PHOENIX VENUS 3

DUAL STEREO AUDIOCODEC IP FULL DUPLEX

WITH LOCAL ANALOG, DIGITAL and DANTE™ AoIP CONNECTIVITY

AEQ

Main features

- Two stereo, bidirectional audiocodecs for communications over IP. Allows two stereo/dual or four mono connections (to one or two destinations).
- Unique advantages from design, such as fully independent dual channels for program and coordination, each one with its respective returns.
- User interface . Control software running on a PC or set of PCs, that allows control of a pair of devices or pool of units. Includes real-time VU meters, contact list management, events log and alarms registering.
- Compatible with third party audiocodecs in the market. Supports SIP and the most popular encoding algorithms. and is 100% compatible with N/ACIP EBU Tech-3326 technical recommendations.
- Additional features when used with other AEQ codecs: When connected to another AEQ codec, it can take advantage of a exclusive set of tools that makes the establishment of communication and the control of the unit a simple task:
  - A selection of OPUS encoding algorithms that ensure high quality audio with a minimum delay.
  - The Smart RTP proprietary call-initiation protocol that simplifies connection to compatible codecs, without requiring manual establishment / hanging of the calls or selection of the coding modes at both ends of the communication.
- 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. Allows for separation of control and audio traffic, thanks to its two independent Ethernet ports.
- 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.
- Webserver management: in order to monitor each channel status and remotely upgrade the device's firmware.
- 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 two embedded auxiliary data via RTP stream over IP, in order to control equipment at the remote end.
- GPIs and GPOs. VENUS 3 incorporates 4 general purpose inputs and outputs for signalling and control.
- Includes encoding algorithms for any purpose. Features a selection of high performance OPUS coding algorithms covering every need, from standard voice quality in narrow band networks to transparent audio quality with a very moderate data rate, always with very low delay. Also it includes the compulsory algorithms according to N/ACIP EBU Tech3326 technical recommendation (G711, G722, MPEG-2), 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.
- Optional IP local audio. The unit optionally offers IP connectivity with DANTE™ technology to provide its two pairs of audio outputs to compatible devices in the local area network, or receive the inputs from other equipment through the network.
- Silent operation. This device is cooled by natural convection, in order to be suitable for Studio installations. No fans are installed.
- Reliable. Includes two universal range AC power supplies, or optionally one AC input power supply plus a 48V DC power supply acting as a backup, or even two 48V DC inputs.

PHOENIX VENUS 3
DUAL FULL-DUPLEX IP AUDIOCODEC
with AoIP INPUTS & OUTPUTS

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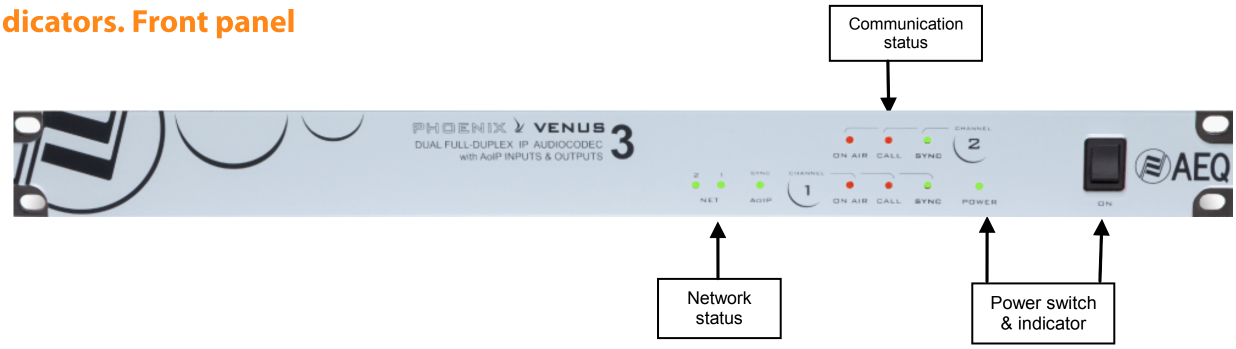
Product details

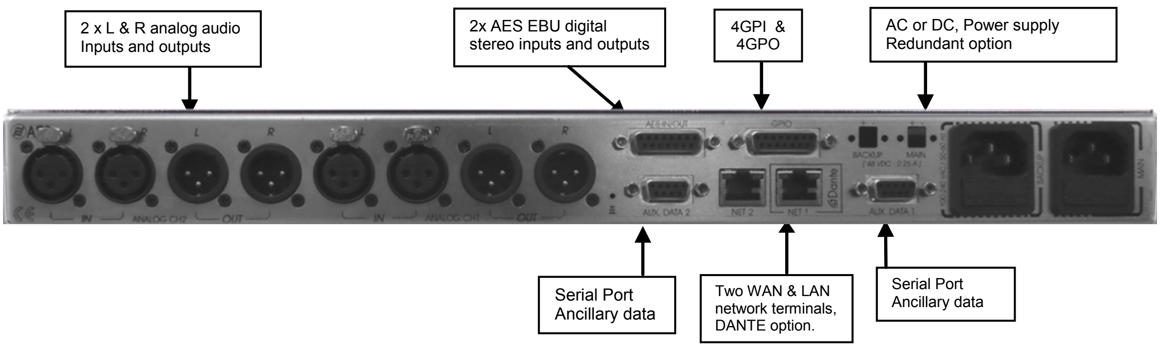
With Phoenix VENUS 3, AEQ provides an IP audiocodec unique features: it is capable of establishing two simultaneous, bidirectional calls with different formats and quality – two audiocodecs within the same device. Software with optional multi-position and multi-device control and remote access to the units.

Analog and Digital inputs and outputs, and AoIP DANTE™ that allows for transportation of the inputs and outputs through a local area network, making them available for any device within the network, being either an AEQ unit or any of the more than 200 compatible manufacturers. GPiOs, redundant supply options from AC and 48VDC. Opus coding algorithm, EBU N ACIP Tech 3326 connection system, and proprietary SmartRTP for ultimate effectiveness. Dual LAN port, dual RS232 ancillary data link.

Indicators. Front panel



Connectivity Rear panel



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.

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 adio 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 VENUS 3 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.

Outside broadcasts and contributions. VENUS 3 can communicate with Phoenix codecs and soft-phones (ALIO, Venus, Stratos, Studio, Mercury), preferably using SmartRTP connection mode, or with other models (Mobile, PC, Pocket or Lite) and even third-party N ACIP compatible devices, in order to incorporate your broadcasting audio from any place.

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.

Remote intercom panel link. Venus 3 can be controlled from a compatible intercom system's (KROMA CONEXIA or KROMA CROSSNET) control software, in order to have access to remote user panels. Venus 3 will establish a communication with the corresponding audio panel in the other end by just pressing a key in either the local or remote panel.

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 http://www.aeqbroadcast.com/products/phoenix-application-notes

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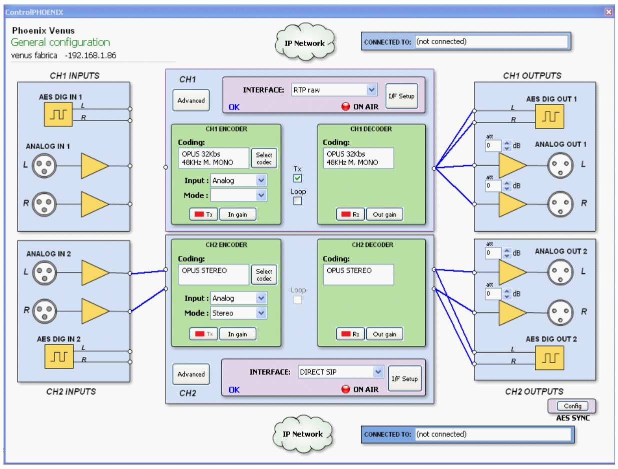
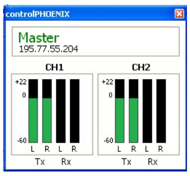
Control Software

VENUS 3 devices have been designed to be controlled locally or remotely through software applications. 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.

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



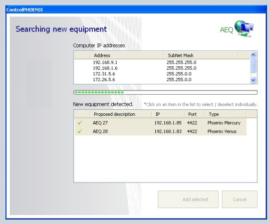


The software includes audio presence indicators as well as remote real-time VU-meters allowing the monitoring of incoming and outgoing audio levels for each device, no matter where physically located.

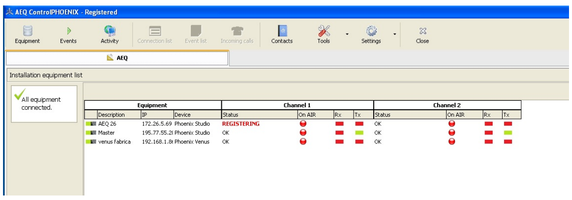
The application also provides a call book management application with copy functions that allows the user to generate a central contacts list and individual subsets for each codec in the network.

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.

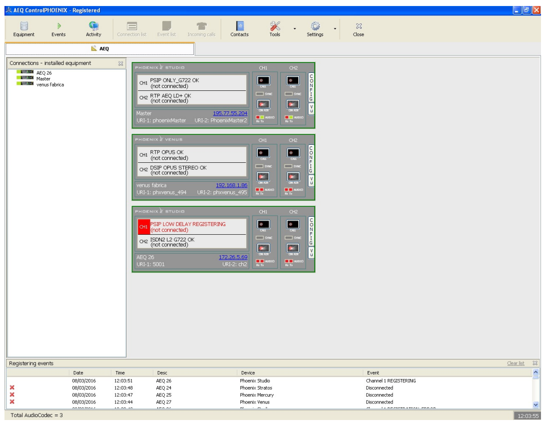
Multi-codec control software allows you to automatically discover all the Phoenix Stratos, Phoenix Mercury, Phoenix Venus Phoenix Venus 3, Phoenix Alio and Phoenix Studio equipment in the same local network in order to co-ordinately control them from a PC or group of PCs.



Remote units can also be controlled over Internet, thus enabling an integral management of the communications network.



A summary list of the main status of all codecs in the system is also available.



Furthermore, if your station has a SNMP management system installed, you will be able to incorporate your VENUS 3 codec units so they can also send the "traps" with issues of interest to your SNMP agent.

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

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Technical Specifications

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.

Optional IP audio inputs / outputs
- 2 stereo inputs and 2 stereo outputs with DANTE™ technology through the RJ45 NET1 (LAN) connector.

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.
- Max. distortion linear audio <0.003%.
- THD + N in SRC @1KHz: -117 dB.
- Dynamic range for linear audio >105dB.
- Crosstalk <-70dB.
- Analog I/O: A/D and D/A 24 bit Sigma-Delta converters, 48 kHz max.
- Modes: Mono, Dual Mono, Stereo

- Frequency response (+/- 0.2dB): 20 Hz- 20 KHz. According encoding algorithm.

Communications interfaces
- LAN & WAN Ethernet ports. 10/100 base T. Connector RJ45.
- Fully compliant N/ACIP EBU Tech 3326.
- SIP: Compliant with EBU-Tech 3326 recommendation.
- AoIP uses DANTE ™ technology. You can find a list of compatible devices at www.audinate.com

ANCILLIARY DATA: Two DB 9, one for each audiocodec.Two individually configurable flows at 1.2, 2.4, 4.8, 9.6, 19.2 ó 38,4 Kbps embedded in the RTP stream.

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

Other features
- 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 48KHz sampling frequency, mono or stereo. 4 selected mono modes and 3 stereo ones with bitrates between 12 and 192kbps and audio bandwidth between 6 and 20kHz.
  OPUS Voice (reduced bw) 12kbps : 6kHz.
  OPUS Voice 20kbps.: 8KHz.
  OPUS Music mono (reduded bw):32 kbps.: 20kHz.
  OPUS Music mono 64kbps.: 20kHz.
  OPUS Music Stereo (reduced bw) 64kbps.: 20kHz.
  OPUS Music Stereo 128kbps.: 20kHz.
  OPUS Music Stereo HQ 192 kbps.: 20kHz

- **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.

General Features

- Range of operational temperature: -10 to + 45 º C (14 to 114 º F).
- 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: Two Power Supply 110 _ 240 V AC, 50 _ 60Hz. 12 w. autoranging. 3 PIN IEC connector*. One or two 48V DC power supplies (optional).
- Ventilation. Natural convection – totally silent. Apt for in-Studio use.

* The AAC encoding algorithms are optional for the Phoenix AudioCodecs.
* Check the application notes in the web site.

Ordering information

- o VENUS 3 Phoenix audiocodec with DANTE™ local connectivity, 80-250V 50/60Hz power supply.
- o VENUS 3 Phoenix audiocodec without DANTE™local connectivity, 80-250V 50/60Hz power supply.
- o DC 48V power supply (substituting one of the AC power supplies).
- o Control Phoenix software with license to simultaneously control more than two Phoenix audiocodecs.

Specifications subject to change without prior notice.