Audio Processor

TS-DP480FIR

Description

This product is a 4-input 8-output high-performance audio processor that uses AEQ automatic equalization, IIR/FIR crossover, equalizer, delay, limiter, compressor and other technologies; by identifying the sound field of the on-site environment, it automatically corrects the audio, improves the fidelity and naturalness of the sound, and makes the sound emitted by the speaker consistent with the sound source; it is mainly used in lecture halls, concert halls, high-end conference rooms and other scenes .

Features

*It has 4 balanced input XLR interfaces and 8 balanced output XLR interfaces; the sampling rate is 96K.
*Each channel input supports noise gate, which can appropriately suppress the obvious background noise caused by the accumulation of previous equipment or improper system settings.
*Each channel input supports a dynamic loudness enhancer, which compensates and enhances the ultra-low and ultra-high frequencies based on the human ear's equal loudness curve.
*Each channel input supports polarity inversion for matching speaker phase or correcting overall phase reversal caused by incorrect signal cable connection.
*Each input and output channel is equipped with an RMS compressor, which can control the signal dynamics in the input channel or be used to shape the sound intensity. The newly designed extremely low distortion peak limiter can prevent sudden large dynamic signals from damaging the speaker unit, effectively ensuring the safety of the system.
*Each channel input and output provides 4 equalizer modes, including 31-band PEQ, FIR+17-band PEQ, 7-band PEQ TARGET + 24-band PEQ, 7-band PEQ TARGET + 10-band PEQ + FIR.
*Input parametric equalizer with 31 bands, output parametric equalizer with 8 bands, 16 filter types, adjustable peak filter, first-order/second-order/variable Q high-shelf/low-shelf filter, notch, first-order/second-order/variable Q high-pass/low-pass filter, and first-order/second-order variable Q all-pass filter.
*Each channel output supports peak limiter/hard limiter to limit peak signals and protect the woofer from mechanical damage caused by voice coil movement exceeding linear stroke.
*Each channel output supports frequency divider, with two built-in frequency divider filters, namely traditional IIR filter and FIR finite impulse response filter; both FIR and IIR filters can be used at the same time to generate the coefficients required for correction .
*Crossover Traditional IIR filters include Bessel, Butterworth, Linkwitz Rayleigh, NXF horn filters with a maximum slope of 48db/oct.
*Each input channel and output channel can import a 48kHz sampling rate 512 tap FIR filter. FIR convolution generated by third-party FIR convolution software in .TXT or .CSV format can be imported for speaker presets to improve phase response and control directivity as required.
*The acoustic correction function can import smaart/fir Maker data or use the sound card in the software to obtain the frequency characteristic curve, and can implement acoustic correction in three ways: PEQ, PEQ+FIR, and PEQ+FIR+PEQ TARGET.
*With various sound cards, more measurement parameters can be measured; the sound field curves of WAVE and ASIO audio driver sound cards can be measured, and the amplitude and phase smoothness from 1 oct to 48 oct can be adjusted, 1 to 16 average points can be supported, and delay can be inserted. Real-time display of amplitude response, phase response and coherence curves.
*IIR automatic equalization amplitude correction; based on the frequency equalization method, the PC client can import smaart curve data to automatically generate filter coefficients, or it can be more detailed after automatic generation by adjusting the target curve changes, real-time dragging of the processing frequency range and increasing the number of filter points to obtain better coefficients.
*FIR automatic equalization amplitude phase correction; based on the frequency equalization method, the PC client can import the smaart curve to automatically generate FIR filter coefficients. It can also adjust the minimum phase and linear phase more carefully after the automatic generation, as well as the target curve change, and display the actual amplitude/phase curve after dragging the processing frequency range in real time. And it can adjust the delay size to meet various needs on site.

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*2.0-inch IPS high-definition screen and encoder knob; with 12 sets of LED digital tubes, each with 3 colors and 8 LED lights, real-time display of 4 sets of LINE IN and 8 sets of LINE OUT level values, and has a tempered glass-covered LED light and IPS high-definition screen integrated structure. It can be used to control and configure device scenes , mute , automatic equalization , crossover , matrix , gain , compressor , noise gate, limiter and delay parameters. And it can display the IP address and current level status.
*Rich touch button resources: The front panel has 12 touch buttons, including home button, return button, scene button, mute button, automatic equalization button, crossover button, matrix button, gain button, compressor button, noise gate button, limiter button and delay button, which can configure device parameters.
*Full matrix variable gain mixing: The input matrix has 4×8 and the output matrix has 8×8, which can send any input channel to the output channel, superimpose and mix several non-adjacent output channels to the physical output, and each mixing channel input and output can be adjusted in the range of -72db to 12db.
*Rich delay timer resources: input delay timer can be adjusted from 0 to 2000ms, output delay timer can be adjusted from 0 to 2000ms.
*Processing delay: When the minimum phase FIR processing function is fully turned on, the delay from input audio to output audio is 388us.
*The device has client software, which can locate devices in the local area network with one click. The located devices will display the location information on the display screen; the software supports centralized control of ≥999 devices; it can be installed in Windows7/Windows10/Windows11 operating systems.

Specification

Input channels	Preamplifier, signal generator, delay, RMS compressor, noise gate, dynamic loudness filter, 31-band parametric equalizer, FIR filter
Output channel	Delay, RMS compressor, peak limiter, hard limiter, 8-band parametric equalizer, FIR filter, IIR divider and FIR divider
Device audio interface	4-channel XLR input interface, using balanced connection; 8-channel XLR output interface, using balanced connection
Communication interface	1-channel RJ45 network port; 1-channel RS485 communication port, which can be connected to the central control system to achieve two-way control.
Power switch	1 rocker power switch
Display	2-inch IPS true color display, resolution 320*240
Panel indicator light	96 panel indicators
Processor	96kHz sampling frequency, 64-bit DSP processor; 96-bit A/D and D/A conversion
Frequency response	20Hz ~ 40KHz
Total harmonic distortion + noise	≤0.003% OUTPUT=20dBu/1kHz
Signal-to-Noise ratio	≥110dB@1kHz 18dBu (A-weighted)
Channel separation	≥100dB@1kHz 18dBu (A-weighted)
Input impedance (balanced)	Balance: 20KΩ
Maximum output impedance (balanced)	Balanced: 100Ω
Input range	≤+18dBu
Delay	Ultra-low latency, the latency is only 0.388ms.
Frequency division - IIR	IIR filters are used to perform high-pass and low-pass processing on channel signals.
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IIR equalizer	It has 31-band parametric EQ and 8-band parametric EQ.
FIR equalizer	With FIR filter, frequency response and phase correction can be achieved.
Power supply range	AC 110V-240V 50-60Hz
Power consumption	≦20W
Operating temperature	-10℃ ~ +45℃
Relative humidity	20% ~ 80% relative humidity, no condensation
Cool down	Fan forced cooling
Product size (LxDxH)	484×209×45mm
Net weight	3.4kg
A/D dynamic range	118dB
Digital/Analog dynamic range	120dB