Features:

- The DSP device adopts the analog device SHARC DSP with 40-bit floating point processing. The main frequency is up to 500MHz, and the 512M memory can have a maximum of 16 fixed analog audio inputs, which are connected through a detachable balanced phoenix connector. The analog input section can support microphones, or line-level signals with nominal levels of 0dBu, 10dBu, 20dBu, 30dBu, and 40dBu. +48VDC phantom power is available for each input. Preamp gain and phantom power are conveniently controlled via the DSP Controller.
- A/D Specifications: Sampling Rate: 48kHz THD+N: 105dB Dynamic Range: 109dB Audio format: 24Bit MSB TDM, Sampling Rate: 48kHz THD+N: -100dB Dynamic Range (A-weighted): 112dB Audio Format: 24Bit MSB TDM.
- The first stage of the analog output section is the D/A converter (DAC). The DSP uses an advanced 24-bit 256X sampling converter. Like A/D converters, a multi-bit architecture is used to achieve a wider dynamic range, but with the same excellent distortion characteristics as conventional single-digit digital-to-analog converters. Unity gain (0dB) is set via the volume control, and the analog output section is corrected to +4dBu with 14dB of headroom. This means that a 0dBFS digital signal is equivalent to a +18dBu output signal. If other signal levels are required, this can easily be achieved by changing the volume.
- 1U standard chassis design. 16 balanced inputs, 16 line outputs Support phantom power
- It has processing modules such as AFC (feedback suppression), AEC (echo cancellation), ANS (noise suppression), ANC (noise gain compensation), AGC (automatic gain), gain sharing, threshold automatic mixing, ducker and other processing modules; bus-type AEC, Tail time: 512ms, convergence rate: 60dB/S, echo cancellation amplitude: 60dB; output per channel: parametric equalizer, delay, frequency divider, high and low pass filter, limiter.
- Built-in one-in-one-out USB sound card, supports music playback, recording and soft video conferencing, supports recording and remote conference full-featured matrix mixing, and the input mixing level can be adjusted.
- Each channel has independent adaptive feedback suppression, automatically finds feedback points, and automatically suppresses.
- Based on the Ethernet network control interface, it solves the network control.
- Control via Ethernet-based network transmission wall panels, WIFI and third-party touch screen.
- 16 groups of presets, each working independently.
- GPIO interface: 4 logic outputs with 4 pairs of common ground pins. When activated, the logic output goes low (0V), and when not activated, it is internally pulled high (5V), which can directly light up the external LED indicator. The logic outputs can be driven by the logic output control block in the device design. Polarity and threshold can be set in software.
- Support RS232 or UDP central control, the UDP port can be freely set, and the control software code can be viewed. The mobile APP can be self-compiled through the host interface, providing end-user customized operation interface, and supporting a maximum of 30 devices for the same interface management.
- It has a central control function, which can control the power supply, signal switching, environmental control, audio and other overall control in the system, and realize the functions required to open the system with one key.
- Applications: conference rooms, stadiums, outdoor performances, multi-purpose halls and other places

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Specification:

Model	DSP1616
Description	DSP Audio Matrix Processor
Channel	16 In 16 Out
Processor	ADI SHARC 21489(x2)
Sampling rate	48K/24bit
Maximum input level	0/10/20/30/40dBu
Maximum output level	+18dBu
Distortion	0.003%@4dBu
Frequency Response	+/- 0.5dB （20Hz to 20kHz.）
Noise Floor	-91 Db A-weighted
Rejection Ratio @60Hz	108 dB
Power consumption	<20W
Frequency Response	+/- 0.5dB （20Hz-20kHz）
Power Supply	AC190-240V/50-60Hz
Power consumption	<20W
Volume	480*220*44mm
Gross weight	4.3kg

Rear Panel:

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