# DSP-1616-DAN

## 16x16 Audio Processor with Dante

## **FEATURE:**

- Provide 12-ch balanced MIC/linear inputs and 12-ch balanced linear outputs
- Provide two standard DANTE network audio interfaces
- Support adaptive feedback suppression function
- Support the full-band adaptive acoustic echo cancellation technology
- Dynamic adaptive noise reduction technology is provided to reduce noise with signal level up to 18dB
- Auto Mixer function is provided to set the order of priority when multiple microphones are input at one time
- Inclusive of Digital signal processing modules such as Expander, Equalizer, Compressor, Auto Gain Control, Limiter, High Pass Filter, Low Pass Filter and Delay
- Capable to switch matrix routings
- Support volume control, meter, scene control, etc.
- 48V phantom power supply for 12-ch MIC inputs
- 48KHz sampling rate, 24-bit for A/D or D/A conversion
- Support 8-ch programmable GPIO function
- Compatible to run on Win 7 and Win 10, with standard RJ45 interface control
- Support RS-232 serial commands control



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This digital audio processor is typically used for video conference, distant learning, and telemedicine. It features 12-ch MIC/linear inputs, and 12-ch linear outputs. Two DANTE ports are also provided to ensure low latency in the audio processor. The product can process audio signals with algorithms, such as, full-band Adaptive Echo Cancellation (AEC), Adaptive Noise Suppression (ANS), Automatic Gain Control (AGC), and Auto Mixer, to output a clear, clean and resonant sound with high Signal-to-Noise ratio. Concise but intelligent, the processor is designed to be applied in scenarios without additional software assistance for debugging.

It is ready to use after installation, perfect for project implementation and testing. The product can be applied in a diverse range of installations and applications across industries, such as, smart system integration in small- medium sized conference room, instruction recording and distance teaching in education, court trial recording and virtual court trial in judiciary, surgery recording and video consultation in healthcare service, and command center establishment in governmental projects.







## **SPECIFICATIONS:**

#### **Technical**

Amplitude-frequency (20Hz~20KHZ@+4dBu)	±0.2dB
THD+N (1KHZ@+4dBu)	≤0.01%
SNR (linear input)	≥90dB
Dynamic Range	≥100dB
Channel Level Difference	±0.5dB
Channel Isolation	≥80dB
Max Input Level	20dBu
Max MIC Gain	40dB
Input Impedance	20ΚΩ
Output Impedance	3000
Sampling Frequency	48KHZ
A/D and D/A Conversion	24Bit
Phantom Power	+48 VDC
Connection	
Input	12 × Balanced MIC/LINE [3-pin phoenix connector] or
	6 × Stereo Audio [3-pin phoenix connector]
Output	12 × Balanced LINE [3-pin phoenix connector] or
	6 × Stereo audio [3-pin phoenix connector]
Digital Audio Interfaces	2 × Dante [RJ45]
Control	1 × LAN [RJ45]
	1 × RS-232 [3-pin phoenix connector]
	8 × GIPO [10-pin phoenix connector]
Mechanical	
Housing	1 x SPDIF IN [Optical audio connector
Color	1 x L/R AUDIO IN [3-pin 3.81mm Phoenix connector
Dimensions	1 x HDMI OUT [Type A, 19-pin female]
Weight	1 x L/R AUDIO OUT [3-pin 3.81mm Phoenix connector]
Power Supply	1 x RS-232 [3-pin 3.81mm Phoenix connector]
Power Consumption	1 x LAN (POE) [RJ45 jack]
Operation Temperature	1 x FIBER [Optical fiber slot]
Storage temperature	2 x USB 1.1 DEVICE [Type-A, 4-pin female]