DSP88-D

Matrix Processor

VAL audio

Die Liebe zur Musik

OVERVIEW

The VAL Audio DSP88-D is a premium 16-input, 16-output Dante enabled matrix DSP processor, delivering high-performance audio processing with cutting-edge hardware and feature-rich control software. At its core, the ADSP SHARC 21489, 40-bit floating-point processor operates at a 48kHz sampling rate, ensuring exceptional audio fidelity and ultra-low latency for professional applications.

Designed for versatile and advanced signal processing, the DSP88-D offers a comprehensive suite of DSP tools on each channel, including high-pass, low-pass, and all-pass filters, parametric EQ (PEQ), delay, expander, compressor and limiter functions. Further to enhance usability and offer versatility the DSP88-D features advanced feedback suppression, acoustic echo cancellation (AEC), noise suppression and an intelligent automixer - all seamlessly accessible through the user-friendly control software.

For seamless integration into modern AV environments, the DSP88-D features an inbuilt USB sound card, enabling direct recording and soft video conferencing codec connectivity. Its extensive control and connectivity options, including USB, 8-way GPIO, Ethernet, RS232, and RS485 allow effortless integration into both new and existing AV setups.

Engineered to handle demanding audio environments, the DSP88-D is the perfect solution for auditoriums, museums, conference rooms, lecture theaters, clubs, entertainment venues, and F&B spaces. When paired with VAL Audio's professional speakers, amplifiers, and microphones, it delivers unparalleled performance, flexibility, and value, making it an indispensable tool for professional audio installations.

KEY FEATURES

- 8 analog and 8 Dante inputs
- 8 analog and 8 Dante outputs
- Complete matrix routing of all inputs and outputs
- High-performance ADSP SHARC 21489, 40-bit floating-point processor optimised for audio processing
- 24bit/48kHz sampling rate for exceptional audio quality and performance
- Comprehensive metering with 16 user presets
- Input channels : signal generator (sine wave, white noise and pink noise), expander, compressor, AGC, PEQ, feedback suppression and 48V phantom power
- Automixer with gain sharing and mic priority setting
- Acoustic Echo Cancellation (AEC) 16x16 for local and remote inputs
- Output channels : Low pass & high pass filters - Bessel, Butterworth, Linkwitz-Riley slopes from 6-48dB/oct, 8 Parametric equalizer (PEQ) - PEQ, low shelf, high shelf, low pass, high pass
- Delay (up-to 408m/1200ms) and limiter on output channels
- In-built USB sound card for USB recording, playback and soft video conferencing codec
- 8-way GPIO (General purpose input output) configured as inputs or outputs, allowing for flexible control of digital signals
- Camera tracking with look at me feature via voice control
- Ethernet, RS232/485 control interface for third-party control
- Full feature control software

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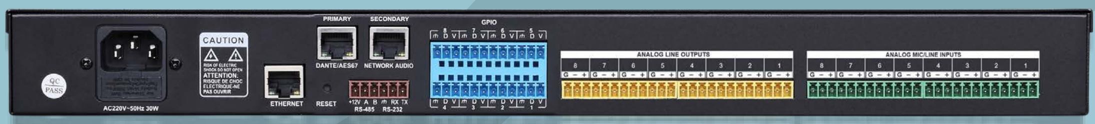

VAL AUDIO CONTROL SOFTWARE

SPECIFICATIONS	
Type	Matrix Digital Signal Processor
Input	8 analog, 8 Dante inputs, USB input
Output	8 analog, 8 Dante outputs, USB output
General Purpose Input Output	8-way GPIO configured as inputs or outputs
Processor	ADSP SHARC 21489, 40-bit floating arithmetic engine
Sampling Rate	48kHz, 24-bit ADC & DAC
Input Gain	0/3/6/9/12/15/18/21/24/27/30/33/36/39/42/45/48 dB
Phantom Power Supply	+48V/10mA max
Frequency Response	20Hz - 20kHz, ±0.3dB
Maximum Level	+18dBu
Input Dynamic Range	110dB
Output Dynamic Range	112dB
THD + Noise	< -94dB @17dBu
Channel Isolation @1kHz	-108dB
Input Impedance	5.4kΩ (balanced), 3kΩ (unbalanced)
Output Impedance	600Ω (balanced)
System delay	<3ms
Power supply	AC 110～240V， 50/60 Hz
Power consumption	55W
Dimension (WxDxH)	482mm x 260mm x 45mm
Net Weight	4Kg