Audio Processor

TS-F0416

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 features 4-channel high-precision acoustic microphone input measurement interfaces, which can perform full-frequency band, fully automated acoustic calibration based on the high-precision acoustic microphone input to measure the characteristics of the acoustic space.
*Use the high-performance computing power of FPGA to design and implement FIR digital filters up to 4096 orders. It can accurately adjust the amplitude-frequency characteristics of the audio signal so that the sound has a flat response in different frequency bands, thereby improving the clarity and accuracy of the mix.
*By analyzing the reflection and absorption of sound at different frequencies, the output of the entire audio band can be dynamically adjusted to reduce standing waves and frequency response unevenness, achieving lower dive, solid low-frequency and high-definition music effects.
*Each input channel can specify 8 microphone test points. After comprehensive calculation, the on-site sound pressure level is more balanced and the audience's listening experience is more consistent.
*It has a custom 16-band parametric acoustic curve correction function; it can perform full-audio segment automatic acoustic correction according to a specific sound curve.
*The advanced phase correction algorithm of the FPGA processor detects and corrects the phase distortion caused by the speaker crossover, ensuring that the phase of the audio signal remains consistent in different frequency bands, thereby eliminating the phase distortion caused by the crossover.
*It has fast audio processing capabilities, with audio processing delay ≤0.3 milliseconds, ensuring real-time sound processing.
*Built-in EQ highpass, EQ lowpass and bandpass functions: used for speaker crossover processing, speaker optimization and adjustment effects.
*Supports 4-channel high-precision acoustic measurement microphone input.
*The device has 4 input channels and 16 output channels; it has 4-input and 8-output matrix distribution control function.
*Supports the integrated automatic quick measurement and optimization curve function, with a fast acoustic environment measurement optimization time of ≤10 seconds.
*The input and output channels can be assigned a 500MS delay to match the delay adjustment of the speaker distribution distance.
*The processor is equipped with a 2-inch display, providing a clear visual interface for status monitoring and function configuration adjustment.
*Built-in high-performance dynamic compressor: It can automatically adjust the dynamic range of the audio signal to ensure that the volume of the audio is balanced and moderate during playback, and increase the clarity and audibility of the audio.
*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 best parameter values for matching speakers.

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Specification

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