Audio Processor

TS-DP1616

Description

It is a high-performance audio processor with 16 analog balanced inputs and 16 analog balanced outputs. It is mainly used in the scenarios of analog microphone sound reinforcement and multi-channel sound source mixing, which can meet the application requirements of sound reinforcement systems in conference rooms, courts, auditoriums, multipurpose halls, classrooms, performance venues and other places.Integrated dynamic range control (DRC), automatic gain control(AGC), auto feedback control (AFC), adaptive noise reduction (ANS), adaptive echo cancellation (AEC), audio filters (GEQ, PEQ, crossover) and other functions.It is mainly used in various professional sound reinforcement scenarios.

Feature

1. High-performance 64-bit DSP processor (1MHz main frequency), 32-bit/48KHz AD/DA, professional DSP processing, providing excellent high-quality sound.
2. With 16-input and 16-output audio matrix, the input sensitivity can be adjusted according to different sound sources. Each input supports 48V phantom power supply, which can be configured to be turned on and off separately, which is flexible and convenient.
3. The input channel supports preamplifier, signal generator, expander, compressor, equalizer (12-band parametric equalizer, optional 10/15/31-band graphic equalizer adjustable, graphic equalizer can be used to adjust bandwidth separately), ducker, AGC automatic gain, AM automatic mixing function (threshold type, gain sharing type), AFC adaptive feedback elimination, AEC echo cancellation, ANC noise elimination, audio matrix; the output channel supports equalizer (12-band parametric equalizer, optional 10/15/31-band graphic equalizer adjustable, graphic equalizer can be used to adjust bandwidth separately), delay, crossover, high-pass and low-pass filters, limiter; 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 switch automatically/manually.
4. It has a ducker function, which is used for background music to automatically duck the microphone speech, and provides a variety of parameter settings for flexible on-site use.
5. 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 at near and far.
6. It has an intelligent mixing function, including gain sharing mixing and threshold automatic mixing. The input channel can choose whether to participate in the intelligent mixing individually, and the corresponding mixing mode can be selected according to the application requirements of different scenarios. It can effectively solve the pain points such as the instability of the sound reinforcement system due to multiple microphones.
7. It has an 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 filters, and the graphic equalizer supports single-point bandwidth adjustment.
8. It has a crossover function, providing three types of filters: Bessel, Linkwich-Rayleigh, and Butterworth for selection, and supports 6/12/18/24/32/40/48db/oct slope settings, and the filter is adjustable across the entire frequency band.
9. It has an expander function to expand the dynamic range of the signal and eliminate the background noise of the device.
10. It has a compressor function to compress the dynamic range of the signal and compress the size of the output signal.
11. It has a limiter function to limit the size of the output signal to prevent the signal from being too large and damaging the sound reinforcement equipment.

12. It has a delay function to provide a maximum delay adjustment of 2000ms, which is used to adjust the delay of each output signal so that each audio signal is synchronized when it reaches the listener's ear.
13. It has an echo cancellation function for remote audio and video conferencing to eliminate echoes and increase voice clarity.
14. It has a noise cancellation function, which can effectively eliminate environmental noise such as air conditioning sound and fan sound, and improve voice clarity.
15. It has a feedback suppression function, and two processing solutions of notch filter + frequency shifter can effectively solve the problem of acoustic feedback.
16. Extremely low system processing delay, the delay is less than 3ms.
17. The panel has a 2-inch IPS true color display screen, which displays device network information, real-time level, channel mute status, matrix mixing and other status.
18. The panel has a USB interface, supports multimedia storage, and can store recording or playback.
19. Supports scene preset, import, and export, and supports 8 scenes.
20. Has the function of restoring factory settings.
21. Has an RS-232 interface, which can be used to connect to an external central control system to achieve centralized control.
22. Has an RS-485 interface, which can be connected to the central control system and camera tracking system, and can realize automatic camera tracking function.
23. Has an 8-channel programmable GPIO control interface (customizable input and output).
24. Supports channel copy, paste, and joint control functions.
25. Enternet multi-purpose data transmission and control port, which can support real-time management of single and multiple devices.
26. The audio processor has cross-platform software and can run 8 operating system versions, including Windows7/10/11, Galaxy Kylin Desktop Operating System (Zhaoxin Edition), Galaxy Kylin Desktop Operating System (Feitian Edition), macOS System, Tongxin UOS, and Ubuntu Desktop Edition Operating System.
27. The product has different control modes 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 the device and realize the synchronization of multi-terminal data.
28. Supports operation and control through Android mobile phones and tablet APP software, device login, scene switching, input and output, matrix routing and channel settings.
29. Supports automatic power-off protection memory function.
30. The rear panel has 16-line audio Phoenix terminal balanced input interface (with 48V phantom power supply), 16-line audio Phoenix terminal balanced output interface, 1 DIP switch, 1 RJ45 interface, 1 RS232 interface, 1 RS485 interface, 8 programmable GPIO control interfaces, and 1 grounding column; the front panel has a 2.0-inch IPS true color display, 1 encoding knob, and 1 USB storage device interface.
31. The device has a unified centralized control function and supports 65,535 devices to be centrally controlled through software.
32. It has a 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.
33. The device has an encoding knob and an IPS screen, which can be used to control and configure the device mute, gain, and scene; the IPS screen can display the IP address and the real-time level of the input and output channels.
34. It has a device positioning function. The client can locate similar devices in the LAN with one click, and the located device will display the positioning information.
35. The audio processor software can be integrated into the conference audio comprehensive management platform to realize the unified management of audio devices. The platform can scan the online status of digital conference hosts, audio processors, mixers, suppressors, and amplifiers. Multiple online devices of the same product can also be scanned, and the device hardware name, hardware IP address, online and offline status information are 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.
36. It has a Web page management function. The control software can be downloaded through the Web page, and the gain of each channel can be adjusted through the Web page.
37. Supports central control function, which can realize gain addition and subtraction, gain limitation, query of each channel level bar, and setting of gain addition and subtraction step value.
38. Has dual-machine hot backup function.

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Specification

Processor	48kHz sampling frequency, 64-bit DSP processor; 32-bit A/D and D/A conversion
Phantom power	DC 48V
Frequency response	20Hz ~ 20KHz
Total harmonic distortion + noise	≤0.002% OUTPUT=18dBu/1kHz
Signal-to-noise ratio	≥110dB@1kHz 18dBu (A weighting)
Channel separation	≥100dB@1kHz 18dBu (A weighting)
Input impedance (balanced)	Balanced 20KΩ
Maximum output impedance (balanced)	Balanced 100Ω
Input range	≤+18dBu
Howling search and suppression method	Fully automatic notch
Notch filter	24 (static and dynamic points can be configured)
Q value range	10-50
Frequency resolution	1Hz
Howling search time	0.1—0.5S
FFT length	1024
Transmission gain	4—10dB
System gain	0dB
Divider	With Butterworth, Bessel, Linkwich-Rayleigh three high and low pass filters
Display	2-inch IPS true color display, resolution 320*240
Power supply range	AC 180V-240V 50-60Hz
Power consumption	≦50W
Operating temperature	-10℃ ~ +45℃
Relative humidity	20% ~ 80% relative humidity, no condensation
Cooling	Fan forced cooling
Product size (L×D×H)	484×298.2×45mm
Net weight	3.69kg
A/D dynamic range	116dB
D/A dynamic range	120dB