

Dante Digital Signal Processor DP88



CATALOG

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Product Introduction

The AIDAP-DP88 is a professional networked digital audio processor from AISpeech, integrating 8×8 analog I/O and 8×8 Dante® I/O with 48 kHz / 24-bit processing. It incorporates core modules such as AEC, AGC, ANS, AFC, auto-mixing, ducker, and parametric EQ, delivering low latency and wide dynamic range for professional venues including meeting rooms, multi-function halls, auditoriums, and administrative centers.

Main Features

I/O Architecture

DP88 provides parallel 8×8 analog and 8×8 Dante® I/O with a visual matrix for flexible routing and bridging. DP88 handles conversion and inter-room links on a single processor, simplifying new builds and retrofits while reducing system complexity and cabling.

DSP Platform

DP88 runs on an ADI SHARC® floating-point platform at 48 kHz/24-bit, delivering low-latency, reliable processing with headroom for complex algorithms and full-scale matrix operations, ensuring consistent results across multi-channel workloads and changing room conditions.

Core Processing Suite

DP88 integrates AEC, AGC, ANS, AFC, auto-mixers, ducker, PEQ, filters, delay, and matrix. Per-channel chain configuration improves far-end clarity, raises gain-before-feedback, and streamlines multi-mic sessions for intelligible conferencing and dependable reinforcement.

Audio Performance

The DP88 delivers voice optimised reproduction and transient response, maintaining natural, stable and low-noise performance even in complex routing and multi-microphone scenarios.

Product Specifications

Function	Specification	
Basic function	• Analog input channels	8
	• Analog output channels	8
	• Dante input channels	8
	• Dante output channels	8
DSP performance	• Processing capability	400 MIPS, 1.6 GFLOPS
Audio performance	• Sample rate	48 kHz ± 100 ppm
	• Quantization bits	24 bit
	• THD+N	<-95dB @17dBu
	• Input dynamic range	113 dB
	• Output dynamic range	113 dB
	• Channel isolation (@1kHz)	108 dB
	• Frequency response (20–20kHz)	± 0.3 dB
	• Maximum input level	+18 dBu
	• Input gain range	0–48 dB (3 dB steps)
I/O characteristics	• Input impedance	5.4 kΩ
	• Output impedance	600 Ω
	• Phantom power	+48 V / 10 mA max
System delay	• Processing delay	< 3 ms

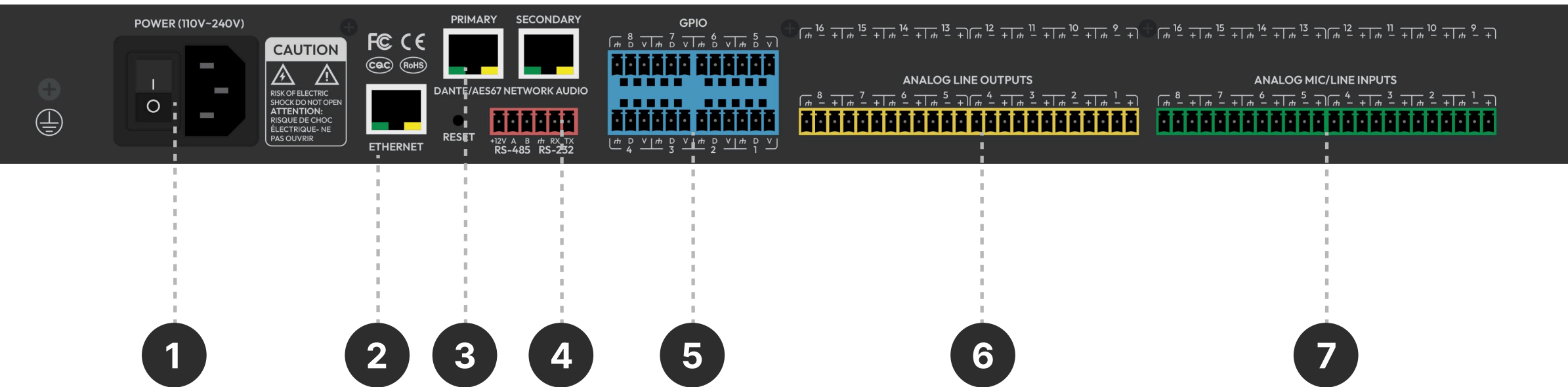
Product Specifications

Function	Specification	
Processing features	• AEC (echo cancellation)	Tail Length: 512 ms, Convergence: 60 dB/s, Magnitude: 60 dB
	• AFC (feedback suppression)	Notch Algorithm, Gain Boost: 10 dB
	• ANS (noise suppression)	SNR Improvement: 18 dB
	• ANC (noise gain compensation)	Supported
	• AMC (auto mixer)	Gain Sharing, Gate Mixer
	• Equalizer	12-band British Parametric EQ, 5 Filter Types
Control & Expansion	• Custom user interface	Supported, up to 30 devices
	• Central control	Power, Signal Switching, Environmental Control, Audio, etc.
	• USB sound card	Built-in 1-in 1-out, Playback, Recording, Videoconferencing
Power & Physical	• Power supply	AC 110–240 V, 50/60 Hz
	• Dimensions (W × D × H)	482 × 260 × 45 mm
	• Shipping weight	4 kg

Component Instructions



- 1 PWR: Power indicator light
- 2 SYS: System status indicator light
- 3 USB AUDIO: Sound card device interface



- 1 POWER: Connects to 110V-240V AC power supply. The toggle switch controls the processor power.
- 2 ETHERNET: Network Control Interface (By connecting to this network port, the client computer can debug and monitor the device)
- 3 DANTE/AES67 NETWORK AUDIO: Used to connect to the Dante audio network.
- 4 RS232+RS485: Connects to control terminals or central control devices.
- 5 GPIO: Logic input/output interface.
- 6 OUTPUTS: Signal Output Interfac (Can connect to amplifiers, active speakers, and other devices)
- 7 INPUTS: Signal Input Interface (Can connect to microphones, DVD players, and other devices)