Audio processor

TF-0UIIH

Description

A high-performance audio processor with 4 analog balanced inputs and 4 analog balanced outputs. Integrated dynamic range control (DRC) , automatic gain (AGC) , feedback suppression (AFC) , adaptive noise reduction (ANS) , adaptive echo cancellation (AEC) , audio filter (GEQ, PEQ, crossover) and other functions , mainly used in professional sound reinforcement scenarios, which can meet the application needs of sound reinforcement systems in conference rooms, courts, auditoriums, multi-function halls, performances, classrooms and other places.

Feature

*The panel has a 2-inch IPS true color display that displays device network information, real-time levels, channel mute status, matrix mixing and other statuses.
*The panel has a USB interface and supports multimedia storage, which can be used for storage, recording or playback.
*High-performance 64-bit DSP processor (800M main frequency), 32-bit/48KHz AD/DA , professional DSP processing , providing excellent high-quality sound.
*Enternet is a multi-purpose data transmission and control port that can support real-time management of single and multiple devices.
*It supports accessing the device through PC software and comes with its own management and control software: the software interface is intuitive and graphical, and can work in Windows 7, 8, 10 and other system environments.
*Supports operation control, device login, scene switching, input and output, matrix routing and channel settings through Android mobile APP software.
*Extremely low system processing delay, delay less than 3ms .
*Supports scene preset , import , and export, and supports 8 scenes . Has the function of restoring factory settings.
*configurable RS-232 interface can be used to connect to external central control systems to achieve centralized management and control .
*The RS-485 interface can be configured, which can be connected to the central control system and camera tracking system, and can realize automatic camera tracking function.
*Configurable 8- channel programmable GPIO control interface (customizable input and output). Supports channel copy, paste, and joint control functions.
*It has a 4-in and 4-out audio matrix, and the input sensitivity can be adjusted according to different sound sources . Each input supports 48V phantom power supply and can be individually configured to turn on or off, making it flexible and convenient.
*It has a ducker function, which is used to automatically dodge microphone speech in the background music. It provides a variety of parameter settings to facilitate flexible use on site.
*It has a microphone automatic gain function, which is used to control the dynamic range of the microphone's pickup signal to achieve consistent sound quality near and far.

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*It has intelligent mixing functions, including gain sharing mixing and threshold automatic mixing. The input channels can be individually selected to participate in intelligent mixing. The corresponding mixing mode can be selected according to the application requirements of different scenarios. It can effectively solve the pain points of the sound reinforcement system being unstable and prone to howling due to too many microphones.
*Equipped with equalizer function, it provides parametric equalizer and graphic equalizer. Each input/output has 12-band parametric equalizer/10-band graphic equalizer/15-band graphic equalizer/31-band graphic equalizer to choose from. . The parametric equalizer supports three types of high-shelf, low-shelf, and peak filters, and the graphic equalizer supports single-point bandwidth adjustment.
*It has a frequency divider function, provides Bessel, Linkwich-Rayleigh, and Butterworth filter types to choose from, and supports 6/12/18/24/32/40/48db/oct slope settings . The whole frequency range is adjustable.
*It has an expander function to expand the dynamic range of the signal and eliminate the noise floor of the device.
*It has a compressor function to compress the dynamic range of the signal and is used to compress the size of the output signal.
*It has a limiter function to limit the size of the output signal and prevent excessive signal damage to the sound amplification equipment .
*It has a delay function and provides a maximum delay adjustment of 2000ms, which is used to adjust the delay of each output signal so that each audio signal remains synchronized when it reaches the listener's ears .
*Equipped with echo cancellation function, it is used for remote audio and video conferencing to eliminate echo and increase speech clarity.
*It has a noise cancellation function that can effectively eliminate environmental noise such as air conditioning and fan sounds and improve speech clarity.
*It has feedback suppression function and two processing solutions of notch + frequency shifter to effectively solve the problem of acoustic feedback .
*Supports automatic power-off protection memory function.

Specification

Processor	48k Hz sampling frequency, 64 -bit DSP processor ; 32-bit A/D and D/A conversion
Phantom power	DC 48V
Frequency response	20Hz ~ 20KHz
Signal-to-noise ratio	≥110dB@1kHz 24dBu (A-weighted)
Total harmonic distortion + noise	≤0.002% OUTPUT=24dBu/1kHz
Channel separation	≥100dB@1kHz 24dBu (A-weighted)
Power supply range	AC 110V-2 20V 50 / 60Hz
Power consumption	≤ 20W
Input impedance (balanced)	Balance: 20KΩ
Maximum output impedance (balanced)	Balance: 100Ω
Input range	≤+24dBu
Input channel	4-way balanced microphone/line, using bare wire interface terminals, balanced connection method;
Output channel	4 balanced line outputs, using bare wire interface terminals, balanced connection method;
Cooling	Fan forced cooling
Relative humidity	20%~80% relative humidity, no condensation
Operating temperature	-10℃ ~ + 45 ℃
Product size (L × D × H)	484 × 298.2 × 45mm
Net weight	3.3 kg