Digital Audio Processor

User Manual

TS-DP880

Please read this manual in detail before using the system

Notification

WARNING

To ensure the reliability of the equipment and the safety of personnel, please observe the following when installing, using and maintaining:

● If any of the following conditions are found, please immediately turn off the power, plug out and quickly contact your nearest dealer. Do not continue using this unit, which may cause a fire or electric shock.

- If you find smoke or have a strange taste from the machine.
- If water or metal falls into the machine.
- If the unit is dropped or the case is damaged.
- If the wire is damaged (wire core exposure, broken wire, etc.).

● If the machine contains high-pressure parts, in order to avoid the fire or electric shock, absolutely don't open the case, if any questions please inform your nearest dealer.
● Do not place cups, bowls, vases or metal and other water-filled substances on the unit. Serious spilled liquid may cause a fire or electric shock.

● Never expose the unit to rain and any moisture or water, which may cause electric shock or fire.

● Do not place metal objects or flammable materials from the vents on the machine cover, nor place coins, which may cause fire or electric shock.

● Do not place heavy objects on the unit to avoid personal injury or property damage when the unit is slipping.

● Make sure that the volume is turned on at the beginning of the boot, and the high volume of the boot may cause hearing problems.

● For long-term accumulation of dust to be cleaned, please inform your dealer to regularly clean the machine, so as to avoid damage to the machine or cause a fire.

● The battery must be replaced with the same type of product and the correct installation should be made in order to avoid electrical damage and explosion hazard.

● The product is a Class III device. The device must be well connected to ground. The power plug must be connected to a power outlet with a grounding device to ensure that the equipment is fully grounded.

● This product uses a power plug or appliance input socket as a disconnecting device with the power supply, and must be disconnected if necessary for safety reasons.

- This equipment is only suitable for safe use at altitudes under 2000 meters.

Precautions

1. The installation environment

When installing the unit, in order to ensure the normal cooling of the host, should avoid the poor ventilation of the place or high temperature environment, to avoid direct sunlight.

Recommend to install cabinet or other well-ventilated place indoor. If you use the machine in the outdoors, please pay attention to waterproof, moisture, lightning protection measures.

Avoid installing in a violent place of vibration; do not place other equipment on the machine.

2. To avoid electric shock and fire

Do not touch the hands and the source with wet hands

Do not spill liquid on the machine, so as to avoid short-circuit or fire inside the machine.

Do not place other equipment directly on the top of the unit.

Non-professional service personnel Do not disassemble the unit yourself to avoid damage and electric shock.

3. Transport and handling

The packaging of the machine is designed and tested to ensure that the host will not be accidentally damaged during transport. It is best to use the original packaging when handling the unit.

Do not move the host device between the place or cold or over hot to avoid condensation inside the machine, affecting equipment life.

4. Please follow the warning instructions on this product, the warning signs on behalf of:

	Applicable to 2000 meters above sea level and below safe use
	Safe use only in non-tropical climates

5. Agreement

Please strictly follow the instructions in this manual. The software, hardware and appearance of this product will be upgraded and updated continually. The above changes will be made without notice.

Non-professional maintenance personnel, do not remove the product, to avoid damage and electric shock.

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1.Product Description

It is a high-performance audio processor with 8 analog balanced inputs and 8 analog balanced outputs. With functions including 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). it is mainly used in professional sound reinforcement scenarios to meet the needs of different places(such as conference rooms, courtrooms, auditoriums, multi-function halls, performances, classrooms) for the application of sound reinforcement systems.

1. 3A(AEC/ANS/AGC, i.e. Acoustic Echo Cancellation, Active Noise Suppression, Auto Gain Control) algorithm solves the acoustic problems of teleconferencing.

2. Shared + threshold auto mixing; AFC algorithm frequency shift + trap to solve the problem of local multi-microphone whistling.

3. Simultaneously support 8 scenes, while each scene is independent of each other.

4. Support each input channel equalizer and output channel equalizer; support independent bandwidth adjustment.

5. Support channel copy. After I/O processing is configured, it can be batch configured to other channels.

6. Support a variety of dynamic range controller DRC algorithm.

Phantom power supply: For capacitive microphone power supply. Do not turn it on in case of line input or non-capacitive microphones to prevent burning;

Ducker: When the level value of a channel exceeds the specified threshold, the level of the other channel will be attenuated.

Expander: In contrast to the compressor in terms of principle, an expander extends the dynamic range of a signal. The most basic difference between these two devices is that a compressor works on signals above the threshold, while an expander works on signals below the threshold. Expanders can make small signals even smaller.

Parametric equalizer: 12-band parametric equalizer. Each frequency point can adjust the Q value (0.002 to 50), gain (-24db to 18db), type, and frequency (20hz to 20khz). Three types of filters are available: high-shelf, low-shelf and peak filters.

Graphic equalizer: 10-band/15-band/31-band graphic equalizer is available. Each frequency point can adjust the Q value (narrowband, normal, wideband) and gain (-24db to 18db).

Compressor: It allows users to reduce the dynamic range of signals higher than the user-set threshold by adjusting threshold, compression ratio, soft knee point, attack time, and release time. The level of signals below the threshold remains unchanged.

Automatic gain: Automatic Gain Control (AGC) is a special case of compressors in which the threshold is set at a very low level, with a medium to slow attack time, long release time, and low ratio. Its purpose is to raise a signal of uncertain level to a target level while maintaining dynamics. Most automatic gain controls include some kind of silence detection to prevent loss of gain reduction during silence. Normalize the level of CD players playing background, foreground or waiting music by using automatic gain control to eliminate some variations in paging microphone levels.

Auto mixing: In conference rooms, if multiple microphones are turned on to the same gain level while only one person is speaking, the sound may not be very clear as other microphones will pick up room noise, reverberation, etc. When these signals are mixed with the normal microphone signal, the mixed audio output quality will be greatly reduced, and the entire sound reinforcement system is very easy to howl and cannot obtain sufficient sound transmission gain. In order to solve this problem, other microphones not in use need to be turned off. An automatic mixer can complete this shutdown process, whose respond speed is much faster than manual operation.

Feedback suppression: Feedback suppression is a device that automatically attenuates the feedback frequency. When acoustic feedback occurs, it will immediately discover and calculate its frequency and attenuation, and execute the command to suppress the acoustic feedback according to the calculation results, preventing infinite amplification of microphone positive feedback.

Echo Cancellation: Acoustic Echo Cancellation, or AEC, is a digital audio signal processing technology, used for audio and video conference calls when a participant in a local conference room and one or more speakers a certain distance away. The AEC program increases the speech intelligibility of the remote speaker by eliminating acoustic echoes generated in the local room.

Noise Suppression: The noise suppression module can effectively remove non-human voices. It can distinguish human voices from non-human voices and treat non-human voices as noise. If a piece of audio containing human voice and noise is processed by this module, then theoretically, only the human voice remains.

Matrix: The matrix has dual operation functions of routing and mixing. Horizontal represents the input channel, vertical represents the output channel, and the default is one-to-one input/output. After setting up auto mixing, echo cancellation, and noise suppression, the matrix also needs to be set up to obtain the correct signal routing relationships.

Delay : The time interval from the signal input to the processor to the output of it. It is generally used to produce effects such as reverberation or echo. It can also be used for the processing of auxiliary speakers used in large occasions, with maximum delay of 2s.

Crossover: Each output channel provides a high and low pass module consisting of a high pass filter and a low pass filter. Three filters are available: Bessel, Linkwich-Riley, and Butterworth, with adjustable frequency and slope (6db/OCT to 48db/OCT).

Limiter: The limiter is a circuit that allows users to clip the signal voltage amplitude by a limited range by means of adjusting the threshold, soft knee point, attack time, and release time to ensure that the signal does not exceed the threshold level.

2.Features

1． High-performance 64-bit DSP processor (800M main frequency), 32-bit/48KHz AD/DA, professional DSP processing, providing excellent high-quality sound.

2． Support 8 in 8 out audio matrix function, the input sensitivity can be adjusted according to different audio sources. Each input supports 48V phantom power supply, which can be individually configured to be turned on and off, flexible and convenient.

3． Support the ducker function, which is used for background music to automatically duck the microphone to speak, and provide a variety of parameter settings, which is convenient for flexible use on site.

4． Support the automatic gain function of the microphone, which is used to control the dynamic range of the pickup signal of the microphone to achieve consistent sound quality from far and near.

5． Support intelligent mixing function, including Gain sharing mixing and Threshold automatic mixing. The input channel can independently choose whether to participate in intelligent mixing, and can choose the corresponding mixing mode according to the application requirements of different scenarios. It can effectively solve the pain points that the sound reinforcement system is unstable and easy to howl due to the multi-opening of the microphone.

6． Support equalizer function, provide parametric equalizer and graphic equalizer, each input/output with 12-band parametric equalizer/10-band graphic equalizer/15-band graphic equalizer/31-band graphic equalizer optional. The parametric equalizer supports three types of EQ highshelf, EQ lowshelf and peak filters, and the graphic equalizer supports single-point bandwidth adjustment.

7． Support crossover function, provide Bessel, Linkwich-Rayleigh, Butterworth three filter types for selection, and support 6/12/18/24/32/40/48db/oct slope settings, the filter is adjustable in the whole frequency band.

8． Support the expander function to expand the dynamic range of the signal and to eliminate the noise floor of the device.

9． Support compressor function to compress the dynamic range of the signal, commonly used to compress the output signal.

10． Support the limiter function to limit the output signal range, and prevent damage to the sound reinforcement equipment.

11． Support the delayer 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.

12．Support the echo cancellation function, which is used for remote audio and video conferences to eliminate echo and increase voice clarity.

13．Support noise cancellation function, which can effectively eliminate environmental noise such as air conditioner sound and fan sound, and improve voice clarity.

14．Support auto feedback control function, two processing schemes of notch filter + frequency shifter, effectively solve the problem of acoustic feedback.

15．With ultra low system processing delay, the delay is less than 3ms.

16．2-inch IPS real color display on the panel, supporting the display of device network information, real-time level, channel mute status, matrix mixing status and other status.

17．Panel with USB interface, supports multimedia storage, and can store, record or play.

18．Support scene preset, import, export, support up to 8 scenes.

19．Support the function of restoring factory settings.

20．Support RS-232 interface, which can be used to connect with external central control system to realize centralized management and control.

21．Support RS-485 interface, which can be connected to the central control system and camera tracking system, and can realize the automatic camera tracking function.

22．8-channel programmable GPIO control interface (customized input and output).

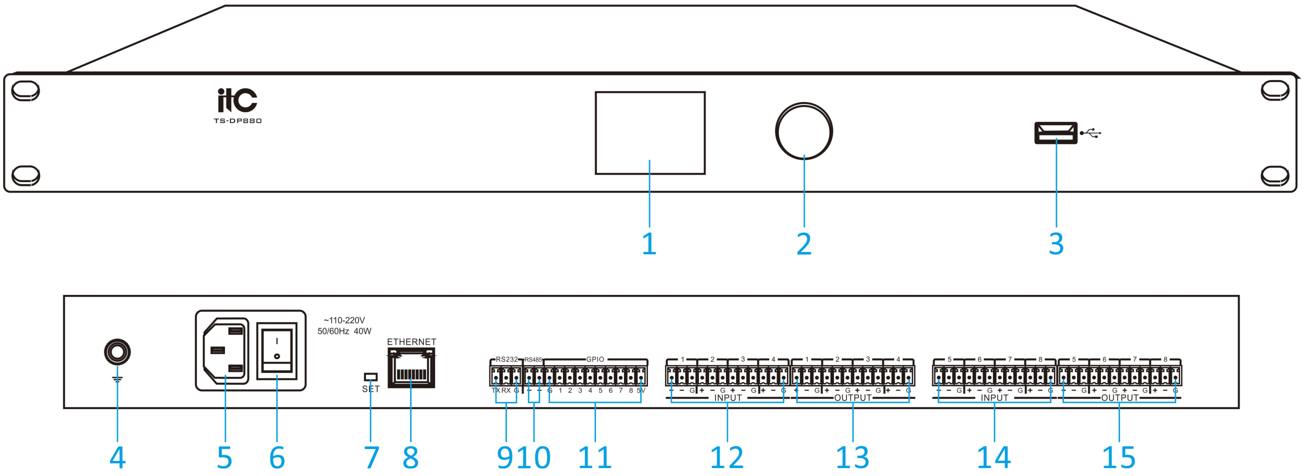
23．Support channel copy, paste, and gang control functions.

24．Enternet multi-purpose data transmission and control port, can support real-time management of single and multiple devices.

25．Support access to equipment through PC software, with management and control software: intuitive and graphical interface, support Windows7, 8, 10 and other system.

26．Support operation control through Android mobile phone APP software, device login, scene switching, input and output,matrix routing and channel setting and other functions through Android mobile APP software.

3.Product Interface Introduction



1． 2.0-inch IPS true color display: Display audio processor function status.

2． Encoder: display function control.

3． USB storage device interface: Support MP3, WMA, SBC, WAV format playback, support MP3 format recording.

4． Grounding pole: Internal device circuitry is grounded through this screw.

5． Power socket: Connect 110-220V50-60Hz AC power supply, to supply power to the device.

6． Rocker power switch: Control the power supply of the device.

7． DIP switch: Used when upgrading the software. Dialing to the RJ45 network port side for normal use.

8． ETHERNET: Network control interface: by connecting this network interface, the client computer can debug and monitor the device.

9． RS232: Support for the central control command, camera tracking and control of external devices. RX: receive data, TX: send data, G: ground wire.

10． RS485: Support for camera tracking.

11． GPIO: GPIO control port (Customizable input and output).

12． 1-4INPUT: Balanced signal input interface. It also supports 48V phantom power supply. It can connect to microphone, DVD and other devices.

13． 1-4OUTPUT: Balanced signal output interface. It can connect to the amplifier, mixer, active speakers and other devices.

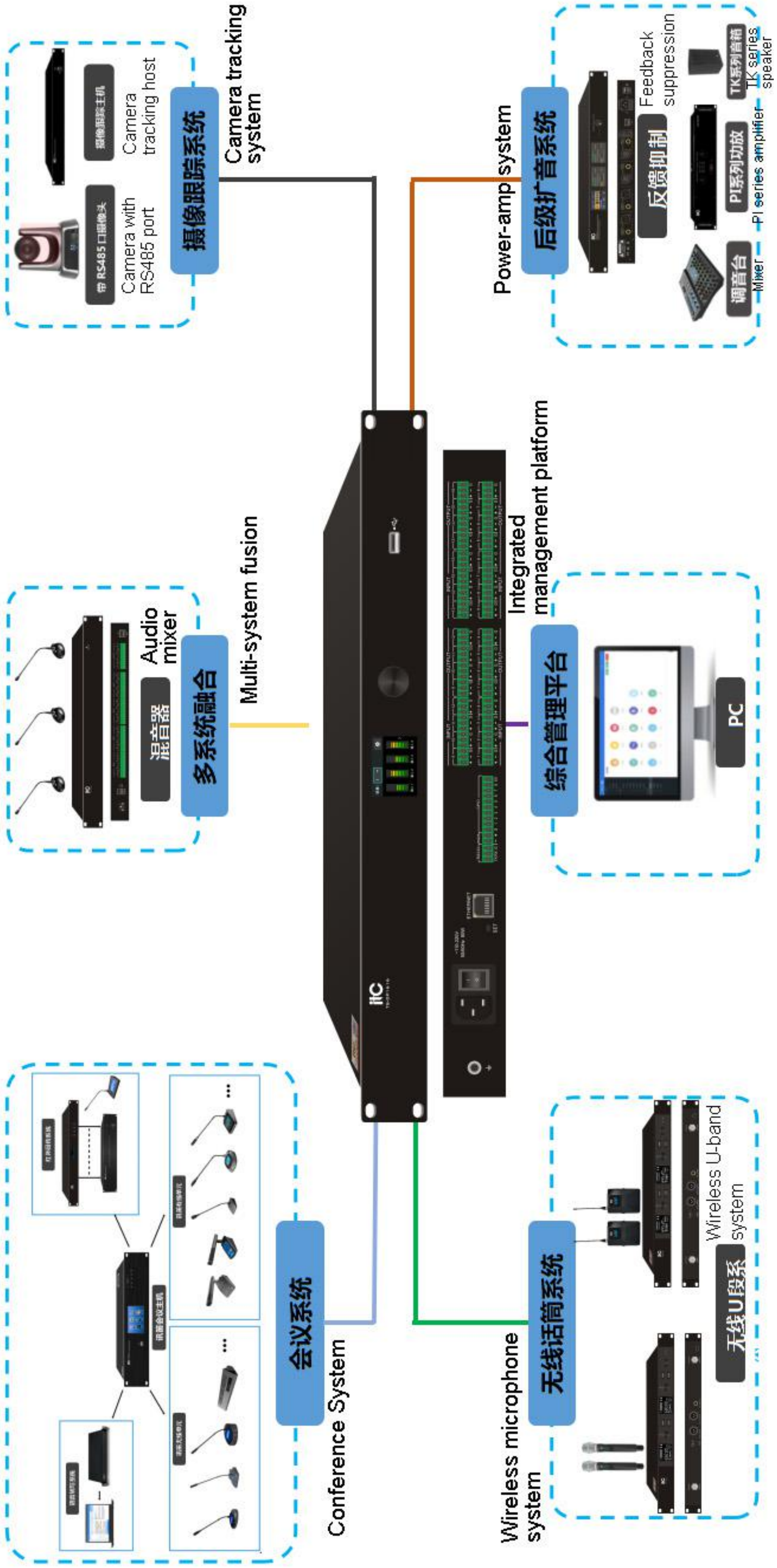
14． 5-8INPUT: Balanced signal input interface. It also supports 48V phantom power supply. It can connect to microphone, DVD and other devices.

15． 5-8OUTPUT: Balanced signal output interface. It can connect to the amplifier, mixer, active speakers and other devices.

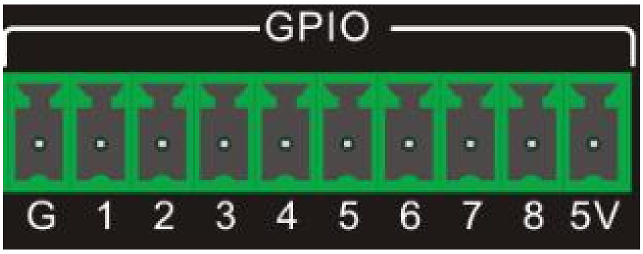
4.Specification

Model	**TS-DP880**
Input channel	8 balanced MIC/LINE inputs, using bare wire interface terminals, balanced connection;
Output channel	8 balanced line outputs, using bare wire interface terminals, balanced connection;
Processor	48kHz sampling frequency, 64-bit DSP processor; 32-bit A/D and D/A conversion
Phantom power	DC 48V
Frequency response	20Hz～20KHz
THD + N	≤0.002% OUTPUT=24dBu/1kHz
SNR	≥110dB@1kHz 24dBu (A-weighted)
Channel isolation	≥100dB@1kHz 24dBu (A-weighted)
Input impedance (balanced)	Balanced: 20KΩ
Maximum output impedance (balanced)	Balanced: 100Ω
Input range	≤+24dBu
Power supply	AC 110V-240V 50-60Hz
Power consumption	≤40W
Working temperature	-10℃～+45℃
Relative humidity	20%~80% relative humidity, no condensation
Cooling method	Fan forced cooling
Dimension (L×D×H)	484×298.2×45mm
Net weight	3.4Kg

5.System Wiring Diagram



6.GPIO Description

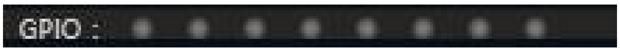


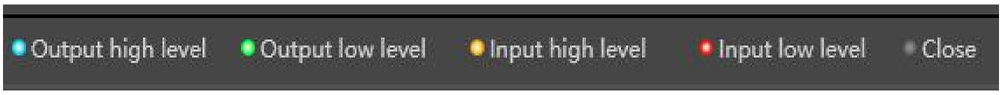
G: Ground

1 to 8: 8 GPIO ports, all freely configurable for input or output.

5V: GPIO default power output pin (5V).

The status of the device GPIO is displayed in real time in the status bar of the main interface of the software:

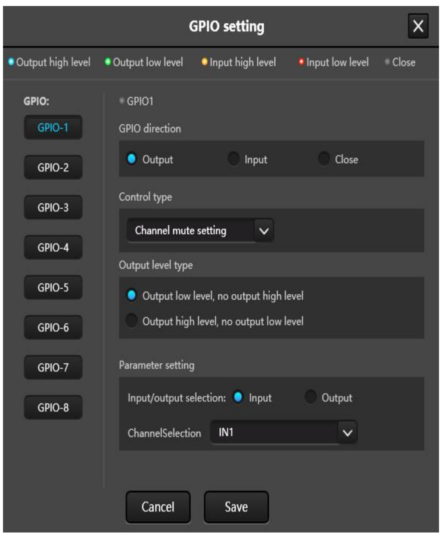




Port input: When the port input level is greater than 3.15V, it is recognized as high level; when there is no external power supply, the port input level is up to 5.5V; and when the input level is less than 1.35V, it is recognized as low level.

The use of GPIO is divided into two types:

1)Output: For example, the control type of the output of audio processor GPIO1 is set to Channel mute setting, the output level type is set to Output low level, no output high level, the parameter is set to Input channel 1 mute; when input channel 1 is mute, the GPIO1 output is low level, otherwise the output is high level.

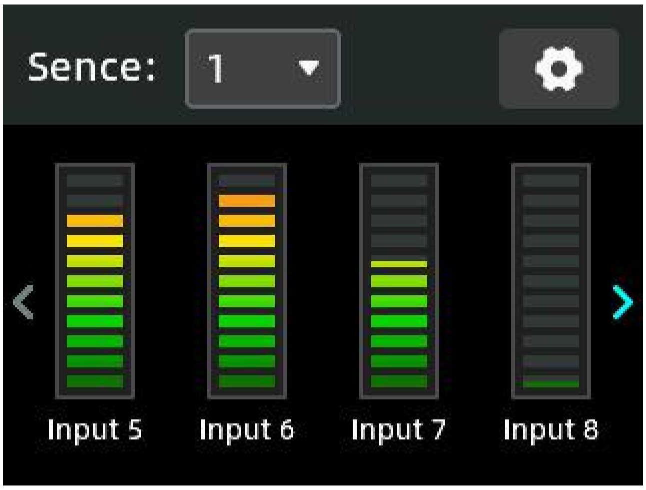


The input channel 1 of the audio processor is Mute -> the GPIO1 output is low level.

2)Input: For example, the trigger type of the input of the audio processor GPIO1 is Rising edge, the parameter is set to Mute input channel 1, and the external circuit connected to GPIO1 changes from low level to high level, causing the GPIO1 input pin level to generate a rising edge, thereby triggering Mute input channel 1.

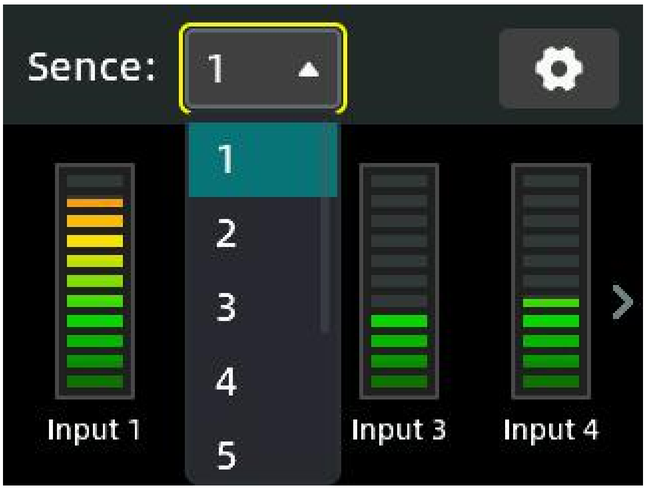
External circuit changes from low level to high level -> GPIO1 pin level generates a rising edge -> Mute input channel 1.

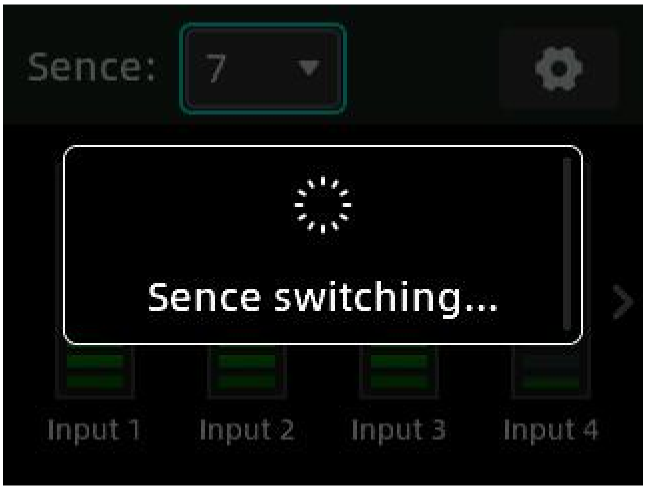
7.Display Operating Instructions



The main functions of the display include scene switching, signal level display, channel gain and mute settings, screen lock, language switching, host information display, and network information display.

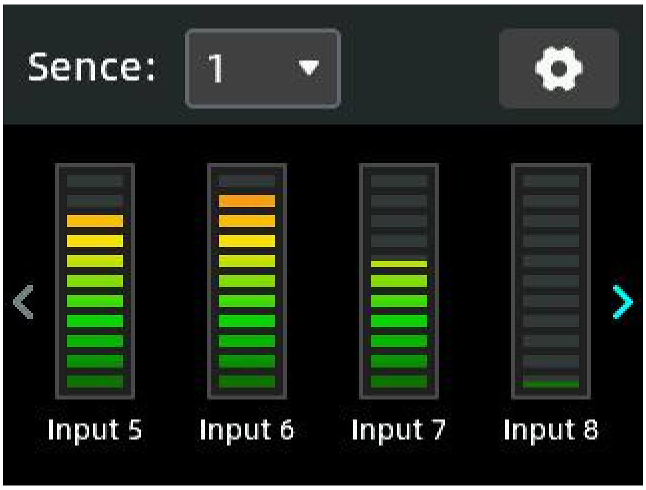
7.1.Scene switching





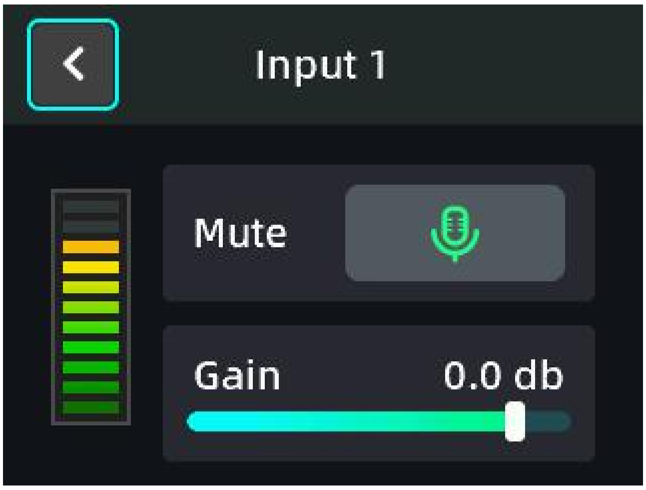
By selecting the scene number through the button, you can switch the corresponding scene and change the parameters of the corresponding scene.

7.2.Signal level display



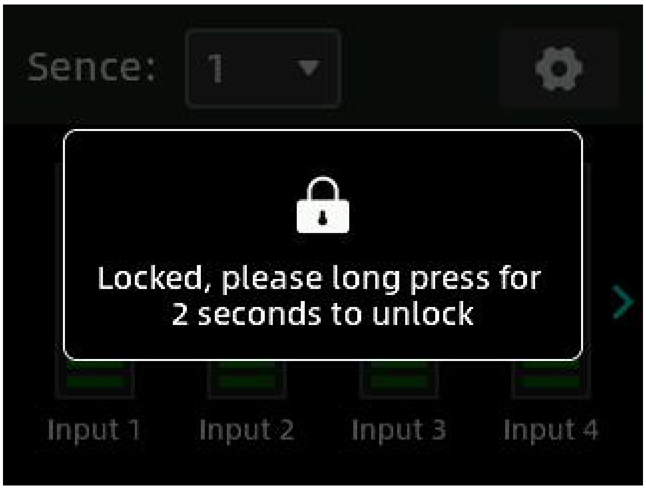
It can display the signal level of local input and remote dante input, and the level value is displayed in real time.

7.3.Channel gain, Mute settings



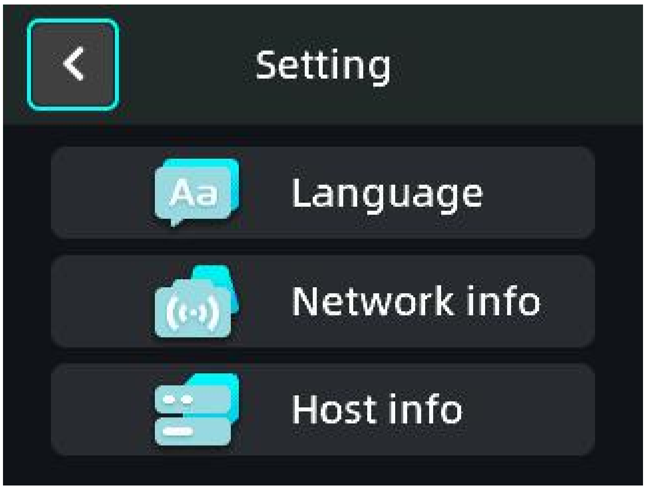
Select the channel number under the corresponding channel level via knob to enter the channel setting page, where you can adjust Mute and gain values of the channel.

7.4.Screen lock



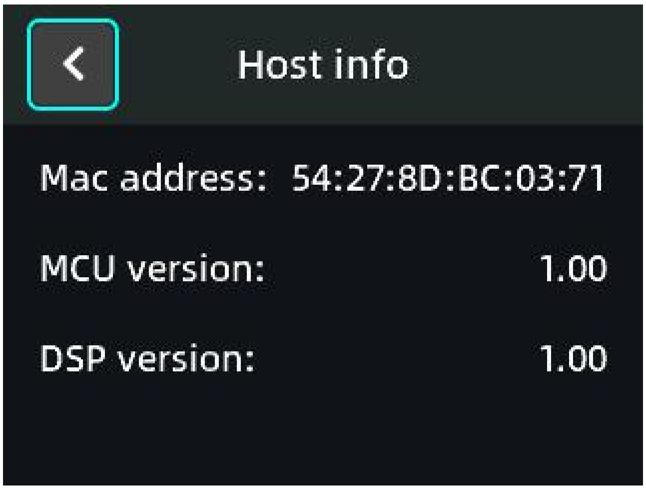
After 30 seconds of no operation, the operating knob will prompt a lock pop-up window, and you need to press and hold the knob for 2 seconds to unlock

7.5.Language switching



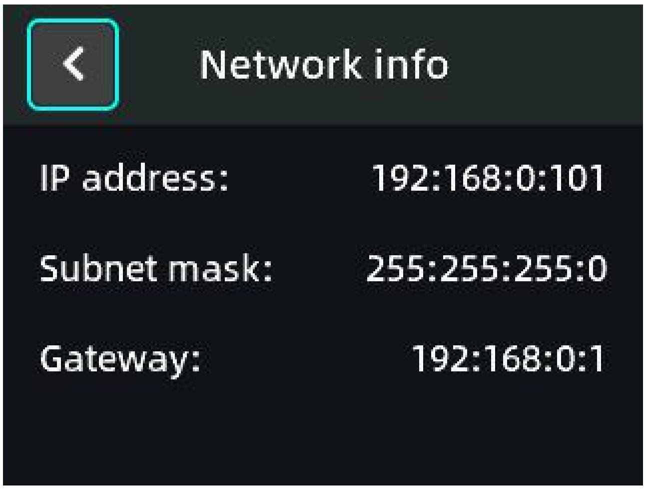
Click the Settings button in the upper right corner of the homepage to enter the settings page. Select the language selection button to enter the language setting. You can choose the language to switch. Currently, only Simplified Chinese and English are supported.

7.6.Host information display



The host information mainly displays the device MAC address, MCU and DSP version numbers.

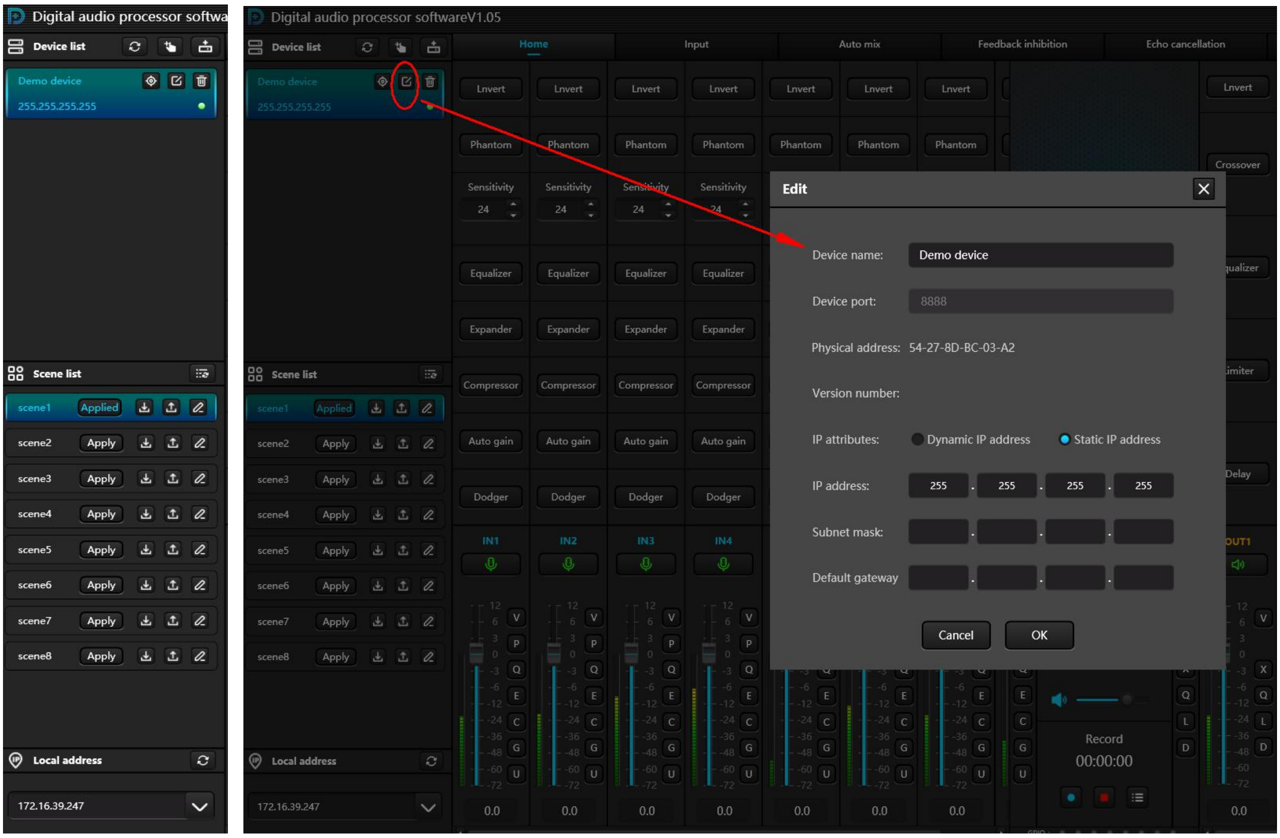
7.7.Network information display



Network information mainly displays the current IP address, subnet mask, and gateway information of the device.

8.PC Software Operating Instructions

8.1.Device settings



Device settings include device list, device scene, local address and other related configurations.

A. Device list

Support device refresh, manual addition of devices, device positioning, device disconnection, device-related parameter editing and device deletion.

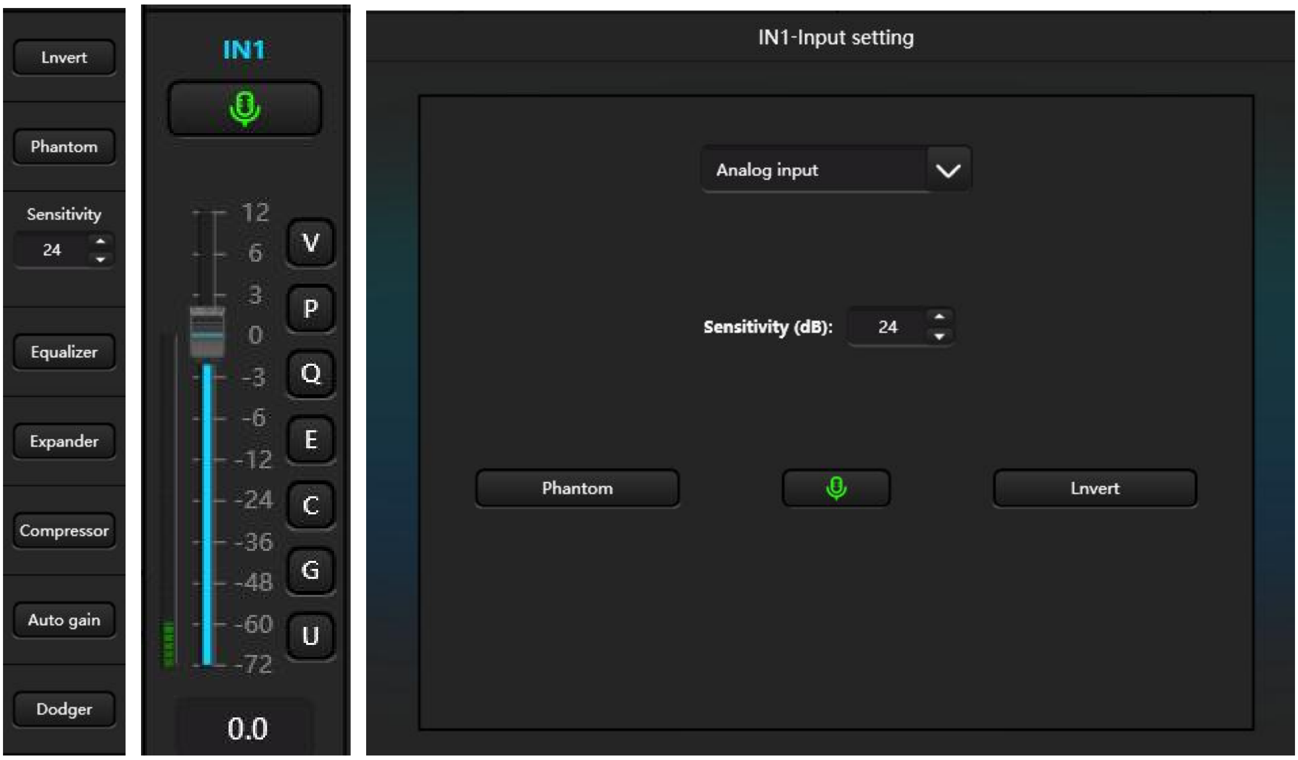
B. Device scene

Support scene import, export and factory settings restoration. When performing scene operations, you need to select the device in the device list.

C. Local address

Display the IP address of the current PC. If the PC is to configure the device, its IP needs to be in the same network segment as the IP of the device.

8.2.Input settings



1．The input signal is divided into analog signal and test signal. You can set whether to supply 48V power to the input source, as well as the control of the inversion, input mode, sensitivity and gain.

2．Test signal, which can generate three types of audio signals (sine wave, white noise, pink noise), and the frequency and level can be set.

3．Specification:

Inversion (V): Turn on/off inversion.

Phantom Power (P): Turn on/off phantom power.

Equalizer switch (Q): Turn on/off the equalizer.

Expander switch (Q): Turn on/off the expander.

Compressor switch (C): Turn on/off the compressor on/off.

Automatic gain switch (G): Turn on/off automatic gain.

Dodger switch (U): Turn on/off the ducker.

Phantom power supply: For capacitive microphone power supply. Do not turn it on in case of line input or non-capacitive microphones to prevent burning.

Input mode: The input source is microphone input or line input. When Microphone input is selected, the standard input level is 120mV; otherwise, the standard input level is 775mV.

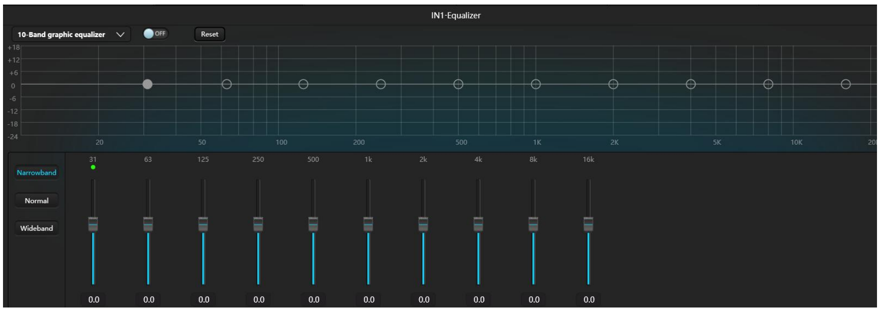
Sensitivity: Input source sensitivity setting, supporting -27~24 adjustment. When the sensitivity is 0, the gain is 0dB; when the sensitivity is 24, the gain is 18dB

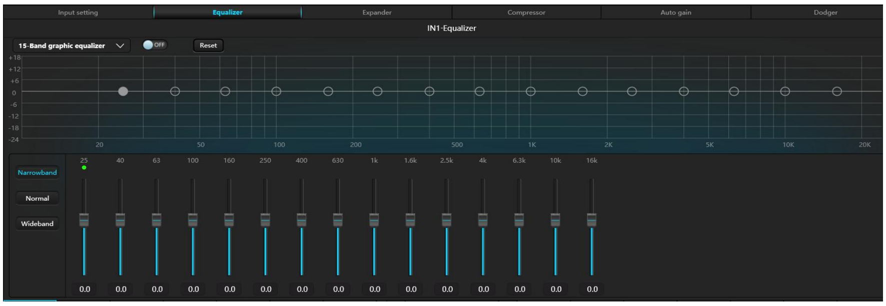
Mute: Channel mutes when it is selected.

Gain: Input source gain, supports -72~12dB adjustment.

Test signal: There are three test signals available: sine wave, white noise, and pink noise. Level ranges from -80~0dBFS.

8.3.Input channel - equalizer





1． The equalizer of the input channels include parametric equalizer (PEQ) and graphic equalizer (GEQ).

2． The main function of the equalizer is to adjust the amplification/reduction of electrical signals of various frequency components, to compensate for the defects of speakers and sound fields by adjusting electrical signals of various frequencies, to compensate and modify various sound sources and other special functions.

3． Specification:

A.Parametric equalizer (PEQ)

Mode: You can choose parametric equalizer (PEQ) or graphic equalizer (GEQ).

Equalizer switch: Turn this function on/off.

Type: High and low shelf filters and parametric filters Optional.

Center frequency: Support 12 frequency bands. Range: 20Hz~20KHz.

Gain: Gain/attenuation at the frequency center point. Range: -24~18.

Q value: The quality factor of the filter. Range: 0.02~50.

Frequency band switch: Turn on/off the frequency band. When turned off, the frequency band will not participate in filtering.

B.Graphic Equalizer (GEQ)

Mode: You can choose 10-segment graphic filter, 15-segment graphic filter and 31-segment graphic filter.

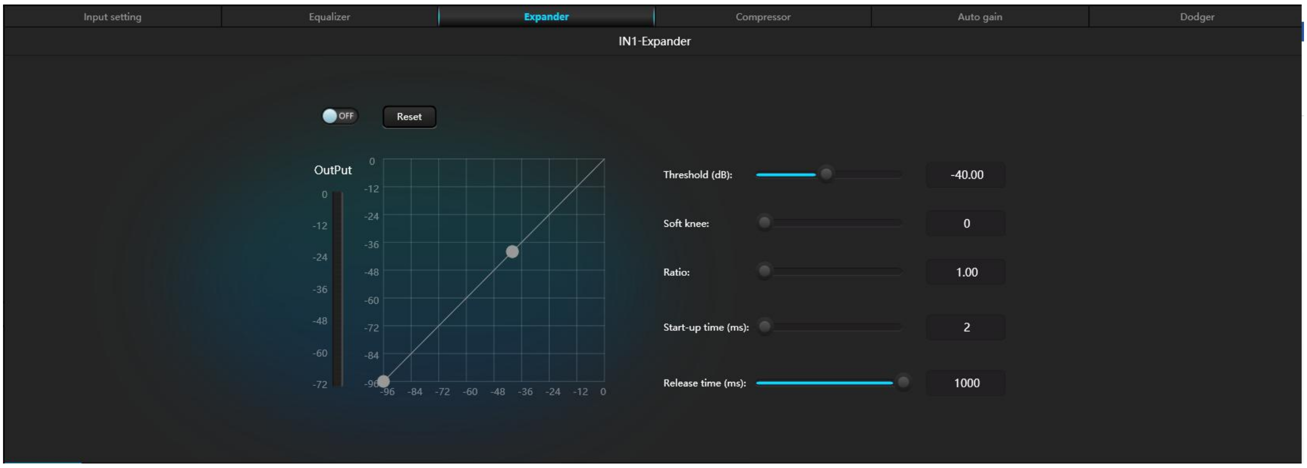
Equalizer switch: Turn this function on/off.

Center frequency: Support 10/15/31 frequency points. Range: 20Hz~20KHz.

Gain: Gain/attenuation at the frequency center point. Range: -18~18.

Bandwidth: Supports 3 different bandwidth modes. Narrowband mode: the bandwidth is lower than ordinary bandwidth; Normal mode: commonly used ordinary bandwidth; Wideband mode: the highest bandwidth.

8.4.Expander



1. Attenuate signals below the threshold, equivalent to a noise gate.
2. Specification:

Extender switch: Turn this function on/off.

Threshold: When the current channel is lower than the threshold, it begins to attenuate.

Soft knee point: It can make the signal attenuation smoother. Range: 0~20.

Ratio: Ratio for attenuation when below threshold. Range: 1~20.

Attack time: The time when the channel signal starts to attenuate after the reference signal is higher than the threshold.

Release time: The time when the attenuated signal returns to the original signal size after the reference signal falls below the threshold.

8.5.Compressor

1. Attenuate signals higher than the threshold. The more above the threshold, the more attenuation.

2. Specification:

Compressor switch: Turn this function on/off.

Gain: Channel gain, supporting -72~0dB adjustment.

Threshold: When the current channel is higher than the threshold, it begins to attenuate.

Soft knee point: It can make the signal attenuation smoother. Range: 0~20.

Ratio: Ratio for attenuation when above the threshold. Range: 1~20.

Attack time: The time when the channel signal starts to attenuate after the reference signal is higher than the threshold.

Release time: The time when the attenuated signal returns to the original signal size after the reference signal falls below the threshold.

8.6.Automatic gain control (AGC)

1. Automatic Gain Control (AGC) function is to maintain the signal at a target level while maintaining dynamics. The application scene is as follows: when the speaker is slightly away from the microphone, the sound picked up by the microphone will be lower than the voice at normal distance. At this time, the AGC will slightly gain the sound. Since it is live sound reinforcement, this gain cannot be increased infinitely, which will cause howling or other adverse situations, and only certain compensation can be made. When the speaker speaks close to the microphone, the sound will be louder than the voice at normal distance, and it will be more obviously especially when the speaker is very close. At this time, the automatic gain will attenuate the sound signal to a certain extent. In this way, automatic gain can maintain the sound within a certain dynamic range.

2．Specification:

Automatic gain on and off: Turn this function on or off.

Soft knee point: It allows the compression ratio to slowly increase as the input signal level increases, making the change from uncompressed to compressed sound less obvious. The larger the value, the larger the influence range of the knee point.

Target Level: Desired output signal level. If the signal is above this threshold, the controller will compress the signal proportionally.; if the signal is below this threshold and above the startup level, gain compensation occurs. For example, the target level is set to -15db and the startup level to -60db, if the microphone level is -10db at this time, the compression will be performed according to the compression ratio; if the microphone level is -17db and the maximum gain is 6db at this time, the gain compensation is 2db.

Compression ratio: The ratio between the change in input signal level above the threshold and the change in output signal level.

Attack time: Reaction time for boosting the input signal level to the target level, that is, when the input level is less than the target level, the signal needs to be boosted. This parameter controls the speed of boosting.

Release time: Reaction time for compressing the input signal level to the target level, that is, when the input level is higher than the target level, the signal needs to be compressed. This parameter controls the speed of compression.

Startup level: minimum required input signal level. If the signal is above this threshold, the signal is compressed/compensated; if the signal is below this threshold, the signal is not processed.

Maximum gain: When the signal level is lower than the target level and higher than the startup level, a certain gain compensation needs to be performed on the signal. This parameter controls the maximum amount of gain compensation to the level.

8.7.Ducker

1． When the level value of one channel exceeds the specified threshold, the level of the other channel will be attenuated. This is the ducking effect.

2． Specification:

Reference signal: Support the remaining 15 channels except this channel as reference signals. When the level value of a selected reference channel is greater than the threshold, the current channel is attenuated. When the level values of all selected reference channels are less than the threshold, the current channel gradually resumes.

Ducking switch: Turn this function on/off.

Gain: Channel gain, supporting -72~0dB adjustment.

Mute: Channel mutes when it is selected.

Threshold: When the reference channel is higher than the threshold, the current channel starts to attenuate, otherwise it resumes when lower than the threshold.

Depth: The amount reduced by the dodge signal.

Attack time: The time when the signal of the ducked channel begins to attenuate after the reference signal is above the threshold value

Release time: The time when the ducked signal returns to the original signal size after the reference signal falls below the threshold.

Holding time: Holding time refers to how long the ducking on the ducked channel will be maintained after the control signal is lower than the threshold.

8.8.Auoto mixing

1． Smart mixing has two modes: Gain sharing mode and Threshold mode. This function can effectively increase the sound transmission gain when multiple microphones are turned on at the same time.

2． Gain sharing mode automatically allocates the gain of one microphone to each microphone. When no one speaks, each microphone allocates the gain of one microphone equally. When only one person speaks, the speaking microphone will obtain all the sound transmission gain. When more than one person is speaking, the gain is allocated according to the volume of each person's voice. The louder the voice, the more gain will be allocated.

3． The Threshold automatic mode is developed based on the noise gate. Each channel has a noise gate, which is either open or closed. The application scenario is as follows: multiple microphones are turned on at the same time, and compared with the host lineout, the mixer can be pulled higher without howling.

4. Specification:

A. Gain-share mixing

(1)Main control parameters

Mode: Gain sharing auto mixinging and Threshold auto mixinging are optional.

Gain: The output gain of the automatic mixer. Range: -72~12.

Gain compensation: Solve the problem of big difference in volume between one microphone and multiple microphones on simultaneously. During the actual debugging process, first turn on only one of the MICs participating in the mixing, and then listen to the volume of the amplifier; then turn on all the MICs participating in the mixing, and adjust it to the extent that the volume of only one

MIC on is the same as that of multiple MICs on.

Slope: Similar to the expansion ratio of an expander. The larger the slope value, the more sound transmission gain the speaking microphone obtains, the more sound transmission gain the non-speaking microphone attenuates. The smaller the slope value, the less sound transmission gain the speaking microphone obtains, the less sound transmission gain the non-speaking microphone attenuates, while the overall attenuation will be increased. A common setting is 2 or a value near 2. If it is set to 1.0, the effect is equivalent to turning off the auto mixinging of all channels; when set to 3.0, it can lead to a larger gain adjustment, which may produce unnatural effects. The larger the value set, the more channels are turned on and the more overall attenuation is achieved.

Response time: The time to acquire all the transmission gain when one microphone is speaking or the time when the transmission gain of other non-speaking microphone attenuates. The longer the setting time, the longer it takes for the speaking microphone to obtain all the sound transmission gain, and the longer it takes for the microphone transmission gain to decay for other non-speaking microphones. Vice versa for for setting shorter time, i.e., if this parameter is set too short, it will result in an unnatural transition of the sound from nothing to something, and too long will have the word header possibly swallowed.

Environmental threshold: The threshold at which the microphone receives ambient noise. When multiple microphones are turned on, if only one microphone is speaking, this microphone cannot get most of the gain, as other microphones not speaking are assigned a certain proportion of gain due to ambient sound. The more microphones are turned on, the greater the proportion. The purpose of this parameter is to recognize which mics are speaking, and the mics that are speaking will receive most of the gain independent of the number of mics turned on.

Switch: Turn gain sharing mixing on or off.

(2) Channel control parameters

Mute: The sound of this microphone will be muted and still participate in smart mixing, while the microphone sound will not be output.

Gain: Mixing channel output gain. Range: -72~12.

Priority: Setting a priority will override a high-priority channel over a low-priority channel, thereby affecting the auto mixinging algorithm. The parameter range is 0~9, and the larger the value, the higher the priority. The impact of this parameter on Gain sharing mixing is that when the volume of the two microphones are the same, the one with higher priority will be assigned much more gain than the other. The impact on Threshold mixing is that when the volume of the two microphones exceed threshold, the one with higher priority will be preferred for remixing.

Smart mixing: Automatic mixing on/off button. Enable this button for channels that need to participate in auto mixinging. It can also be turned off and the channel will not participate in auto mixinging. Channels that do not participate in auto mixinging will be output directly superimposed with the output after automatic mixing and then output.

B.Threshold mixing

(1) Main control parameters

Mode: Gain sharing auto mixinging and Threshold auto mixinging are optional.

Gain: The overall output gain of the automatic mixer. Range: -72~12.

Switch: Turn Threshold mixing on or off.

(2) Channel control parameters

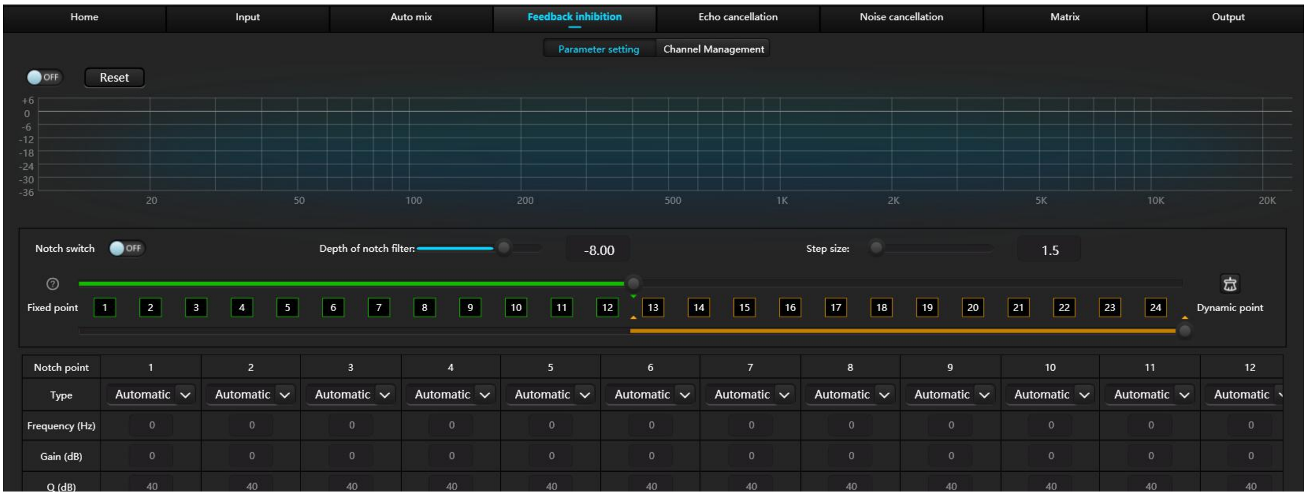
Mute: The sound of this microphone will be muted and still participate in smart mixing, while the microphone sound will not be output.

Gain: Mixing channel output gain. Range: -72~12.

Sensitivity: The noise gate threshold of the mix channel. Range: -72~0.

Smart mixing: Automatic mixing on/off button. Enable this button for channels that need to participate in auto mixinging. It can also be turned off and the channel will not participate in auto mixinging. Channels that do not participate in auto mixinging will be output directly superimposed with the output after automatic mixing and then output.

8.9.Feedback suppression



1．The sound signal collected by the microphone includes the sound signal amplified by the speaker. This signal is continuously superimposed and amplified in the acoustic feedback loop. The positive feedback produces oscillation and howling. AFC obtains the frequency of howling by analyzing the power spectrum of fixed-size audio frame by frame. After detecting the components, use a notch filter or frequency shift filter to change the amplitude and phase of the audio to achieve howling suppression.

2．The fixed point of the notch is mainly used to capture the inherent howling point of the sound field; the dynamic point is used to capture the howling point formed after the sound field changes. When the fixed point of the trap is full, the grabbing howling point will be set as a dynamic point. According to the actual sound field environment settings on site, the corresponding fixed point/dynamic point of the trap.

3．If both the notch filter and the frequency shifter are set to "Enable", the frequency shift function will only be enabled when the fixed points and dynamic points of the notch filter are full. The frequency shift function is always enabled when the notch is set to "off" and the frequency shifter is set to "on".

4. Specification:

Feedback suppressor switch: Turn on/off the feedback suppressor.

Notch filter switch: Turn on/off the notch filter.

Notch depth: The notch point can attenuate the maximum gain. Range: -24~0db.

Step size: The attenuation gain of the notch point captured for the first time is 3db. When the same frequency point is detected again, the attenuation gain of the notch point is increased by this value.

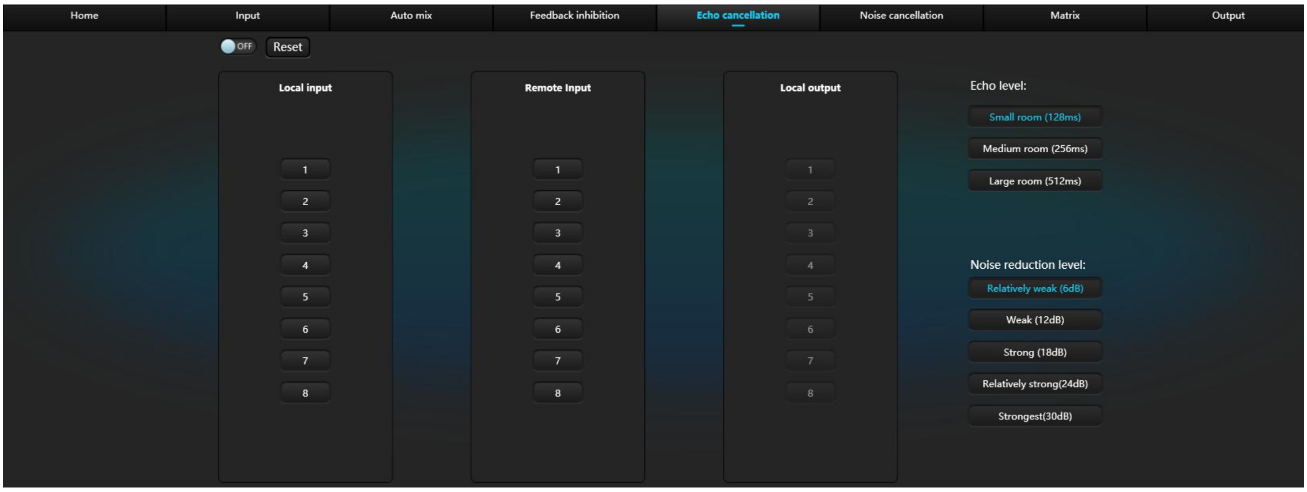
Range: 1.5~9.

Notch point: It is divided into Manual and Automatic type; the frequency, gain and Q value of manual notch point can be set manually; the frequency and gain of automatic notch point are automatically set according to the frequency of the captured notch point.

Frequency shift switch: Turn on/off frequency shift.

Frequency shift strength: The frequency of the overall mobile signal. Range: -10~10HZ.

8.10.Echo cancellation



1． The far-end speaker's voice is collected by the far-end microphone and transmitted to the communication device. After wireless or wired transmission, it reaches the near-end communication device and is played through the near-end speaker. This sound will be collected by the near-end microphone to form an acoustic echo. After transmission, it is returned to the far-end communication device and played through the far-end speaker, so that the far-end speaker hears his/her own echo. The adaptive algorithm is used to adjust the iterative update coefficients of the filter to estimate a desired signal that approximates the echo signal that passes through the actual echo path, which means to simulate the echo signal, and then this simulated echo is subtracted from the mixed signal captured by the microphone, to achieve the function of echo cancellation.

2. Specification:

Echo cancellation switch: Turn on/off echo cancellation.

Far-end input: The far-end speaker reaches the near-end input channel after wireless or wired transmission.

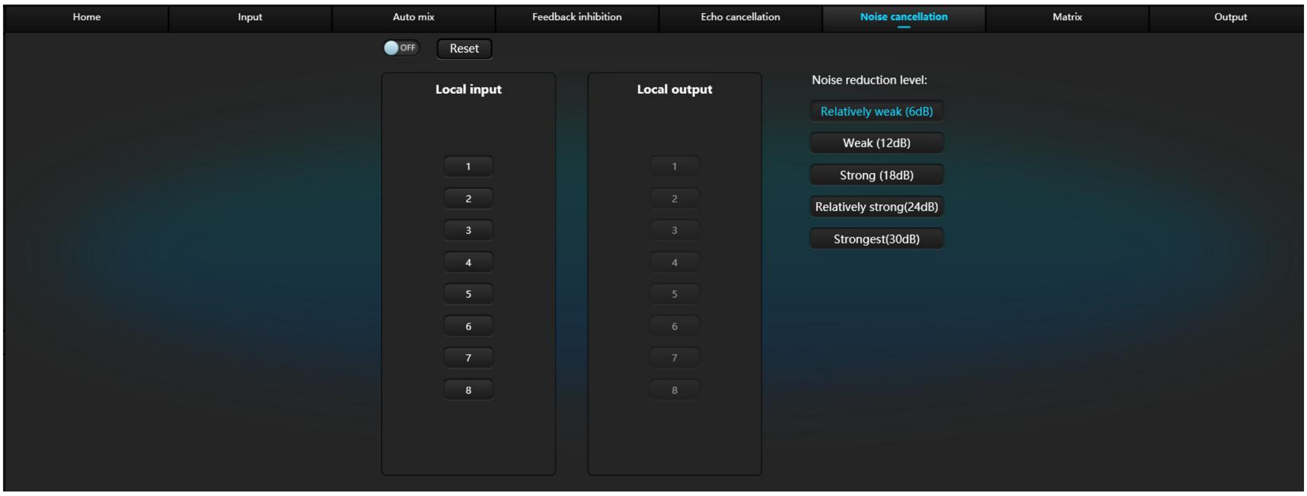
Local input: The input channel where the local near-end microphone captures the audio.

Local output: The channel where the local near-end microphone captures the audio and outputs the audio after adaptive echo cancellation processing.

Echo level: Three echo levels of small room, medium room and large room are supported.

Noise reduction level: It is divided into five noise reduction levels: Faintish, Weak, Strong, Fairly strong, and Strongest.

8.11.Noise cancellation



1. A process in which a noise-mixed signal passes through a filter to suppress the noise, allowing the signal itself to pass through without distortion.

2. Specification:

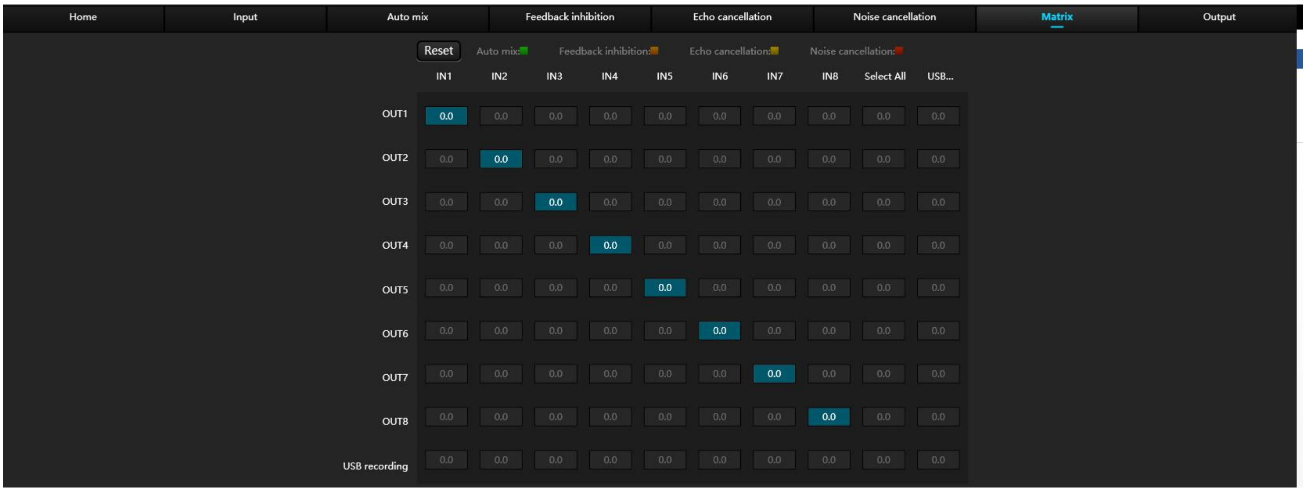
Noise cancellation switch: Turn on/off noise cancellation.

Local input: The input channel where the local near-end captures the audio.

Local output: The channel where the local near-end captures the audio and outputs the audio after noise cancellation processing.

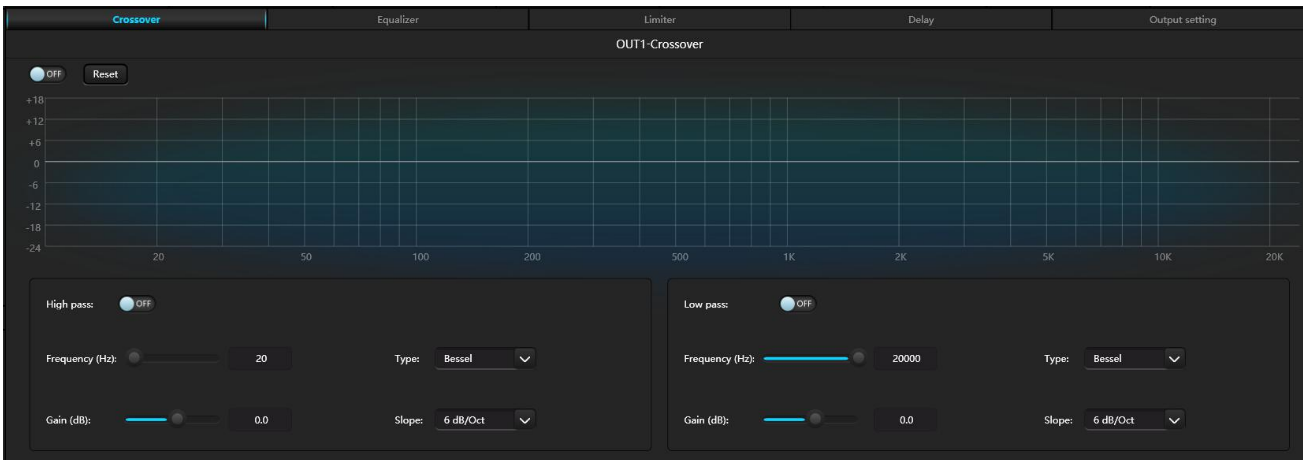
Noise reduction level: It is divided into five noise reduction levels: Faintish, Weak, Strong, Fairly strong, and Strongest.

8.12.Matrix



The matrix has dual operation functions of routing and mixing, and the mixing here uses overlay mixing. The matrix has dual operation functions of routing and mixing. Horizontal represents the input channel, vertical represents the output channel, and the default is one-to-one input/output, as shown in the figure. If the sound of Input Channel 1 and Input Channel 2 needs to be mixed to Output Channel 1, just click the horizontal 1 and 2 on Output Channel 1. If automatic mixing is selected, the sounds of Input Channel 1 and Input Channel 2 **after smart mixing processing should be superimposed and mixed to Output Channel 1.** Otherwise, the sounds of input channel 1 and input channel 2 **before smart mixing processing should be superimposed and mixed to Output Channel 1.** Ditto for the echo cancellation, noise suppression and feedback suppression modules.

8.13.Crossover



1. Each output channel provides the high and low pass module, consisting of a high pass filter and a low pass filter.

2. Specification:

Crossover switch: Turn on/off the crossover.

High pass/low pass switch: Turn on/off high pass/low pass.

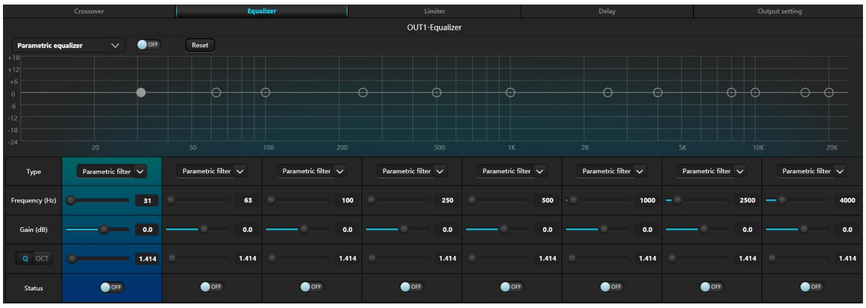
Frequency: The cutoff frequency of the filter. The cutoff frequency of Bessel and Butterworth is defined at -3dB, while the cutoff frequency of Linkwitz-riley is defined at -6dB.

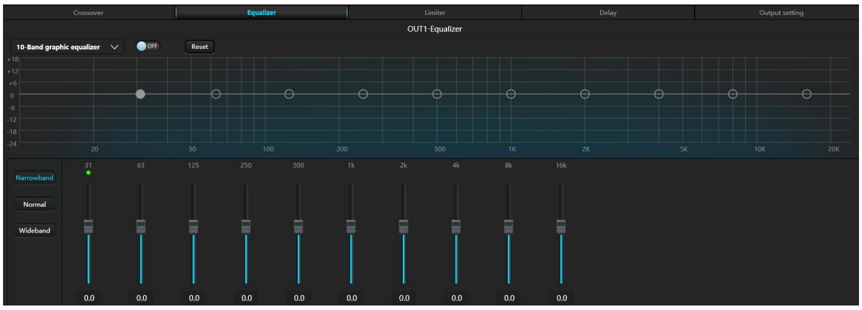
Type: Three filter types are available: Bessel, Butterworth and Linkwitz-riley.

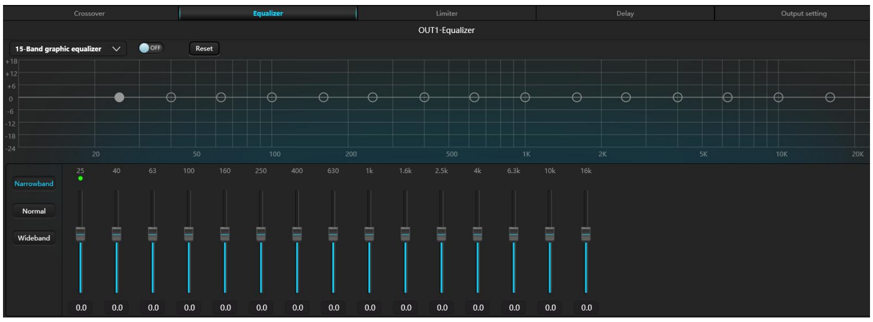
Gain: Gain/attenuation of the filter's cutoff frequency. Range: -15~15.

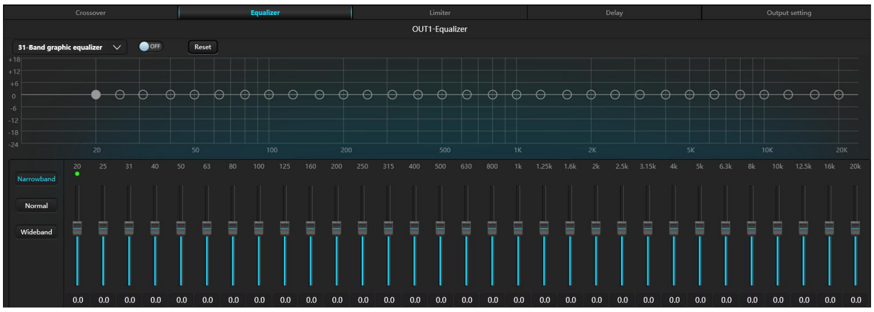
Slope: The attenuation size of the filter's transition band. There are eight choices: 6, 12, 18, 24, 30, 36, 42, and 48dB/Oct. For example, 6dB/Oct means that in the transition band, the amplitude is attenuated by 6 dB per octave difference in frequency.

8.14.Output channel - equalizer









1． The equalizer of the output channels include parametric equalizer (PEQ) and graphic equalizer (GEQ).

2． The main function of the equalizer is to adjust the amplification/reduction of electrical signals of various frequency components, to compensate for the defects of speakers and sound fields by adjusting electrical signals of various frequencies, to compensate and modify various sound sources and other special functions.

3. Specification:

A. Parametric equalizer (PEQ)

Mode: You can choose parametric equalizer (PEQ) or graphic equalizer (GEQ).

Equalizer switch: Turn this function on/off.

Type: High and low shelf filters and parametric filters Optional.

Center frequency: Support 12 frequency bands. Range: 20Hz~20KHz.

Gain: Gain/attenuation at the frequency center point. Range: -24~18.

Q value: The quality factor of the filter. Range: 0.02~50.

Frequency band switch: Turn on/off the frequency band. When turned off, the frequency band will not participate in filtering.

B. Graphic Equalizer (GEQ)

Mode: You can choose 10-segment graphic filter, 15-segment graphic filter and 31-segment graphic filter.

Equalizer switch: Turn this function on/off.

Center frequency: Support 10/15/31 frequency points. Range: 20Hz~20KHz.

Gain: Gain/attenuation at the frequency center point. Range: -18~18.

Bandwidth: Supports 3 different bandwidth modes. Narrowband mode: the bandwidth is lower than ordinary bandwidth; Normal mode: commonly used ordinary bandwidth; Wideband mode: the highest bandwidth.

8.15.Limiter

1．Set a threshold level and any signal above that threshold will be automatically downgraded. This helps prevent the signal from being too loud, which will cause distortion or damage to the device, thus ensuring the audio signal remains within safe and acceptable levels.

2．Specification:

Limiter switch: Turn this function on/off.

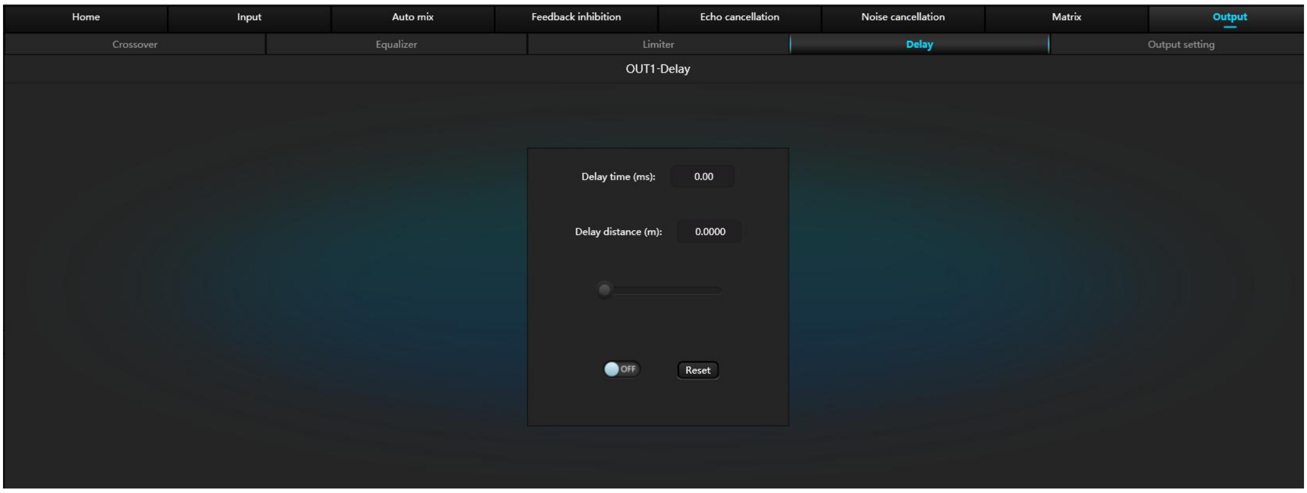
Threshold: The startup level of limiting. When the signal is higher than this limit value, limiting processing is performed.

Soft knee point: It can make the signal attenuation smoother. Range: 0~20.

Attack time: The time when the channel signal starts to attenuate after the reference signal is higher than the threshold.

Release time: The time when the attenuated signal returns to the original signal size after the reference signal falls below the threshold.

8.16.Delay



1. The time interval between the input and output of the signal to the processor, generally used to produce effects such as reverberation or echo.

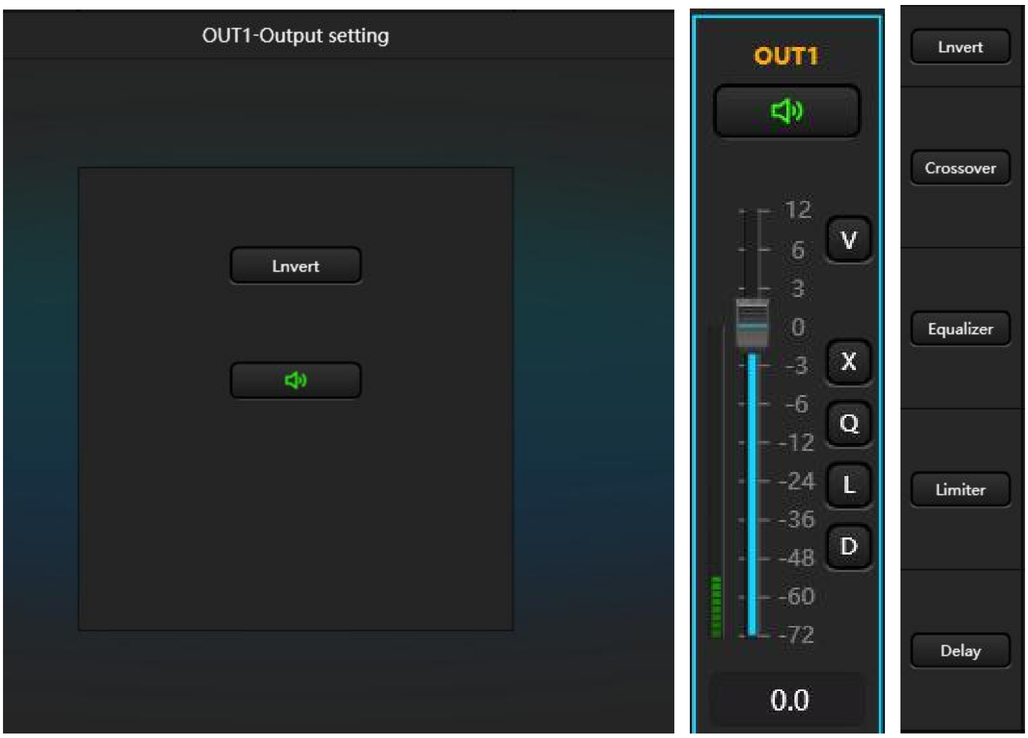
2. Specification:

Delay switch: Turn on/off the delay.

Delay time: Signal output delay, range: 0~2000ms.

Delay distance: Signal output delay distance, range: 0~680m.

8.17.Output settings



1. Functions such as Inversion, Mute and Gain can be set.

2. Specification:

Inversion (V): Turn on/off inversion.

Crossover (X): Turn on/off the crossover.

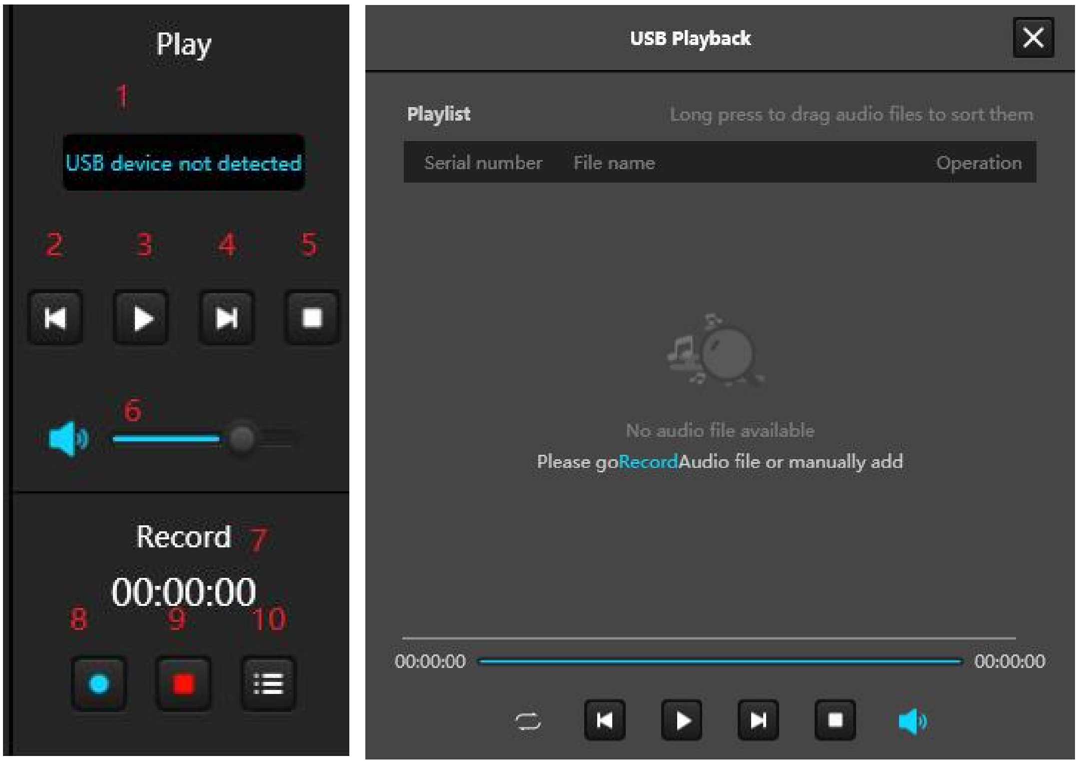
Equalizer switch (Q): Turn on/off the equalizer.

Delay switch (D): Turn on/off the delay.

Mute: Channel mutes when it is selected.

Gain: Output source gain, supporting -72~12dB adjustment.

8.18.USB recording and playback function



1. Support USB recording and playback functions.

2. Interface introduction:

A.Playback settings

1: Playlist, and display whether a USB flash drive is inserted.

2: Previous.

3: Pause.

4: Next.

5: Stop.

6: Volume adjustment.

B.Recording settings

7: Recording duration display.

8: Start recording.

9: Stop recording.

10: Recording list.

C.USB playlist

11: Song list

12: Playback mode, supporting Loop playback, Single playback, and Single cycle.

13: Previous.

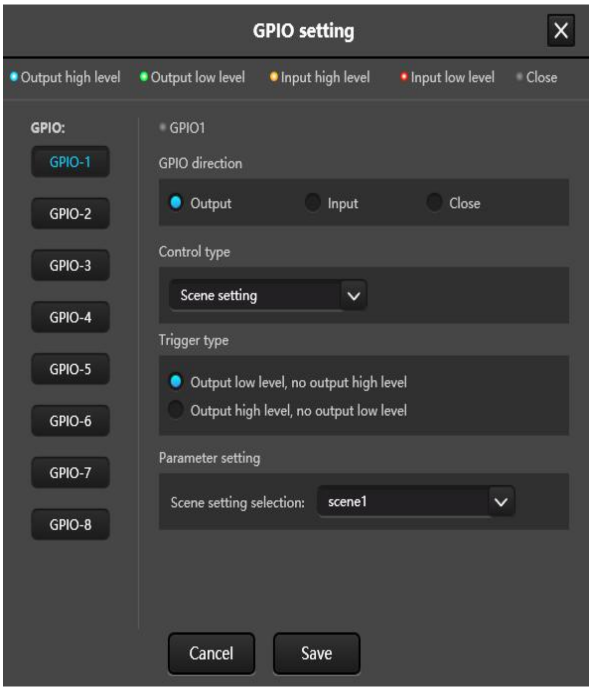
14: Pause.

15: Next.

16: Stop.

17: Mute.

8.19.GPIO settings



1. The device has a total of 8 GPIOs, which can be independently configured as inputs or outputs.

2. Scene, Volume, Channel mute, System mute, Mixing settings, and Serial commands are available

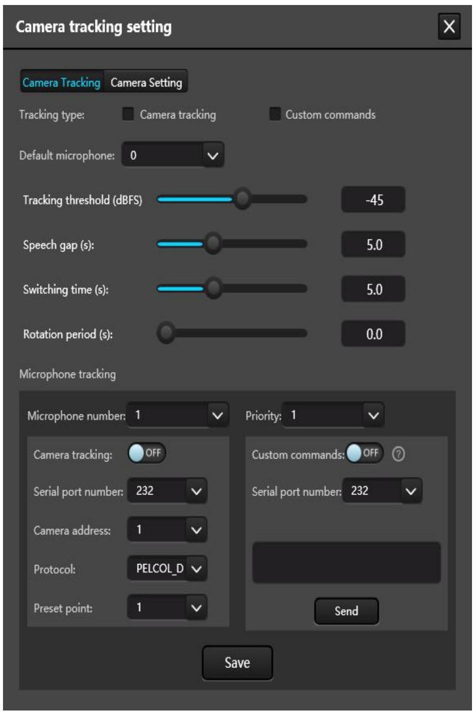
for the input GPIOs.

3. Scene, Volume, Channel mute, System mute are available for the output GPIOs.

8.20.Serial port setting

Support baud rate, data bit, stop bit and parity bit settings for 485 and 232 serial ports.

8.21.Camera tracking settings



Support camera tracking and camera settings. It is mainly based on the voice amplitude to determine whether someone is talking on the microphone.

Digital Audio Processor

Version 0.1