DSP9275

Intelligent Digital Audio Processor

Description

In indoor sound reinforcement, the signal sent from the speaker and the signal fed back from the building are returned to the pickup device to be superimposed with the original signal, and then amplified by the system and sent out through the speaker, and so on and so forth, forming oscillations, which is manifested as acoustic feedback (howling), and even burned sound reinforcement equipment in serious cases. In particular, the conference teaching and live sound reinforcement have always been a headache for audio experts.

This machine adopts adaptive environmental howling suppression algorithm, high-speed floating-point digital audio processor and advanced Sub-Band Echo Cancellation technology to automatically and completely eliminate howling and echo at high speed, while allowing live audio to be accurately and clearly delivered in real time. It has a built-in adaptive dynamic noise filter to filter out the background noise of the live environment without affecting the high-quality transmission of voice signals. The intelligent level control technology of the system can be used to get a clear and continuous voice signal without obvious fluctuations in signal levels. The built-in AGC bidirectional control, with the ability to distinguish between enhanced signals and soft signals, can be used to keep the speech tone coherent and the voice easy to be heard, thus maintaining the comfort of listening, and improving the gain by 6-15dB. It is a brand-new feedback suppression solution.

Features

- Built-in 24-bit A/D and D/A conversion; 32-bit DSP processor. Adopt high-speed floating-point digital audio processor and state-of-the-art Sub-Band Echo Cancellation technology to effectively eliminate echo and howling.
- Support fully automated working mode, free of manual debugging. Regardless of the change in location, temperature, humidity and decoration in the room environment, the system can be installed without the need for sound field debugging, which is accurate, reliable, and easy to use.
- With a built-in adaptive dynamic noise filter to filter out the background noise of the live environment without affecting the high-quality transmission of voice signals, improving the signal-to-noise ratio and sound quality.
- With a built-in automatic gain control (AGC) to obtain a clear and continuous voice signal.
- With built-in digital high-pass and low-pass adjustment control to limit the voice frequency response.

- With a built-in digital intelligent adjustment to improve the distance of sound pickup.
- With a built-in 10-band graphic equalizer for more accurate frequency control.
- With a built-in voice stimulation function to improve speech intelligibility.
- With automatic priority function for music and microphone in a specific channel, and automatic switching function for multi-channel microphones, the equipment can be used more flexibly and applied to more engineering application scenarios.
- With a 2.4" color display screen to display parameters and states more comprehensively and clearly, easy for debugging. All functions can be set through the machine or by connecting a computer.

Specifications

Model	DSP9275
Analog Input	
MIC IN Impedance	＞5KΩ
MIC IN Level	20mV
MIC Frequency Characteristic	50Hz~18KHz (-3dB, band-pass off)
LINE IN Impedance	＞10KΩ
LINE IN Level	500mV, 2 channels
LINE Frequency Characteristic	20Hz~20KHz (+/-1dB)
Performance	
Dynamic Range	＞102dB (A-weighted)
Distortion	＜0.1%
Feedback Suppression Frequency Response	+/-2dB (50Hz to 20KHz)
Bypass Frequency Response	+/-2dB (40Hz to 20KHz)
Phantom Power	+48V/6V (DC)
Operating Power	AC220-240V/50-60Hz
Maximum Output Power	250W*2 (constant-resistance bridge)
Dimensions	480mm×390mm×66mm
Net Weight (KG)	4.5kg
Package Dimensions	545mm×400mm×150mm
Gross Weight (KG)	5.2kg
Packing List	1 host, 1 USB cable, 1 user manual, 1 warranty card, 1 certificate, 2 Phoenix connectors

Front / Rear Panel

Front Panel

①. Power Switch: Used to turn on or off the power supply of the machine.

②. Color LCD Screen: The 2.4" color screen can display parameters and states, input signal level indication and works with the encoder.

③. Mute Button

Press this button to switch between mute and non-mute states, and at the same time, the display screen will display the corresponding state. In the mute state, there is no output at the output terminal of the amplifier.

④. Menu Selection Button

Press this button to select the function menu of the device, and set specific parameters and functions by the setting encoder ⑤. On the main interface, press and hold for 3 seconds to lock or unlock the system. In the locked state, other operations are not available.

⑤. Encoder for Setting Functional Parameters

It works with the Menu button. When a menu is selected, the encoder can be turned for function and parameter settings.

⑥. MIC VOL. Control Knob

Used to control the volume of all microphones. Turn the encoder clockwise to turn up the volume, counterclockwise to turn down the volume. At the same time, the corresponding volume parameter value (with an adjustable volume range from 0dB to 20dB) can be seen on the color screen. When the volume is set to 0dB, the machine still has an output. Only by the microphone potentiometer on the rear panel can the music be set to the minimum.

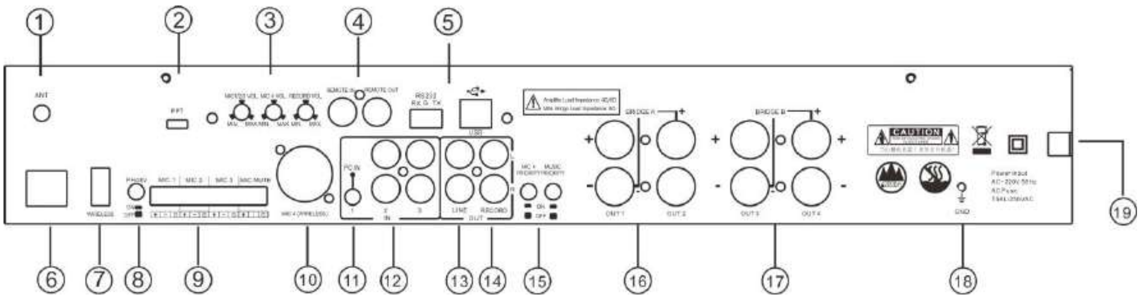
⑦. LINE VOL. Control Knob

Used to control the volume of the line input (or music). Turn the encoder clockwise to turn up the volume, counterclockwise to turn down the volume. At the same time, the corresponding volume parameter value (with an adjustable volume range from -79dB to 0dB) can be seen on the color screen.

⑧. USB Interface

Connect a computer through the USB interface for adjustment and settings.

Rear Panel



①. Built-in Wireless MIC Antenna Interface (Optional, certain models with this function).

②. PPT Page Turning Function Control Interface. Connect to the computer through the corresponding USB cable (Optional, same as the function ①).

③. Volume Control Potentiometer for All MICs and Recording Output (The potentiometer is not adjustable for the volume of MIC 3.)

④. Remote Input and Output Interfaces for Recording and Broadcasting Teaching (Optional, certain models with this function).

⑤. RS232 and USB Interfaces. The device can be controlled by the central control system via the RS232 interface, and the functional parameters of the device can be set on the computer via the USB interface.

⑥. Network Interface. Can be used for remote transmission of IP public address (Optional, certain models with this function).

⑦. External 2.4G Wireless MIC Interface (Optional, certain models with this function).

⑧. Phoenix Terminal, XLR MIC +48V Phantom Power Switch. When the switch is pressed down, it is connected to +48V, otherwise it is disconnected from +48V.

⑨. MIC 1/MIC 2/MIC 3 Phoenix Terminal, 1 MIC Mute External Port. The MIC input interface is connected according to the screen printing; short-circuit the “+” terminal and the “G” terminal of the MIC Mute external port to mute all channels of

microphones.
⑩. MIC 4 IN Composite XLR Interface. TRS (6.35) cable is connected to DC +6V phantom power, while XLR (Cannon) cable is connected to 48V phantom power. When the switch is turned on, +48V phantome voltage will be provided.
⑪. 3.5 LINE IN Interface for 2-Channel PC Audio Signals. The input terminal has a built-in audio isolator (optional, certain models with this function), which can effectively isolate the noise caused by the potential difference between the computer or mobile digital device and the device.
⑫. LINE IN Interface. This interface allows the input of various music signals.
⑬. LINE OUT Interface. The output signal of this interface is a mixed signal of microphones and music.
⑭. Recording Output Interface. It outputs the microphone signal only. The output level can be adjusted by the recording volume potentiometer.
⑮. Music Priority and MIC 4 Priority Function Switches. When the switch is pressed down, the corresponding function is enabled, otherwise it is disabled. When the music priority function is enabled, if there is a line signal input, the system will automatically mute the microphone signal, and if there is no line signal input, the system will automatically resume the microphone signal. When the MIC 4 priority function is enabled, if the MIC 4 interface has a signal input, the system will automatically mute the signals of MIC 1, MIC 2 and MIC 3, and if the MIC 4 has no signal input, the system will automatically resume the signals of MIC 1, MIC 2 and MIC 3.
⑯/⑰. 4-Channel Amplifier Output Terminals. Among them, OUT 1 and OUT 2 can be bridged as a group, and OUT 3 and OUT 4 can be bridged a group for output. Note: Please follow the minimum amplifier load impedance as marked on the screen printing for system connection. When not bridged, each output can be connected to a minimum impedance of 4Ω; when bridged, each group of outputs can be connected to a minimum impedance of 8Ω.
⑱. Grounding Screw. For the safety of users, be sure to ensure safe grounding to prevent power leakage and electric shock.
⑲. Power Cord. Please connect the power supply voltage as marked on the screen printing to prevent damage to the device.