



Description

The product uses a high-end audio processor based on FPGA computing core, which has extremely low delay, high phase uniformity, and can intelligently adjust the sound field; It can quickly and efficiently improve the sound quality of the speakers and the overall sound effect, reduce environmental audio interference, remove audio howling points, and evenly and automatically distribute the sound pressure of the speakers to enhance the immersive feeling; It is suitable for places with high sound quality requirements such as multi-function halls, theaters, high-end conference rooms, hotel banquet halls, etc.

Feature

- * The FPGA audio processor uses advanced algorithms and digital signal processing technology to perform fully automatic acoustic calibration in the full audio band based on the characteristics of the acoustic space measured by 4-channel high-precision acoustic microphone inputs.
- * By analyzing the reflection and absorption of sound at different frequencies, the output of the full audio band can be dynamically adjusted to reduce standing wave phenomena and frequency response unevenness, achieving lower divergence and solid low-frequency and high-definition music effects.
- * Use the high-performance computing power of FPGA to design and implement FIR digital filters. It can accurately adjust the amplitude-frequency characteristics of the audio signal, so that the sound has a flat response in different frequency bands, improving the clarity and accuracy of the mix.
- * Each input channel can specify 8 microphone test points. After comprehensive calculation, the on-site sound pressure level will be more balanced and the audience's listening experience will be more consistent.
- * Support custom 16-segment parametric acoustic curve correction function; full-audio band automatic acoustic correction can be performed based on specific sound curves to achieve high-definition music effects.
- * Advanced phase correction algorithm of FPGA processor. Support detection and correction of phase distortion caused by speaker crossovers, ensuring that the phase of audio signals in different frequency bands remains consistent, and eliminating phase distortion caused by crossovers.
- * With fast audio processing capabilities, the audio delay is less than 0.3 milliseconds, ensuring real-time sound processing.
- * Built-in high-pass, low-pass and band-pass functions: used for speaker frequency division processing, speaker optimization and adjustment effects.
- * Support 4-in and 8-out matrix distribution control function, and free signal distribution and processing.
- * The input and output channels can be assigned a delay of 500MS to match the speaker distribution distance delay adjustment.
- * Equipped with a 2-inch display screen, providing a clear visual interface for status monitoring and function configuration adjustment.
- * Built-in high-performance dynamic compressor: support automatic adjustment of the dynamic range of audio signals to ensure balanced and moderate audio volume during playback, increasing audio clarity and audibility.
- * Multiple devices can be cascaded through PC software: Users can use one computer to control multiple devices at the same time, which can help users manage and operate multiple devices more efficiently.
- * Users can export and import scene parameters, and can also pre-store the optimal parameter values for supporting speakers.



Specification

Input channel function	4 channel
Output channel function	16 channel
Input resistance	20K Ω (balanced), 10K Ω (unbalanced)
Input quantization	48KHz/24bit
A/D dynamic range	116dB
D/A dynamic range	127dB
Phantom power	DC +48V
Frequency response	20 ~ 20kHz (\pm 0.5dB)
Dynamic Range	105dB @THD < 1% / 1kHz
Total harmonic distortion (THD+N)	\leq 0.005% @1kHz , +4dBu
Channel isolation	\geq 86dB
SNR	\geq 110 dB
Output impedance	100 ohm
Maximum output level	+18dBu
Output quantization	48KHz/24bit
Working power supply	\sim 100V - 230V, 50Hz /60Hz
Power consumption	45W
Weight	3.2kg
Dimensions	484 \times 258 \times 44mm