Audio Processor

TZ-8UIIL

Description

A high-performance audio processor with 4 analog balanced inputs and 4 analog balanced outputs. It integrates dynamic range control (DRC) , automatic gain control (AGC) , feedback suppression (AFC) , adaptive noise reduction (ANS) , adaptive echo cancellation (AEC) , audio filters (GEQ, PEQ, crossover) and other functions . It is mainly used in professional sound reinforcement scenes and can meet the needs of conference rooms, courtrooms, auditoriums, multi-function halls, performances, classrooms and other places for sound reinforcement system applications.

Features

1. The panel has a 2-inch IPS true-color display that displays device network information, real-time level, channel mute status, matrix mixing and other status.
2. The panel has a USB interface and supports multimedia storage, which can store recording or playback.
3. The input channel supports preamplifier, signal generator, expander, compressor, equalizer (12-band parametric equalizer, optional 10/15/31-band graphic equalizer, the graphic equalizer can be used to adjust the bandwidth individually), ducker, AGC automatic gain, AM automatic mixing function (threshold type, gain sharing type), AFC adaptive feedback cancellation, AEC echo cancellation, ANC noise cancellation, and audio matrix .
4. High-performance 64-bit DSP processor (800MHz main frequency), 32-bit/48KHz AD/DA , professional DSP processing , providing excellent high-quality sound.
5. Enternet multi-purpose data transmission and control port can support real-time management of single and multiple devices.
6. Supports operation control through Android mobile phone APP software, including device login, scene switching, input and output, matrix routing and channel setting, etc.
7. The audio processor has cross-platform software and can run 8 operating system versions , including Windows 7/10/11, Kylin Desktop Operating System (Zhaoxin Edition), Kylin Desktop Operating System (Feitian Edition), macOS, Tongxin UOS, and Ubuntu Desktop Operating System.
8. Extremely low system processing delay, less than 3ms .
9. Support scene preset , import , export, support 8 scenes . Has the function of restoring factory settings.
10. With RS-232 interface, it can be used to connect to external central control system to achieve centralized management and control . Supports 65535 devices to be centrally controlled through software.
11. It has RS-485 interface, which can be connected to the central control system and camera tracking system to realize automatic camera tracking function.
12. It has 8-channel programmable GPIO control interface (customizable input and output). It supports channel copy, paste, and joint control functions.

13.Based on the howling detection threshold update method, it has frequency shift + notch combination feedback suppression, can use 24 programmable notch points, can freely assign dynamic/static points, and automatically/manually switch.
14.It has a 4-input 4-output 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 be turned on or off, which is flexible and convenient.
15.With matrix gain adjustment function, the gain of each input channel participating in the mixing is adjustable, and the gain adjustment range is -72db to 12db.
16.It has a ducker function, which is used to automatically duck the microphone speech during background music, and provides a variety of parameter settings for flexible on-site use.
17.It has equalizer function, providing 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 filter, and the graphic equalizer supports single-point bandwidth adjustment.
18.The output channel supports equalizer, delay, crossover, high-pass and low-pass filters, and limiter .
19.The product has different control methods for PC client, mobile phone and Android tablet. You can log in to the APP software and PC client at the same time to connect to the device and realize the synchronization of multi-terminal data.
20.The device has an encoded knob and IPS screen that can be used to control and configure the device mute, gain, and scenes.
21.It has an automatic microphone gain function, which is used to control the dynamic range of the microphone's pickup signal to achieve consistent sound quality near and far.
22.It has intelligent mixing function, including gain sharing mixing and threshold automatic mixing. The input channel can choose whether to participate in intelligent mixing individually. The corresponding mixing mode can be selected according to the application requirements of different scenarios. It can effectively solve the pain points such as unstable sound reinforcement system and easy howling caused by multiple microphones.
23.The audio processor software can be integrated into the conference audio comprehensive management platform to achieve unified management of audio equipment. The platform can scan the online status of digital conference hosts, suppressors, mixers, amplifiers, and audio processors. Multiple online devices of the same product can also be scanned, and the device hardware name, hardware IP address, online and offline status information will be displayed. It has the function of uploading configuration information to the cloud or saving it locally for backup and restoring configuration information with one click.
24.It has a crossover function and provides three filter types: Bessel, Linkwich-Rayleigh, and Butterworth for selection. It also supports 6/12/18/24/32/40/48db/oct slope settings , and the filter is adjustable across the entire frequency band.
25.With the device positioning function, the client can locate similar devices in the local area network with one click and display the positioning information.
26.It has an expander function to expand the dynamic range of the signal and is used to eliminate the background noise of the device.
27.It has a compressor function to compress the dynamic range of the signal and is used to compress the size of the output signal.
28.It has a limiter function to limit the size of the output signal to prevent excessive signals from damaging the sound reinforcement equipment .
29.It has a delay function, providing 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 reaching the listener's ears .
30.It has echo cancellation function, which can be used for remote audio and video conferencing to eliminate echo and increase voice clarity.
31.It has a noise cancellation function, which can effectively eliminate environmental noise such as air conditioning sound, fan sound, etc., and improve voice clarity.
32.It has feedback suppression function and two processing solutions: notch filter + frequency shifter, which can effectively solve the problem of acoustic feedback .
33.Supports automatic power-off protection memory function.

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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- 220V 50 / 60Hz
Power consumption	≤ 20W
Input impedance (balanced)	Balance: 20KΩ
Maximum output impedance (balanced)	Balanced: 100Ω
Input range	≤+24dBu
Input Channels	4 balanced microphone/line, bare wire interface terminal, balanced connection
Output Channel	4 balanced line outputs, using bare wire interface terminals, balanced connection
cool down	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