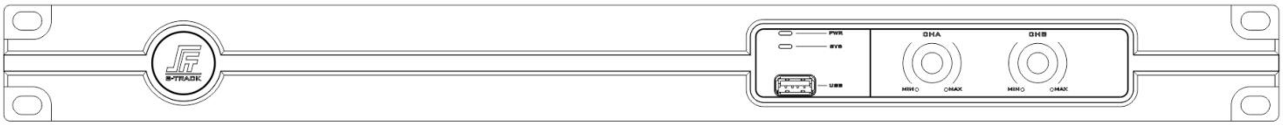
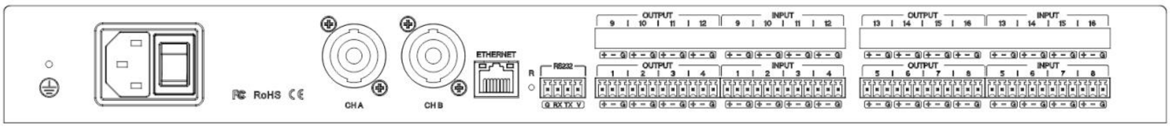
S-TRACK





LION X Series
Digital Signal Processor

User Manual

Preface

The purpose of this section is to ensure that the user is able to use the product correctly through this manual in order to avoid danger in operation or property damage. Before using this product, please read the product manual carefully and keep it for future reference.

Outlined

This manual applies to Digital Signal Processors.

This manual describes the functions and use of the various functional modules of the Digital Signal Processor, and guides you through the installation and commissioning of the Digital Signal Processor.

Symbol Conventions

The symbols that may be found in this document are defined as follows.

Symbol	Description
Note	Provides additional information to emphasize or supplement important points of the main text.
Caution	Indicates a potentially hazardous situation, which if not avoided, could result in equipment damage, data loss, performance degradation, or unexpected results.
Danger	Indicates a hazard with a high level of risk, which if not avoided, will result in death or serious injury.

Safety Instructions

Danger

To ensure reliable use of the equipment and the safety of personnel, please observe the following during installation, use and maintenance:

- During the installation and use of the equipment, all electrical safety regulations of the country and the region of use must be strictly observed.
- When installing the equipment, make sure that the power supply of the equipment is AC 100V-240V, 50/60Hz.
- Keep the working environment well ventilated so that the heat generated by the equipment during operation can be discharged in time to avoid damage to the equipment due to excessive temperature.

- Always unplug the unit's power adapter from the AC power outlet before: A. Removing or reinstalling any part of the equipment; B. Disconnecting or reconnecting any electrical plug or connection of the equipment. Do not operate with electricity.
- There are AC high-voltage parts in the equipment, non-professionals should not disassemble them without permission to avoid the risk of electric shock. Do not repair the equipment privately to avoid aggravating the damage.
- Do not spill any corrosive chemicals or liquids on or near the equipment.
- If the unit emits smoke, odour or noises, turn off the power immediately and unplug the power cord, and contact your dealer or service centre.
- If the appliance is not working properly, contact the shop where you purchased the appliance or the nearest service centre and do not disassemble or modify the appliance in any way. (We cannot be held responsible for problems caused by unapproved modifications or repairs).

Caution

- Do not drop objects on the equipment or vibrate the equipment vigorously, and keep the equipment away from locations with magnetic field interference. Avoid installing the equipment in a place where the surface vibrates or is susceptible to shock (neglecting this may damage the equipment).
- Do not use the equipment in high temperature, low temperature or high humidity environments. Refer to the equipment's data sheet for specific temperature and humidity requirements.
- Use the unit indoors, not in an exposed installation where it may be exposed to rain or extreme humidity.
- When the equipment is not used for a long period of time or in a humid and dewy environment, the main power supply of the equipment should be switched off.
- When cleaning the equipment, please use a sufficiently soft dry cloth or other alternatives to wipe the internal and external surfaces, do not use alkaline detergent to wash, and avoid hard objects to scratch the equipment.
- Please keep all the original packaging materials of the equipment properly, so that in case of problems, use the packaging materials to pack the equipment and send it to the agent or return it to the manufacturer for processing. We will not be responsible for any accidental damage in transit not caused by the original packaging materials.

Note

- Requirements for the quality of installation and commissioning personnel

  Qualifications or experience in the installation and commissioning of audio and video systems and qualifications to perform related work, in addition to the knowledge and operational skills listed below.

  - Basic knowledge and installation skills of audio and video systems and components.

- Basic knowledge and skills in low voltage cabling and wiring of low voltage electronics.
- Basic audio and networking knowledge and skills and the ability to read and understand the contents of this manual.

TABLE OF CONTENTS

Chapter 1 Product Introduction

1.1 Introduction

The Digital Signal Processor equipped with a high-performance 32-bit floating-point DSP processor and A/D ~ D/A converter, support for 24bit/48KHz sampling frequency, high-quality 21-stage preamplifier circuit, DSP processing bus structure built-in Acoustic Feedback Canceler, Acoustic Echo Canceler, Adaptive Noise Suppressor, Automatic Mixer and other audio core algorithms, to restore high-quality sound, with a comprehensive matrix mixing function It supports multiple scene presets, scene saving and other functions, and user-friendly control software interface. Mainly used in a variety of large places, can meet the theatre, concert halls, remote video conferencing, stadiums, churches, conference center, theme parks, public sound reinforcement systems and other aspects of the application needs.

1.2 Product Features

- ➢ Highly integrated, integrating a variety of traditional Analog audio processing equipment in a Digital Signal Processor;
- ➢ High-performance 32-bit floating-point DSP processor, all-digital processing, fast response to AGC (Automatic Gain Control), AM (Automatic Mixer), AFC (Acoustic Feedback Canceler), AEC (Acoustic Echo Canceler), ANS (Adaptive Noise Suppressor) and other audio processing;
- ➢ High-performance A/D, D/A converter, 24bit/48KHz sampling frequency, high-quality Analog → Digital, Digital → Analog conversion;
- ➢ Up to 8 Analog input channels and 8 Analog output channels, very small distortion and ultra-low background noise;
- ➢ Integrated design of audio processing and amplifier output;
- ➢ Rich interface expansion;
- ➢ Humanization, graphical, intuitive and easy-to-operate control software interface;
- ➢ Comprehensive matrix mixing functions;
- ➢ Scene storage is different from the Analog equipment is one of the most practical and significant features, can store 100 complete scenes, all the scenes can be exported to an external storage device for storage backup, so that the later call at any time.

1.3 Functions

- ✧ Comprehensive matrix mixing function, 48kHz sampling rate, 24-bit high performance A/D, D/A converter and 32-bit floating point DSP processor;

- ✧ DSP audio processing, built-in automatic mixing console, including mixing and automatic mixing functions, but also has a mixing component control function; at the same time with AFC, AEC, ANS module; AFC: Support notch feedback cancellation algorithm, manual and automatic notch feedback canceler, with manual, dynamic, fixed three modes, can automatically capture the feedback point or manually set the feedback point, the maximum support for the capture of 16 feedback points, the maximum depth of inhibition up to 24dB;
- ✧ Inputs per channel: Preamplifier 51dB, Invert, Signal Generator, Expander, Equalizer (5/8/12-band Parametric Equalizer, 10/15/31-band Graphic Equalizer), Compressor, Automatic Gain Control (AGC), AM (Gain Sharing Automatic Mixer or Gating Automatic Mixer), Ducker, Acoustic Feedback Canceler (AFC), Acoustic Echo Canceler (AEC), Adaptive Noise Suppressor (ANS), Parametric Equalizer filter type selectable (Low Shelf, High Shelf, Low Pass Filter, High Pass Filter);
- ✧ Outputs per channel: Delay, Crossover, Equalizer (5/8/12-band Parametric Equalizer, 10/15/31-band Graphic Equalizer), Limiter, Invert, Parametric Equalizer filter type selectable (Low Shelf, High Shelf, Low Pass Filter, High Pass Filter);
- ✧ Integrated design of audio processor and amplifier output;
- ✧ Display shows the IP address;
- ✧ Test signal generator, Sine wave, Pink Noise, White Noise, frequency and level magnitude selectable;
- ✧ Input phase button, mute button, phantom power button;
- ✧ Output mute button, phase button per channel;
- ✧ Supports Simplified Chinese, English and Traditional Chinese languages;
- ✧ One-click display of all function modules;
- ✧ Storing user manual and software with the device;
- ✧ Central control code generated in the control software; power failure automatic protection memory function; one-click reset function;
- ✧ Channel copy, paste, group control function;
- ✧ Supports setting the maximum and minimum volume range for each channel;
- ✧ The same host allows 10 users to manage;
- ✧ Device name can be modified;
- ✧ Editable preset mode, new, delete, modify, one-click reset, preset mode can be stored to computer and one-click reset;
- ✧ With camera tracking function, can independently adjust the preset position of a camera, compatible with VISCA, PELCO-D, PELCO-P three control protocols, support for custom commands;

- ✧ Convenient and fast web control: Built-in web controller for fast operation on Windows, macOS, Linux, Android, iOS and other OS platforms;
- ✧ Ethernet multi-purpose data transmission and control port, can support real-time management of single and multiple devices;
- ✧ Intuitive image, simple and easy to understand the graphical software control interface, for customers to bring fast, real-time operating experience;
- ✧ The device does not need a CD, comes with installation software, a device for a software version, to solve the troubles caused by the loss of the installation CD and the confusion of multiple software versions;
- ✧ Extendable USB interface, support U disk recording and playback, or USB sound card, please refer to the actual device;
- ✧ Configuration of bi-directional RS232 interface, RS485 interface, standard Ethernet control interface, 8-channel programmable GPIO control interface (customisable inputs and outputs), level support for external inputs 3.3～24V;
- ✧ Support 100 groups of scene presets, scene new, save, delete and other functions;
- ✧ Intuitive, graphical software control interface, works on Windows XP, 7, 8, 10, 11, etc.;
- ✧ Support mobile iOS, iPadOS, Android control software.

Chapter 2 Specification

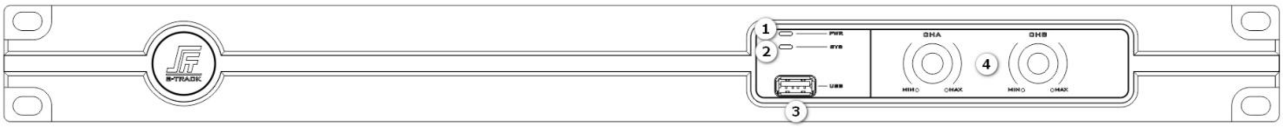
Category	Parameter Item	Parameter Description
Peripherals	Input Interfaces	4/8 Analog
	Output Interfaces	4/8 Analog
	Digital Amplifier Channels	2x150W
	Control Interfaces	1 RJ45 interface, 1 RS232 interface
Audio processing	Processor	TI 456MHz FLOPS dual-core 32-bit DSP processor; 24-bit A/D and D/A converter, 48kHz sampling rate

	Input Channel	Functional module: Preamplifier 51dB, Signal Generator, Expander, Equalizer (5/8/12-band Parametric Equalizer, 10/15/31-band Graphic Equalizer), Compressor, Automatic Gain Control (AGC), AM (Gain Sharing Automatic Mixer or Gating Automatic Mixer), Ducker, Acoustic Feedback Canceler (AFC), Acoustic Echo Canceler (AEC), Adaptive Noise Suppressor (ANS), Parametric Equalizer filter type selectable (Low Shelf, High Shelf, Low Pass Filter, High Pass Filter). Physical interface: Balanced Phoenix terminals.
	Output Channel	Functional module: Delay, Crossover, Equalizer (5/8/12-band Parametric Equalizer, 10/15/31-band Graphic Equalizer), Limiter, Parametric Equalizer filter type selectable (Low Shelf, High Shelf, Low Pass Filter, High Pass Filter). Physical interface: Balanced Phoenix terminals.
	Phantom Power	DC 48V
	Input Impedance	Balanced: 20KΩ
	Output Impedance	Balanced: 100Ω
	Common Mode Rejection Ratio	>60dB@50Hz
	Input Dynamic Range	108dB
	Frequency Response	20Hz～20KHz, ±0.2dB
	Noise Floor	-90dBu
	Signal-to-Noise Ratio	106dB
	THD+N	≤0.003% @1kHz, +4dBu
	Channel Isolation	>100dB@1kHz

	Input Range	≤+18dBu (A-Weighting)
	Crossover	Three types of high and low pass filters: Butterworth, Bessel and Linkwitz-Riley
	Equalizer	Parametric Equalizer: Frequency: 20 to 20kHz, Gain: -15 to +15dB, Bandwidth: 0.02 to 4 Graphic Equalizer: Frequency: 20～20kHz, Gain: -15～+15dB
	System Latency	≤9ms
	Maximum Output Level	18dBu
	Maximum Input Level	18dBu
	Analog/Digital Dynamic Range	114dB
	Digital/Analog Dynamic Range	120dB
	Equivalent Input Noise	≤-125dBu
General specification	Operating Voltage	AC 100V～240V, 50Hz/60Hz
	Maximum Power	330W
	Operating Temperature and Humidity	0℃～40℃, 10%～90%RH, No condensation
	Chassis	1U
	Product Dimensions (L×W×H)	482.4mm×260.5mm×44mm
	Net Weight	3kg
	Package Dimensions (L×W×H)	590mm×430mm×110mm
	Package Weight	3.5kg

Chapter 3 Interface Description

3.1 Front Panel



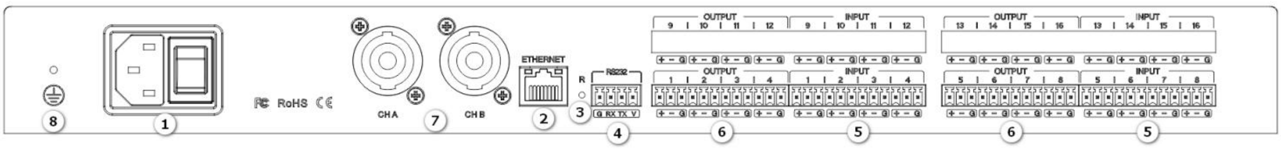
① PWR: Power indicator, the indicator light is always on to indicate that the device is powered normally;

② SYS: System operation indicator, the indicator will flash once per second to indicate that the system is operating normally;

③ USB: USB interface, function optional, support U disk recording and playback, or USB audio card, please refer to the actual device;

④ Amplifier Volume Adjustment Knob: Adjust the volume of CHA and CHB amplifier outputs respectively.

3.2 Rear Panel



① POWER: Power supply interface, connect AC 100V-240V power supply, warp switch control processor power;

② ETHERNET: Network control interface, through the connection of this network port, the client computer can debug and monitor the device;

③ RESET: Reset button, long press to restore factory settings and restart the processor;

④ RS232: Connect to the control terminal or central control device;

⑤ INPUT: Analog input interface, can be connected to mixer, microphone, PC and other devices;

⑥ OUTPUT: Analog output interface, can be connected to the amplifier, active speakers and other devices;

⑦ Amplifier output interface: Output CHA, CHB channel signals amplified by amplifier;

⑧ Ground screw: Used to ground the chassis, play accidental leakage safety protection, electrostatic balance and other protective measures.

Note: This manual takes 8-channel device as an example for illustration, please refer to the actual device for details.

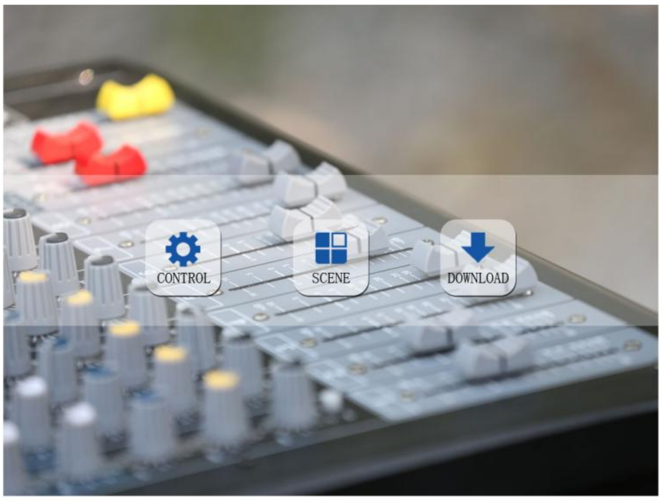
Chapter 4 Instructions for Use

4.1 Software Download

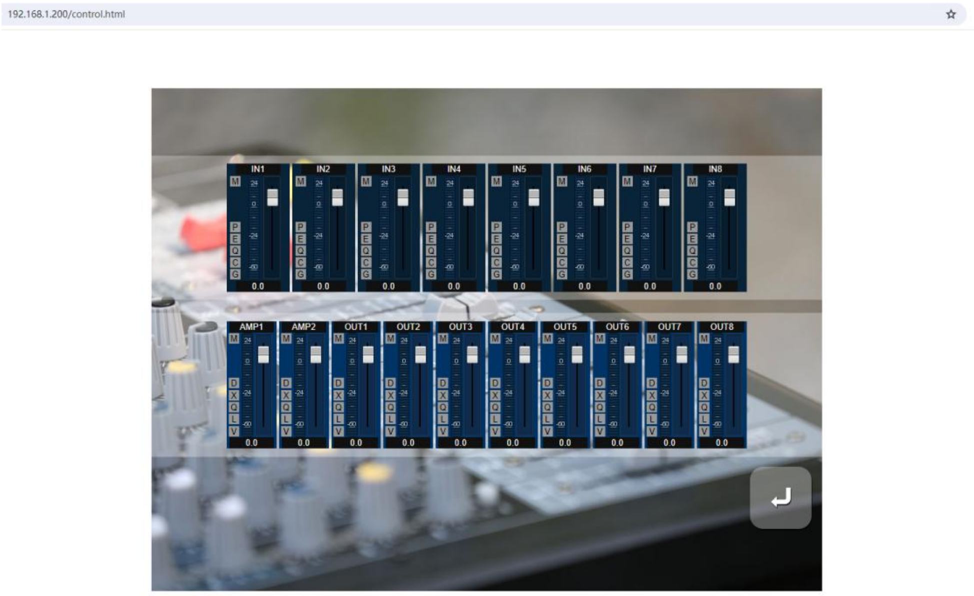
Installation software source files embedded in the Digital Signal Processor device, download the software simply by entering the device's factory default IP address (default IP: 192.168.1.200) information in the URL address bar of the browser, enter will be able to navigate to the download interface, according to the content of the web interface information click on the download of the software can be, in addition to pay attention to the installation of software before the PC side of the PC, please make sure that PC clients have been Net Framework3.5 or above for Windows system.

Note: Make sure the PC client is in the same network segment as the device IP address (default IP: 192.168.1.200 subnet mask: 255.255.255.0) when you download the software, otherwise you will not be able to access it.

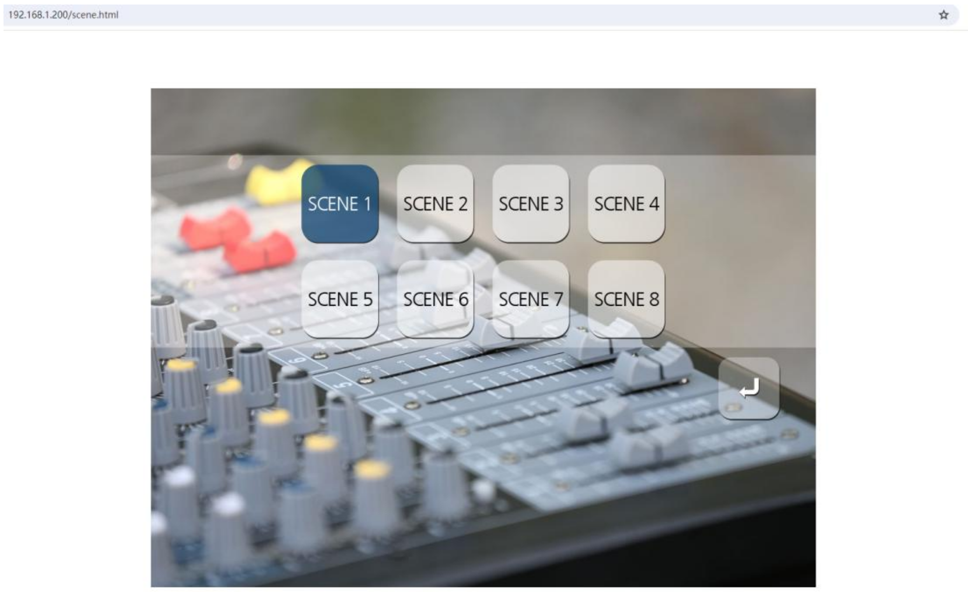
Web page controls as well as downloads:



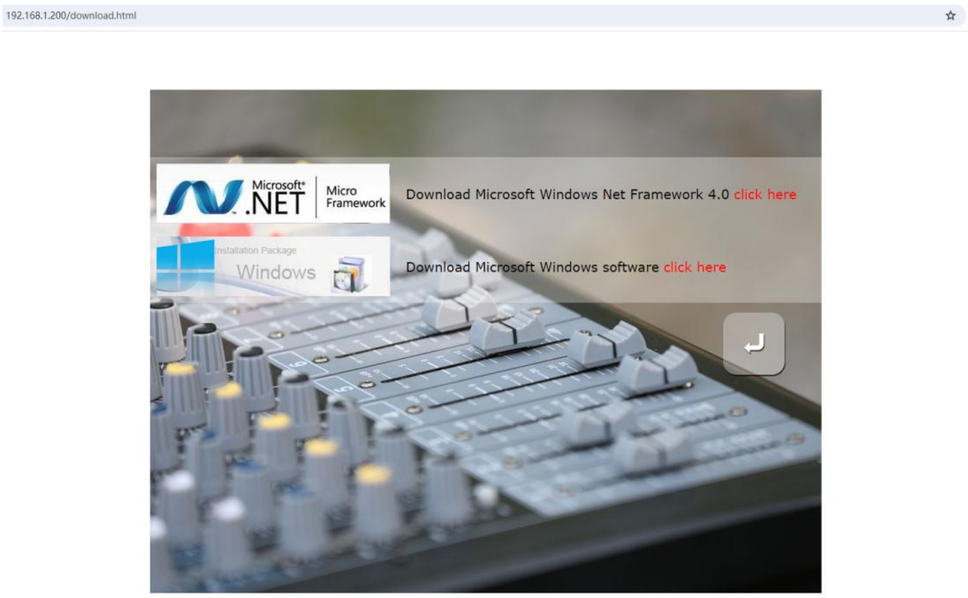
Web CONTROL:



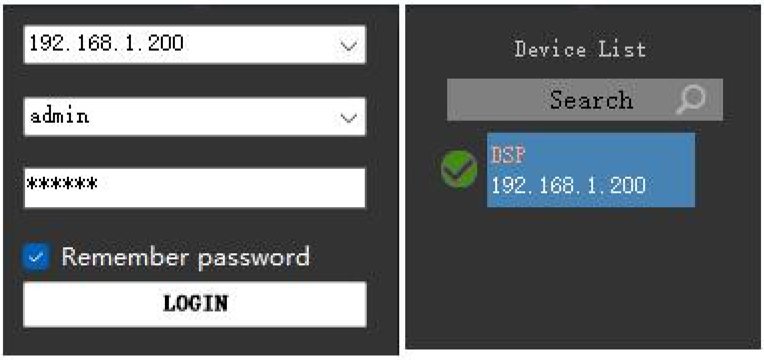
SCENE:



Software DOWNLOAD:



4.2 PC Software Login Connection



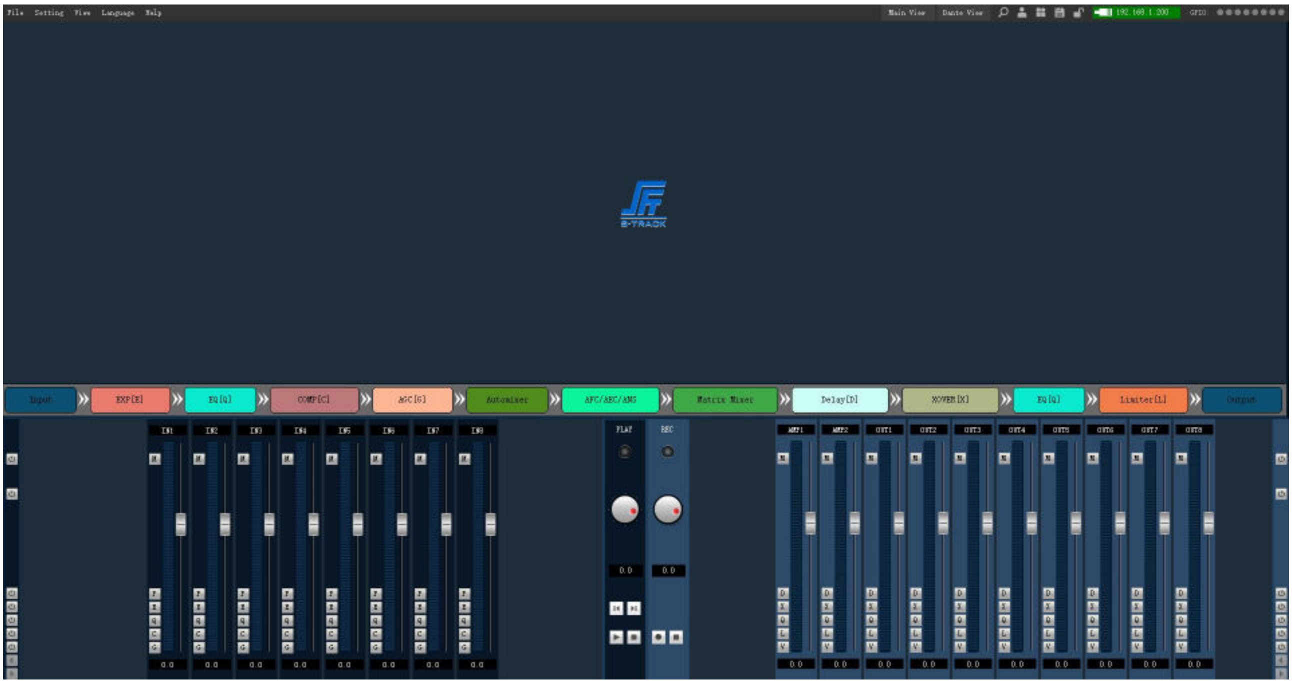
4.2.1 Login

Click the [Search] button, when the device list is refreshed, the online device [Model Name] will be displayed in the list column, double-click the corresponding device [Model Name] in the list column to bring up the 'left figure' login box, enter the user name/password input box (default user name: admin, password: 123456), click the [LOGIN] button to complete the login connection to the device. After successful connection, the status bar of the software will display the user name and IP address of the connected device.

4.2.2 IP Address Modification

When the IP information of the device is not in the same network segment as the client, an 'exclamation mark' will appear in front of the model name of the device in the list of settings, at this time, you only need to double-click on the [model name] to bring up the IP information modification [dialogue box] and then you can modify the IP address.

4.3 Main Interface



① Menu Bar;

② Button Function Area;

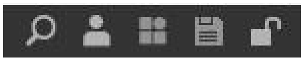
③ Input...Output: Audio Processing Modules;

④ IN1...IN8: These are the analog input channels;

⑤ AMP1, AMP2: These are the amplifier output channels;

⑥ OUT1...OUT8: These are the analog output channels.

4.3.1 Button Function Area



① Button: Device Search Button: Click to search for connectable devices and display device IP;

② Button: Device Connection Button: The IP of the device is known, and you can connect directly by entering the IP address, user name and password in the pop-up box;

③ Button: Scene List Button: You can select and view the saved scenes through the list;

④ Button: Save Scene button: saves (overwrites) the parameter changes to the selected scene;

⑤ Button: Interface lock button: locks the current interface, which must be unlocked with the administrator password.

4.3.2 File

① New: Create a new scene, the parameters are factory configured and only available offline.

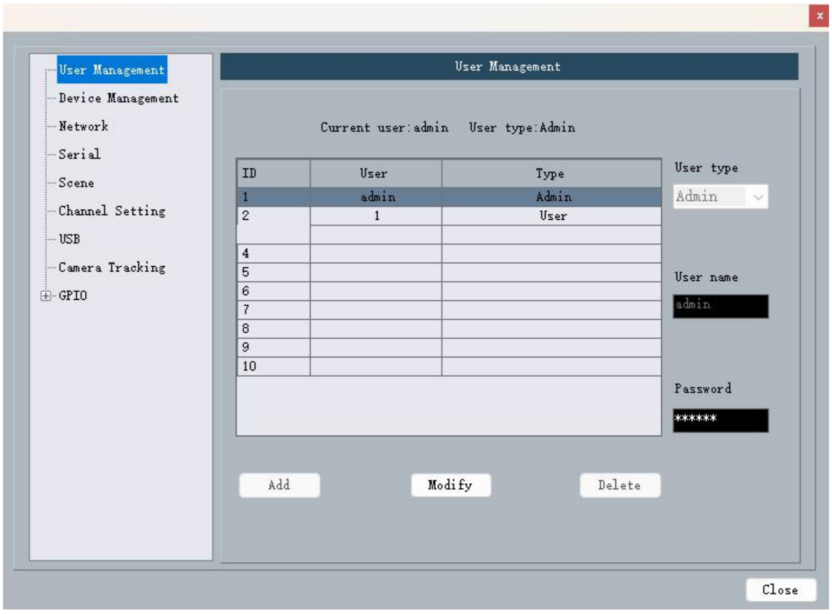
② Open: Open the locally saved scene.

③ Save as: Save the current configuration (i.e. scene) as a file locally.

④ Exit: Close the software.

4.3.3 Setting - Device Setting

I. User Management



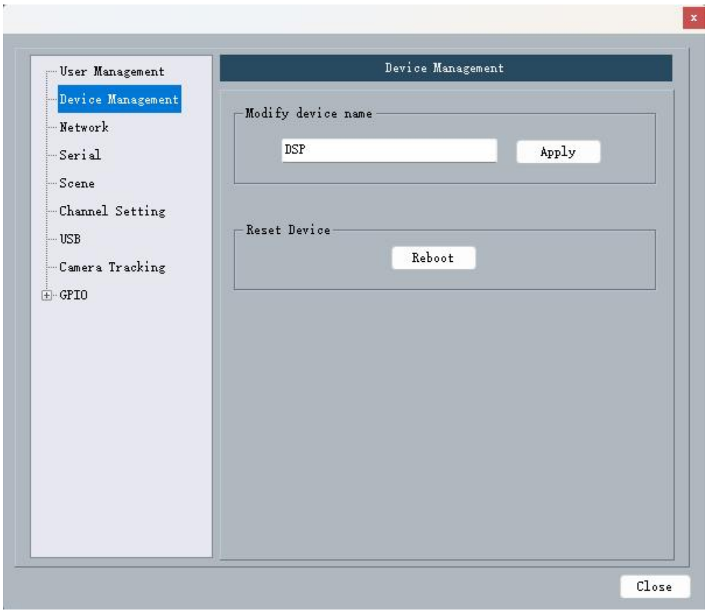
① Initial user name of the device: admin, password: 123456, administrator can add, delete, modify all user information, ordinary users can only modify personal information.

② Add User: Select an empty line in the left list, and enter the new user's information in the right user name and password edit box (should be empty), click "Add" button to add a new user.

③ Modify User: First select the user you want to modify in the user list, the user name and password edit box will display the information of the currently selected user, enter the new information and click the "Modify" button.

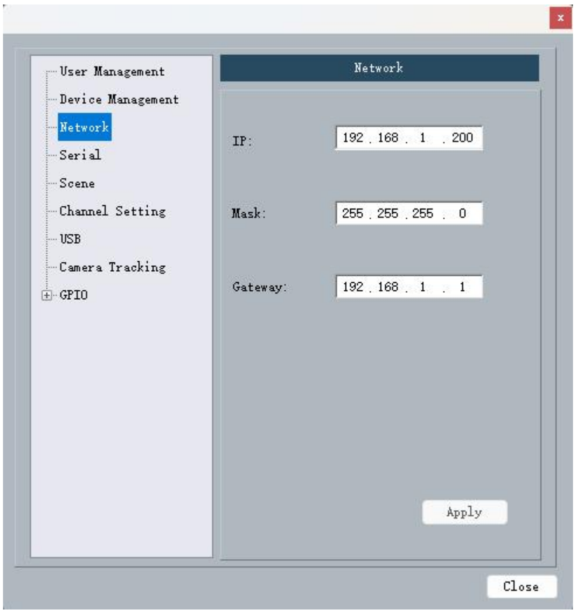
④ Delete User: Select the line in the user list to be deleted, click "Delete" button to delete the user.

II. Device Management



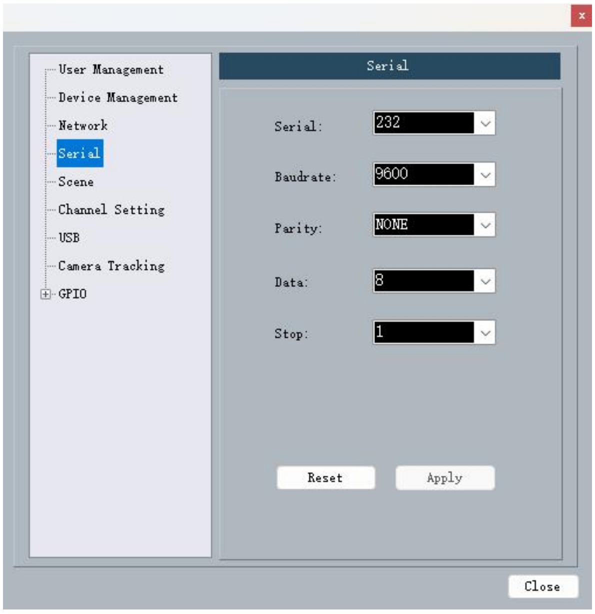
View and modify the name of the device, enter the name of the device in the corresponding location, click the [Apply] button to complete the modification, [Reboot] can be in the window to control the device restart.

III. Network



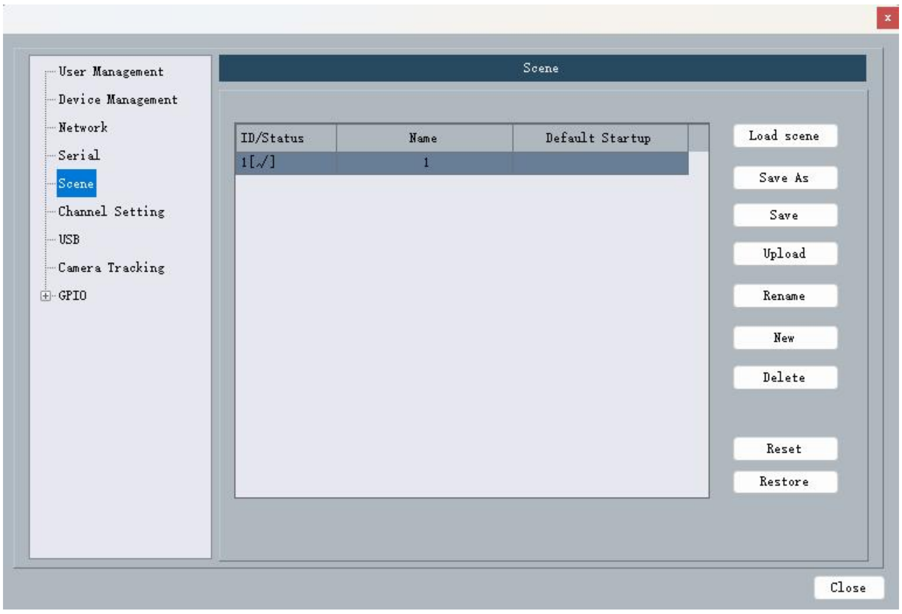
To view and modify the network address information of the device, enter the Network window, type in the IP address, subnet mask and gateway in the corresponding input boxes, and click the [Apply] button to complete the modification.

IV. Serial



Serial window can view or modify the RS232 serial port number baud rate, check bit, data bit, stop bit settings are complete, click on the [Apply] button to modify the current device's serial port information, such as the need to restore to the initial default value, click on the [Reset] button, you can restore the settings of the parameter can not be empty.

V. Scene



① Load Scene: Enable the currently selected scene, usually used for scene replacement;

② Save As: Save the selected scene locally;

③ Save: Save the currently running parameters to the selected scene;

④ Upload: Upload the scene from PC and overwrite the selected scene;

⑤ Rename: Modify the name of the selected scene;

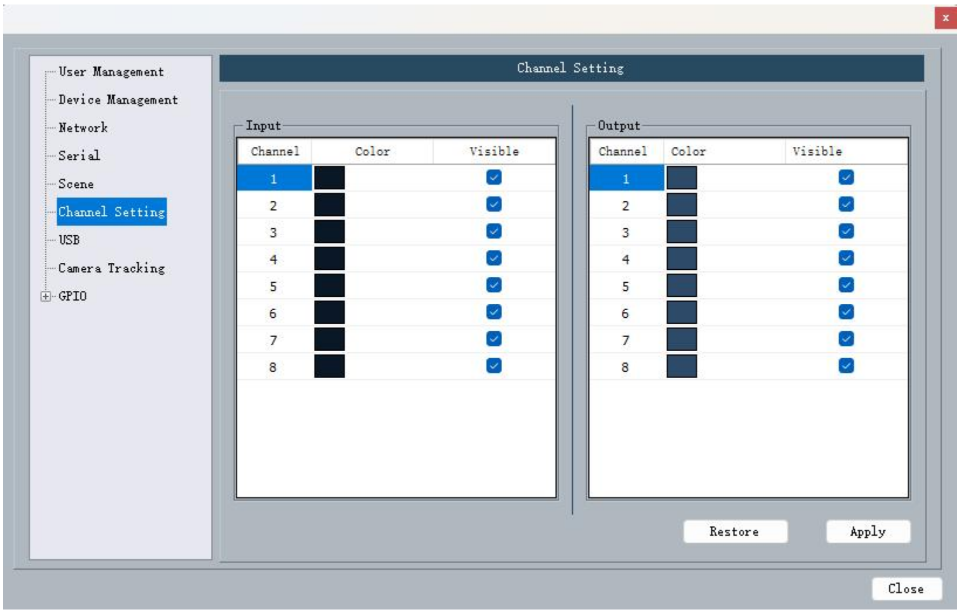
⑥ New: Create a new scene file, you can customize the new scene, support up to 100 groups of scenes;

⑦ Delete: Select any scene in the list and delete it;

⑧ Reset: Restore the currently selected scene to the factory default state;

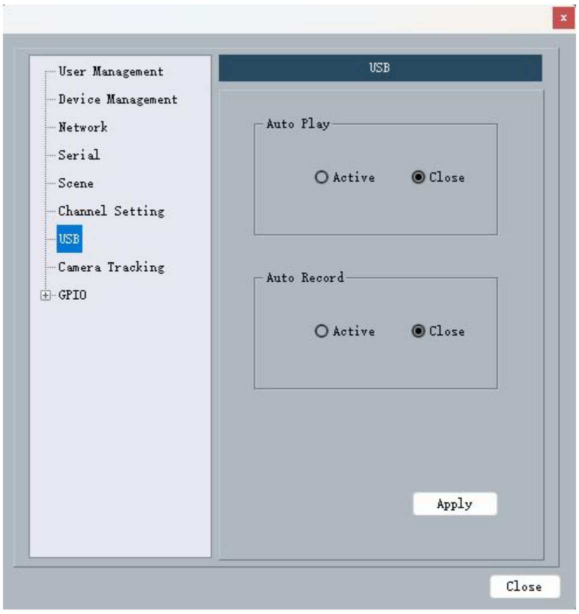
⑨ Restore: Restore all scene configurations to the default configuration and clear all new scenes, only retain the factory default 8 groups of scene files, please use with caution.

VI. Channel Setting



Users can customize the input/output channel UI colour scheme according to the usage scene.

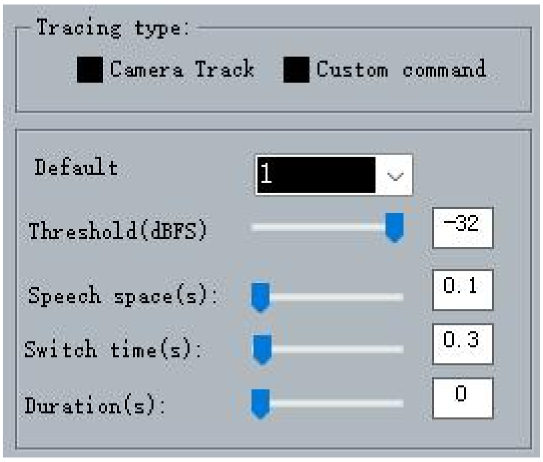
VII. USB



Users can customize to enable or disable the automatic playback/recording function according to the use of the scene, configure the playback channel or recording channel matrix routing in the matrix mixing interface and save the current scene, the device will automatically playback and record after powering on the device again, no need to manually configure separately.

VIII. Camera Tracking

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



① Tracking type: There are camera tracking and custom commands. Camera tracking is used to control camera rotation by channel input signal; custom command sending is used to send corresponding custom commands to the corresponding port by channel input signal.

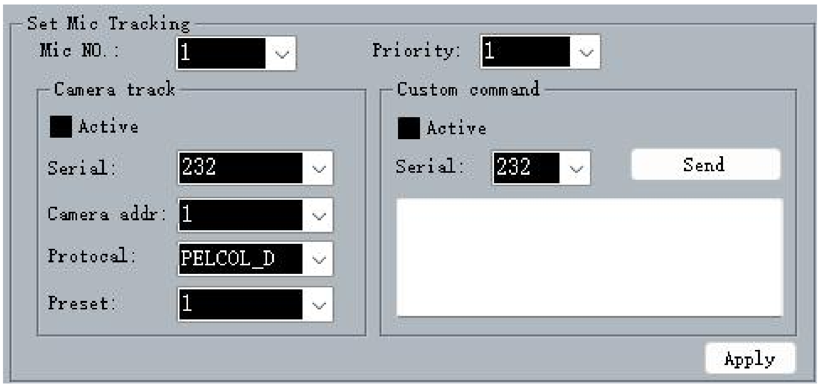
② Default: When there is no input from all microphones, rotate the camera to the position set by the default Mic or send the associated command defined by the default Mic. The one with # sign indicates the virtual number, which can only be used to set the default microphone.

③ Threshold: Means the detected input signal must be greater than or equal to the Tracking Threshold, and the system automatically enables the tracking parameter.

④ Speech space: Maximum intermittent time for a valid signal. If you use the microphone to speak, set the response time to 3 seconds, the signal is still regarded as continuously valid within 3S pause in the middle of the speech, and the signal is regarded as invalid if it exceeds 3S.

⑤ Switch time: The shortest speaking time required for the camera to switch to a valid position. 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 Duration: The interval time between sending camera switching commands or custom commands, such as 0 means special treatment, only triggered once.



⑦ The microphone number generally corresponds to the input channel of the device, i.e. it is the channel number to which the microphone is connected. The microphone number with # is a virtual number, which can only be used to set the default microphone.

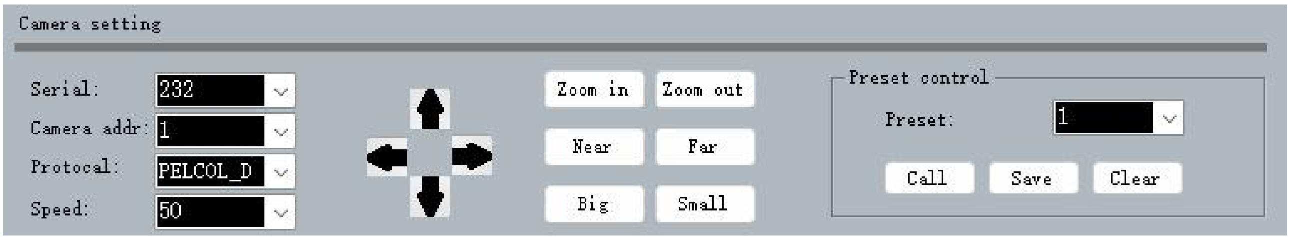
⑧ The smaller the priority number is, the higher the priority level is. When the priority level is the same, it will be processed in accordance with the triggering priority order; for example, if two microphones are speaking at the same time, the camera will automatically rotate to the preset bit corresponding to the microphone with the small priority number (i.e., the high priority level) or send the command corresponding to the microphone with the small priority number (i.e., the high priority level); however, if the two microphones are with the same priority level, the signal that is checked first will prevail.

⑨ Enable this Mic setting: you can set all the microphone parameters in full in advance, but when you use it, only some of them will be enabled according to the actual situation.

⑩ Preset points, serial port numbers, camera addresses, protocols and camera-related, must correspond to the actual connection of 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 secondly, you can also pre-set the command, but do not check "Enable Custom Command", the device will not send it automatically, but you can still click the "Send" button, and the command in the input box will be sent to the specified serial port at any time.

⑫ Click on "Save" to save the parameters to the device, so that the microphone for the channel is now 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 Setting is a camera debugging interface, generally debug the camera position before tracking starts, and finally the parameters of this part will be saved on the camera.

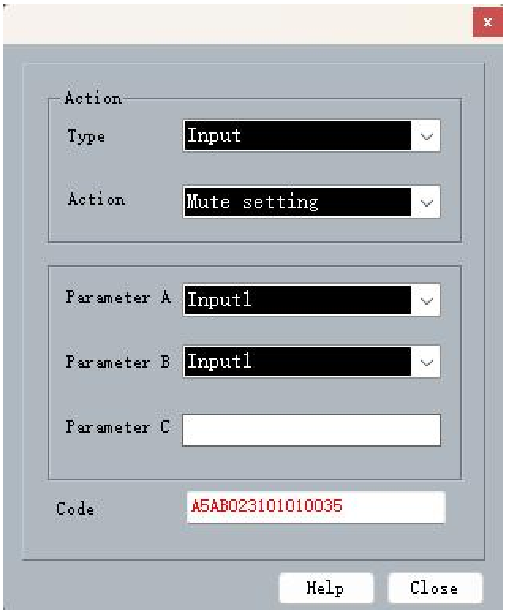
⑭ Firstly, serial port setting, there are 2 serial ports (232, 485), which correspond to the back panel port that the PTZ is connected to;

⑮ Next is 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 Lastly the preset point number is the user-defined identification for the camera, and then the adjustment of 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 to delete the information of the current preset point, "Recall" is used to view the camera saved by the current preset point. "Clear" is to delete the information of the current preset point, and "Recall" is used to view the camera position saved by the current preset point.

IX. Center Control



The Central Command Generator is able to convert frequently used operations into a 16-character command code for easy invocation by external devices.

Control command types: Scene, Input, Output, Mixer, Parametric Equalizer, Graphic Equalizer, Expander, Compressor, Automixer, Delay, Crossover, Limiter, AEC, ANS.

4.3.4 Viewing

① Open All: Open all function module interfaces to display them;

② Open Input: Open all the input function module interfaces and display them;

③ Open Output: Open all the output function module interfaces and display them.

4.3.5 Help

① Help: View the embedded user manual of the device;

② Upgrade: For updating the system software version;

③ About: Display software version information.

4.4 Pre-processing

4.4.1 Input Setting

The Analog Input provides line-level input for devices with line-level outputs, and inputs for microphones. The Analog Input converts the analog input signal to a processed digital and provides software controls before and after the convertor. Connections are made using one three-terminal 3.5mm Euro style connectors.

① Model type: Analog input or Test Signal;

② Sensitivity (24~-27): Analog signals can be adjusted by adjusting the sensitivity of the input can be selected, from 24 ~ -27dBu, step 3dBu a grade;

③ Mute: Mutes the input signal;

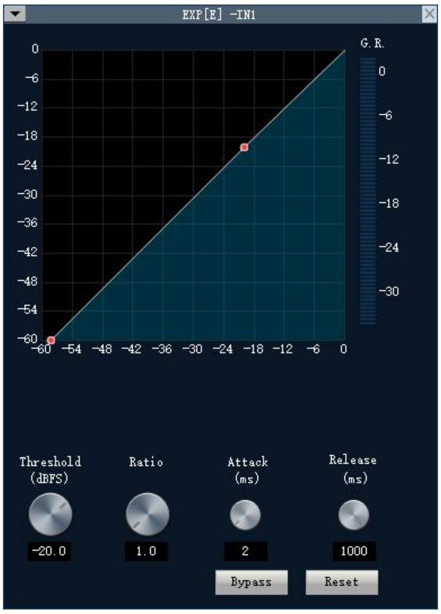
④ Invert: Inverts the polarity of the output signal;

⑤ Phantom: Toggle turning on and off phantom power (+48VDC) to the condenser microphone. Please do not turn on the Line Input or non-condenser microphone to prevent burning;

⑥ Test Tone: Including Sine, Pink Noise, White Noise, the system will automatically block the Analog input signal when the Test Signal is enabled.

4.4.2 Expander

The Expander is 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 will compress 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 will be 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.



① Threshold (-60~20): Sets the point from which the attenuation is calculated based on the Ratio setting. This is where the Expander starts working. Assuming only one Input, a level below the Threshold Level is attenuated, anything above the Threshold Level is not attenuated;

- **Example:**
- If the: Threshold Level is -15 dB; Ratio is 2.5; Input level is -25 dB
- Then the Adjusted Output is:
- [(Input Level - Threshold Level) * Ratio] + Threshold Level = Output Level
- {[-25 dB - (-15 dB)] * 2.5} + (-15 dB) = -40 dB.

② Ratio (1~20): The ratio between the Input and the Output as measured from the Threshold Level;

③ Attack Time (1~1000): The time required for an input signal less than the Expander Threshold Level to enter the expansion state and to output at the set expansion ratio;

④ Release Time (1~1000): The time required for the input signal level to return from the extended state to the original non-extended state;

⑤ Bypass/Active: Bypass or Active the Expander for the current channel; When the Expander is bypassed, audio is passed through without any change;

⑥ Reset: Resets the parameters to the default.

4.4.3 Parametric Equalizer

The Parametric Equalizer is a variable equalizer allowing you to individually adjust the Gain, Bandwidth and center Frequency of up to 5 frequency bands. You can also Bypass individual bands.

The Parametric Equalizer is mainly used to modify over-emphasised or missing frequency ranges. Whether the frequency range is narrowed or widened, the Parametric Equalizer can help to repair the narrowed frequencies or widen the frequency range to achieve the ideal signal tone.

① Bypass All/Active: Bypass All or Active the Parametric Equalizer for the current channel; When the Parametric Equalizer is bypassed, audio is passed through without any change;

② Reset: All band filter parameters are restored to the default;

③ Bypass (1~5)/Active (1~5): Bypass or Active the Parametric Equalizer for an individual frequency band; When an individual frequency band is bypassed, audio is passed through without any change;

④ Frequency (20~20000): Sets the center Frequency of an individual band;

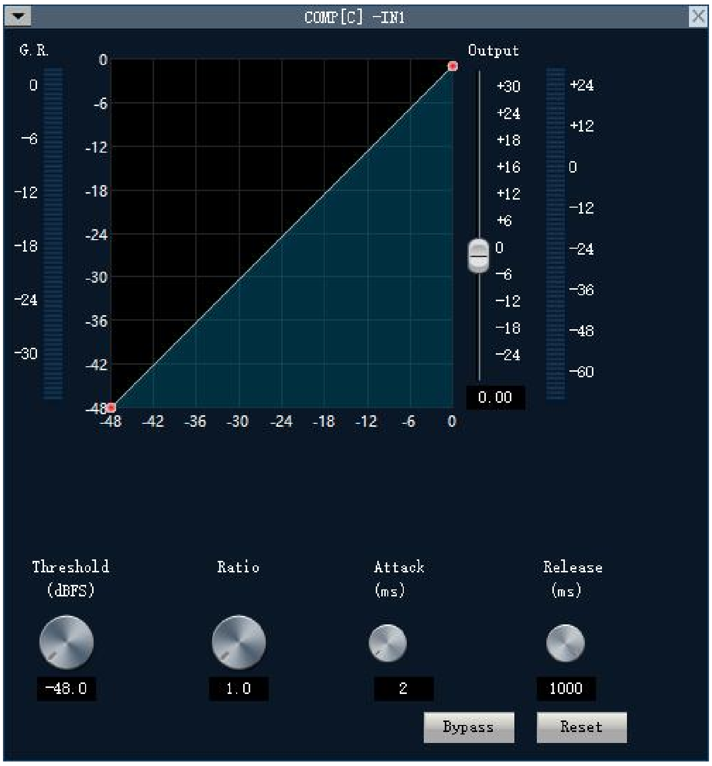
⑤ Gain (-15~15): Controls the Gain for an individual frequency band;

⑥ Bandwidth (Octave): Sets the bandwidth of an individual band of the equalizer, from 0 octaves to 4.00 octaves (default is 1.00). This is not active when either Low-shelf or High-shelf Type is selected. Adjusting bandwidth generally adjusts Q-Factor in an inverse manner.

4.4.4 Compressor

The purpose of the Compressor is to control the dynamic range of the Output above a set Threshold Level.

The Compressor can be adjusted from unity (1:1) with the Input, to an almost flat (20:1 - very little amplitude variation) Output.



① Threshold (-48~0): Sets the level where compression begins. This is the point from which the amount of attenuation is calculated based on the Ratio setting. Assuming only one Input, a level below the Threshold Level is not compressed, anything above the Threshold Level attenuation is applied;

- Example:
- If the:Threshold Level is -15 dB; Ratio is 2.5; Input level is 10 dB
- Then the Adjusted Output is:
- [(Input Level - Threshold Level) / Ratio] + Threshold Level = Output Level
- {[10 dB - (-15 dB)] / 2.5} + (-15 dB) = -5 dB.

② Ratio (1~20): The ratio between the Input and the Output as measured from the Threshold Level. The closer the Ratio is to 20, the smaller dynamic changes in the Output level. As the Ratio is adjusted closer to 1, the dynamic range of the Output increases;

③ Attack Time (1~1000): Time between when the input level reaches the threshold and when the compressor is activated;

④ Release Time (1~1000): Time between when the input level is less than the threshold and when the compressor stops working completely;

⑤ Output Gain (-24~30): Controls the Gain of the output;

⑥ Bypass/Active: Bypass or Active the Compressor for the current channel, When the Compressor is bypassed, audio is passed through without any change;

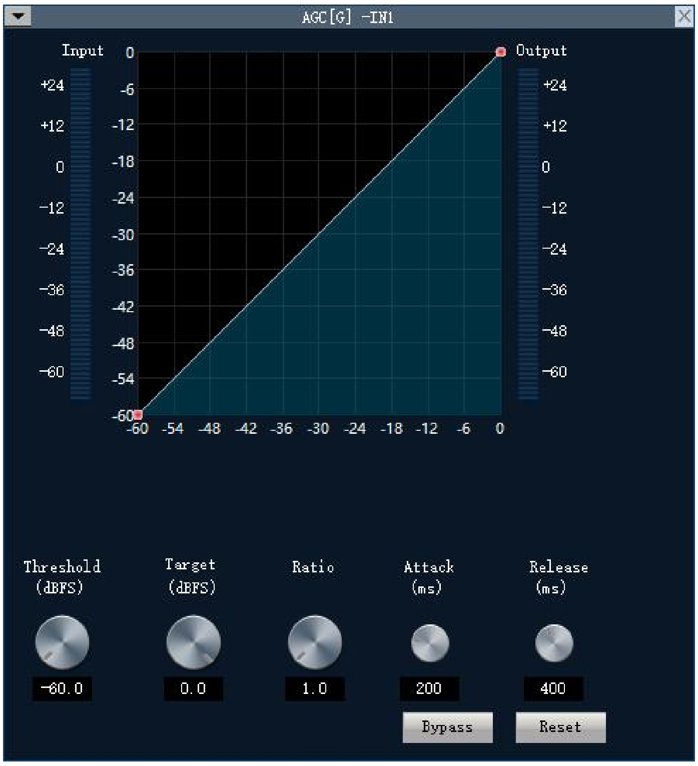
⑦ Reset: Resets the parameters to the default.

4.4.5 Automatic Gain Control

The purpose of the Automatic Gain Control is to control the overall dynamic range of the Output when the Input level changes.

The AGC can be adjusted from unity (1:1) with the Input, to an almost flat (20:1 - very little amplitude variation) Output. The AGC automatically adjusts the Gain to the Target Level by compensating for low Inputs, compressing for high Inputs.

Typical use: For example, when the user speaks in front of the microphone, the distance between the mouth and the microphone will be far away and close to each other, which will cause the output volume to go up and down, or even feel that the speech is intermittent. AGC is to set the threshold value for the input signal below the threshold in accordance with the ratio of 1:1 output, for the level above the threshold is in accordance with the ratio of direct enhancement, set the target level, the sound signal can be stable output.



① Threshold (-60~0): The level at which the AGC Component becomes active. This should be set at a level so that the anticipated background noise does not activate the AGC. When an Input exceeds the Threshold Level, the Gain for that input is adjusted to calculated level based on the Ratio and the Target Level;

② Target Level (-60~0): Sets the point from which the Gain is calculated based on the Ratio setting. Assuming only one Input, a level below the Target Level has a positive Gain applied, a level above the Target Level has a negative Gain (attenuation) applied;

- **Example:**
- If the Target Level is -15 dB; Ratio is 2.5; Input level is 10 dB
- Adjusted Output is
- [(Input level - Target Level) / Ratio] + Target Level = Output Level
- {[10 dB - (-15 dB)] / 2.5} + (-15 dB) = -5 dB.

③ Ratio (1~20): The ratio between the Input and the Output as measured from the Target Level. The closer the Ratio is to 20, the closer the Output will be to the Target Level, which also means smaller dynamic changes in the Output level. When the Input is below the Target Level and the Gain applied by the Ratio setting is greater than the Maximum Gain setting, the Output is clipped per the Maximum Gain setting;

④ Attack Time (1~1000): Controls the response time of signals above the threshold level;

⑤ Release Time (1~1000): Controls the response time of signals below the threshold level;

⑥ Bypass/Active: Bypass or Active the AGC for the current channel. When the AGC is bypassed, audio is passed through without any change;

⑦ Reset: Resets the parameters to the default.

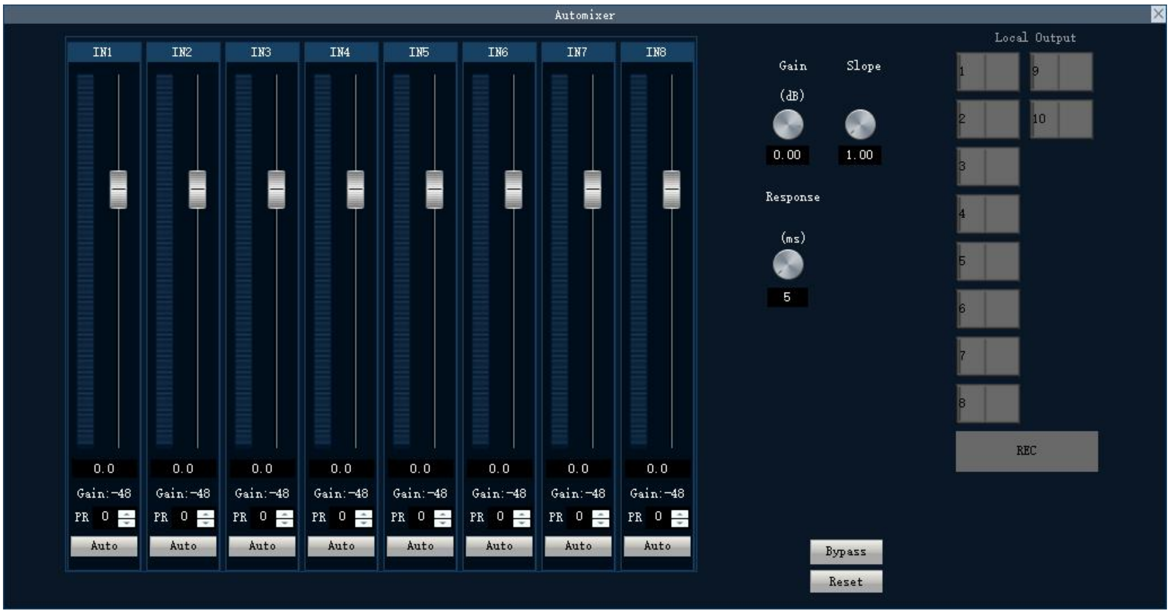
4.4.6 Automixer

Automixer is a mixer optimized for solving the problem of multiple live microphones operating together as a system – as found in boardrooms, classrooms, churches, courtrooms, etc. The Automixer controls the live microphones by turning up microphones when someone is talking and turning down microphones that are not used; thus, it is a voice-activated, real-time process without an operator. The Automixer controls the additive effect of multiple microphones being on at the same time and adapts to changing background noise conditions, thereby maintaining the natural ambience of the room.

The Automixer is implemented as a gain-sharing type. The gain of each microphone input is calculated as the ratio of its RMS level to the combined RMS levels of all inputs. This insures unity system gain at all times.

Consider a typical conference room scene with ten participants, each with a microphone, if ten microphones are switched on at the same time and only one person is speaking as a result, then the output will definitely not be ideal as the other nine microphones pick up room acoustics, reverberation, etc., which will reduce 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 Automixer. Channel Mute and Fader are both Auto Gain types. To mute a signal and prevent it from going into the Automixer, turn Mute on and off. The channel fader controls the mix level and direct output level of the channel.



① Priority Gain (-72~12): Adjusts the gain or attenuation applied to a particular Input channel. For example, you can boost the gain of a mic being used by a soft spoken person and attenuate the gain of a louder person so they can be heard equally in the room;

② Priority PR: Selectable Priority that adds the Priority Gain to the automix function of the corresponding input – this gives the corresponding input more share of the available gain, but does not change the level of the actual input. The higher the value the more gain acquisition capability and the higher the priority, able to reduce the ability of channels with a low priority level to acquire transmission gain. This control defines the priority level with a value between 0 (lowest priority) and 10 (highest priority), and the default value is 5 (standard priority). If all channels have equal priority, set the priority of all channels to 5;

③ Auto: Each channel has an automix Enable/Disable button, which should be turned on for channels that need to participate in automix. It can also be turned off so that the channel does not participate in the automix;

④ Gain (-72~12): Adjusts the Mix output channel gain. This applies to all Input channels on the Mix output, but does not affect the individual output channels;

⑤ Slope: The slope control affects the attenuation of lower levels. At higher slopes, channels with lower levels are also attenuated more. Ratio similar to Expander, the larger the slope, the more the speaking microphone acquires the transmission gain, the more the non-speaking microphone attenuates the transmission gain; the smaller the slope, the less the speaking microphone acquires the transmission gain, and the less other non-speaking microphone attenuates;

⑥ Response Time: A master Response control adjusts the speed at which the mixer changes its automatic gain. The time for a microphone to acquire all the transmission gain when it is speaking or the time for other non-speaking microphones to attenuate the transmission gain. The longer the setting, the longer it takes for the speaking microphone to acquire the full transmission gain and the longer it takes for the other non-speaking microphones to attenuate the transmission gain. The shorter the setting, the opposite is true;

⑦ Local Output: Selects the channel for local output;

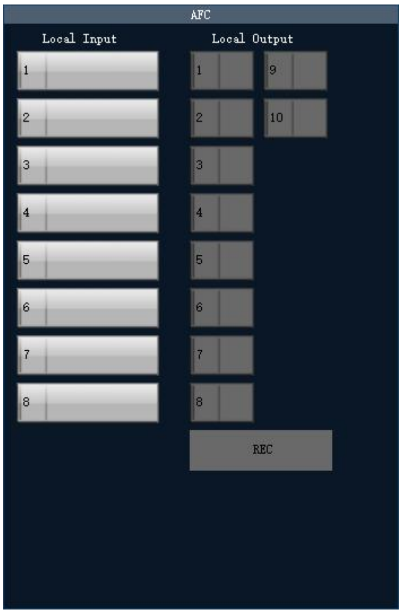
⑧ Bypass/Active: Bypass or Active the Automixer. When the Automixer is bypassed, audio is passed through without any change;

⑨ Reset: Resets the parameters to the default.

4.5 AFC, AEC, ANS

4.5.1 AFC

The Acoustic Feedback Canceler (AFC) is used to cancel the whistling generated between the microphone and speakers in the sound reinforcement system, thus capturing the frequency that causes the whistling for attenuation to ensure the quality of the sound as well as to prevent burning out of the amplifier or speakers.



Feedback: Selects the signal that needs to be processed by the Acoustic Feedback Canceler, and the processed signal selects the output channel in the mixer;

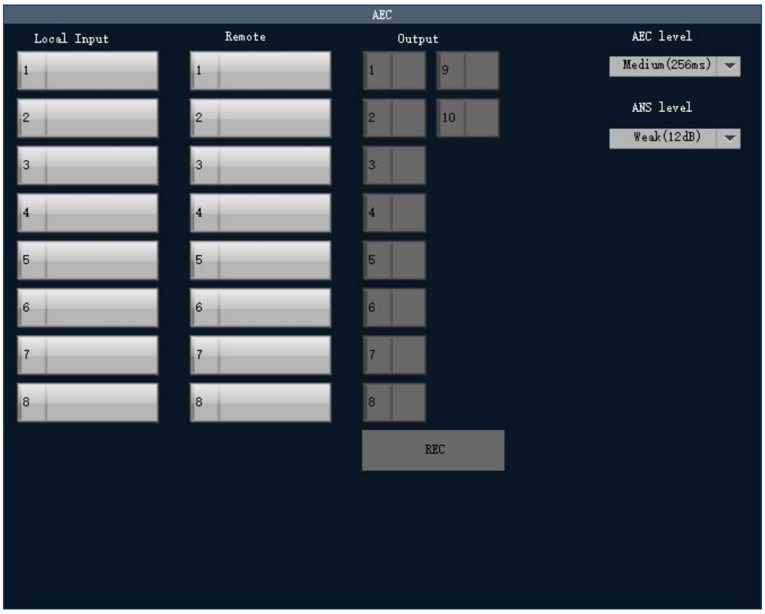
① Local Input: The channel for local Mic output, i.e. the signal that needs to be processed by the Acoustic Feedback Canceler;

② Local Output: The signal processed by the Acoustic Feedback Canceler, output to the local output channel.

4.5.2 AEC

The Acoustic Echo Canceler (AEC) is used in conference rooms (Near-End) and other installations where people call in from remote locations. The remote caller's (Far-End caller's) voice is broadcast over loudspeakers in the conference room. The sound is picked up by microphones in the conference room and echoed back to the Far-End caller. The purpose of the AEC is to eliminate these echoes while at the same time allowing the Far-End caller to hear clearly, what people in the room are saying.

Each microphone in the conference room is plugged into one channel of the AEC. Each channel also receives the loudspeaker signal that carries the remote talker's voice. This is called the AEC reference signal. To remove the echoes, the AEC subtracts a filtered version of the reference signal from the microphone signal.



Acoustic Echo Canceler: Set the signal that needs to be processed by the Acoustic Echo Canceler, and the processed signal selects the output channel in the Mixer;

① Local Input: local Mic output channel, i.e. the signal that needs to be processed by the Acoustic Echo Canceler;

② Remote Input: The echo remote input, i.e. the reference signal;

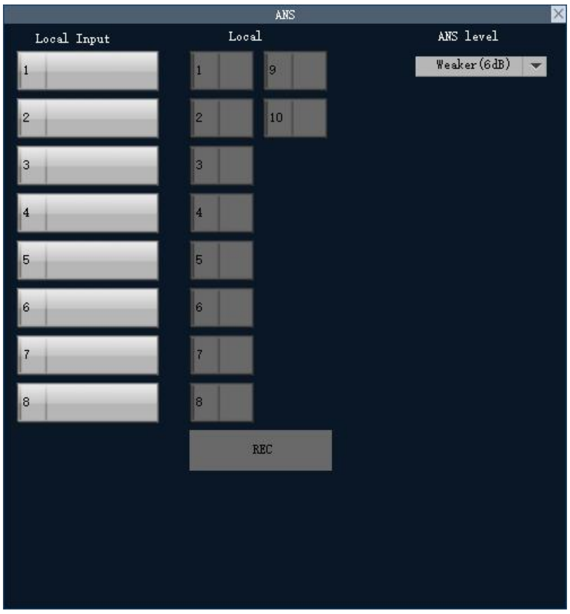
③ Local Output: The signal after Acoustic Echo Canceler processing is output to the local output or output to the remote end;

④ AEC level: Echo level range [small room (128ms), medium room (256ms), large room (512ms)];

⑤ ANS level: Noise reduction level range (6~30dB).

4.5.3 ANS

The Adaptive Noise Suppression (ANS) effectively removes non-vocal sounds. Distinguish between human voice and non-human voice, and treat non-human voice as noise. A piece of audio containing both vocals and noise is processed by this module, and theoretically, only the vocals remain.



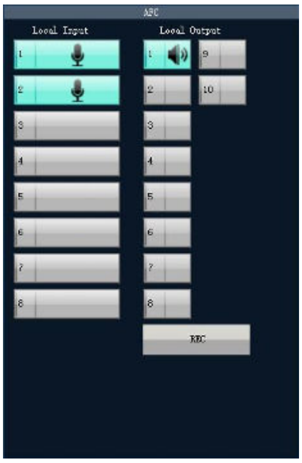
Adaptive Noise Suppression: Select the signal that needs Adaptive Noise Suppression processing, and the processed signal is output in the mixer by selecting the corresponding channel;

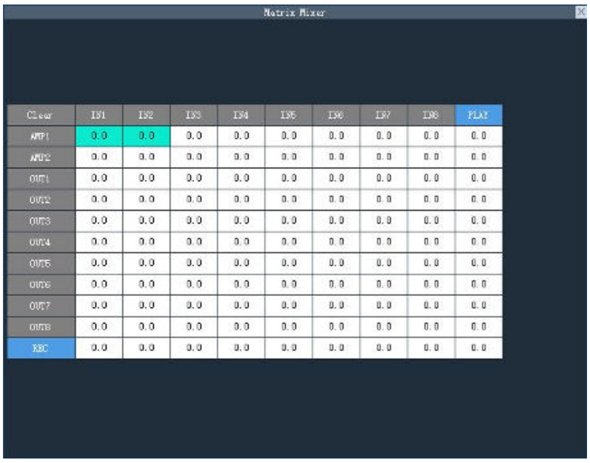
① Local Input: Local Mic output channel, i.e. the signal that needs Adaptive Noise Suppression processing;

② Local Output: The signal after Adaptive Noise Suppression processing is output to the local output channel;

③ ANS level: Range of noise reduction level (6~30dB).

Example 1 AFC with Matrix Mixer association operation



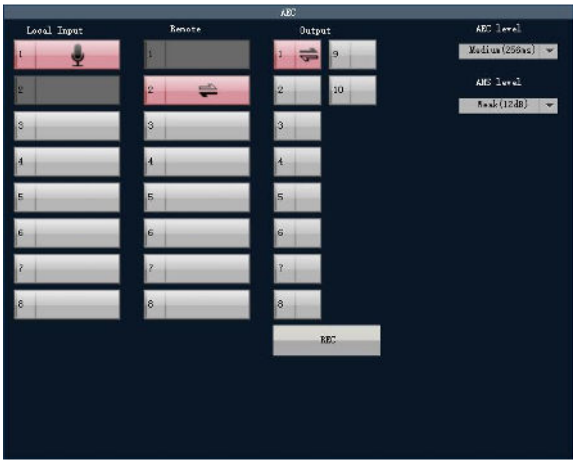


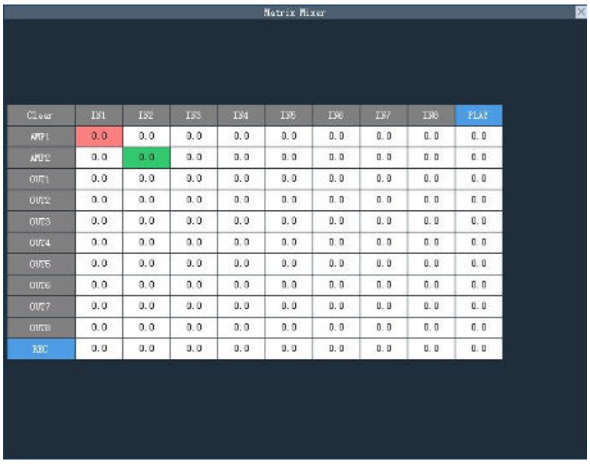
The signals from input channels IN1 and IN2 are processed in the AFC and output in output channel OUT1, configured as shown above:

① AFC window [Local Input] list selects the input channels IN1 and IN2, indicating that the signals of input channels IN1 and IN2 will be sent to the AFC for processing;

② The [Local Input] list selects the point corresponding to the output channel OUT1, indicating that the input signal is routed to the OUT1 channel for output after being processed by the AFC. After the AFC is enabled, the corresponding channel of the Matrix Mixer Window list is displayed in cyan.

Example 2 AEC with Matrix Mixer association operation



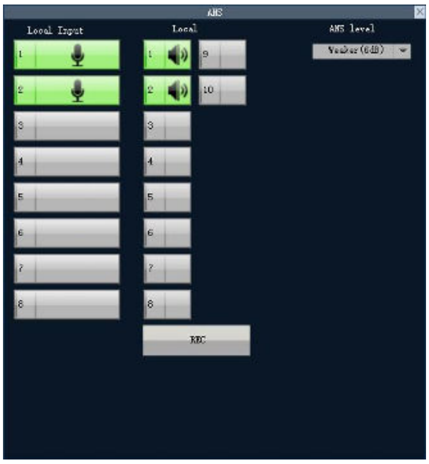


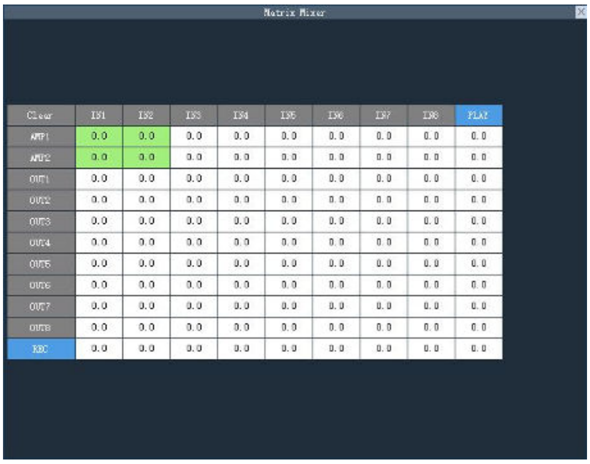
The local input signal is IN1 channel and the remote input signal is IN2 channel, at this time, the local input and the corresponding channel of the remote input will be grayed out to prevent the AEC from being activated abnormally due to the checking, and the local input signal will be AEC from OUT1 channel to the remote place, and the configuration will be as described in the above figure:

① AEC window to select the local input IN1 channel and then select the remote input IN2 channel, that is, the remote input IN1 channel and the remote input IN2 reference signal will be sent to the AEC for processing;

② The local input IN1 signal is then output to the remote end through the OUT1 channel, and the remote input IN2 reference signal is then routed through the matrix to the OUT2 channel for output to the local loudspeaker.

Example 3 ANS with Matrix Mixer association operation





The local input IN1 and IN2 channel signals are processed for ANS and output on OUT1 and OUT2 channels, configured as above:

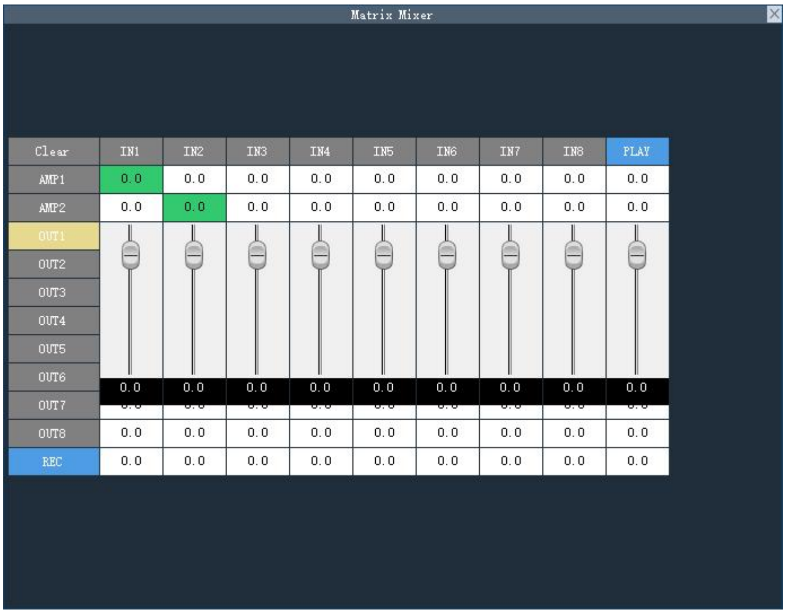
① Input IN1 and IN2 channels are selected in the ANS window, indicating that the input signals of input IN1 and IN2 channels are sent to the ANS for processing;

② Check OUT1 and OUT2 in the local output list, which means that the result of the ANS processing will be sent to the output OUT1 and OUT2 channels for output. After enabling the ANS, the corresponding channels in the Matrix Mixer window list are shown in green.

4.6 Matrix Mixer

The Matrix Mixer receives an audio input and distributes the audio signal to each Output channels.

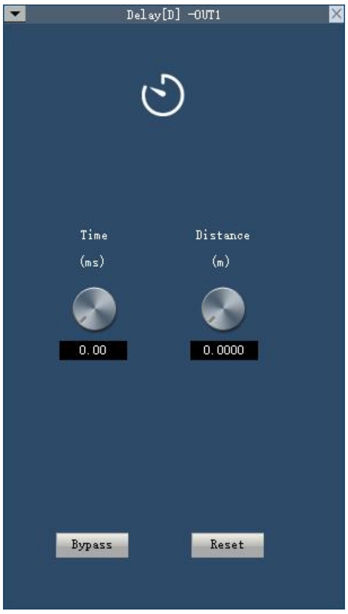
Matrix both signal routing and mixing double multiplexing function, X-axis is the input channel, Y-axis is the output channel, the matrix initialization state is (as shown in the figure above: one-to-one) input and output.



The OUT channel of the mixer is equipped with a gain fader, which allows you to individually control the gain of any output channel with a gain range of (12~-72dB).

4.7 Post-processing

4.7.1 Delay



The time interval between the input of a signal to this processor and the output of this processor is generally used to produce other effects such as reverberation or echo, and can equally be used to act as a treatment for auxiliary loudspeakers in larger rendition situations.

① Delay time: Delay time range (0~2000ms);

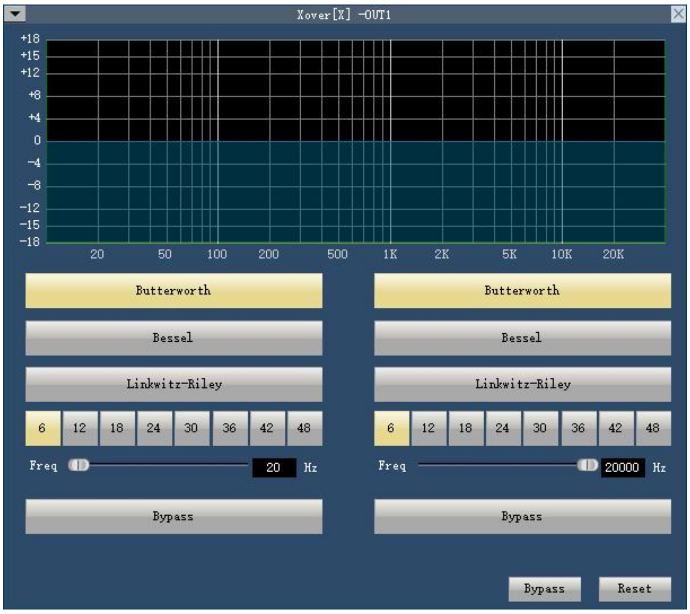
② Delay distance: Delay distance range (0~680m);

③ Bypass/Active: Bypass or Active the Delay for the current channel. When the Delay is bypassed, audio is passed through without any change;

④ Reset: Rests the parameters to the default.

4.7.2 XOVER

The XOVER divides the audio input signals into 3 frequency bands: low-pass, band-pass and high-pass, you can set the Slope rates and filter types (Butterworth, Linkwitz-Riley, Bessel) for each filter in each band.



① Filter Type: The Type setting includes: Butterworth, Linkwitz-Riley, Bessel filters. You can select any combination of two of these filters for a band's high-pass and low-pass frequencies;

② Slope: Determines the rate of change of attenuation at the high-pass and low-pass frequencies of the band. The slope setting establishes the crossover region between two adjacent bands. The Slope includes 6, 12, 18, 24, 32, 36, 42, 48;

③ High Pass frequency: High Pass filter cutoff frequency;

④ Low Pass frequency: Low Pass filter cutoff frequency;

⑤ High Pass Bypass/Active: Bypass/Active the High Pass filter;

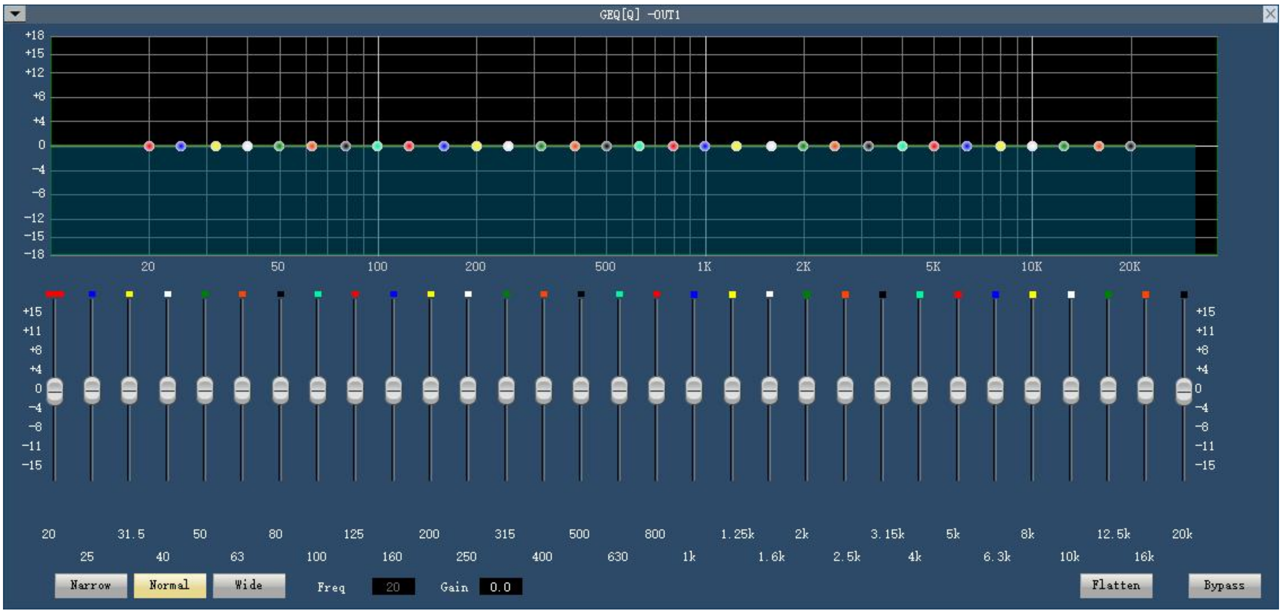
⑥ Low Pass Bypass/Active: Bypass/Active the Low Pass filter;

⑦ Bypass/Active: Bypass or Active the XOVER for the current channel. When the XOVER is bypassed, audio is passed through without any change;

⑧ Reset: Rests the parameters to the default.

4.7.3 Graphic Equalizer

The Graphic Equalizer Component provides from 31 bands with 1/3th octave per band. Each band can be adjusted from -18dB to +18 dB.



① Narrow: Narrowband equalization filter; Normal: Normal equalization filter; Wide: Wideband equalization filter;

② Freq.: Frequency indication of the current equalization filter;

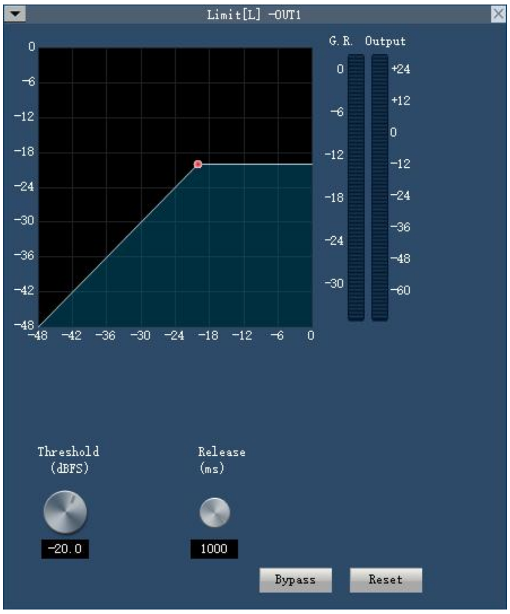
③ Gain: Gain indication or control of the current equalization filter;

④ Flatten: Rests all the band gains to the default;

⑤ Bypass/Active: Bypass or Active the Graphic Equalizer for the current channel. When the Graphic Equalizer is bypassed, audio is passed through without any change.

4.7.4 Limiter

The Limiter limits the output level to the Threshold Level, prevent signal overload and transient interference.When the input signal is above the threshold, the output signal is equal to the threshold; when the input signal is below the threshold, the output signal is equal to the input signal.



① Bypass/Active: Bypass or Active the Limiter for the current channel. When the Limiter is bypassed, audio is passed through without any change;

② Threshold (-48~0): Sets the level at which the Limiter has an effect, and the level at which the output is held;

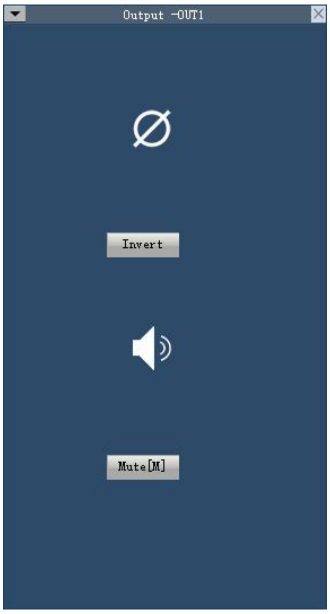
③ Release Time (1~1000): When the input signal falls below the Threshold Level, the sound channel is not turned off immediately, but is delayed for the Release Time. During this time, the sound channel stays on as long as there is a signal above the Threshold Level.

④ G.R.: Graphically displays the amount of attenuation applied to the Channel;

⑤ Reset: Resets the parameters to the default.

4.7.5 Output Setting

The Analog Output provides one channel of line-level output for device. The Analog Output converts the processed digital signal to analog and provides software controls before and after the convertor. Connections are made using one three-terminal 3.5mm Euro style connectors.



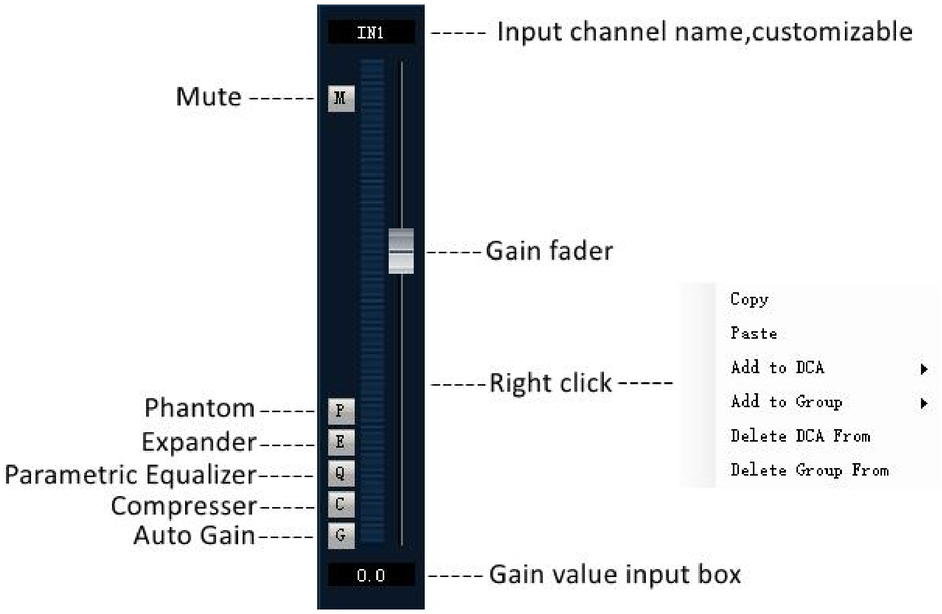
① Mute: Mutes the output signal;

② Invert: Inverts the polarity of the output signal.

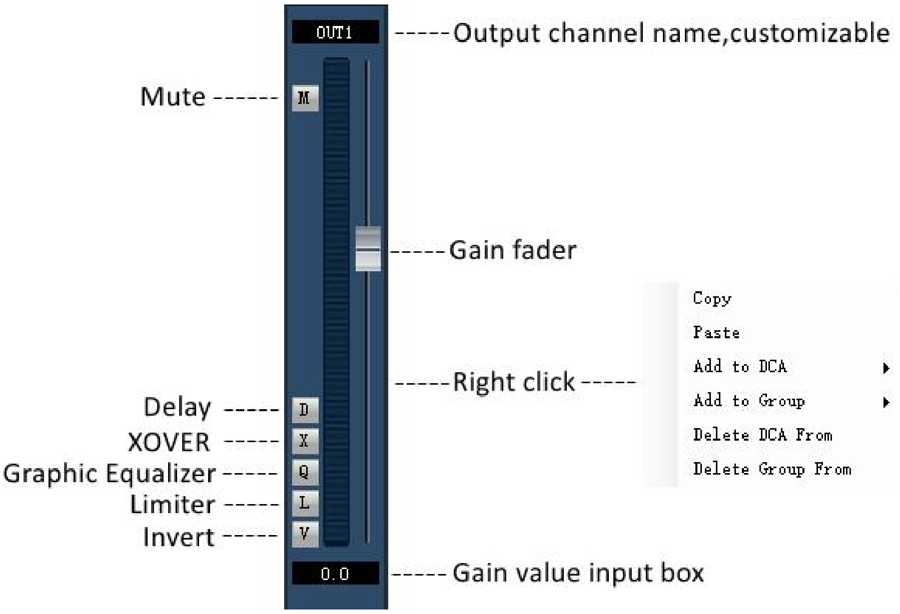
4.8 Other Functions

4.8.1 Channel Control

I. Input Channel Control and Shortcut



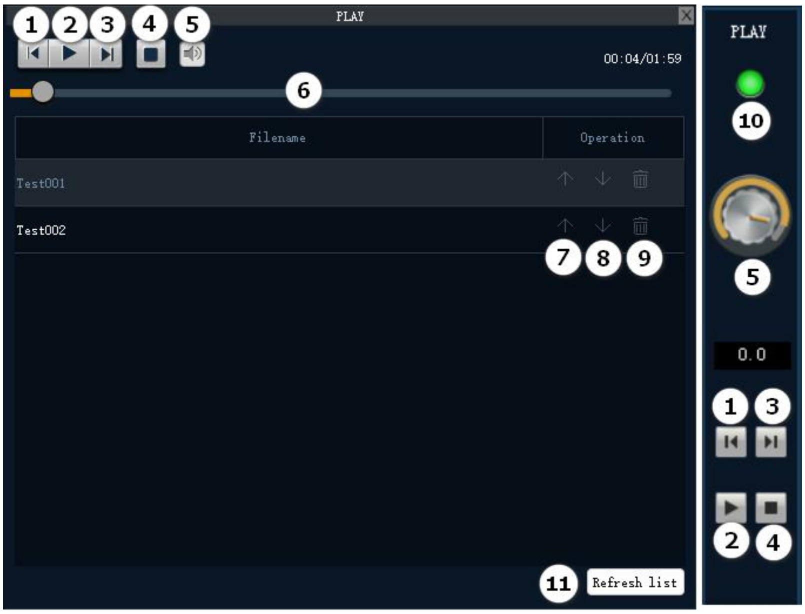
II. Output Channel Control and Shortcut



4.8.2 USB Playback and Recording

I. USB Playback

The USB Player component provides playback of audio files from a USB flash drive inserted into the USB A port of the device, supporting USB flash drive format FAT32 and audio file formats MP3 and WAV.



① Previous: Click this button to play the previous file in Play List order. If the current song is the first in order, this button is inactive;

② Play/Pause: The Play button starts playing the selected audio file from either the beginning, or from where it was paused; The Pause button temporarily stops playing the selected file. You can resume playing the file by clicking either the Play or Pause buttons;

③ Next: Click this button to play the next file in Playlist order. If the current song is the last in order, this button is inactive;

④ Stop: The Stop button stops playing the selected audio file. The file is reset to the beginning;

⑤ Controls the Gain of the output signal;

⑥ Countdown progress, HH:MM:SS;

⑦ Move upwards;

⑧ Move down;

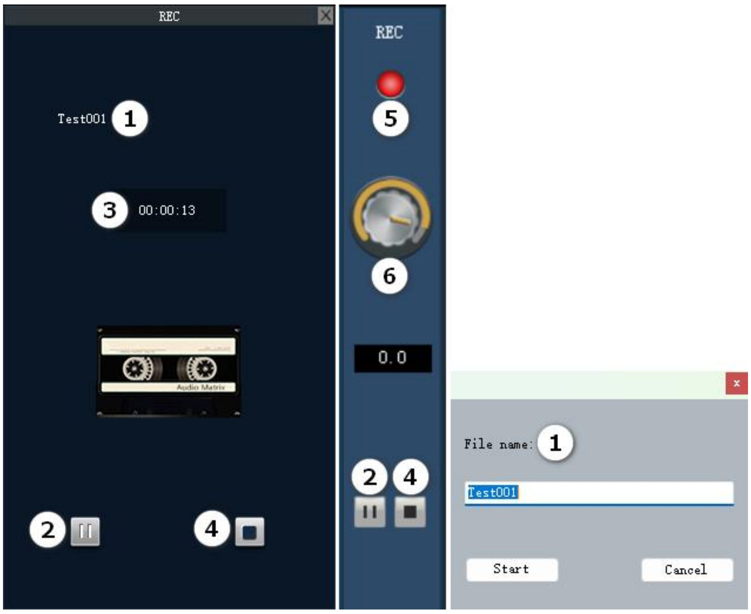
⑨ Delete the Play List files;

⑩ Playback indicator: Gray (not playing) Green (playing);

⑪ Refresh list.

II. USB Recording

The USB Recording allows you to record one channel of audio.



① File Name: Displays the name of the recorded file in date time, audio file format is WAV;

② Start/Pause the recording;

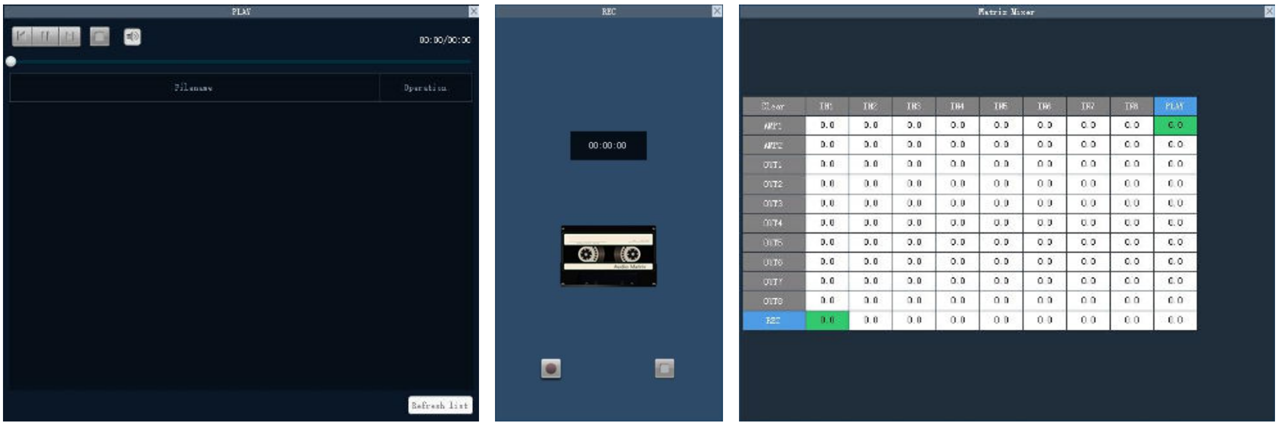
③ Displays the length of the recording. HH:MM:SS;

④ Stop the recording;

⑤ Recording indicator: Grey (not connected to the USB flash drive) Red (ready to record) blinking is considered to be recording;

⑥ Controls the Gain of the input signal.

III. USB Playback Recording Matrix Mixer Example



Playback Recording Example

When the device is connected to a mobile USB memory stick, return to the Matrix Mixer window and select OUT1 channel output under the corresponding input type [Play], then click the [Play] button to play it, and support audio file (MP3, WAV) format. Note: The USB flash drive audio file must be placed in the parent directory, otherwise the device will not be able to identify the audio file and thus can not be played;

Recording is through the input channel audio signal saved to the USB flash drive device, as shown in (above), select the IN1 channel audio signal pass for recording, matrix configuration such as (above) in the IN1 channel corresponds to the recording channel for the matrix routing, in the click on the [Start] button, the pop-up dialogue box to modify the name of the recording file can be selected to customize the name of the audio file, and finally click on the [Start] button to record.

Chapter 5 FAQ

5.1 Abnormal power indicator (PWR)

No light: First, check whether the power connection and device power supply are normal; Second, check whether the power switch on the rear panel of the device is on.

Blinking: Reboot the device.

If the problem is not solved, please contact the distributor or manufacturer.

5.2 Abnormal status indicator (SYS)

After 18 seconds of power-on, the system is working normally and the system light of the device should be flashing once per second.

Does not light up, often light up, or blinks rapidly:

First, system error, contact the after-sales service to upgrade the software version; Second, long press the reset button (R hole on the rear panel of the device) for more than 6 seconds, the device will restore the factory settings and restart automatically. If the problem is not solved, please contact the distributor or manufacturer.

5.3 Channel no sound

First, check whether the audio source, audio input and output wiring is normal; Second, check whether the mute function of the corresponding audio channel is enabled, if the mute switch has been turned on, please turn off the mute switch; Third, check whether the settings of the corresponding channel's input processing, matrix mixing and output processing are normal. If the problem is not solved, please contact the distributor or manufacturer.

5.4 The control software cannot search the device

First, check whether the system light of the device is in normal blinking state; Second, check whether the network connection is normal; Third, ensure the network accessibility between the configuration host and the device; Fourth, press and hold down the reset button (R-hole on the rear panel of the device) for more than 6 seconds, and the device will restore the factory settings and restart automatically. If the problem is not solved, please contact the distributor or manufacturer.

5.5 Network connection failure

Network connection failure is usually caused by different network segments of the device. If the LAN and the processor network segments are different, you can connect the processor directly via PC, login to the device configuration interface, change the processor network segment to be the same as the LAN and then access the LAN. (**Note:** If the LAN is automatically obtaining IP, please use 0.0.0.0 for the processor IP).

5.6 Current noise in output channel

Please check whether the processor is well grounded, which usually requires the grounding screw on the left side of the rear panel of the chassis to be connected to the metal enclosure such as the cabinet through a metal wire. If the problem still exists, please check the wiring of the input devices. If the input devices are unbalanced (two wires), please connect the "+" and "G" of the input connector of the processor.

5.7 How to recognize system noise

After the system is set up, there is noise troubleshooting: First, unplug the device output audio cable, there is noise, please check the causes of the back stage equipment; Second, restore the output wiring, mute the corresponding output channel, there is noise, if unbalanced connection, try to shorten the connecting line, to avoid the introduction of interference, if balanced connection, try to disconnect the ground wire; Third, cancel the corresponding channel mute, unplug the device input audio cable, there is noise, long press the reset button (R hole on the rear panel of the device) for more than 6 seconds, the device will restore the factory settings and restart automatically;Fourth, restore the input wiring, turn off the audio source, there is noise, check the input connection, refer to the second point of the processing; Fifth, check the audio source is there is noise. If the problem is not solved, please contact the distributor or manufacturer.

5.8 RS232 center control command does not work

First, check whether the connection is normal, the central control host TX connected to the device's RX, the central control host RX connected to the device's TX, the central control host and the device's ground interconnect; Second, check the software configuration of the device interface items: baud rate, start bit, stop bit, etc. Settings are the same as the interface configuration of the central control host. If the problem is not solved, please contact the distributor or manufacturer.

5.9 U disk audio can not be played

The device only supports FAT32 format U disk, only supports MP3 or WAV format audio files, audio files need to be stored in the root directory; ensure that the U disk is writable, the device needs to create a playlist. U disk has more than one partition only recognizes the first partition.

Chapter 6 Packing List

Device	Power Cable	Quick Guide	12pin Phoenix Connector	4pin Phoenix Connector	Small Screwdriver
1PCS	1PCS	1PCS	-	1PCS	1PCS

Warranty Regulations

The warranty period of this product is 1 year.

In the warranty period of non-man-made damage caused by the product performance failure can enjoy three packages of service.

Warranty card by the sales unit stamped after the effective. The alteration is invalid!

The following conditions (including, but not limited to, this) are not covered by the three-package service:

1. No warranty card or missing valid invoice or the date has exceeded the validity period of the three packages of services;
2. Not in accordance with the requirements of the product instructions for use, maintenance, management and damage caused;
3. The product model or code on the warranty voucher does not match the physical goods;
4. Damage caused by the dismantling and repair of non-authorised service providers;
5. Normal discolouration, wear and tear and consumption during the use of the product are not covered by the warranty;
6. The product cannot be used due to the user's own network reasons, please consult customer service staff.

SHENZHEN S TRACK SCIENCE TECHNOLOGY CO., LTD

Web: www.s-track.com.cn Tel:+86 755 29983191 Mail:service@s-track.cn

Add: 9F, 1B, Shangzhi Technology Park, Guangming District, Shenzhen City, Guangdong Province, China 518107