NET2 interfaces, outgoing control connection configuration (studio IP and port), verifying the Fw versions installed in the equipment, and REBOOTing are also possible.

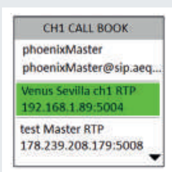
Access to a call menu in which you can choose to call from:

a) the phone-book, previously stored in the equipment.

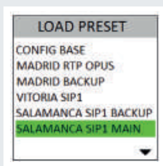
b) call history.

c) dialing directly using the numeric keypad.





PRESET key. Pressing it will access a menu in which we will select the preset to launch.



Applications and Technical Specifications

USES AND APPLICATIONS

STL LINK	BROADCASTING NETWORKS	REMOTE CONTRIBUTIONS
(Studio to Transmitter Link). Up to two stereo or four mono programs can be sent (to one or two destinations) through private VLAN IP connections, IP Radio links, WiMAx, WiFi, ADSL, Cable Modem, etc., as well as remote control and command services, between production centers and broadcasting centers. Thanks to multiple-unicast, each of the two stereo or dual channels can be sent to up to four destinations (approx.), with return from one of them.	Different broadcast stations can communicate to distribute various programs, while simultaneously establishing contribution circuits in the opposite direction through IP networks, when quality of service is negotiated with an operator. Distribution of each channel's input signal to a group of correspondents (with return from one of them) is possible using the Multiple Unicast mode, thus reducing the number of audiocodecs in the central headquarters of a network.	It can communicate over IP networks with AEQ Talent and Phoenix (Alio, Venus, Stratos, Studio, Mobile, Mercury) audiocodecs or third-party units-provided that they are compatible with N/ACIP-, and even with softphones, in order to incorporate broadcast audio from anywhere. There is a wide variety of IP networks which can be used for contribution: private VLANs, IP radio links, WiMAx, WiFi, ADSL, Cable MODEM, satellite or similar IP links, etc.

CONFIGURATION GUIDES FOR IP APPLICATIONS WITH AEQ CODECS

A collection of application notes using AEQ IP audiocodecs is available, showing more than 10 connection examples for fixed and mobile uses, suggesting network access equipment, and describing how to configure these and the codecs, in order to facilitate system implementation.
http://www.aeqbroadcast.com/products/phoenix-application-notes

TECHNICAL SPECIFICATIONS

INPUTS AND OUTPUTS

Analog audio inputs.
• 4 x XLR female. Zin> 9Kohm. Electronically balanced. Professional line level.
Analog audio outputs
• 4 x XLR male. Zout < 100 ohm. Electronically balanced. Professional line level.
Digital audio I/O
• DB15 connector. 2 AES/EBU interfaces. SRC on inputs.
Optional IP audio inputs / outputs
• 2 stereo inputs and 2 stereo outputs with DANTE / AES67 technology. Dedicated 1 Gbps RJ45 port.
External synchronization:
• AES/EBU output can synchronized with the inputs sampling frequency. Synchronization can also be transported through the AoIP network.

AUDIO SPECIFICATIONS

• Nominal input level. 0dBu.
• Maximum input level. +20dBu.
• Nominal output level. 0dBu.
• Maximum output level. 20dB above nominal level.
• Distortion at maximum level (encoder to decoder loop in PCM mode): <0.003%
• THD + digital input noise at SRC @1KHz: -117 dB.
• Dynamic range for linear audio >100dB.
• Crosstalk <-70dB. at 1kHz.
• Analog I/O: A/D and D/A 24 bit Sigma-Delta converters, 48 kHz max.
• Working modes: Mono, Dual Mono, Stereo.
• Frequency response (+/- 0.2dB): Up to 20 Hz- 20 KHz. According encoding algorithm.

COMMUNICATION INTERFACES

• 10/100 baseT Ethernet ports (NET1 and NET2) for combined or separate use of control and WAN. RJ45 connector. Fully compliant with EBU Tech 3326 recommendation of N/ACIP.SIP.
• AoIP option using DANTE / AES67 technology. Additional independent 1Gbps port.

GENERAL FEATURES

• Range of operational temperature: -10 to + 45 º C (14 to 114 º F).
• Dimensions and weight: 1RU, 482 x 44 x 200 mm ; 19" x 1.75" x x 7.87").
• Weight: 2,8 kg (6,172 lbs) aprox.
• Input power: One or optionally two power Supplies: 110 - 240 V AC, 50-60Hz. 12W. Auto-ranging. 3 PIN IEC connector*. One or two 48V DC power supplies (optional).
• Ventilation. Natural convection – totally silent. Apt for in-studio use.

ANCILLIARY DATA

• Two DB 9, one for each communications channel.
• Two individually configurable flows at 1.2, 2.4, 4.8, 9.6, 19.2 or 38.4 Kbps embedded in the RTP stream.

GPIO

• DB15 connector, 4 optically-coupled general purpose inputs (GPI), 6 optically coupled, open collector GPO outputs.

OTHER FEATURES

• Multicast IP: transmission and reception with subscription management by IGMP protocol.
• Multiple-unicast in RTP-raw mode: allows the unit to send a same stream to several different destinations (depending on the encoding algorithm).
• SIP: according to EBU-Tech 3326 recommendation. Possibility of operation with or without SIP Proxy server.

ENCODING ALGORITHMS *

• OPUS with 48KHz sampling frequency, mono or stereo. 5 selected mono modes and 3 stereo ones, with bitrates between 12 and 192kbps, very low delay and audio bandwidth between 6 and 20kHz.
• OPUS Voice (reduced bw) 12kbps : 6kHz.
• OPUS Voice 20kbps.: 8KHz.
• OPUS Voice 192kbps: 8kHz.
• OPUS Music mono (reduced bitrate):32 kbps.: 20kHz.
• OPUS Music mono 64kbps.: 20kHz.
• OPUS Music Stereo (reduced bitrate) 64kbps.: 20kHz.
• OPUS Music Stereo 128kbps.: 20kHz.
• OPUS Music Stereo HQ 192 kbps.: 20kHz.
(The receiver is synchronized and can decode the received stream of other OPUS modes as long as they are sampled at 48 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.
• 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 16 and 20 KHz.

*For other options, contact us.
Features subject to change without prior notice.

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