VX222e-S – Stereo Sound Card

Key Workflow: Linear PCM stereo stereo sound card for professional stereo audio workstations

VX222e-S is a professional linear (PCM) stereo sound card based on the PCI Express bus interface. It is designed for use in any professional PC-based audio system running under Windows and requiring balanced analog and AES/EBU connectivity.

At a glance

VX222e-S is the reference stereo PCM sound card designed for operating in continuous 24/7/365 use-environments as part of professional audio systems. It can be used with software applications based on standard driver interfaces such as WDM Kernel streaming, DirectSound, Wasapi, and ASIO for Windows. Compared to VX222e, it adds an AES11 sync input, a hardware sample rate converter on the AES/EBU input, and a word clock synchronization input.

- Developed for the broadcast industry
- High audio quality sound card that supports balanced analog and AES/EBU audio
- Interoperable with most of the third party software applications for audio production under Windows

Key features

- Stereo Linear PCM card
- PCI Express bus interface
- Two balanced analog inputs and outputs, +24 dBu max level
- One AES/EBU input/output, with high quality hardware sample rate converter on the input.
- Adjustable input and output analog and digital gains
- On-board 3-band parametric EQ and Maximizer effects
- Two GPI (dry contact) and two GPO (relays)
- AES11 synchronization input, and Word Clock sync input (uses a GPI input)
- Support for Windows 32-bit and 64-bit versions (Wasapi, DirectSound, ASIO)
- Breakout cables with XLR connectors for audio connectivity

Configuration

Bus/Format: PCI EXPRESS**TM** (PCIe®) x1 (x2, x4, x8 compatible)

Size: 168 mm x 99 mm x 20 mm

Power requirements (+3.3V/ +12V): 1 A / 0.2 A

Operating: temp / humidity (non-condensing): 0°C / +50°C • 5% / 90%

Storage: temp / humidity (non-condensing): -5°C / +70°C • 0% / 95%

Inputs

Analog line inputs (mono): 2 balanced

Maximum input level/impedance: +24 dBu / >10 kOhms

Digital inputs (stereo): 1 AES/EBU with hardware sample rate converter (SRC); conversion ratio from 1:8 to 7,5:1

Programmable input gain:

- analog: from –94 dB to +16 dB **Ä**
- digital: from –110 dB to +18 dB

AES11 synchronization: AES/EBU Sync (up to 200 kHz),

Other inputs

- WordClock (up to 192 kHz),
- 2 GPI dry contacts (1 GPI available if WordClock input is used).
- LTC

Outputs

Analog line outputs (mono): 2 servo-balanced

Maximum output level / impedance: +24 dBu / < 100 Ohms

Digital outputs (stereo): 1 AES/EBU, up to 200 kHz

Programmable output gain:

- analog: from –24 dB to +24 dB
- digital: from –110 dB to +18 dB

Other outputs:

- 1 stereo headphone output (600 W)
- 2 GPO (relay, 0.5 A, 48 VCC)

Connectors

Internal connector: inter-board synchronization

External connectors:

- 15-pin Sub-D for analog I/Os
- 15-pin HD Sub-D for digital I/Os, Sync., and GPIO
- 1 mini jack headphone stereo output (3.5 mm TRS female jack)

Audio specifications

Sampling frequencies available: Programmable from 8 to 192 kHz

A/D and D/A converter resolution: 24 bits

Supported audio formats: PCM (8, 16, 24 bits), Float IEEE754

Analog audio performance

Frequency response (record + play):

- at 48 kHz: 20 Hz – 20 kHz: +0 /-0.3 dB
- at 96 kHz: 20 Hz – 40 kHz: +0 /-0.4 dB
- at 192 kHz: 20 Hz – 80 kHz: +0 /-1.1 dB
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Channel phase difference: 20/20kHz: <0.2°/2°

Dynamic range (A-weighted):

- analog In: >104 dB
- analog out: > 106 dB

THD + noise 1 kHz at -2 dBfs:

- analog In: >-97 dB
- analog out: <-95 dB

Crosstalk (Analog in or out):

- 1 kHz at 24 dBu: <–115 dB
- 15 kHz at 24 dBu: <-100 dB

Sample rate converter performance

Maximum frequency: 192 kHz

Frequency ratio: from 1:8 to 7,5:1

THD + noise 1 kHz at –2 dBfs: <-130 dB

Development environments

OS supported: Windows versions from Windows 7 and Windows server 2003 (32-bit and 64-bit versions), Linux

API under Windows: WASAPI, ASIO, DirectSound, Digigram proprietary “np” SDK

API under Linux: Alsa

Main on-board processing features (accessible through the Digigram “np” SDK): PCM play & rec, Float IEEE754,direct monitoring, real-time mixing, level adjustment, panning, cross-fade, punch-in/punch-out, scrubbing, 3-band parametric equalizer, maximizer, frequency conversions