Preface

Thank you for purchasing our products. Please read this manual first to familiarize yourself with the product.

Note: This manual provides relevant information for all models in this series. Since different models have different configurations, the actual configuration and functions of the product you purchased are covered in this manual.

Important Safety Instructions

1. Please read these instructions carefully.
2. SAVE THESE INSTRUCTIONS.
3. Heed all warnings.
4. Follow all instructions.
5. Do not use this product near water or in places with high humidity.
6. Clean only with a dry cloth.
7. Do not block the ventilation holes, and leave at least 50mm of ventilation space for each of the left and right ventilation holes.
8. Do not install near any heat sources such as radiators, heat registers, stoves, or other products (including amplifiers) that produce heat.
9. Do not defeat the safety purpose of the grounding plug. A grounding plug has two blades and a third grounding prong. The third prong is provided for your safety. If the provided plug does not fit into your outlet, have an electrician replace the outlet.
10. Protect the power cord from being walked on or pinched, particularly at plugs, electrical outlets, and around the product where they enter and exit. Never pull, rip, or forcefully twist the power cord.
11. Use only attachments/accessories specified by the manufacturer.
12. Use only with cabinets, racks, or equipment tables designed to provide adequate mechanical strength, heat dissipation, and stability to the building structure.
13. Unplug the power cord of this product when there is lightning outdoors or when it is not used for a long period of time.
14. Servicing is necessary when the product has been damaged in any way, such as when the power cord or plug has been damaged, when it has been contaminated with liquid or foreign objects have fallen into the

product, when it has been exposed to rain or moisture, when it does not operate properly, or when it has been dropped and damaged. Please contact after-sales service personnel for all repair services.

Graphic symbol explanation:

Lightning: Hazardous live voltages are present during operation of this equipment. Do not touch terminals marked with this symbol while the equipment is connected to a power source.

Exclamation mark: Components (e.g. fuses) may only be replaced in accordance with the technical parameters specified by the manufacturer. Otherwise, the operational safety of the device may be impaired.

Hazardous Moving Fan Blades: Disconnect power before servicing and keep clear of moving fan blades.

WARNING - To reduce the risk of fire or lightning shock, do not expose this product to rain or moisture or to dripping or splashing. No objects filled with liquids, such as vases, should be placed on the product.

Warning- The product powered by the mains adopts the safety grounding technology and must be connected to a properly grounded mains socket to provide safety grounding protection.

Disconnect device - The AC mains plug or appliance coupler is the AC mains disconnect device and must remain readily accessible after installation.

Caution - Unless you are a professional technician, you cannot perform any service that is not included in the operating instructions. Before opening the product, the professional must disconnect the AC mains power supply.

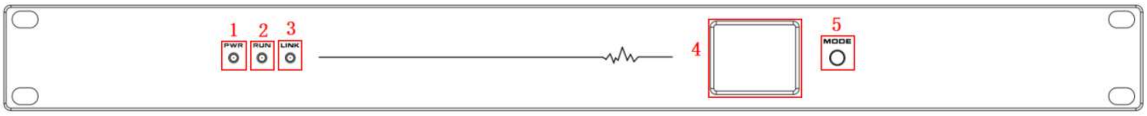
CAUTION - Any procedures regarding physical equipment installation and removal are for qualified personnel only and must comply with all local regulations.

Table of contents

1. Product Introduction

1.1 Device Interface

Device front panel introduction:



(1) PWR: Power indicator light. When the power is turned on, the light is on, indicating that the device is powered normally. The light is off, indicating that there is an abnormality in the power supply of the device.

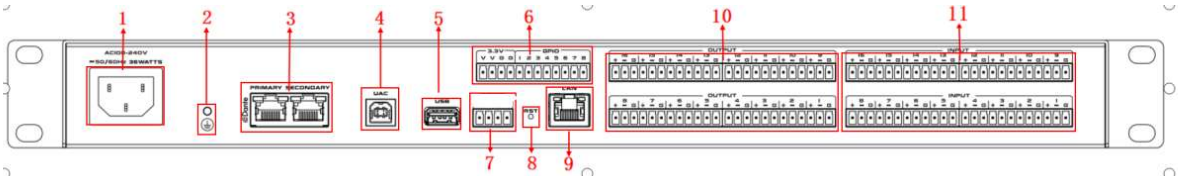
(2) RUN: Status indicator light, flashing indicates normal operation;

(3) LINK: When it is always on, it means the device is started up successfully;

(4) LED display screen: displays current device information, such as device IP information , device name, software version, current scene , device input and output channel mute status and level display (A represents analog channel, D represents Dante digital channel, and U represents USB channel) ;

(5) MODEL button: switch display information button;

Device rear panel interface:



(1) Working power supply: AC100V ~ 240V, 50Hz/60Hz ;

(2) Threaded grounding port: M3 ;

(3) Dante: In switching mode, connect any of the PRIMARY/SECONDARY network ports; in redundant mode,

The PRIMARY/SECONDARY network ports are distinguished by their names and connected to different switches;

(4) UAC: The sound card has 2 input and 2 output channels, which transmit the two-channel sound source to the audio processor through the computer.

You can also output other input channel signals to the computer for recording;

(5) USB: External USB interface, supports FAT16, FAT32 file systems, supports the playback of MP3 and WAV file formats;

(6) GPIO: input or output defined by software, receiving external signal control and controlling external devices, with DC3.3V power supply and ground port;

(7) Control serial port : bidirectional communication interface, receiving external central control command control and controlling external devices through central control command, supporting camera

tracking , with DC5V power interface ;

(8) RESET: Restore factory settings button . Method: When the device is online, press and hold the RESET button for 5 seconds and then release it . The device will automatically restart and all parameters will be restored to factory defaults .

(9) LAN: Network interface, connected to PC, online editing and command sending and receiving control;

(10) OUTPUT: Analog audio output interface, balanced Phoenix port;

(11) INPUT: Analog audio MIC\LINE input interface, balanced Phoenix port;

1.2 Software Interface

This series of products uses a B/S architecture and controls and manages the device through WEB login. The device supports login through browsers such as Google , Firefox Edge, 360 , and Kylin . Secondly, at least one of the IP addresses on the PC side must be in the same network segment as the device IP address (the factory default IP address of the device is " 192.168.20.100 ") , otherwise it cannot connect normally.

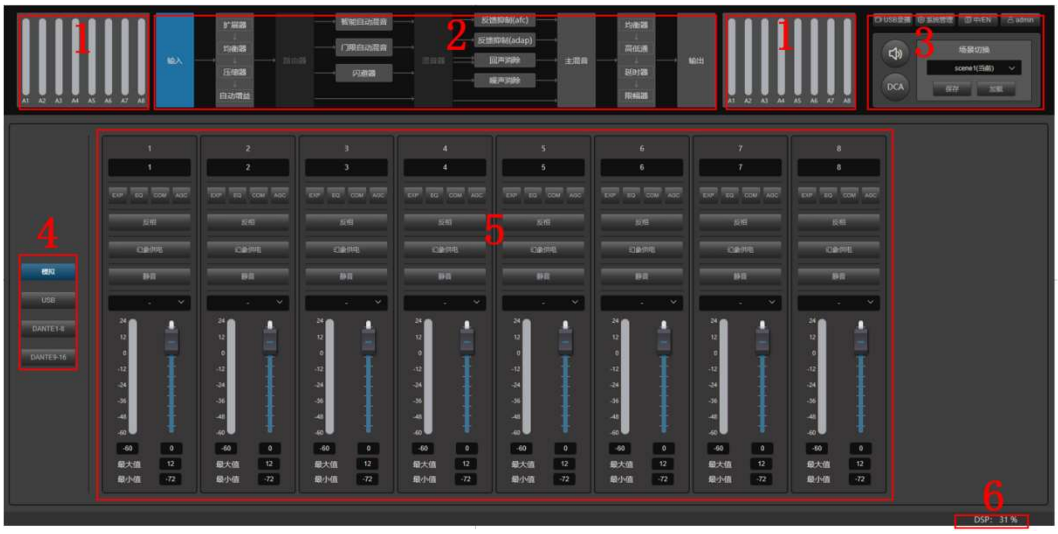
Enter 192.168.20.100:8080 in your browser .

Admin administrator (PC side) login: User name: admin, Password: admin,

User login (PC and mobile tablet) : Username: user , Password: user ,

You can switch the language selection control interface by logging in (supports Simplified Chinese/English), as shown in the figure:

After successful login, the initial control interface is as shown below :



(1) Level display area: The left side is the input channel level display area, and the right side is the output level display area. You can slide the mouse or roll the mouse wheel to view the real-time levels of all channels.

(2) Signal processing and algorithm navigation area: front-end and rear-end signal processing and algorithm navigation area . Clicking a module will immediately switch to the setting interface of the corresponding algorithm .

(3) System function area: The system function area includes USB recording and playback function, system management, Chinese and English control interface switching, and user parameter settings. Click the above buttons to jump to the setting interface of specific functions; this area also includes shortcut keys for the following functions:

System mute: control all output channels to mute with one button;

Scene: switch scenes and save and load;

DAC grouping: Input and output channels support group control function, multiple channels can be set to one group, when the gain of the group is adjusted, the gain of the subordinate channels of the group will change at the same time, after the group is set, the group logo will appear on the gain fader. DCA grouping, adjust the percentage of gain.

(4) Channel type switching area: including analog channels, USB (UAC sound card, USB recording and broadcasting) channels, and Dante channels;

(5) Channel parameter control area : displays the parameters of each functional module of each channel;

(6) DSP occupancy rate display area: displays the DSP occupancy rate of the current device in real time.

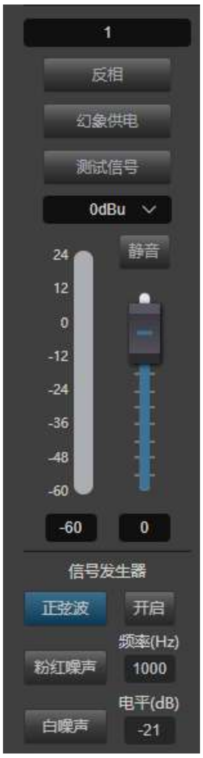
2. Audio Processing

2.1 Input Module

1. The edit box above displays the channel name and can be modified.
2. Quick keys for setting parameters of the pre-processing module.
3. Invert: Press this button to invert the phase of the input signal by 180 degrees.
4. Phantom power: Provides DC48V phantom power to the microphone for powering condenser microphones . Do not turn on line input or non-condenser microphones to prevent burning.
5. Mute: Pressing it means that the channel is muted, which is equivalent to shielding the input signal. Popping it up means that the channel signal is not muted.
6. Level display and input gain adjustment fader.
7. Max/Min: Set the gain range for the channel fader.
8. Grouping: Set any number of input channels into a group. After setting, adjust the volume of any channel in the group, and the faders in the group will move synchronously.

2.2 Pre-processing module

2.2.1 Input Source



of microphone input and line input can be selected . High sensitivity means that the microphone has high efficiency in sound-to-electricity conversion and is sensitive to weak sound signals.

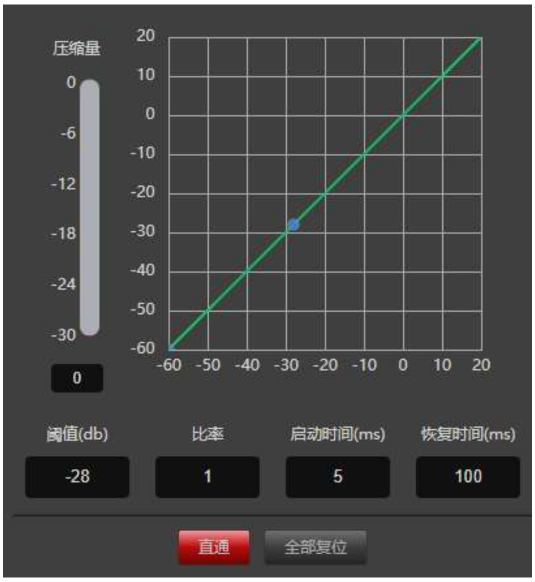
Mute: Pressing it means that the channel is muted, which is equivalent to shielding the input signal; popping it up means that the channel signal is not muted.

Invert: Reverses the phase of the input signal by 180 degrees.

Phantom power: It can provide phantom power to the microphone , which is used to power condenser microphones . Do not turn on line input or non-condenser microphones to prevent burning.

Test signal: including sine wave, pink noise and white noise. When pressed, the parameters of the signal generator can be adjusted and the signal of the signal generator can be connected to the channel. Other input signals will no longer be output.

2.2.2 Expander



The expander increases the dynamic range of the input according to user needs.

When the input signal is less than the "threshold", the expander will reduce the input signal at the set "ratio"; when the input signal is greater than the "threshold", it will be output at a 1:1 ratio, with the output level = input level; when the ratio is adjusted to the maximum (∞), the expander becomes a noise gate.

Bypass: Bypasses the input signal and sends it directly to the next processing module. When Bypass is pressed (grey), the expander is enabled. When Bypass is popped up (red), it is disabled.

Threshold: The level value at which the expander function is enabled. When the signal is less than this limit value, the expander processing mod ule is started and the signal is expanded and output ; when the signa l is greater than this limit value, the expander processing module is not started and the input signal is directly output .

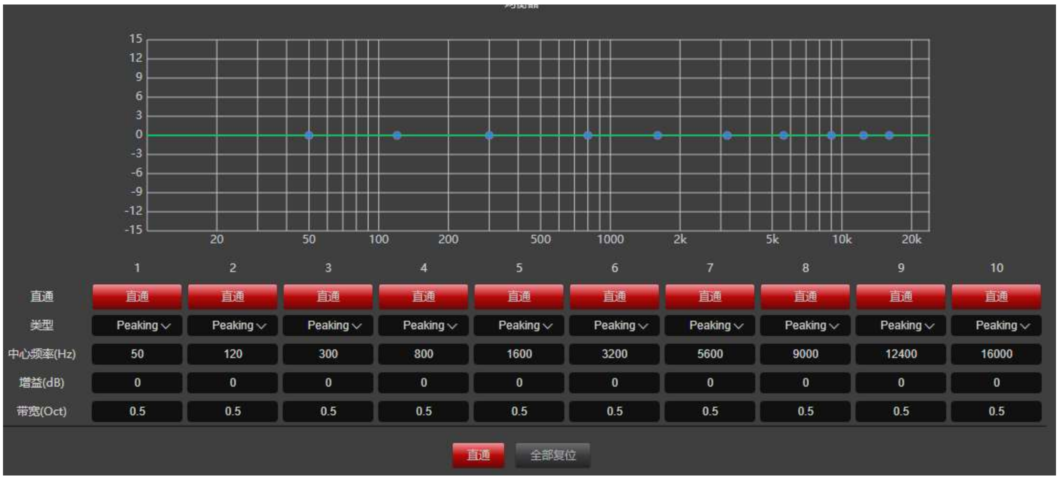
Ratio: The number of decibels of dynamic changes in the expander input signal / the number of decibels of dynamic changes in the expander output signal.

Attack time: The time required for an input signal less than the expander's "threshold" to enter the expanded state and be output at the set expansion ratio.

Recovery Time: The time required for an input signal to return from an expanded state to its original non-expanded state.

Reset All: Reset all parameters to factory settings.

2.2.3 Equalizer



The equalizer compensates and corrects the frequency characteristics to achieve a relatively flat frequency response characteristic. Different models have different equalizer types, and the specific type is subject to the actual model of the device.

Band pass-through: After making EQ adjustments, you can cancel the bypass of certain set frequency bands, which will only affect that band.

Center frequency: The frequency that needs to be equalized.

Gain: Gain/attenuation value at the frequency center point. When this value is 0, the center frequency and Oct value are invalid.

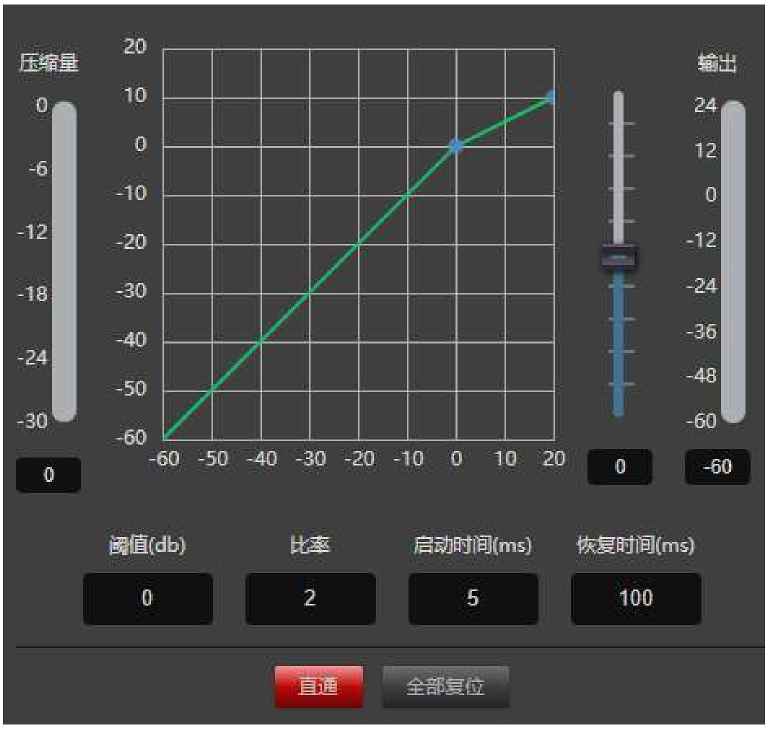
Bandwidth: The bandwidth of the frequency point, that is, the influence range of this segment around the center frequency. The larger the value, the larger the bandwidth and the larger the influence range.

Type : The equalizer has six built-in filter types: Peak, Notch, H-shelf, L-shelf, HP, and LP.

Direct pass: bypass all frequency band signals, and the signals go directly to the next processing module; popping up the direct pass (red) means that the direct pass is effective, that is, the equalizer is not activated, and all frequency band signals are directly passed to the next processor; pressing the direct pass (gray) means that the equalizer is activated, and the signal is processed by the equalizer before being output

Reset All: Reset all parameters to factory settings.

2.2.4 Compressor



The compressor compresses the signal greater than the threshold value at a predetermined ratio and outputs it .

When the input level is greater than the preset threshold, the output level is compressed according to the set ratio, and the output signal = threshold + (input signal - threshold) / ratio; when the input level is less than the preset threshold , the signal is output directly, and the output signal = input signal; when the ratio is adjusted to ″ +∞″ , the compressor at this time becomes a limiter.

Bypass: Bypass the input signal without processing.

Threshold: The starting level of compression. When the signal is higher than this limit value, the compression processing module is started to compress the signal greater than this value; when the signal is less than this limit value, the compression processing module is not started and the input signal is directly output.

Ratio: The compression ratio of the signal above the threshold.

Attack Time: The time from when the input level reaches the threshold to when the compressor starts to compress .

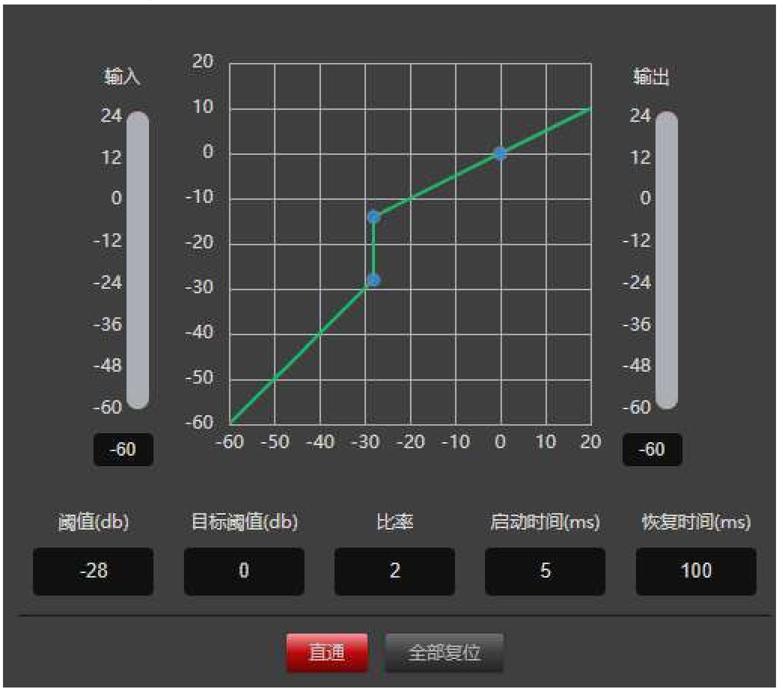
Release Time: The time from when the input level drops below the threshold until the compressor stops working completely.

Output: Adjusts the gain of the output level and the level actually compres

sed after the compressor processing.
Reset All: Reset all parameters to factory settings.

2.2.5 Automatic Gain

First, let's explain the role of automatic gain. For example, when a user speaks to a microphone, the distance between the mouth and the microphone may vary, causing the output volume to fluctuate, or even make the speech sound intermittent. Automatic gain is to set a threshold, output the input signal below the threshold at a 1:1 ratio, and directly increase the level above the threshold according to the ratio. Once the target level is set, the sound signal can be output stably.



Automatic gain control automatically controls the gain amplitude by changing the input-output compression ratio. When a weak signal is input, the signal is amplified to ensure the strength of the output sound signal; when the input signal strength reaches a certain level, the signal is compressed to reduce the sound output amplitude.

Input: Input level to the AGC.

Output: Output level of the automatic gain controller.

Threshold: The starting level of automatic gain. When the signal is lower than this limit value, the automatic gain processing module will be started to gain the signal that is too small.

Target threshold: The starting level of automatic gain. When the signal is lower than this limit value, the automatic gain processing module will be started to gain the signal that is too small.

Ratio: When the signal reaches the threshold starting level, the signal is gained according to the ratio.

Attack Time: The time from when the input level reaches the threshold to when the automatic gain starts .

Recovery time: The time required for the input signal to return from the automatic gain state to the original non-automatic gain state.

Bypass: Bypass the input signal without processing.

Reset All: Reset all parameters to factory settings.

twenty three Intelligent automatic mixing

Intelligent automatic mixing is an algorithm designed based on network audio automatic gain attenuation control of large conference room microphones, but it is not limited to network audio applications. It is suitable for almost any mixer, news broadcast, talk show, auditorium, film and television Dialogue recording or any discussion, meeting scenario involving multiple microphones and participants. This algorithm mainly involves microphone active speech processing technology based on sound field balance, which can automatically control the gain of multiple microphones in real time, thereby significantly reducing feedback, environmental noise, and comb filtering effects produced by adjacent microphones. Even when multiple speakers speak at the same time, it can maintain consistent system gain and achieve perfect cross-fade transitions. The gain connection is smooth and traceless, and there is no unnaturalness that may be caused by an expander. noise reduction effect.

(1) Common module:

1. On: The red state bypasses the input signal and does not process it.
2. Reset: Reset all parameters to factory settings .
3. Level search: channel level detection index. (Dante models have "analog level search" and "dante level search" switching modes)
4. Output level and automatic gain: This button can switch the output and gain value display.

(2) Channel control

Input and output level meters: Each channel has 2 level meters , which display the level of the signal before entering the algorithm and after passing through the algorithm in real time .

Weight fader: Adjust the weight fader of the incoming channel. The intelligent mixing algorithm can assign different output gains according to different weights. The adjustment range of weight is -30 ~ 15 .

Automatic : Indicates that the signal of this channel has been added to the intelligent mixing algorithm for processing.

Mute: Indicates that the channel signal does not participate in the intelligent mixing algorithm.

Direct: Bypass the input signal and output it directly without automatic mixing algorithm processing .

(2) On : The intelligent automatic mixing algorithm is enabled .

(3) Reset : Restore all adjusted parameters to factory settings.

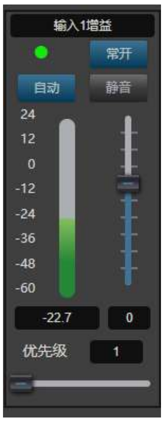
(4) Output : Click the output button and the button will change to gain state. You can view the real-time gain status of each channel .

2. 4 -threshold automatic mixing

The Gain Sharing Automixer has fast, smooth sounding operation and requires little adjustment and setup time. However, for some users, a gating algorithm that limits the total number of open microphones (NOM) is preferred. The Gated Automixer module is built around an adaptive threshold algorithm that is easy to set up and does not require setting fixed thresholds.

The Gated Automixer analyzes all microphone inputs and attenuates or shuts off any microphone that is not passing speech audio. Microphones that are passing speech audio are heard normally. The Gated Automixer does this by setting a dynamic threshold just above the background noise floor. Microphones below this threshold are gated. If a qualified speech signal exceeds the threshold, the channel gate opens and the signal passes. The threshold is updated in real time if the background noise floor changes.

The Gated Auto-Mixer uses an adaptive noise threshold for each channel. A channel’ s signal must exceed the threshold by a programmable amount. Therefore, a channel will not appear in the presence of stationary noise, such as equipment fans or HVAC . In addition, additional module functionality prevents multiple microphones from gating when a single person is speaking. As more channels come in, the overall
The total NOM level is reduced to prevent feedback. This is called NOM reduction and is also programmable. The total NOM may also be limited. When this limit is reached, a multi-level priority scheme determines which microphones can be turned on.



Indicator light: When the channel is selected and turned on by automatic mixing, the indicator light is on.
Normally open: When the normally open button is selected, the channel does not participate in the automatic mixing algorithm, nor does it affect the automatic mixing algorithm. It is only mixed and output together with the normal microphone sound and the microphone after automatic mixing.

Level meter: Each channel has 1 level meter, which is the pre-auto gain level value. The -60dBu level meter shows the level of the input signal, which is the level value before the fader and auto gain.

Auto: Each channel has an "Auto" button. Press it to add the channel to the automatic mix.

If you want to set a background music channel to a fixed level, or keep a "chairman" microphone always on, the automatic key needs to be selected as "pop-up". When the automatic key of a channel is selected as "pop-up", its gain will not be automatically adjusted, and the signal level of this channel will no longer affect the gain of other channels (no longer affect the automatic mixing gain calculation).

Mute: Channel muting is pre-auto gain. That is, turning on mute eliminates the effect of that channel on the gain of other channels. It mutes the channel in the mix and mutes the direct output of that channel.

Faders: Channel faders are auto-gain. They control the mix level and direct out level for that channel, but do not affect the automixer algorithm. You can also precisely control the channel level by clicking in the text box and entering a dB value. Channel mutes and faders are auto-gain, meaning that no matter what you do, it has no effect on the automix. This means that even if a channel is muted, the levels of the remaining channels will be reduced if a high level signal is present on that channel. To mute a signal without affecting the automix, turn Mute on and uncheck Auto.

Priority PR: Allows channels with higher priority to override channels with lower priority, thus influencing the automix algorithm. This control defines the priority as a value between 1 (lowest priority) and 10 (highest priority). You can adjust the priority by either using the slider control or clicking on the edit box . If two channels are receiving signals of the same level, the one with the higher priority will have a higher gain. For every 1 priority "unit" difference between the channels, the gain will differ by 2dB (assuming the priority step value is set to 2.0). For example, if channel 1 is set to priority 6 and channel 2 is set to priority 3, when both channels are receiving signals of the same level, channel 1 will have a 6dB higher gain than channel 2. Obviously, the priority step value setting affects the weight of the priorities. If all channels have equal priority, set the priority of all channels to 5.

Enable Last Mic: Pressing the Enable Last Mic button ensures that the last microphone remains on, even if no speech is detected. This guarantees that at least one microphone will always remain on to prevent dead air.

Mute and Gain: The Master section includes master gain and mute settings. These only affect the main mix output, not the channel direct outputs.

Bypass: Mutually exclusive with the Bypass button of the gain sharing automatic mixer, that is, only one algorithm can participate in them.

The Hold slider adjusts how long the channel remains open after the microphone stops speaking . Use a time long enough so that brief pauses between words and sentences do not cause the channel to close. A longer hold time often helps if excessive chatter (frequent opening and closing) occurs. However, a hold time that is too long can prevent the next talker from starting to pick up when the NOM count is reached.

Threshold: Sets the level at which all channels are turned off. A higher setting (e.g. -20dB) may be useful to ensure that some room sound is present when all mics are off, if Enable Last Mic is not used. Muted channels will be fully attenuated regardless of this setting.

Noise Sensitivity: The Sensitivity slider controls the adaptive noise threshold that a channel must be above to be gated. If a channel gates quiet noise too easily, increase this setting. In our experience, noisy rooms (e.g. with a lot of fans) require lower settings than quiet rooms. This is because in a quiet room, even low-level sounds can be much louder than the noise.

Priority Step Value: The Priority Step Value slider affects the amount of attenuation on all channels as more microphones are turned on. This parameter represents the attenuation when two microphones are gated, plus the additional attenuation for each doubling of open microphones.

Open mic number: Set the number of open mics

Last microphone number: Displays the last microphone number that was turned on.

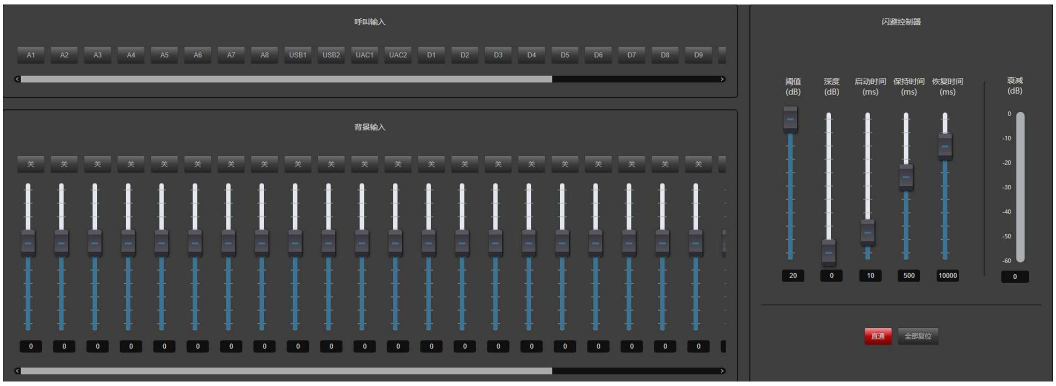
2. 5 Dodgers

The main function of the ducker module is to reduce the signal of the channel next to it when a certain channel is selected as the ducking signal. Effect, mainly used in conference rooms and other occasions.

If the voice input of the rostrum is selected as the control signal, when the main speaker speaks, the volume output of other speakers will be reduced, so that the effect of only one speaker is achieved;

For example, a shopping mall will play background music during normal business hours, but when there is a notification or reminder that needs to be played, the music will be turned on.

It automatically turns off when vocals are inserted, and automatically restores after the vocals disappear. This is when the ducker is working.



In the input channel "On" / "Off" button shown in the figure above (the number of channels is the same as the number of physical input channels of the device), select the input channel that needs to be ducked "On" (that is, the background music signal channel mentioned above).

In the call input as shown in the figure, select the input channel as the ducking signal (i.e. the MIC voice

insert into the signal channel).

Threshold: The starting level of ducking. When the signal is higher than this limit, the ducking processing module will be activated.

signal to dodge.

Depth: This value can be adjusted to reduce the gain once the signal of the adjacent signal exceeds the set threshold.

Attack Time: The time it takes for the gain to be reduced when the signal in the adjacent channel exceeds the threshold.

Hold Time: When the gain is reduced, the time the setting condition is maintained.

Release time: When the input signal is lower than this setting value, the sound channel will not be closed immediately, but will be closed according to this setting.

The value delays the closing time. During this time, as long as the signal is higher than the "Thd" limit value, the sound channel can be kept open.

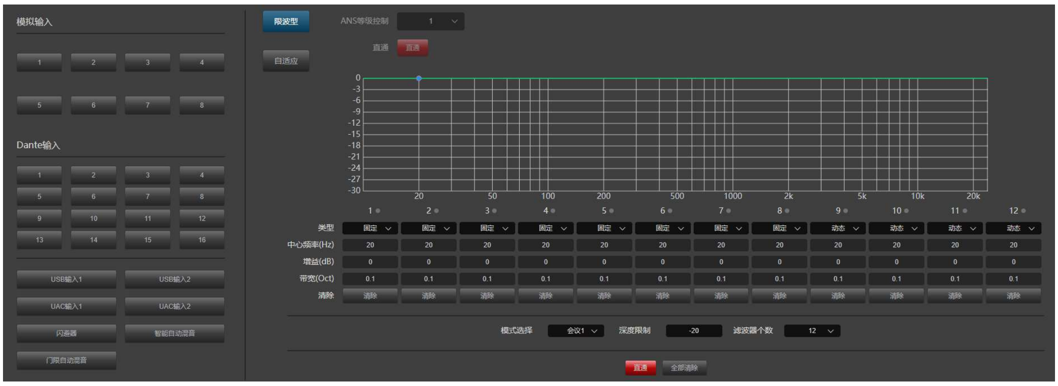
Attenuation: The output level after being processed by this module for users to view.

Pass Through: Bypass the input signal without ducking.

Reset All: Reset all parameters to factory settings.

2.6 Feedback Elimination

Feedback eliminator, automatically detects and suppresses acoustic feedback in audio systems, can distinguish feedback from audio, and when feedback is detected, adds a notch filter or adaptive filter to attenuate the frequency of the feedback.



Mode selection: There are four modes to choose from: Conference 1, Conference 2, Concert 1, and Concert 2. The detection sensitivities of the four modes are different, and the notch gain and bandwidth values of the notch filter are different.

Analog Input: Route the analog input signal to the feedback cancellation module.

USB input: Route the USB input signal to the feedback elimination module.

Ducker: Indicates that the signal processed by the ducker enters the AFC module.

Intelligent automatic mixing: Indicates that the signal processed by the intelligent automatic mixing module enters the AFC module.

Threshold automatic mixing: indicates that the signal processed by the threshold automatic mixing module enters the AFC module.

Number of filters: The default is 12 filters.

Depth limit: The maximum depth limit of the filter notch, adjustable from 0dB to -20dB.

Clear: Clear the data of a single filter. Center Frequency: Displays the frequency value of the feedback howling point.

Gain: Displays the gain value of the filter notch.

Bandwidth: Displays the bandwidth value of the filter notch.

Type: Each filter has three types to choose from: **fixed**, **dynamic**, and **manual**.

Fixed and **dynamic** are automatically detected and displayed, and manual is a user-manual parameter setting. Fixed points cannot be modified or overwritten after setting until they are intentionally cleared; dynamic points cannot be modified, but when the filter is fully used and feedback is still occurring, the new howling point will overwrite the original point from the first dynamic point in sequence. When the manual type is selected, the user can modify the values of the filter frequency, gain, and bandwidth.

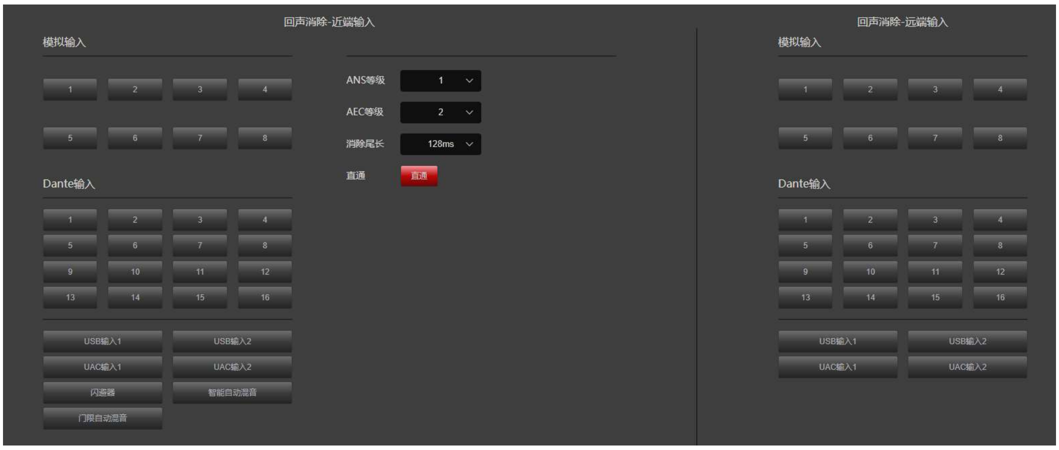
Bypass: bypasses all input signals without processing. Bypass (red) is disabled, and Bypass (grey) is enabled.

Clear All: Clear the data of all filters.

ANS Level: There are 0-6 levels to choose from (only adaptive feedback cancellation has this option)

2. 7 Echo Cancellation

Acoustic echo is caused by the sound of the speaker being fed back to the microphone multiple times in hands-free or conferencing applications (the microphone collects the sound from the speaker as well as the near-end sound). Some echo is necessary, but most echoes have a negative impact and reduce the quality of communication.



Set the signal that needs to be processed by the echo canceller. The processed signal can be output in "Source" of the output part.

Near-end input: local MIC input, that is, the signal that needs echo processing.

Remote input: reference signal.

USB input: Indicates that the USB input signal is selected to enter the echo cancellation module.

Ducker: Indicates that the signal processed by the ducker enters the echo cancellation module.

Automatic mixing: Indicates that the signal processed by the automatic mixing module enters the echo cancellation module.

Threshold automatic mixing: indicates that the signal processed by the threshold automatic mixing module enters the ANS module.

Control level: There are 6 levels to control, level 1, the noise suppression amount is -3dB, 2/3/4/5/6 each level decreases by 3dB, the maximum suppression amount is -18dB.

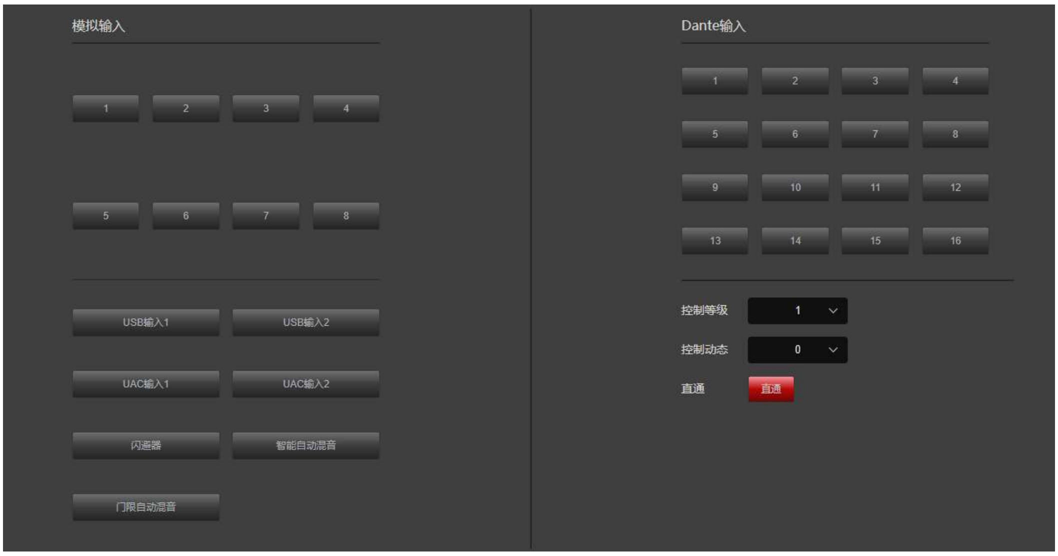
ANS level: There are 4 controllable levels. Level 1 noise suppression is -9dB, and each level 2, 3, and 4 decreases by 3dB, with a maximum suppression of -18dB.

AEC level: includes 3 echo attenuation levels. The larger the value, the higher the echo attenuation level.

Direct: Indicates that the signal goes directly to the next module without echo cancellation processing.

2.8 Noise Suppression

The purpose of the noise suppressor is to eliminate the steady-state environmental noise picked up by the microphone, such as ventilation systems, air conditioners, fans, motors or other mechanical equipment. There are 6 levels of noise suppressors for users to choose from. We recommend using the lowest level to achieve the expected environmental noise elimination effect. For example, using level 2 to achieve the expected effect is not recommended to use level 3. The noise suppressor uses an adaptive algorithm. It is normal that the noise suppressor does not work when it is just turned on.



Input source: You can select any one channel or multiple channels of signals or their combination from the input sources to enter the noise suppression module.

Analog input: 1-8 button: analog input channel number.

USB input: routes the USB input signal to the ANS module.

Ducker: Indicates that the signal processed by the ducker enters the ANS module.

Intelligent automatic mixing: Indicates that the signal processed by the intelligent automatic mixing module enters the ANS module.

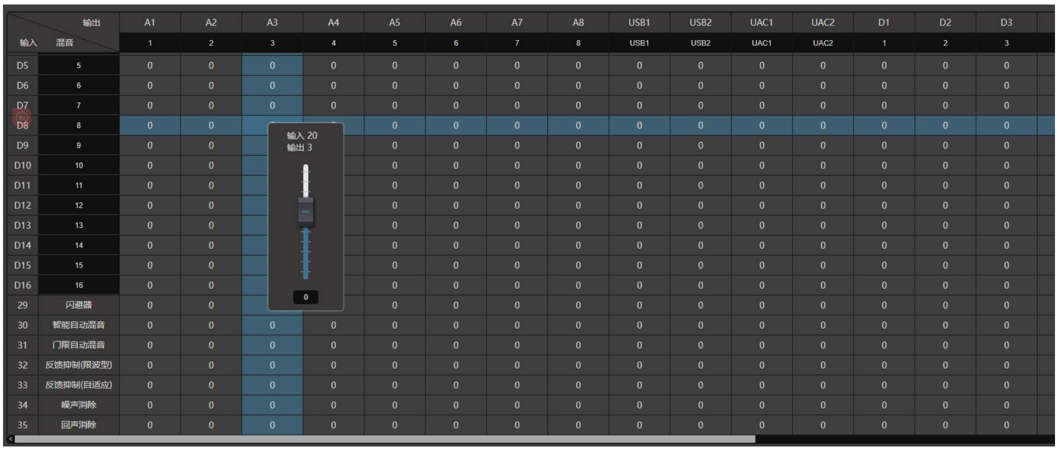
Threshold automatic mixing: indicates that the signal processed by the threshold automatic mixing module enters the ANS module.

Control level: There are 6 levels to control, level 1, the noise suppression amount is -3dB, 2/3/4/5/6 each level decreases by 3dB, the maximum suppression amount is -18dB.

Control dynamics: The dynamic level is divided into 0-15 levels, which means smoothing the real-time changing noise signal. The larger the value, the smoother the mutation noise processing.

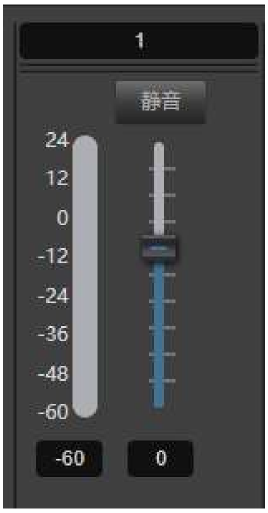
Direct: Indicates that the signal goes directly to the next module without being processed by ANS.

2.9 Main Mix



The mixing matrix intuitively displays the connection relationship between input and output channels in the form of a matrix array. If you need to output the sound of input 1 to output 1, just light up the node button in the first row and first column of the matrix. The right-click mixing button can also adjust the gain of the mixing output.

3.0 Post-processing



Mute: Pressing it means that the channel is muted, which is equivalent to shielding the output signal; popping it up means that the channel signal is not muted.

3.0.1 Speaker Manager

(1) EQ

The equalizer compensates and corrects the frequency characteristics to achieve a flatter frequency response characteristic.

Pass-through : Restore the frequency point in this band to 0 dB.

Type: The equalizer has four built-in filter types: Peak, Notch, H-shelf, and L-shelf;

Gain: Gain/attenuation at the center frequency. When this value is 0, the center frequency and Q value are invalid.

Bandwidth: The bandwidth of the frequency point, that is, the influence range of this segment around the center frequency. The larger the value, the larger the bandwidth and the larger the influence range.

(2) Frequency divider

The frequency divider is realized by high-pass filter and low-pass filter. Please make frequency division according to the actual situation when using it.

High Pass Bypass: High frequency band bypass switch.

Low Pass Bypass: Low frequency band bypass switch.

(3) Delay device

The time interval from the signal input to the signal output of the processor is generally used to produce effects such as reverberation or echo. It can also be used to process auxiliary speakers in larger occasions.

(4) Setting the number of equalizer bands

The parametric equalizer can be set to 10, 15, or 31 segments.

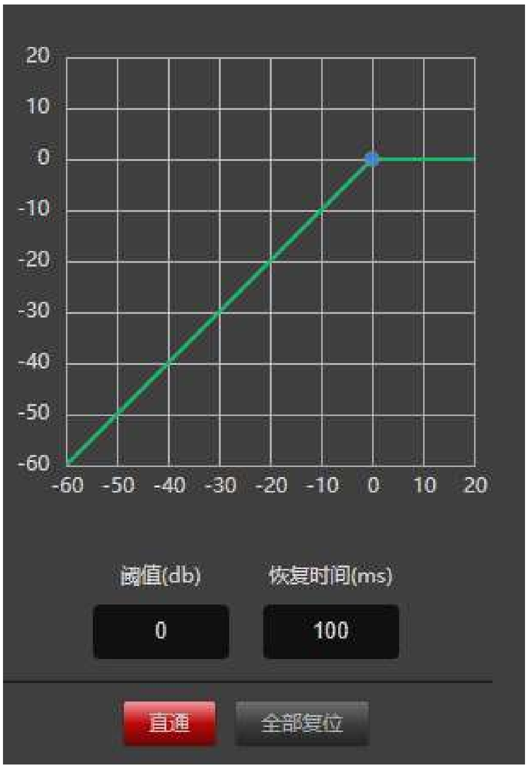
3.0.2 Limiter

The limiter performs a limiting process on the signal that is greater than the threshold. When the input signal is greater than the threshold, the output signal is equal to the threshold, and when the input signal is less than the threshold, the output signal is equal to the input signal.

Bypass: Bypass the input signal without compression.

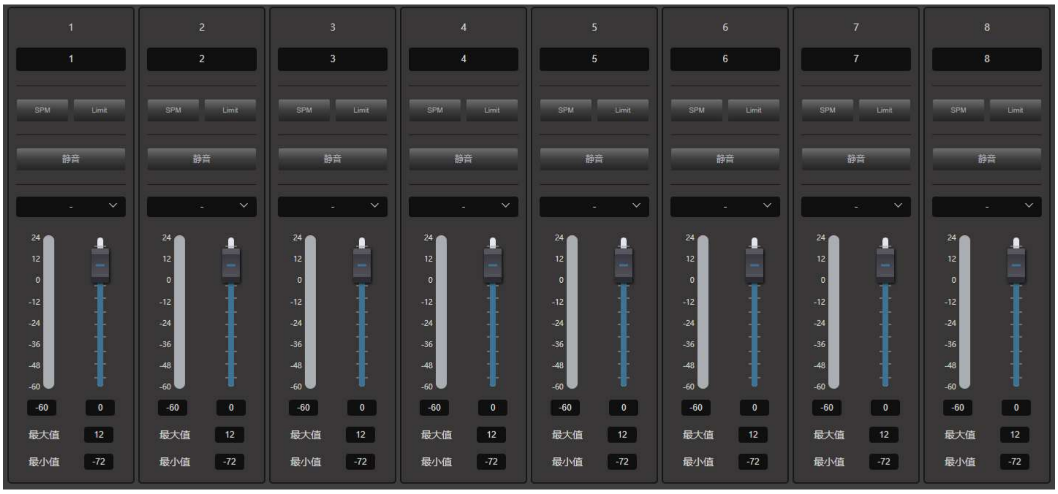
Threshold: The starting level of the limit. When the signal is higher than this limit value, the limit processing module is started to limit the excessive signal.

Recovery Time: The time required for the input signal to return from a



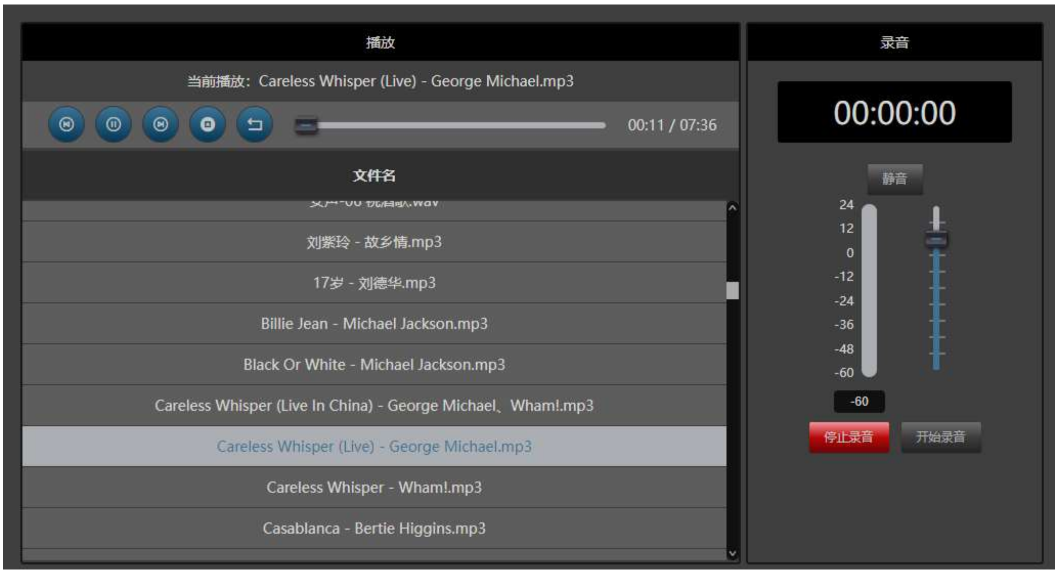
limiting state to its original non- limiting state.

3.1 Output Module



1. The edit box above displays the channel name and can be modified.
2. Shortcut keys for setting parameters of post-processing module.
3. Mute: Pressing it means that the channel is muted, which is equivalent to shielding the input signal. Popping it up means that the channel signal is not muted.
4. Level display and input gain adjustment fader.
5. Max/Min: Set the gain range for the channel fader.
6. Grouping: Set any number of output channels into a group. After setting, adjust the volume of any channel in the group, and the faders in the group will move synchronously.

3.2 USB recording and playback



The USB playback function is to connect a removable storage device (USB flash drive) for direct playback

Audio files in removable storage devices.

Support standard mp3 and wav format files

Support sampling rate (KHz): 8/11.025/12/16/22.05/24/32/44.1/48

Support FAT32 format, do not support NTFS, exFAT and other formats, support up to 64G USB flash drive (laboratory test

maximum)

Support recording while playing

Supports plugging and unplugging USB disks during recording, playing, and pausing, and supports recording to blank USB disks

Supports multiple partitions on a USB flash drive; when all partitions are in FAT32 format, only the last partition is valid; when there are multiple

When partitioning, only FAT32 partitions will be searched for files.

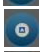
Play: After connecting the USB flash drive, the interface will automatically identify the .mp3 or .wav format files in the USB flash drive and display them in the play list.

middle.

 : Play the previous song

 : Play / Pause

 : Play next song

 : Stop playing

 : Play mode, loop play - random play - sequential play - single play

: Display playback progress

Recording: After mixing the channels to be recorded to the recording channels in the mixing matrix, click the "Start Recording" button to start recording or stop recording. Click "Pause Recording" to pause or continue recording. After recording is completed, a recording file will be generated in the root directory of the U disk. The recorded music is saved in the root directory of the U disk in the format of the file name rec- xxxx (current recording time) .wav .
Note: When using the USB playback function, in order to effectively identify the files in the USB disk, it is recommended to store
The file is in a format that the device can recognize. The recorded file is played in mono.

3.3 Scene Switching

Save: Save the current scene to the selected device scene number.

Load: Enable the currently selected scene, usually used for scene change

3. System Management

3.1 User Management

The initial user list of the device is:

type	username	password	Maximum number of connections	Can I change my password?
administrator	admin	admin	8	able
user	user	user	8	able

3.2 Environmental Control Management

3.2.1 Serial port settings

Used to view and modify the serial port information of the currently connected device. After the settings are completed, click the "Save" button to modify the serial port information of the current device;

3.2.2 Voice Tracking

A. Voice tracking settings are divided into camera tracking and custom command sending. Camera tracking is used to control the camera rotation when the microphone is speaking; custom command sending is used to send corresponding commands to the corresponding port when the microphone is speaking.
The tracking threshold means that the detected voice signal must be greater than or equal to the tracking threshold to enable the tracking parameter, otherwise no tracking will be performed.

The default microphone means that when all microphones have no input, the camera is rotated to the default MIC setting position or the default microphone definition associated command is sent. The vc in front indicates a virtual number and can only be used to set the default microphone.

B. The camera setting is a camera debugging interface. Generally, the camera position is debugged before tracking begins. Finally, the parameters of this part will be saved on the camera. First, set the serial port. The serial port number is 232, which corresponds to the backplane port connected to the PTZ. Then select the camera address and protocol type. Please refer to the actual address of the camera for the camera address. The protocol is related to the camera model. Finally, the preset position number is the user-defined identifier for the camera. Then adjust the up, down, left, right, and focal length, aperture and other parameters to define the camera position and settings.
Note: A camera address can contain multiple preset positions, but a preset position only corresponds to one camera address. Finally, click "Save" to save the parameters to the camera, "Clear" is to delete the information of the current preset point, and "Call" is used to view the camera position saved by the current preset position.

C. The microphone number generally corresponds to the input channel of the device, that is, the channel number to which the microphone is connected.

The microphone number represented by is a virtual number and can only be used to set the default microphone.

The larger the priority number, the higher the priority level. When the priorities are the same, they are processed in the order of trigger priority. For example, if two microphones are speaking at the same time, the camera automatically moves to the preset position corresponding to the microphone with a larger priority number (i.e., a higher priority level) or sends a command corresponding to the microphone with a larger priority number (i.e., a higher priority level). However, if the priorities of the two microphones are the same, the signal of whichever is detected first will prevail.

When the MIC setting is enabled, if there are multiple microphones, all microphone parameters can be set in advance, but when using, only some of them can be enabled according to the actual situation.

Preset position, serial port number, camera address, and protocol are generally related to the camera and must correspond to the actual connection of the camera. Custom commands mean that when the matrix microphone detects an input signal (usually when someone is talking), it automatically sends the corresponding command to the defined serial port. Secondly, you can also pre-set the command, but if you do not check "Enable custom commands", the device will not send it automatically, but you can still click the "Send" button to send the command in the input box to the specified serial port at any time.

Note: There are several parameters such as preset position, serial port number, camera address, and protocol in the camera settings and they need to be used according to the actual situation. Click "Save" to save the parameter information to the device. At this time, the microphone of this channel has been associated with the corresponding camera address. Then use the "Enable Microphone Settings" option to determine whether the microphone settings are valid when tracking is enabled.

Control example:

Example: Connect a camera with address 2 to the RS232 interface, and connect a microphone to input channels 1 and 8 respectively (requires that preset position 1 be associated with the microphone of channel 1, preset position 2 be associated with the microphone of channel 8, and set the camera to switch to the microphone of the corresponding input channel). The setting process is as follows:

Connect the PC and the device, open the serial port settings in the device settings, select 232 for the serial port number, select 9600 for the baud rate, leave the check bit, data bit, and stop bit unchanged, and click "Save".

A. Preset position 1 is associated with the microphone of channel 1: (1). Camera settings: select 232 for the serial port number; select 2 for the camera address; set the preset point to 1; select the correct protocol based on the actual device; adjust the up, down, left, right, focal length, aperture and other parameters to aim the camera at the microphone of input channel 1, and click the Save button to save the preset position;

B. (2) Preset control settings: Select 1 for microphone number; Select 2 for camera address; Select the correct protocol based on the actual device; Select 232 for serial port number; Select 1 for preset point setting. Click to enable this microphone setting; Click OK. B. Preset 2 is associated with the microphone of channel 8: (1). Camera settings: Select 232 for serial port number; Select 2 for camera address; Set the preset point to 2; Select the correct protocol based on the actual device; Adjust the up, down, left, right, focal length, aperture and other parameters to aim the camera at the microphone of input channel 8, and click the Save button to save the preset; (2). Preset control settings: Select 8 for microphone number; Select 2 for camera address; Select the correct protocol based on the actual device; Select 232 for serial port number; Select 2 for preset point setting. Click to enable this microphone setting; Click OK. After the two microphones are associated with the camera, select Enable tracking and set the tracking threshold to achieve voice tracking control.

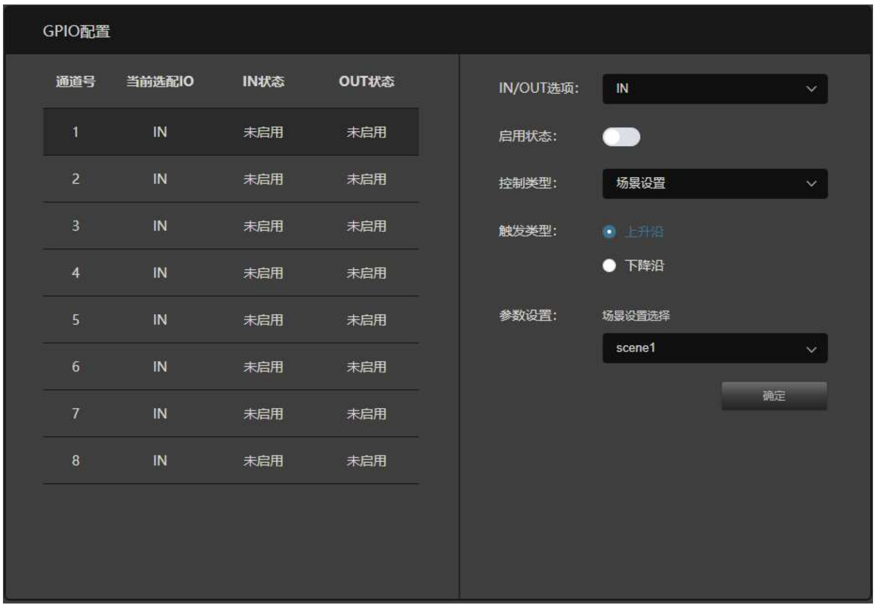
3.2.3 Scene Management

Scene management meets customers' demands for multi-scene applications, including functions such as saving scenes, loading scenes, importing scenes, and exporting scenes, achieving the effect of one-click scene switching, which greatly facilitates customer use. Scene import and export can not only serve as a backup but also quickly build a second system, making debugging and maintenance easy and convenient. Default startup item: Set the current scene to the scene after startup.

Save: Save the current scene to the selected device scene number.
Load: Enable the currently selected scene, usually used for scene change.
Reset: Restore the selected scene configuration to the default configuration.
Select file: Select the file to upload.
Upload: Click to select the file, then click to upload the selected file.
Export: Save the currently selected device scene to the PC.

3.2. 4 GPIO

The 8 GPIO channels on the rear panel of the device can be defined as channel input or output through the following configuration interface.



A. GPIO output settings

Internal changes in the audio matrix --- →GPIO pin level changes --- →drive external circuits

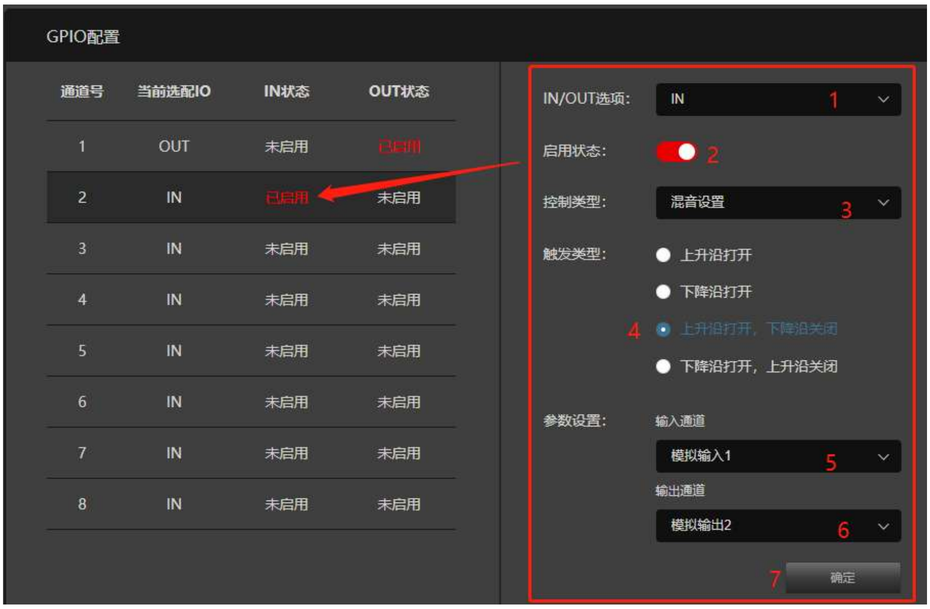
For example: Use GPIO 1 channel, processor input channel 1 is muted, outputs high level, non-muted outputs low level. The configuration is shown in the figure below:

GPIO配置
通道号
当前选配IO
IN状态
OUT状态
1
OUT
未启用
已启用
2
IN
未启用
未启用
3
IN
未启用
未启用
4
IN
未启用
未启用
5
IN
未启用
未启用
6
IN
未启用
未启用
7
IN
未启用
未启用
8
IN
未启用
未启用
IN/OUT选项:
OUT
1
启用状态:
2
显示类型:
通道静音显示
3
输出电平类型:
输出低电平，无输出高电平
4
输出高电平，无输出低电平
参数设置:
输入/输出选择
5
输入
输出
通道选择
6
模拟输入1
7
确定

B. GPIO input settings

External circuit status changes --- →GPIO pin level changes --- →audio matrix internal changes

For example: Use GPIO 2 channel, external input high level, control the audio processor mixing input channel 1 to route to output channel 2; external input low level, control the processor input channel 1 to route to output channel 2 mixing. The configuration is shown in the figure below:



3.3 System Configuration

3.3.1 System Information

Displays system configuration information. The device name can be modified.

3.3.2 Network Configuration

You can choose manual or automatic acquisition mode, and set IP, mask, gateway, and DNS server.

3.3.3 System Maintenance

System upgrade: first select the required upgrade file, then click Upgrade.

To modify the theme and customize the icon: Select the modified ZIP compressed file and click Upload. After the upload is successful, the interface theme and LOGO will take effect immediately after refreshing the interface, and the device model information will take effect only after restarting the device.

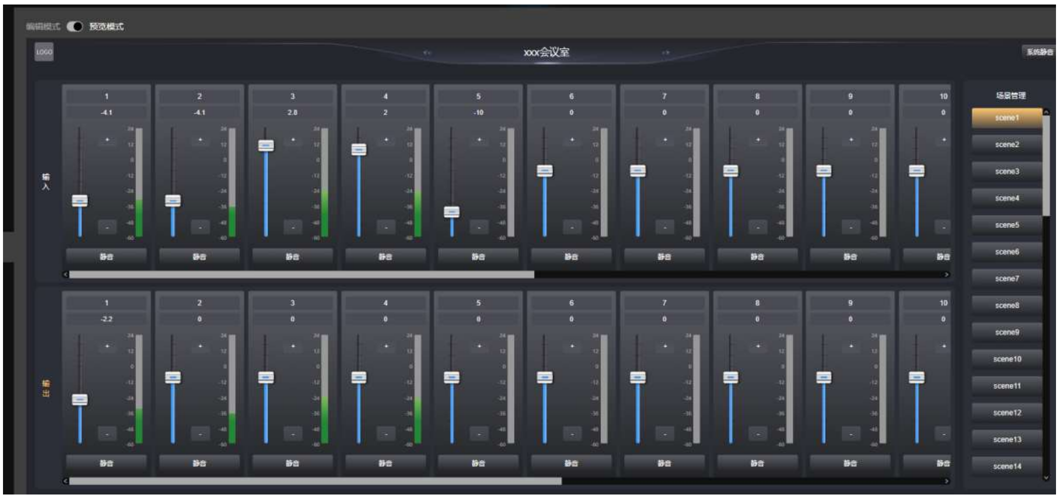
Restore factory settings: You can restore the device to factory default settings.

Check "Restore to default IP" and click "Restore factory settings" to restore the device to factory settings including the device IP.

3.3.4 Resource List

Resource list: stores help documents and central control documents.

3.3.5 Custom interface



The custom interface is mainly used to design the user login interface, and supports running tablets: Android, Hongmeng, IOS system; WIN-PC, Kylin system defense version. Through all displayed controls, control the gain, mute, system mute of the device input and output channels, save and call of scenes, and channel routing.

Edit Mode & Preview Mode:

1. Edit mode: Design the user's control interface.
2. Preview mode: The control interface display is consistent with that of different operating systems. You can adjust the volume of the fader, turn the mute button on and off, call scenes, save and route the matrix, etc.

Level display & control:

1. Right-click the button and a pop-up window will appear with hide/show options, which can set the channel to hide or show;
2. Right-click the button and a pop-up window will appear with hide/show options. You can set the volume adjustment method, fader or volume up and

down buttons or a combination of the two;

Mixing Matrix:

1. Right-click a row or column to have a hide/show option, and edit whether the input or output channel should hide/show the entire row of the array;
2. The row and column hidden intersections are in disabled mode, and the editable button is operable. If it is not operable, it is grayed out;

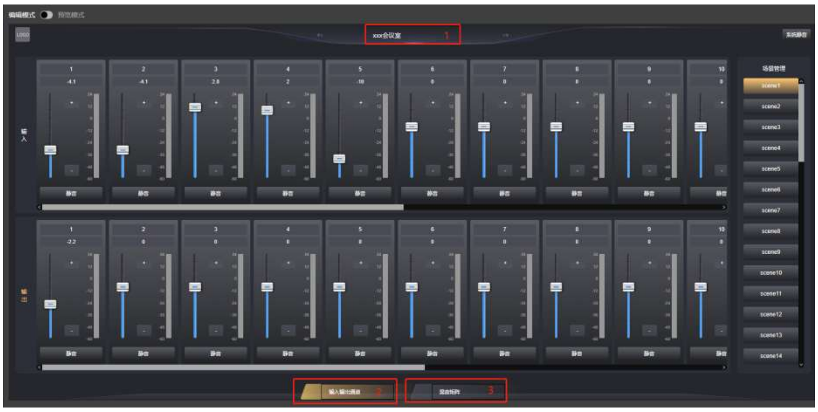
Scene saving and calling:

1. Right-click the button and a pop-up window will appear with the option to hide/show the button;
2. Click the scene button to pop up the save/call selection page and select the execution operation;
3. The current scene button is highlighted;

LOGO theme:

1. In the editing state, click LOGO to insert the picture and display it in the specified area;
2. Image size and format: (file size: less than 500k; image format: svg; resolution 1:1);

The editable part of the client interface name is as follows:



3.3.6 Interface Configuration

Custom profile management:

Save the parameters configured in the edited custom interface as a file, which is mainly used to restore device configuration parameters. For example, import parameters of different

devices in the same usage scenario.

User client control interface base map management: