Digital Audio Processor Help Manual

Notes:
This manual uses the Full-Featured Audio Processor Matrix as an example of how to use it, and can be used as a reference for other processor models.
This manual is intended as a user operating instruction only and is not intended for maintenance service purposes.

catalogue

Guidelines for Safe Operation

To ensure reliable use of the equipment and the safety of personnel, during installation, use and maintenance.
Please observe the following:
1. When installing the equipment, make sure that the ground wire in the power cord is well grounded and the chassis grounding point is well grounded, do not use a two-pronged plug. Ensure that the input power supply of the equipment is 100V-240V, 50/60Hz AC.
2. Keep the working environment well ventilated, so as to facilitate the equipment in the work of the heat generated by the timely discharge, so as to avoid excessive temperature and damage to the equipment.
3. Turn off the main power supply of the equipment when it is not used in a humid and dewy environment or for a long time.
4. Always unplug the unit's AC power cord from the AC power outlet before: A. Removing or reinstalling any part of the unit.
B. Disconnect or reconnect any electrical plugs or connections to the equipment.
5. There are AC high-voltage parts in the equipment, non-professionals should not disassemble the equipment without permission to avoid the risk of electric shock. Do not repair the equipment privately to avoid aggravating the degree of damage.
6. Do not spill any corrosive chemicals or liquids on or near the equipment.
7. The disconnecting device of this product is the appliance coupler, the appliance coupler is located at the back shell position of the product, when the product is not in use, please unplug the power cord from the product through the appliance coupler, and do not place any other objects near the appliance coupler, so as not to prevent the processor from disconnecting with the power supply.

1. Product overview

This equipment supports a maximum of 16 analog channels into 16 out and 1 USB extension recording channels, high-quality 21-stage preamplification circuit, DSP processing bus structure, built-in feedback eliminator, noise eliminator, echo eliminator and other functions, restore high-quality sound, mainly used in various large-scale venues, to meet the theater, concert halls, remote video conferencing, stadiums, churches, conference centers, Theme parks, public sound reinforcement systems and other aspects of the application needs. Its operation is simple and its performance is powerful;

- Comprehensive matrix mixing, 24bit/48KHz sampling frequency, high-performance A/D D/A converter and 32-bit floating-point DSP processor.
- High-precision input sensitivity adjustment, totaling 21 stops in 3dB steps, with a maximum input gain of 60dB;
- Efficient algorithmic processing: AFC, AEC, ANS, AUTOMIXER, EQ, GATE, AGC and more;
- Rich interface expansion: support for 8-channel custom input and output GPIO, level support for external input 3.3 ~ 24V; USB interface support for recording and broadcasting; RS-485 support for automatic camera tracking function, easy to realize video conferencing; RS-232 bidirectional serial control interface, you can send or accept the control of, such as video matrices, cameras and other equipment;
- Support multiple groups of scene presets, scene saving and other functions, humanized operation software interface;
- Convenient web control: Built-in web control port for fast operation on Windows, Android, iOS and other platforms;
- Control software developed specifically for the iPad and iPhone is now available;

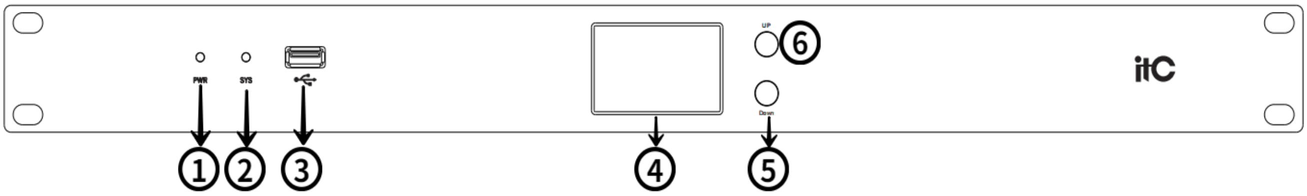
2、Product parameters

Maximum number of	16
Maximum number of	16
232 Number of serial	1
485 Number of serial	1
Number of GPIOs	8, freely configurable inputs and outputs
RJ45 Quantity	1
Number of USBs	1, can support recording and broadcasting
Maximum Analog Gain	-51dB
phantom power	48V
Input/Output	48KHz/24bit
A/D Dynamic Range	120dB
Input common mode	80 dB @ +24dBu@60Hz
Input Impedance	20kΩ balance, 10kΩ unbalance
Maximum Input	24dBu
D/A Dynamic Range	120dB
Channel Isolation	100dB
frequency response	20 to 20kHz (±0.25dB)

Total Harmonic Distortion (THD+N)	≤0.002% @1kHz, +4dBu
Output Impedance	100 Ω balance, 50 unbalance
maximum output	24dBu
Operating power	AC 110V/220V 50Hz/60Hz
operating temperature	0~40℃

3. Front and rear panel interface description

3.1 Front panel description



① PWR: power indicator, light indicates that the device power supply is normal, otherwise the power supply is abnormal;

② SYS: Status Indicator, light flashing indicates that the device is running normally, otherwise the device is faulty;

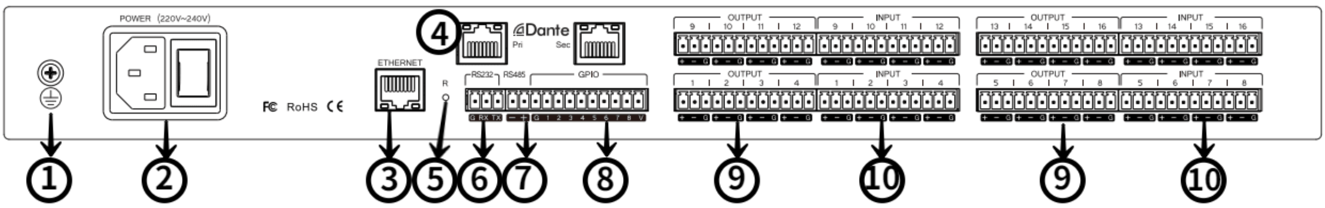
③ USB: Supports recording and broadcasting functions;

④ TFT screen: embedded GUI display;

⑤ UP button: turn the page up;

⑥ Down button: Turns the page down;

3.2 Backplane Interface Description



① Chassis grounding point;

② POWER: 220V AC power input connector and power switch;

③ ETHERNET: 10M/100M Ethernet interface for connection to a console (PC, router, etc.);

④ Dante:Dante network interface for redundant backup;

⑤ RESET: Restore factory settings, long press for 5 seconds;

⑥ RS232: support center command and camera tracking,RX: receive data,TX: send data,G: ground;

⑦ RS485: Supports camera tracking;

⑧ GPIO: GPIO control;

⑨ OUTPUT: Analog output;

⑩ INPUT: Analog input;

4. Software Operation Guide

The source files for the installation software on this machine are downloaded by accessing a processor on the same network segment on the LAN. The source files are accessed by typing the URL in the address bar of the IE browser (usually: http://192.168.1.200/) in the address bar of IE browser to access the processor The installation software is downloaded locally by typing the URL (usually:) into the address bar of your Internet Explorer browser to access the processor and find the download connection.

4.1 Web control and software downloads

The factory default IP address of the device is: 192.168.1.200 Subnet mask: 255.255.255.0. Please add the address of this network segment in your PC first, so that the device can be connected normally.

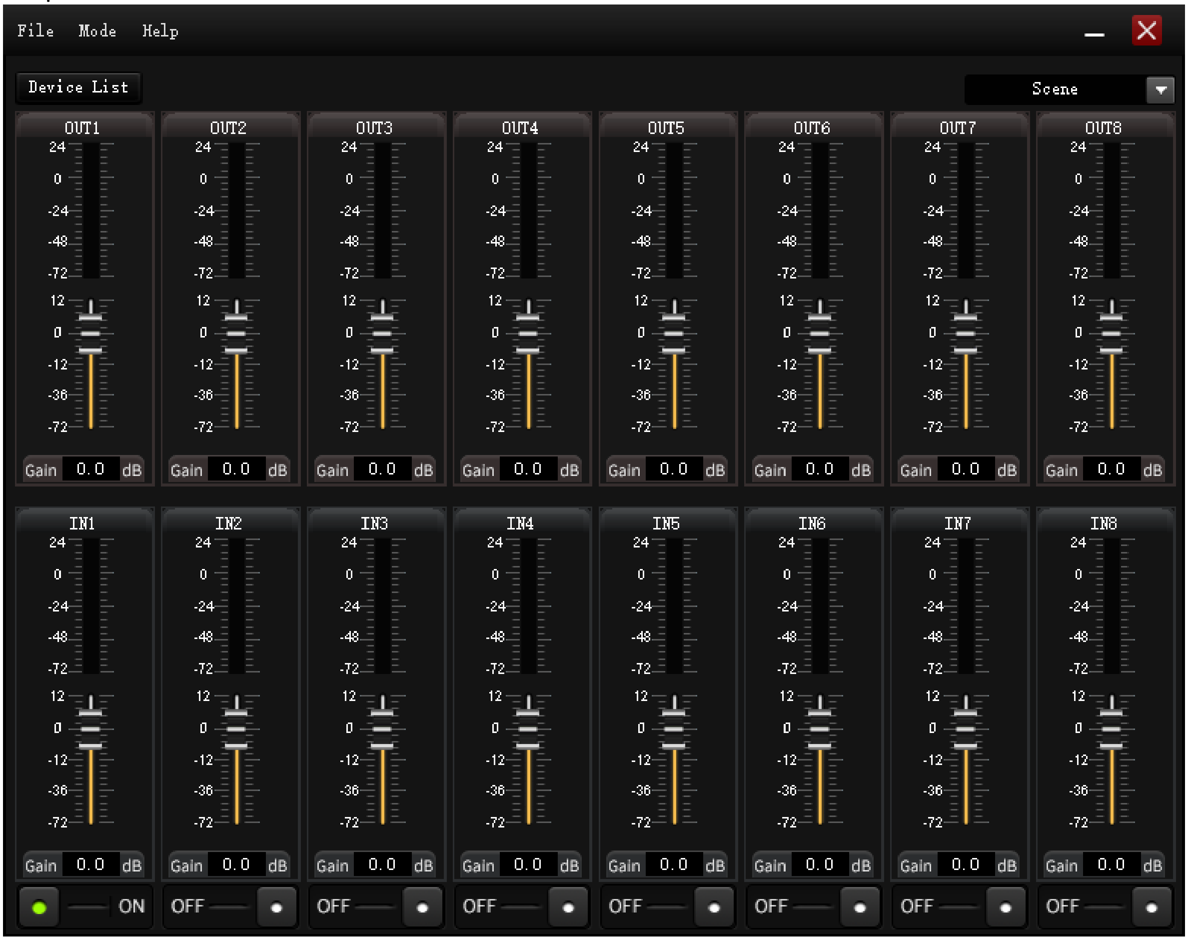
After the device has finished booting up, use a web browser to access the address "http://192.168.1.200/" , as shown in the following figure

Download: The download link provides the download of the PC side software of the .Net framework which supports XP, Win7, Win8 operating systems;

Before installing the software on PC, please make sure that Microsoft . When installing the software, some systems (e.g. WIN8) will pop up a prompt: "User account control information", please click "OK" button to elevate the privileges of the software;

4. 2 Main interface

simple mode



expert model

① Menu bar: File, Settings, Mode, View, Language, Help;

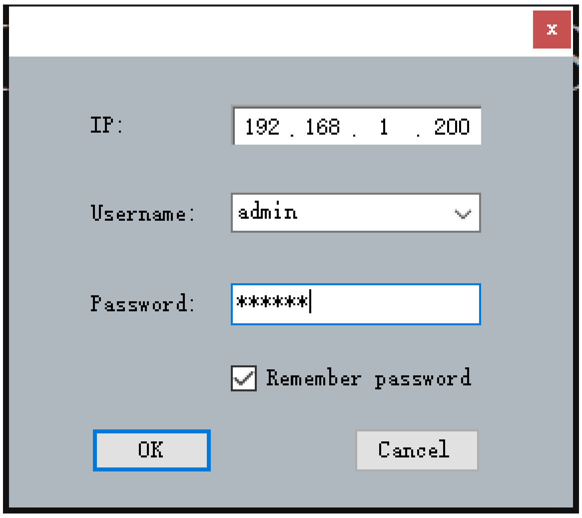
② Process control area: audio data flow chart, you can click on the icon to set the parameters of each processing in detail;

③ System lock: to prevent misuse, (Default password: 123456);;

④ Scene drop-down box: switch scenes;

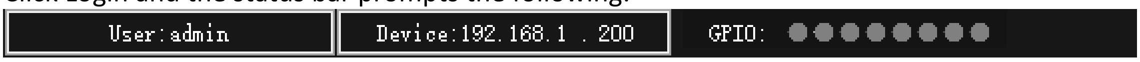
⑤ Save Scene: Users can choose to save the current scene according to their needs;
⑥ Connecting the device: Enter the IP address, account/password to log in;
⑦ Login Device: Under the device list, double click to select the device name to enter account/password to login;
⑧ Device List: Displays the current online devices;
⑨ ⑩ Input/output channel quick control area: displays the level and gain of each channel, as well as the quick enable/disable of each processor press to support the copy/paste function;
When the playback/recording channel is set: When a portable hard disk device is connected to the USB port on the front, playback or channel recording can be performed;

Log in to the operator interface:



Default account/password: admin/123456, account passwords can be changed by entering under the settings module;
Before logging in, please make sure that the IP address of the device and the IP address of the client are in the same network segment. If they are not in the same network segment, double-click the name of the device in the device list to modify it or change the IP address of the client to be the same as that of the device;

Click Login and the status bar prompts the following:



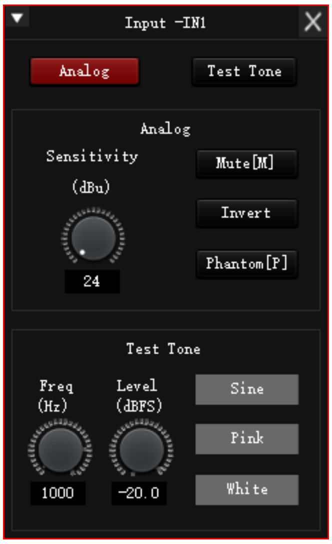
4.3 System processes

Signal Processing Flowchart

standard configuration	Inputs: Test Signal/Mute/Expander/5-band EQ/Compressor/Auto Gain; Outputs: Delay / Crossover / 31-band Graphic EQ / Limiter / Output Reverse / Mute
Advanced Configuration	(AFC) Feedback Canceller (AEC) Feedback Eliminator (ANS) Noise Suppressor (AutoMixer) AutoMixer

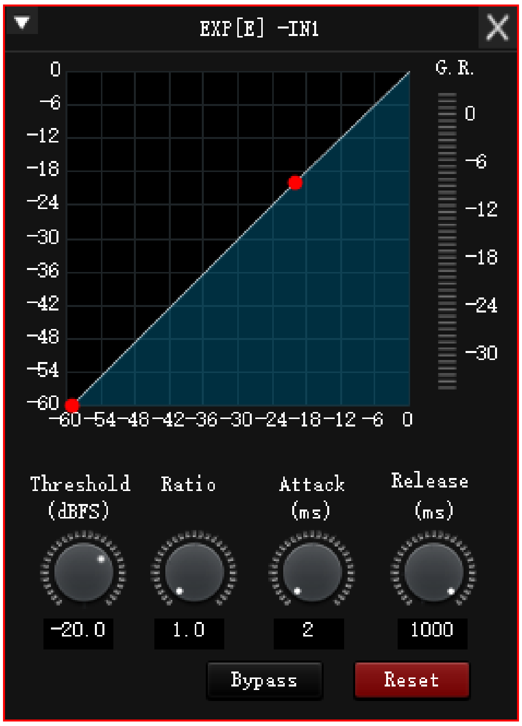
4.4 Input section

4.4.1 Input settings



The input signal can be an analog signal, a test signal generated internally by the device, or, in the case of the network version with Dante, a network digital signal;
Analog signals can be adjusted by adjusting the sensitivity of the input can be selected; from -60 ~ 0, every 3dB;
Mute: mutes the channel when selected;
Invert: 180 degree processing of the signal phase;
Phantom power: For condenser microphone power supply, do not turn on the line input or non-condenser microphone to prevent burnout;
Test signals: including sine, pink, and white noise. Enabling test signals automatically masks the analog input;

4.4.2 Extenders

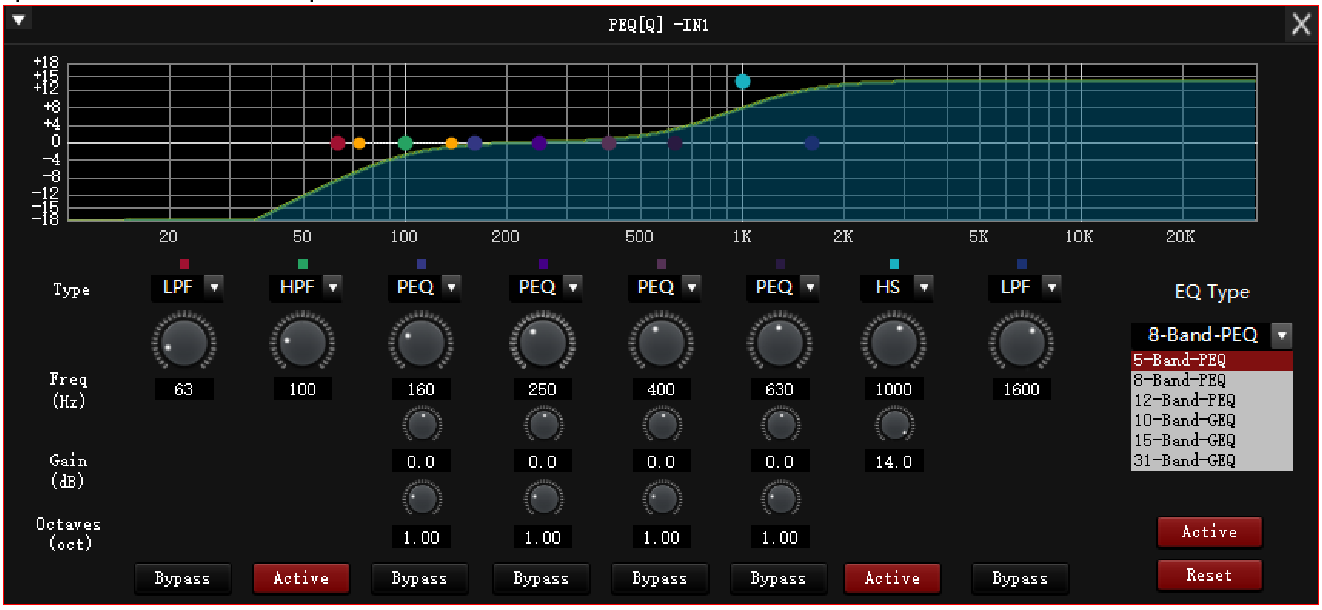


The expander is designed to increase the dynamic range of the input according to the user's needs. When the input signal is less than the "Threshold", the expander compresses the input signal according to the set "Ratio", the output level = Threshold - (Threshold - Input Level)/Ratio; when the input signal is greater than the "Threshold", it is output at 1:1, the output level = Input Level. When the input signal is greater than the "Threshold", the output is 1:1 and the output level = input level.
Straight-through/Enabled: whether the extender is active or not.
Ratio: the number of decibels in which the expander input signal dynamically changes / the number of decibels in which the expansion output signal dynamically changes;
Start-up time: The time required for an input signal less than the expander's "threshold" to enter the expansion state and output at the set expansion ratio;
Recovery Time: The time required for the input signal to return from the extended state to the original non-extended state;

4.4.3 Equalizer type selection

4.4.3.1 Parametric equalizer

Any frequency point and all electrics in the bandwidth can be controlled by adjusting the knob or by entering specific numbers in the input box.



Parametric equalizers are divided into 5-band parametric equalizers, 8-band parametric equalizers, and 12-band parametric equalizers.

Types: low, high, low pass, high pass types added

1) Low Shelf: applies quantized boost or attenuation to all frequencies below the cutoff frequency.
2) Elevated: applies quantized boost or attenuation to all frequencies above the cutoff frequency.
3) Lowpass: the cutoff frequency of the low-pass filter.
4) High-pass: the cutoff frequency for high-pass filtering.

Reset: all filter parameters are reset to their default values;

Straight-through/Enabled: whether the equalizer is active or not;

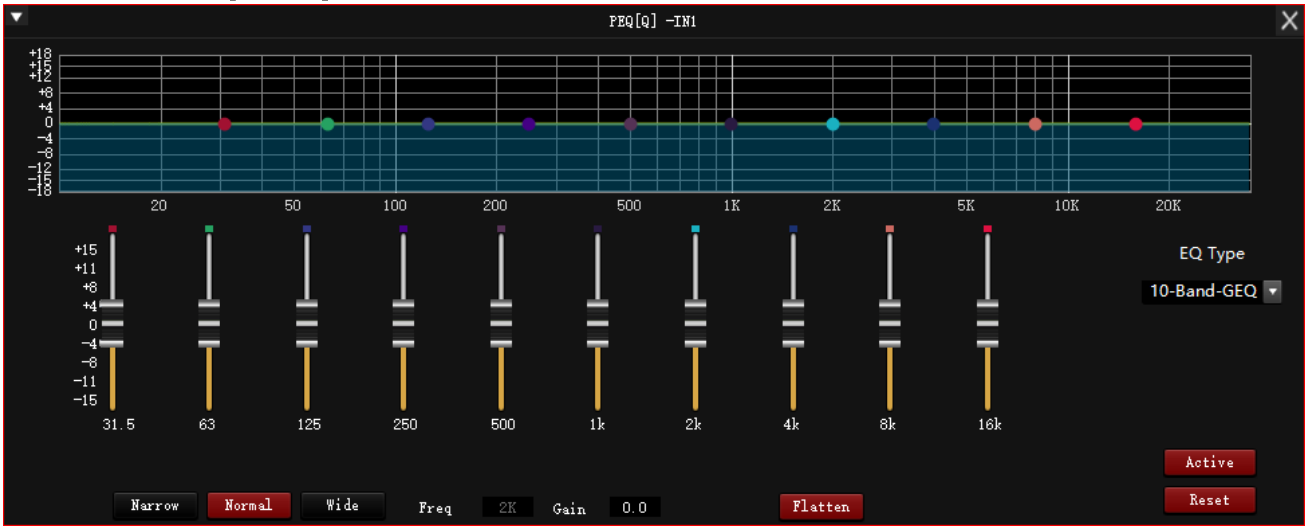
Segment Pass-Through/Enable: whether the segment equalizer is active or not;

Center Frequency: The center frequency at which equalization needs to be done;

Gain : The gain/attenuation value at the center of the frequency;

Bandwidth : That is, the influence range of the segment around the center frequency, the larger the value of the bandwidth is, the larger the influence range is;

4.4.3.2 Graphic equalizer



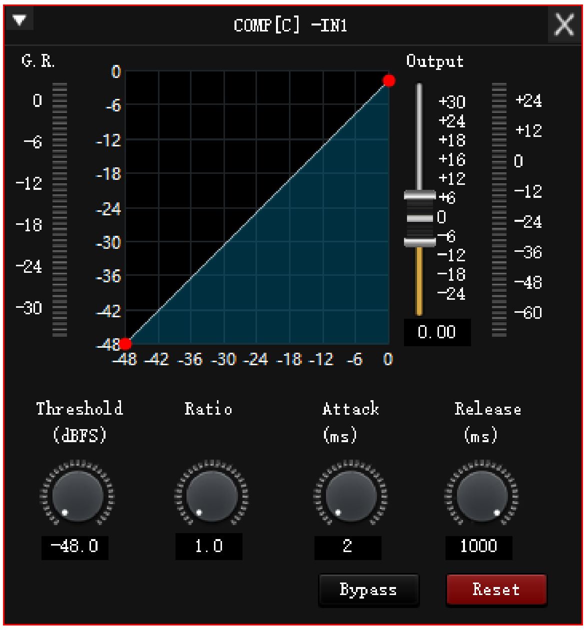
The graphic equalizer is divided into 10-band graphic equalizer, 15-band graphic equalizer, and 31-band graphic equalizer.

Narrowband: narrowband equalization filter;

Normal: Regular equalization filter;

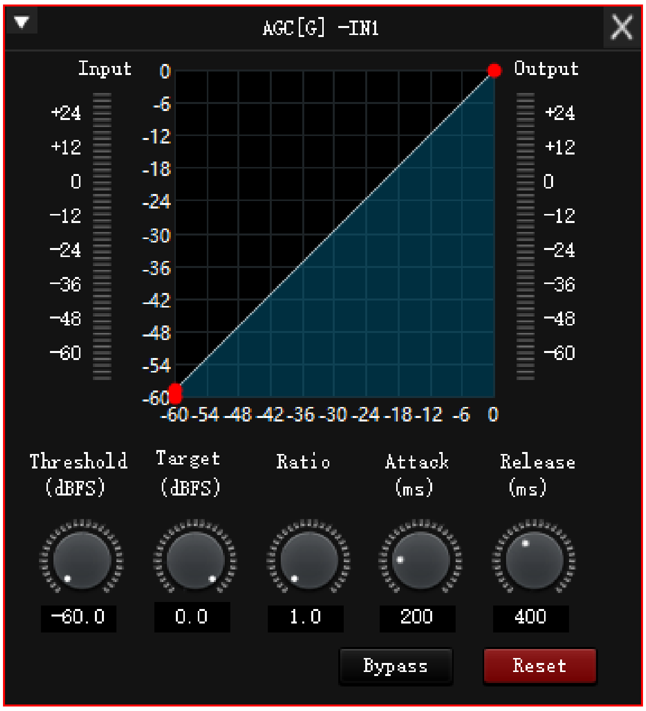
Wideband: wideband equalization filter;
Center Frequency: Indication of the center frequency of the current equalization filter;
Gain: Gain indication or control for the current equalization filter;
Flat: Indicates or controls whether the equalizer is enabled;
Reset: restores all band gains to their default state;

4.4.4 Compressor



The compressor is used to reduce the dynamic range of a signal above a user-determined threshold. Signal levels below the threshold remain unchanged.
Threshold: The signal level above this threshold starts to reduce the gain. This point is at the inflection point in the input/output curve. For Peak Stop, the threshold to be stopped is set just below the peak level;
Ratio: the compression ratio value of the input and output.
Startup Time: The processing speed of the gain reduction associated with the start of the compressor. The shorter the start-up time, the greater the instantaneous change in signal, and the short gain decay makes listening uncomfortable.
Release Time: The release time determines the moment-to-moment gain change of the compressor. Fast release times increase the subjective level, while slow release times are more useful for keeping the level under control;
OUTPUT FADER: The fader controls the output gain of the module. If the compressor reduces the signal level significantly, a boost in output gain may be required to maintain the perceived volume;

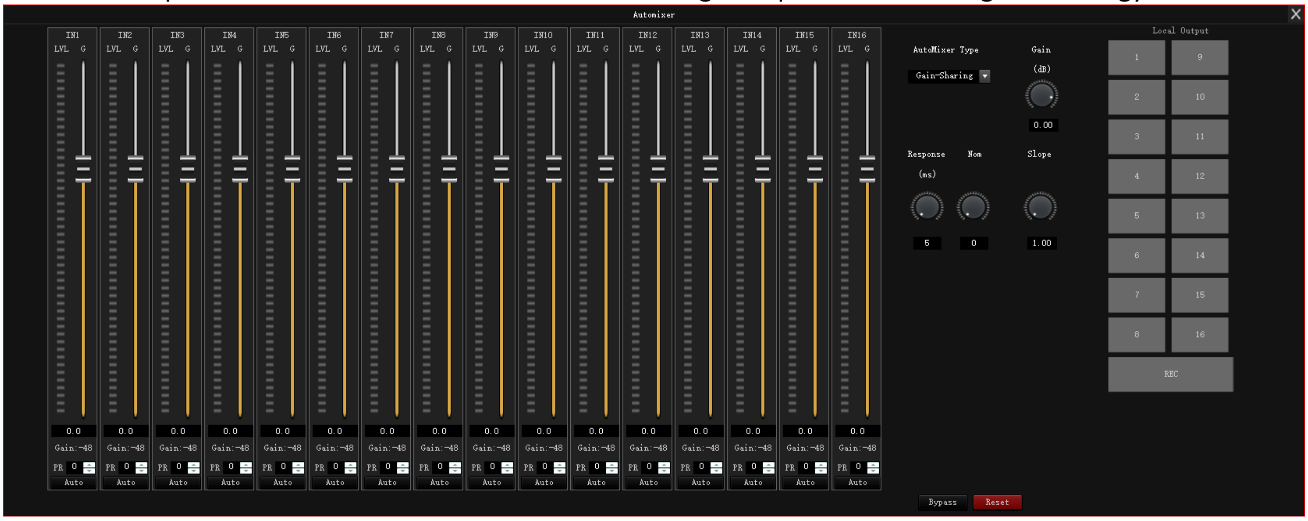
4.4.5 Automatic gain



The purpose of automatic gain control is to bring the signal of uncertain level to the target level while maintaining the dynamic range of the volume. Typical use: For example, when the user is speaking in front of a microphone, the distance between the mouth and the microphone will be far away and close, which will cause the output volume to go up and down, or even feel that the speech is intermittent.
Auto Gain is to set the threshold value to output the input signal below the threshold according to the ratio of 1:1, and for the level above the threshold, it will be boosted directly according to the ratio, set the target level, and the sound signal can be output stably;
Threshold: 1:1 input/output ratio when the signal level is below the threshold. input/output = ratio when the signal level is above the threshold. Set this threshold level slightly above the noise ratio of your input signal;
Target Level: the desired output signal level;
Automatic gain control is to automatically control the amplitude of gain by changing the ratio of input and output compression. When the weak signal input signal amplification processing, to ensure that the strength of the output sound signal; when the strength of the input signal reaches a certain level, the signal compression processing, so that the sound output amplitude is reduced;

4.4.6 Automatic mixing

This series of products now offers an automatic mixer using "Adaptive Gain Sharing" technology.



An automated mixer is used to automate the operation and control of a traditional mixer with a large number of speech inputs to produce the desired results. Consider a typical conference room scenario with ten participants, each with a microphone. If all ten microphones are turned on at the same time, and only one person is speaking, then the output will not be ideal because the other nine microphones are picking up the room acoustics, reverberation, etc., which will degrade the output of the entire system.

Each channel of the automixer has an input, gain level meter and an auto gain, channel fader, priority, and channel mute. Channel Controls Each channel has an "Auto" button that is pressed to add the channel to the automix. Channel Mute and Fader are both Auto Gain types. To mute a signal and prevent it from going into the Auto Mix, turn Mute on and off. The channel fader controls the mix level and direct output level of the channel. Priority Control PR: Allows a channel with a high priority level to override a channel with a low priority level, thus affecting the automix algorithm. This control defines the priority level as a value between 0 (lowest priority) and 10 (highest priority), with a default value of 5 (standard priority). If all channels have equal priority, set the priority of all channels to 5;

4.5 Feedback, Echo, Noise Cancellation

Feedback: Selects the signal that needs to be processed through the feedback eliminator, and the processed signal selects the output channel in the mixer;

Echo: Set the signal that needs to be processed through the echo canceller, and select the output channel in the mixer after processing;

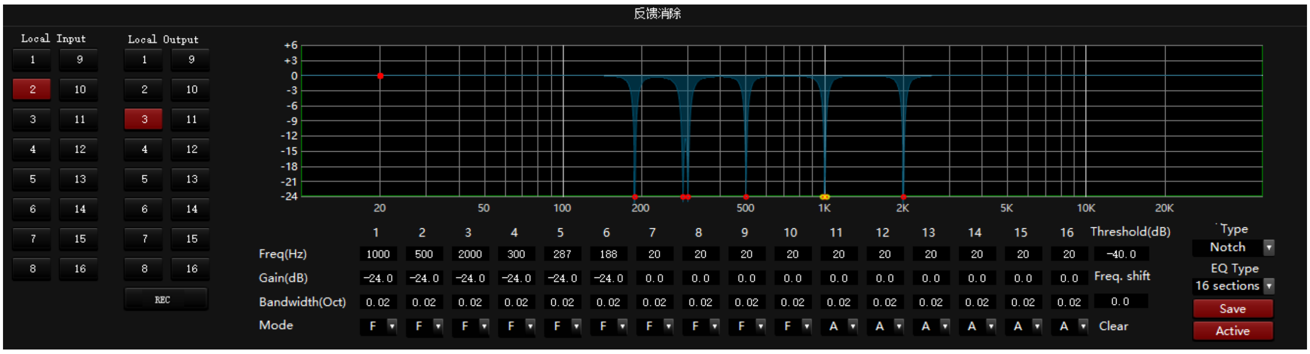
Echo Near Input: the local MIC output, i.e. the signal to be echo processed. Echo far input: the reference signal;

Noise: Select the signal that needs noise elimination processing, and select the corresponding channel output in the mixer after processing;

Mix: Mixes the signals from the selected input channel to the corresponding output channel;

Feedback suppression: Used to suppress the whistling generated between the microphone and speaker in the sound reinforcement system, thus capturing the frequency that causes whistling for attenuation, thus ensuring the quality of the sound as well as preventing burnout of amplifiers or speakers;

Example 1



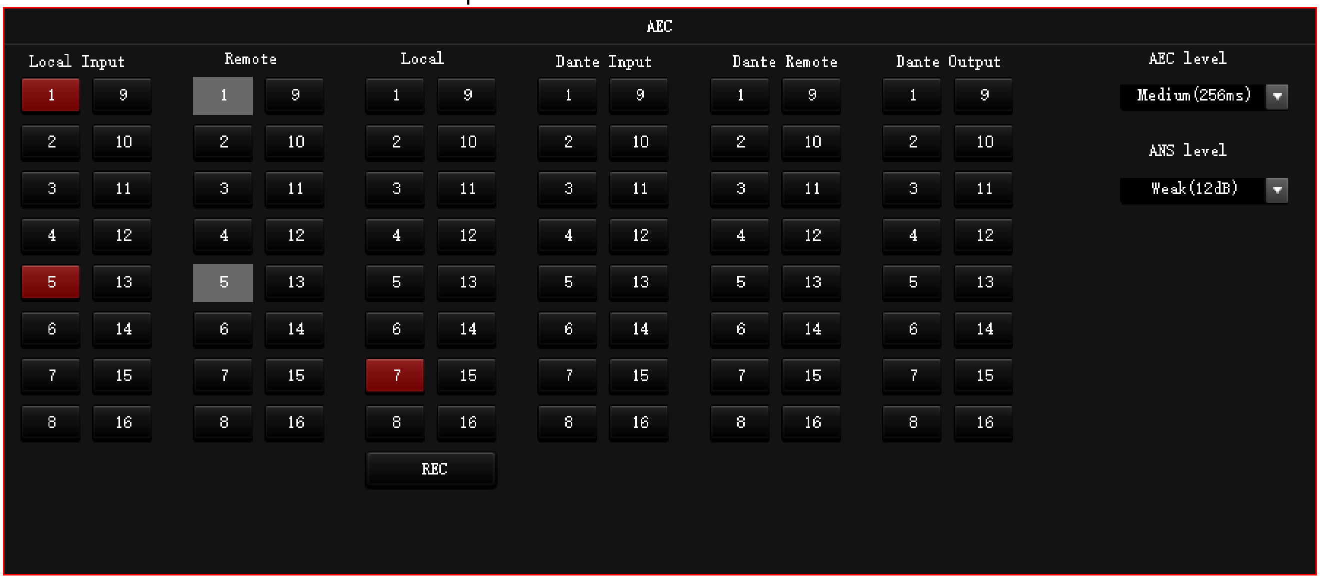
Trap feedback eliminator and mixer association operation:

The signal from channel 2 is processed with trap feedback and output in channel 3, configured as shown on the left:

1. Select input channel 2 in the Trap Feedback Eliminator to indicate that the signal of input channel 2 will be sent to the Feedback Eliminator for processing;
2. Select the point corresponding to OUT3 in the "Feedback Elimination" column of the Mixer, indicating that the result of the feedback eliminator processing will be sent to output channel 3.

Example 2

Noise canceler and mixer association operation:

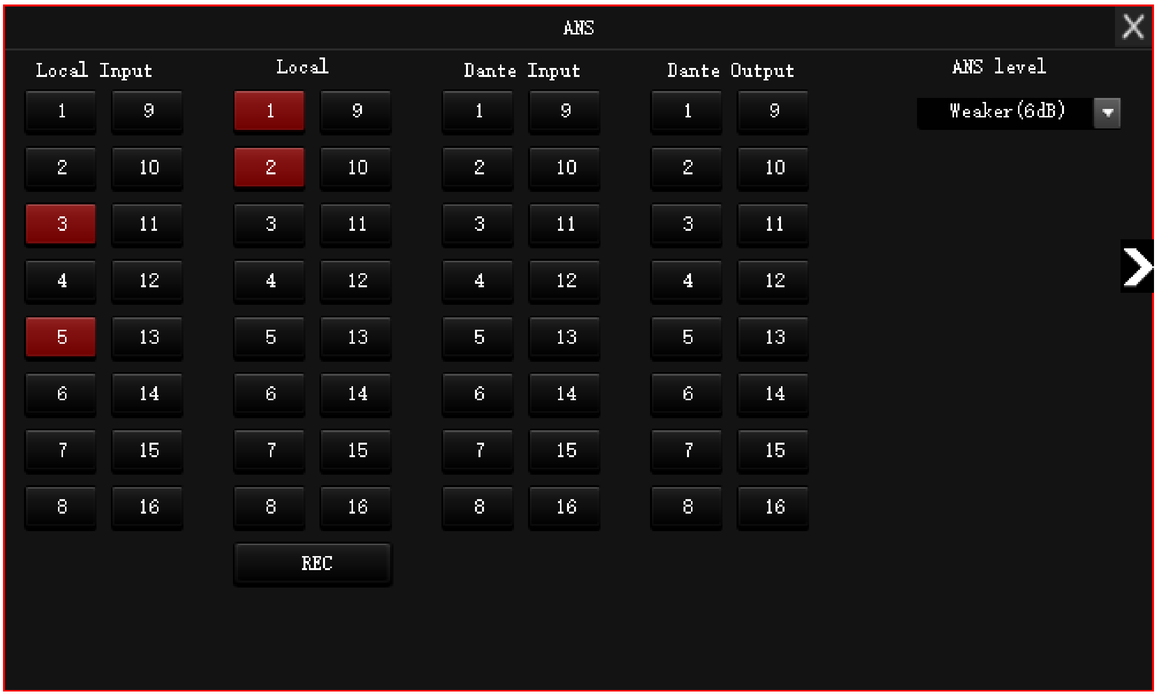


The local signal is input channel 1 and the remote signal is input channel 5, that is, the signal about channel 5 in channel 1 is removed and output in channel 7, the configuration is as shown in the figure:

1. Selecting input channels 1 and 2 in the Feedback Eliminator indicates that the signals from input channels 1 and 2 are sent to the Feedback Eliminator for processing;

Select the point corresponding to OUT7 in the "AFC/Feedback Elimination" column in the Mixer to send the result of the feedback eliminator processing to Output Channel 7 for output;

Exhibit 3



Noise canceler and mixer association operation:

The signals from channels 3 and 5 are noise-canceled and output on channels 1 and 2 in the configuration shown:

1. Selecting input channels 3 and 5 in the Noise Canceller indicates that the signals from input channels 3 and 5 are fed into the Noise Canceller for processing;

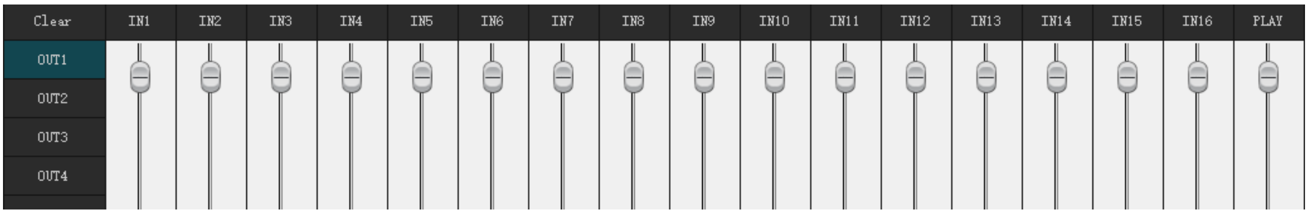
Select the points corresponding to OUT1 and OUT2 in the "ANS/Noise Canceling" column of the Mixer to indicate that the results of the noise canceling process will be sent to Output Channel 1 and Output Channel 2 for output;

4.6 Matrix Mixing

Control of mixing logic

Columns: Input channels

Row: Output channel



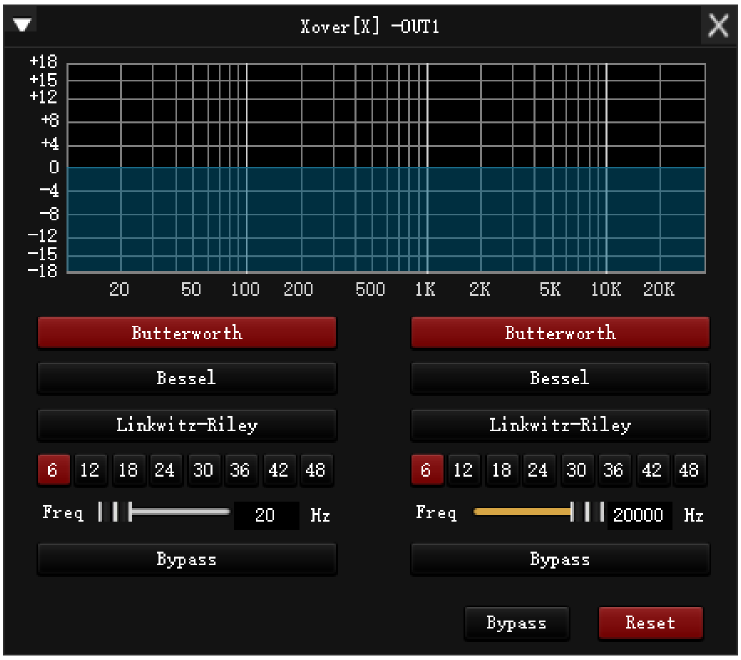
Mix Output Gain: Adjust the gain via the faders (12~-72dB);

4.7 Output section

4.7.1 Delay timer

The time interval between the input and output of the signal to the processor is generally used to produce effects such as reverberation or echo, and can also be used to process auxiliary speakers for larger applications;

4.7.2 Crossover

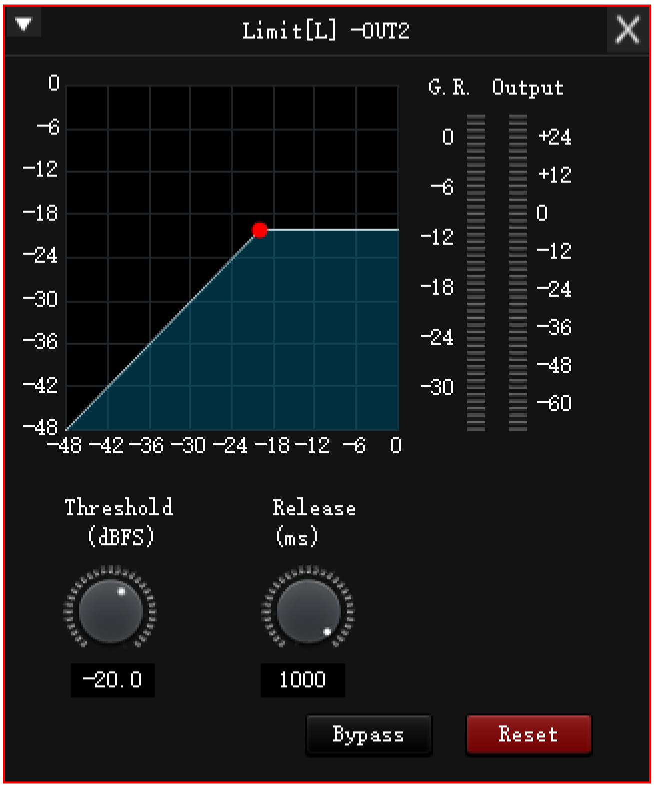


HF Passthrough/Enable: enables and disables the high pass filter;
Low Frequency Pass-Through/Enable: enables and disables the low pass filter;
High-pass frequency: the cutoff frequency for high-pass filtering;
Low-pass frequency: the cutoff frequency of the low-pass filter;

4.7.3 Equalizer

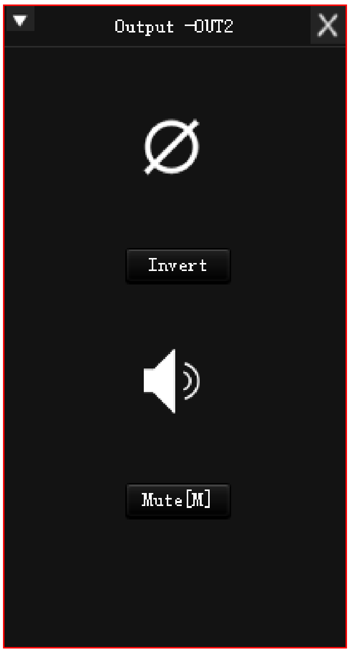
Please refer to section 4.4.3.

4.7.4 Limiter



Straight Through/Enable: enables or disables the limiter;
Threshold: the starting level of limiting, when the signal is higher than this limit value the limiting processing module is activated;
Recovery Time: When the input signal falls below this setting, the sound channel will not be turned off immediately, but will be delayed by this setting. During this time, the sound channel will be turned on continuously as long as there is a signal above the "Threshold" limit.
Compression: The difference between the signal after processing by the limiter and the input signal;

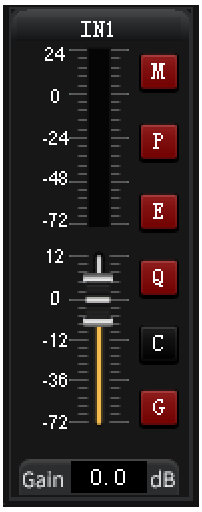
4.7.5 Output settings



Mute and invert can be set for the outputs;

5. Other functions

5.1 Channel control



Input:

1) Channel name that can be modified by double clicking on the channel name;

(2) M, P, E, Q, C, G indicate the corresponding input channel shortcut operation mode:

M is checked for mute	Q Checked to enable and disable
P is checked to turn phantom	C Checked to enable and disable
E Check to enable and disable the	G Checked to enable and disable

3) The level meter displays the input level of the current output channel;

4) The faders adjust the digital gain of the current output channel;

(5) The level meter displays the input level of the current input channel, you can drag left and right or roll the middle mouse button to display the hidden channel, click the channel to copy and paste the channel parameters;

Output:

1) Double-click the channel name to modify the channel name;

(2) M, D, X, Q, L, V indicate the corresponding input channel shortcut operation mode:

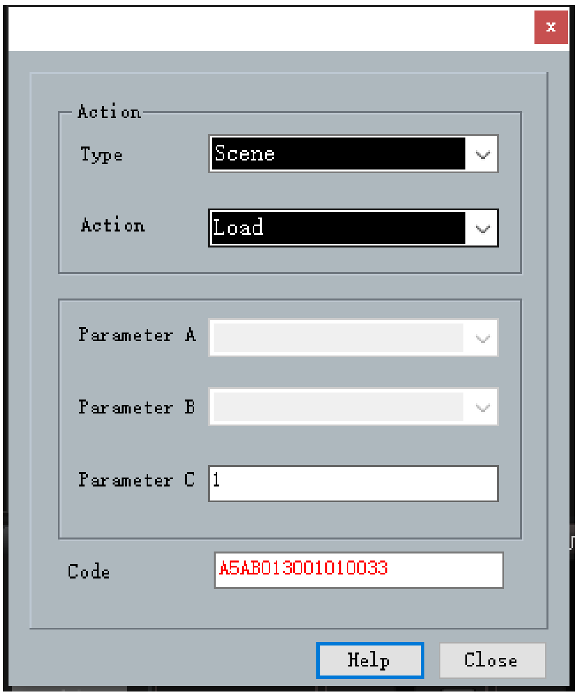
M is checked for mute	Q Checked to enable and disable the
D Check to enable and disable	L is checked to enable and disable
X is checked to enable and	V is checked to enable and disable

3) The level meter displays the output level of the current output channel;

4) The faders adjust the digital gain of the current output channel;

(5) The level meter displays the output level of the current output channel, you can drag left and right or scroll the middle mouse button to display the hidden channel, click the channel to copy and paste the channel parameters;

5.2 Central Control Commands

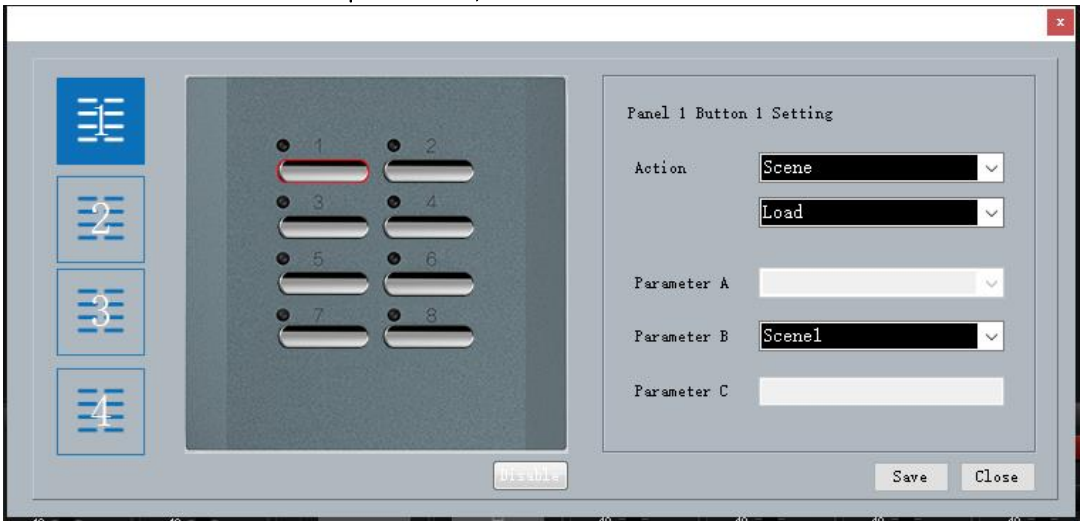


The Center Command Generator converts frequently used operations into a 16-character command code for easy calling by external devices. Each command contains three or fewer different parameters;

Control Command Types: Scene, Input, Output, Mix, Parametric EQ, Graphic EQ, Expander, Compressor, Auto Gain, Delay, Crossover, Limiter;

5.3 Control Panel Configuration

When the processor becomes connected to the console we provide, it is also necessary to set the function of the buttons above the control panel here;



Currently, up to 4 panels can be connected in series, and the panels are automatically numbered from 1 to 4 according to the order in which they are connected in series. Select the panel that needs to be set up, and then select the corresponding button on the center panel, and then set up the function of this button in the right function setting column.

For example, in the above figure, select the first button of Panel 2, select "Input" and "Volume Up" for the function, and select "Input 1" for the parameter 1.

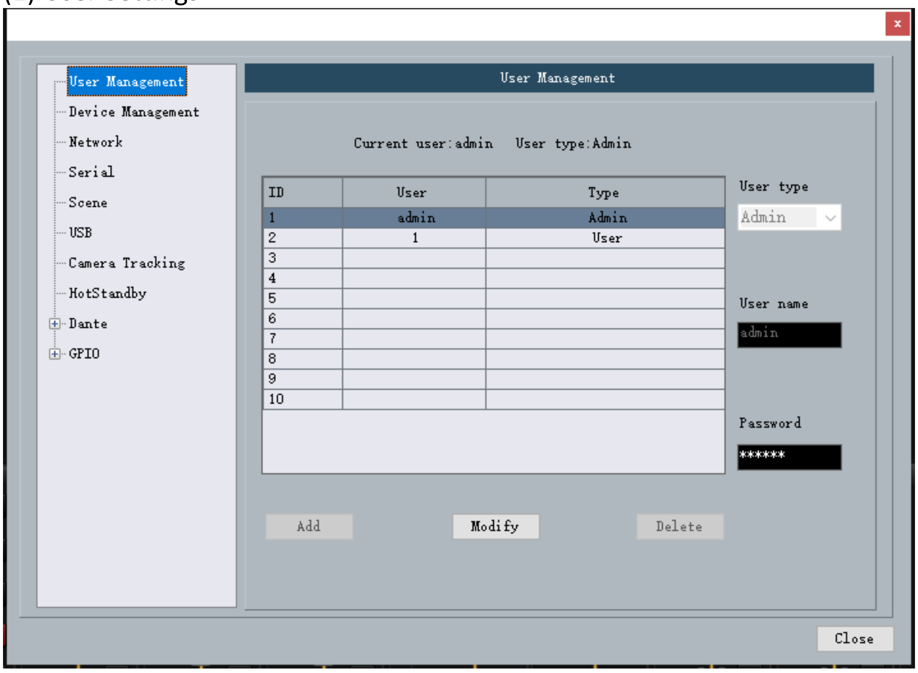
Channel", select "Input 4 Channel" in Parameter 2, enter "1" in Parameter 3, and tap Save to complete the first button of Panel 2.

Function Setting, pressing this button increases the volume of all input channels 1 through 4 by 1 dB;

5.4 Equipment setup

Device settings include user settings, network settings, serial port settings, scene settings, camera tracking, and GPIO;

(1) User Settings

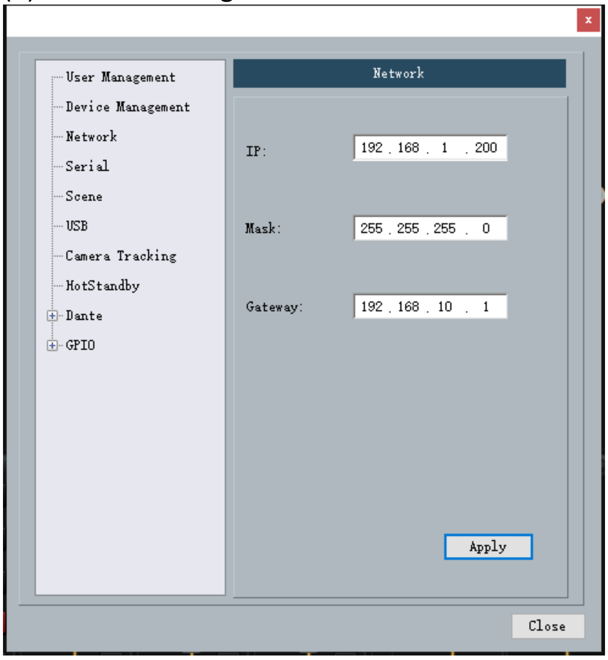


1, the initial user name of the device admin / password 123456. administrator can add, delete, modify all user information; ordinary users can only modify personal information.

2、Modify the user: First, select the user you want to modify in the user list, the user name and password edit box will show the information of the currently selected user, enter the new information and click the "Modify" button.

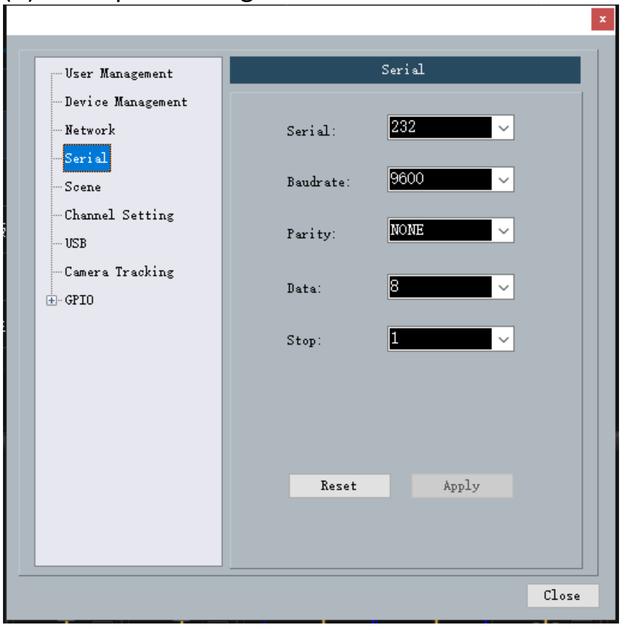
3、Delete User: Select the line to be deleted in the user list and click the "Delete" button to delete the user.

4, add users: in the list on the left to select an empty line, and in the right side of the user name and password edit box (should be empty) to enter the new user's information, click "Add" button to add a new user;
(2) Network settings



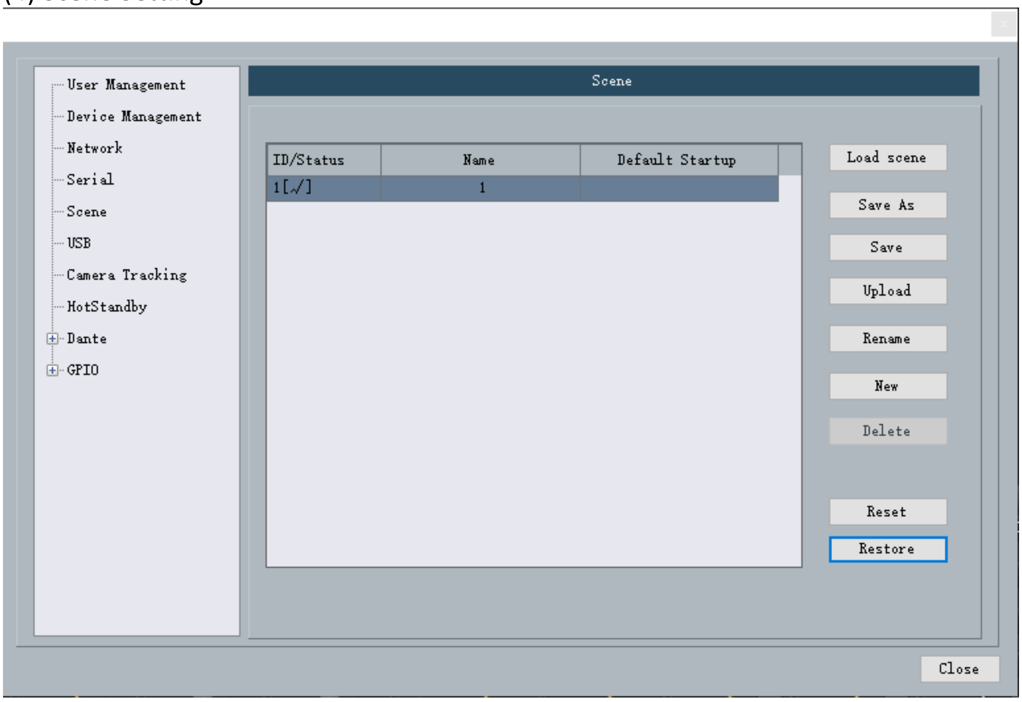
To view and modify the network address information of the device, enter the IP address, subnet mask, and gateway in the corresponding positions, and click the Apply button to complete the modification;

(3) Serial port setting



View and modify the serial port information of the current device, click the "Apply" button to modify the serial port information of the current device after the setting is completed; if you need to restore to the default value, click the "Reset All" button directly, and the items can not be empty during the setting;

(4) Scene Setting



1、 Modify name: Modify the name of the selected scene;

2、 Upload Scene: Upload the PC scene and cover the selected scene;

3、 Save Scene: Save the current running parameters to the selected scene;

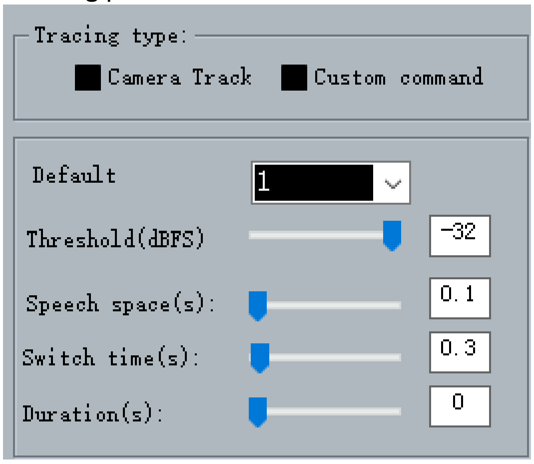
4、 Save As： Save the current running parameters to PC in the way of scene;

5、 Load Scene: Enable the currently selected scene, usually used for scene replacement;

6. Restore Factory Settings: Restore all scene configurations to the default configuration;

(5) Voice tracking

Save voice tracking parameters: Each scene can save different voice tracking parameters, firstly, click "Apply" after setting in voice tracking interface; then click "Save" in "Scene Control" interface, the voice tracking parameters will be automatically saved to the corresponding scene; then, the voice tracking parameters will be automatically saved to the corresponding scene. Then click "Save" in the "Scene Control" interface, the voice tracking parameters will be saved to the corresponding scene automatically;



1. Camera tracking types: camera tracking and custom commands. Camera tracking is used for channel input signal to control camera rotation; custom command sending is used for channel input signal to control sending corresponding custom commands to the corresponding port;

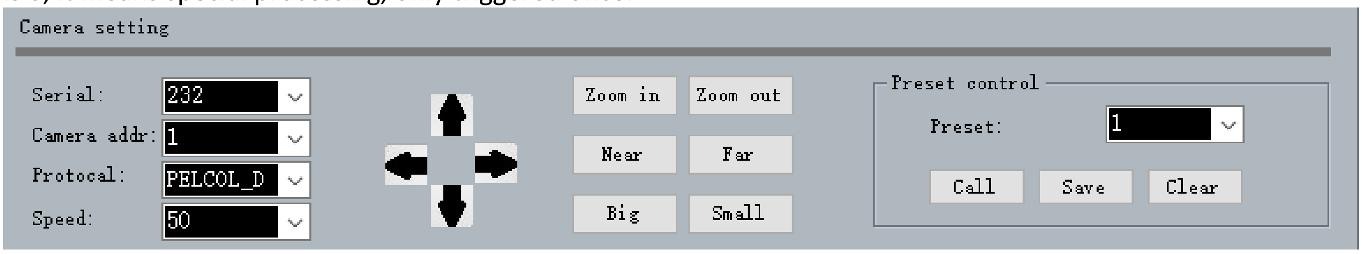
2、 Tracking Threshold: It means the detected input signal must be greater than or equal to the tracking threshold, and the system automatically enables the tracking parameter.

Default MIC: When there is no input from all microphones, rotate the camera to the position of the default MIC setting or send the associated command defined by the default MIC. The commands with # are virtual numbers and can only be used to set the default microphone;

4. Reaction time: the maximum intermittent time of valid signal. If you use microphone to speak, set the reaction time as 3 seconds, the signal is still regarded as valid within 3S pause in the middle of the speech, but the signal is regarded as invalid if it exceeds 3S;

5, switching time: the camera to switch to a valid position of the shortest speech time required. If you use the microphone to speak, the length of speech must be greater than the "switching time", the channel signal is considered valid, and then the camera will automatically turn to the set position. Usually the "Switching Time" is greater than the "Reaction Time";

6、 Rotation time: the interval time between sending camera switching commands or customized commands, if it is 0, it means special processing, only triggered once.



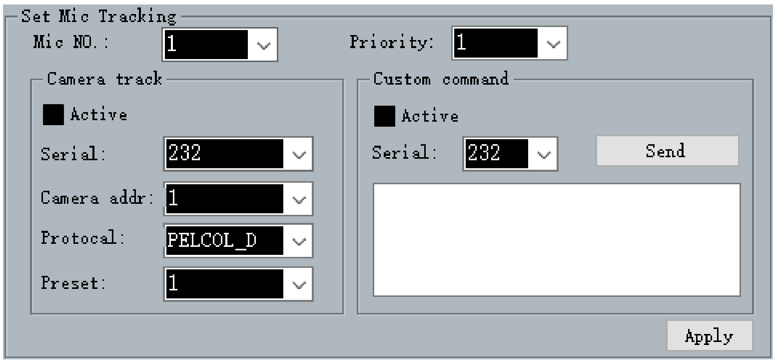
The number of the microphone generally corresponds to the input channel of the device, that is, the channel number to which the microphone is connected. The microphone number indicated with # is a virtual number and can only be used to set the default microphone;

8, the smaller the priority number, the higher the priority level, when the priority level is the same, according to the trigger priority order; such as two microphones talking at the same time, the camera automatically rotates to the preset position corresponding to the microphone with the small priority number (i.e. high priority level) or sends the command corresponding to the microphone with the small priority number (i.e. high priority level); however, if the two microphones have the same priority level, the signal that is checked first shall prevail;

9、 Enable this MIC setting: You can set all the MIC parameters in advance, but when you use it, you can enable only some of them according to the actual situation;

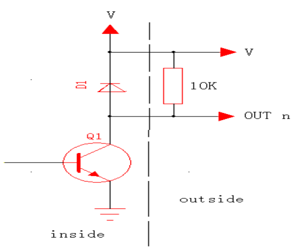
10, preset points, serial port number, camera address, protocol and camera-related, must correspond to the actual connection with the camera

11、Custom command means that when the microphone of the matrix checks the input signal (usually when someone speaks), it will automatically send the corresponding command to the defined serial port, and you can also set the command in advance, but if you don't check the box of "Enable Custom Command", the device won't 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; "Send" button will send the command in the input box to the specified serial port at any time. However, you can still tap the "Send" button to send the commands in the input box to the specified serial port at any time;
12. Click Save to save the parameters to the device, then the microphone of the 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;

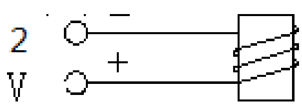


13、Camera Setup is a camera debugging interface, generally debugging the camera position before the tracking starts, and finally the parameters of this part will be saved on the camera;
14, first of all, set up the serial port, there are 2 serial ports (232, 485), and the backplane port corresponding to the PTZ connected to the backplane;
15, followed by the camera address and protocol type, please refer to the actual address of the camera for the camera address, and the protocol is related to the camera model;
16. The final preset point number is the user-defined identification for the camera, and then adjusting the up, down, left, right, and focal length, aperture, and other parameters will define the camera's position and settings;
17、Finally click "Save" to save the parameters to the camera, "Clear" is the current prefabricated point to delete the information, "Call" is used to view the current preset point of the preset camera position. camera position saved in the current preset point.
Note: A camera address can contain several preset points, but a preset point corresponds to only one camera address. The camera settings and microphone settings have several parameters, such as preset point, serial number, camera address, and protocol, so you need to consider the actual situation when applying them;
232 and 485 port switching method:
If a camera is first connected to the audio processor in the 232 port debugging, in the case of the position remains unchanged, disconnect and then with another audio processor.
Connect the 485 port with the processor. At this point, the parameters in the camera are retained without resetting, only the microphone settings need to be adjusted, but the 485 port should be selected for this port;

(6) GPIO Setting

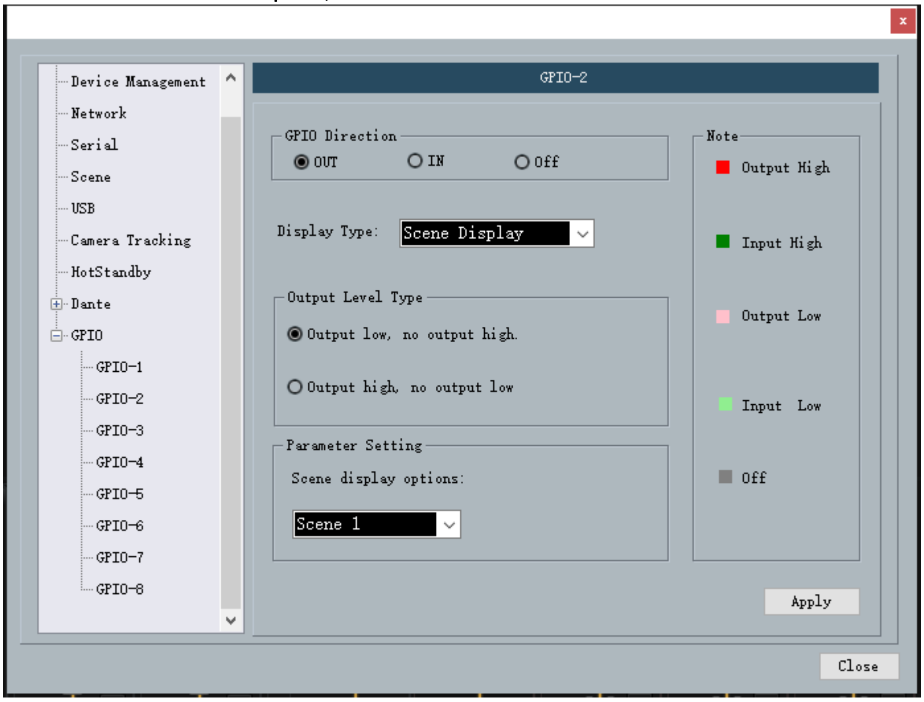


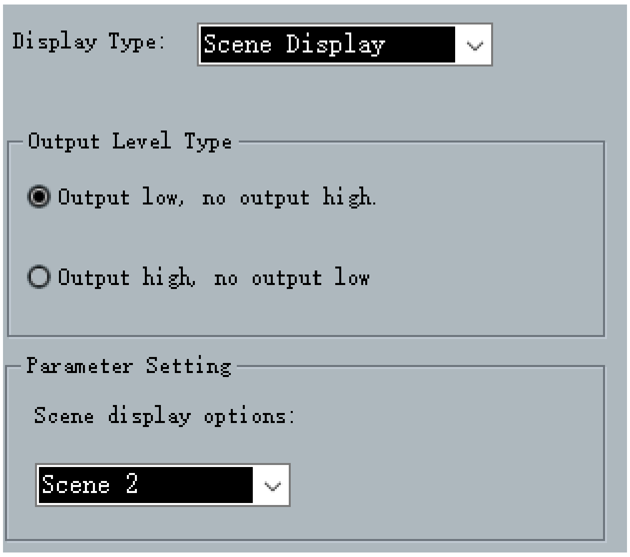
Output Connection I:
First, connect a 10K/0.25W resistor between a GPIO pin (e.g. port 2) and "V" on the device (as shown in the figure), the pin will output a low level 0 or high level 1 according to the matrix state, and the level can be used to trigger another GPIO or other devices;



Output Connection 2 (for control)
Drive relays: relays can be
Used for controlling alarm devices, etc., with built-in diode
Output setup and use:

PC connects to the device -> Setup menu -> Device settings -> corresponding GPIO (gpio-2 in this example) -> direction is set to "Output";



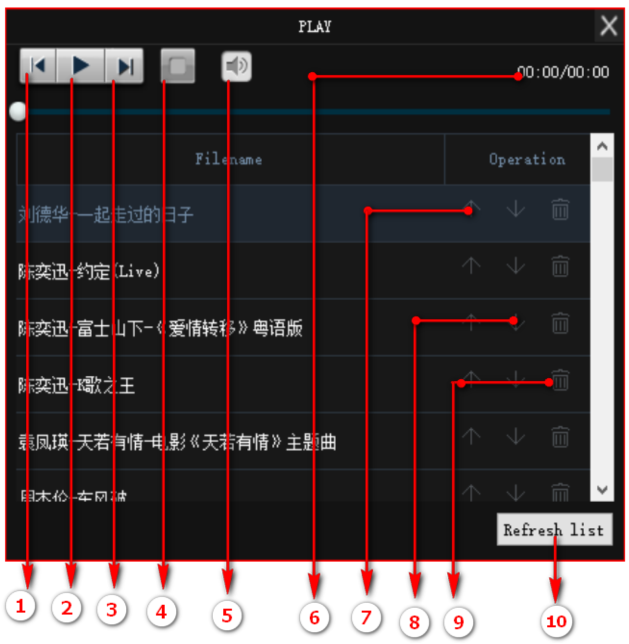


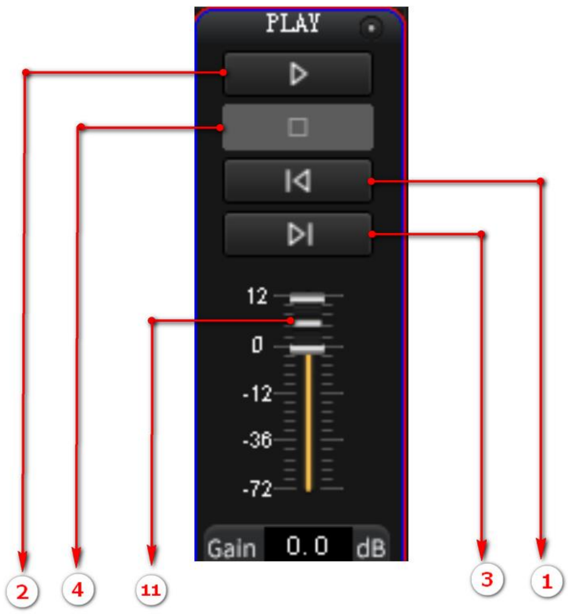
Display: Select the scene to display, as shown in the picture, click "Apply" to make the settings take effect;
When the matrix is loaded in scene 4, GPIO(n) outputs low level relay with current flowing through it, and when the matrix is switched to other scenes (e.g., scene 3), GPIO(n) does not have current flowing through it and does not operate. If "Output high, no output low" is selected, the action is reversed;
If GPIO(n) is bound to other parameters, such as level display, mute display, system mute display, etc., the relay can also switch the operating state according to the parameters set by the matrix;

5.5 USB Recording Function

USB Playback Function: The processor automatically reads and selects audio files in MP3 and WAV formats from USB flash disks for playback through the USB interface.

USB Playback interface opening method: Click the playback window of the quick operation interface to open the playlist, the interface is as follows:

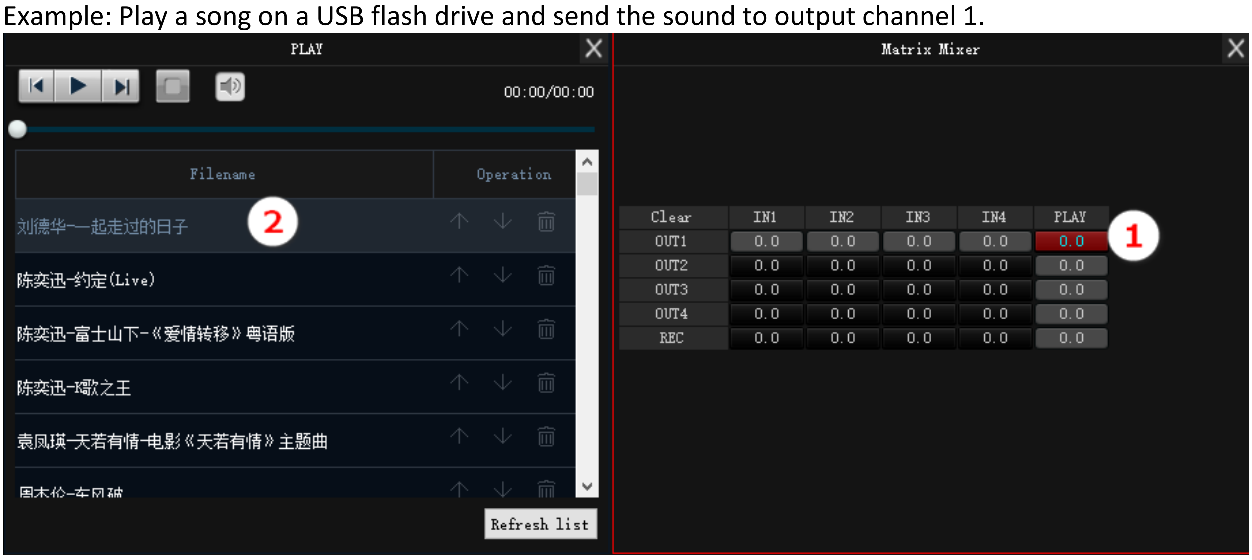




1. Previous song button.
2. Play and pause buttons.
3. Next song button.
4. Stop button.
5. Mute and volume control buttons.
6. Displays the total song time and played time.
7. Move up button.
8. Move down the button.
9. Delete button.
10. Refresh the playlist.
11. Volume control buttons.

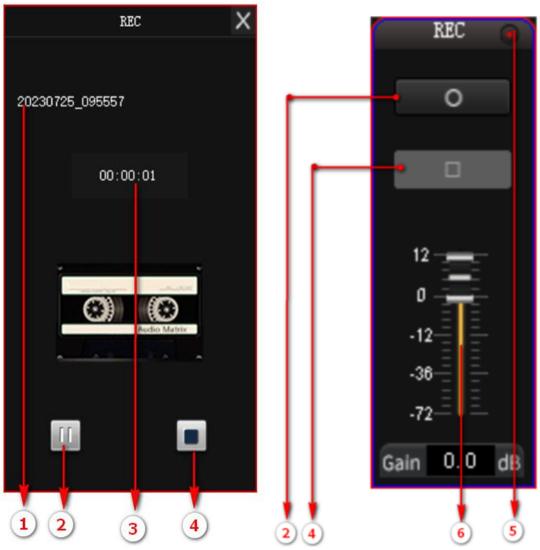
Instructions for using the USB playback function:

Example: Play a song on a USB flash drive and send the sound to output channel 1.



1. Select the switch corresponding to "Output 1" in the "play" column of the matrix mixing interface.
2. Double-click the song name or the play button to start playing the song;

USB Recording Function: Save the audio signal of the channel to the USB disk and other storage media through USB interface. How to open: Click the Record window in the Quick Operation Interface to open the recording interface, as shown below:

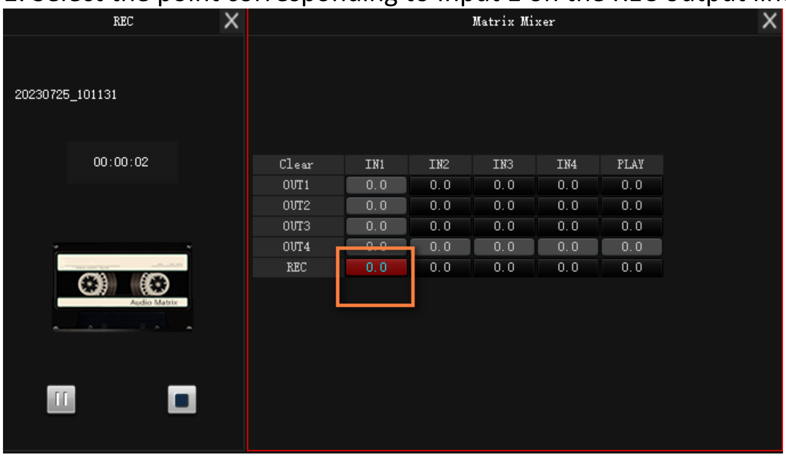


① Filename;
② Record Start Pause button;
③ Recording time;
④ Record stop button;
⑤ The indicator light has three states: gray means the device is not connected to a USB flash drive, red means the device is connected to a USB flash drive, and flashing means it is recording.
⑥ Record volume control button;

Instructions for using the USB recording function:

Example: Record the sound of input channel 1 to a USB flash disk;

1. Select the point corresponding to Input 1 on the REC output line of the Matrix Mix screen;



2. Click the Record button, enter the recording file name in the pop-up dialog box, and then click Start Recording;