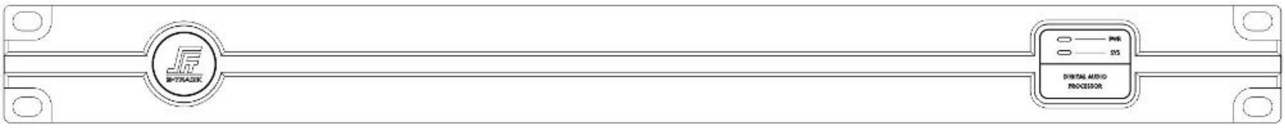

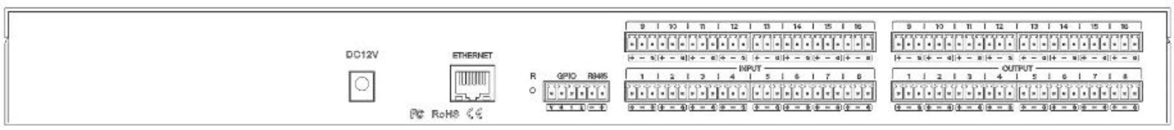

SWIFT X Series
Digital Signal Processor

User Manual

This user manual applies to software version V4.x

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 input power of the equipment power adapter is 100V-240V, 50/60Hz AC power.
- 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.

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Chapter 1 Product Introduction

1.1 Introduction

The Digital Signal Processor equipped with a high-performance 32-bit floating-point DSP processor and 24-bit A/D～D/A converter, 48kHz sampling rate, 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;
- 24-bit high-performance A/D, D/A converter, 48kHz sampling rate, high-quality Analog → Digital, Digital → Analog conversion;
- 16 Analog input channels and 16 Analog output channels, very small distortion and ultra-low noise floor;
- 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

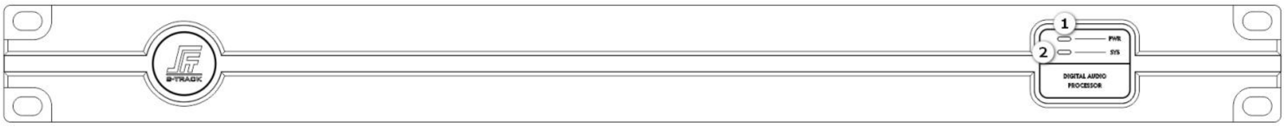
- Comprehensive matrix mixing function, 48kHz sampling rate, 24-bit high performance A/D, D/A converter and 32-bit floating point DSP processor;
- DSP audio processing, built-in automatic mixing console, including mixing and automatic mixing functions, but also has a mixing component control function; at the same time with AFC, AEC, ANS module;

- Inputs per channel: Preamplifier, Invert, Signal Generator, Expander, 5-band Parametric Equalizer, Compressor, Automatic Gain Control (AGC), AM (Gain Sharing Automatic Mixer or Gating Automatic Mixer), 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, 31-band Graphic Equalizer, Limiter, Invert,;
- 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;
- 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;
- 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;
- Configuration of bi-directional RS485 interface, standard Ethernet control interface, 2-channel programmable GPIO control interface (customisable inputs and outputs);
- 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 Interface Description

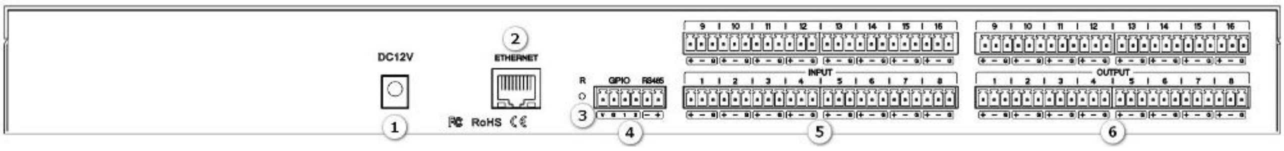
2.1 Front Panel



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.

2.2 Rear Panel



① **DC12V:** Power connector to connect DC 12V/2A power adapter;

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 reboot the processor;

④ **RS485+GPIO:** Connect to the control terminal or central control devices;

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.

Note: This manual takes 16-channel device as an example for illustration, please refer to the actual device for details.

Chapter 3 Control Software Download

3.1 Software download preparation

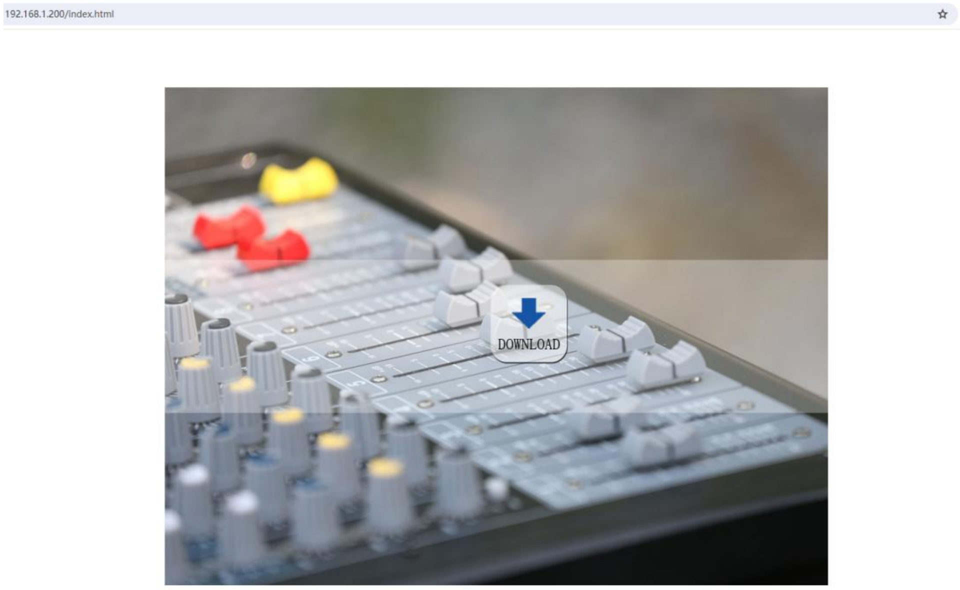
Please ensure that your computer and device are connected to the same network. The device's default IP address is 192.168.1.200, with a subnet mask of 255.255.255.0 and a gateway of 192.168.1.1. If your computer's IP address is not in the same subnet, you will be unable to access the device. You can check and adjust your computer's IP address using the following steps: Open the "Control Panel," select "Network and Sharing Center," click on the currently connected network, select "Properties," double-click "Internet Protocol Version 4 (TCP/IPv4)," and choose "Obtain an IP address automatically" or manually set an IP address within the same subnet as the device (e.g., 192.168.1.x).

3.2 Check the system environment

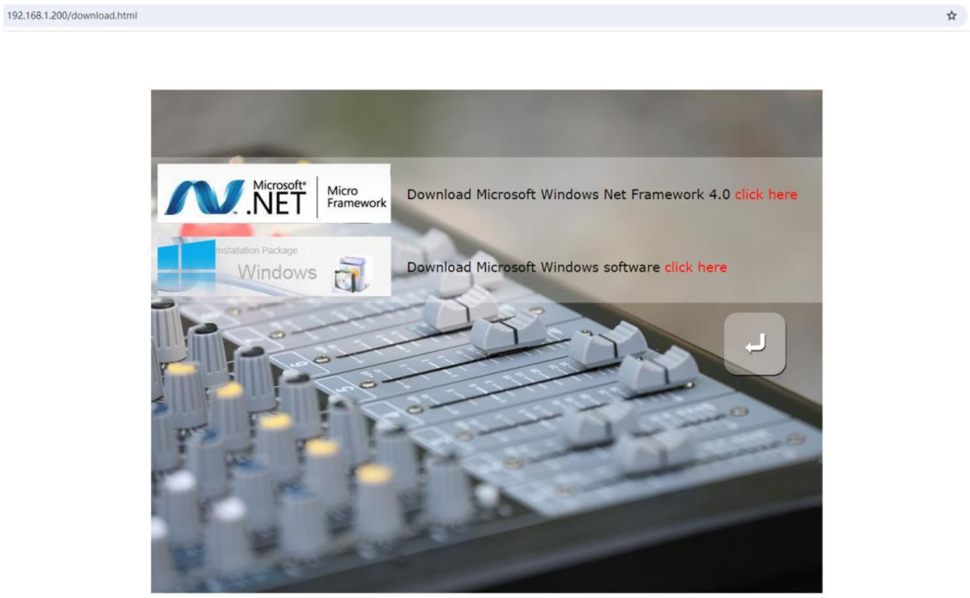
Before installing the audio processor control software, ensure that your computer has the Microsoft .NET Framework 4.0 or higher version of the Windows system runtime library installed. If not installed, you can download the Windows system Microsoft .NET Framework 4.0 runtime library from the software download interface and follow the library file installation wizard to install it.

3.3 Download and installation steps

Go to the download interface, open your browser, enter the device's factory default IP address (192.168.1.200) in the address bar, and press Enter. The webpage will automatically load the device's download interface. Download and install the package: Locate the "DOWNLOAD" button on the webpage and click it. Wait for the installation package (file name: app.msi) to download completely. Once downloaded, the installation package will be saved in the default download folder on your computer (typically "C:\Users[Your Username]\Downloads"). Install the control software: Locate the downloaded app.msi installation package, double-click to run it, and follow the installation wizard's prompts to complete the installation process.

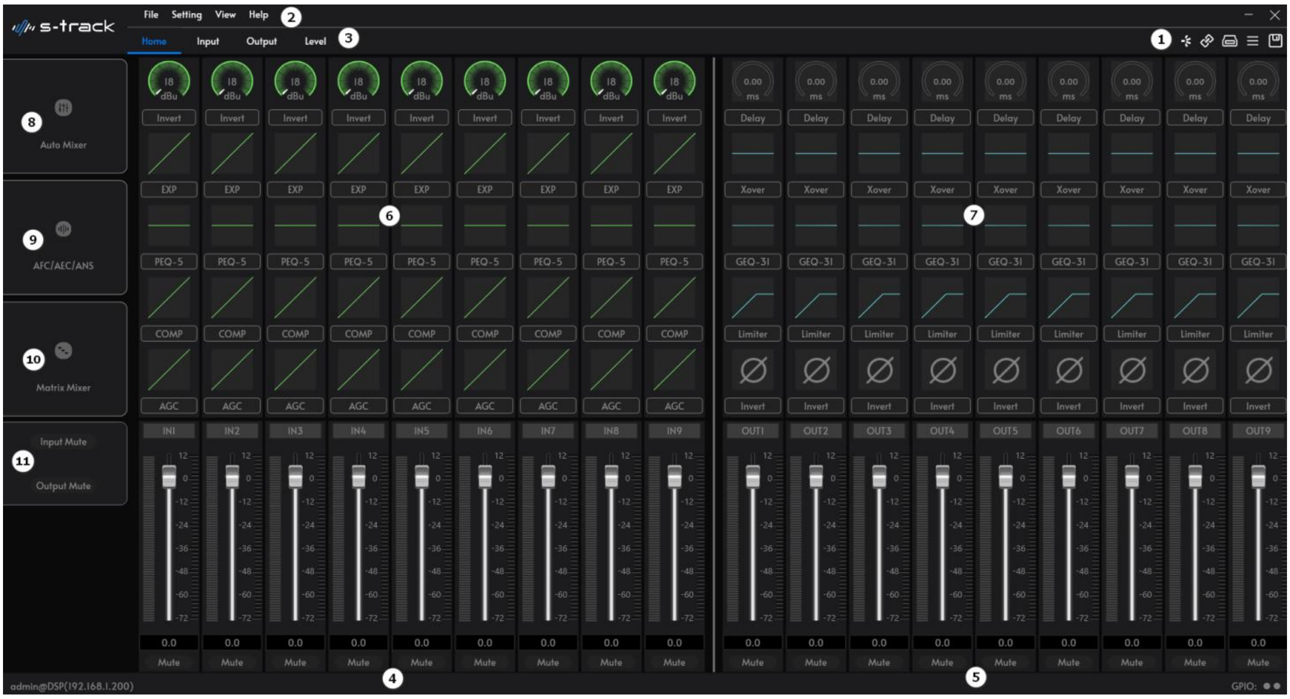


Software DOWNLOAD:



Chapter 4 Control Software User Guide

4.1 Main Interface



① Button function area;

② Menu;

③ View page selection area;

④ Input Channels: IN1～IN16: 16 Analog Input channels;

⑤ Output Channels: OUT1～OUT16: 16 Analog Output channels;

⑥ Input channels signal processing components preview area, you can display hidden channels by dragging the mouse or using the left and right arrow keys;

⑦ Output channels signal processing components preview area, you can display hidden channels by dragging the mouse or using the left and right arrow keys;

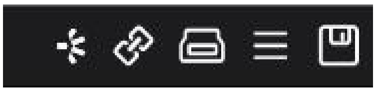
⑧ Auto Mixer component;

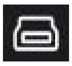
⑨ AFC/AEC/ANS: Acoustic Feedback Canceler (AFC) component, Acoustic Echo Canceler (AEC) component, Adaptive Noise Suppressor (ANS) component;

⑩ Matrix Mixer component;

⑪ Input Mute: Mute all input channels; Output Mute: Mute all output channels.

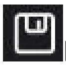
4.2 Button Function Area

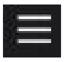


① Button: Device List Button;

② Button: Device Search Button: Click to search for connectable devices and display device IP;

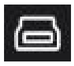
③ Button: Device Connection Button: The IP of the device is known, and you can connect directly by entering the IP address, user name and password in the pop-up box;

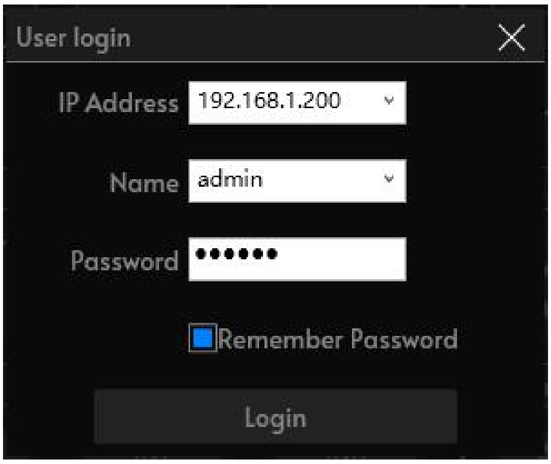
④ Button: Save Scene Button: Saves (overwrites) the parameter changes to the selected scene;

⑤ Button: Scene List Button.

4.3 Software Login Connection

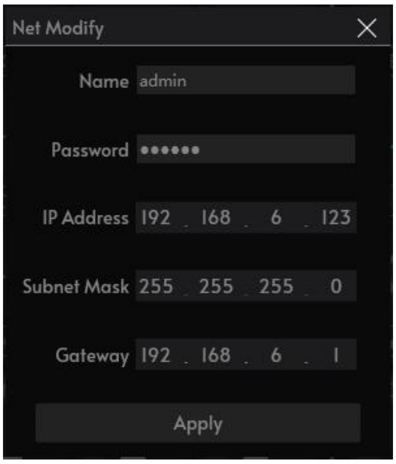
4.3.1 User Login

After opening the audio processor control software, in the Button Function Area, click the  button and  button, and the system will automatically scan and refresh the device list. Once the device list has been refreshed, you will see the [Model Name] of the online devices in the list bar. Find the device you need to log in to, double-click its [Model Name], and a login window will pop up. In the login window, enter the default username and password: the username is "**admin**" and the password is "**123456**". After entering the information, click the [Login] button. If the connection is successful, the software's status bar at the bottom will display the username and IP address information of the currently connected device.



4.3.2 IP Address Modification

When checking the device status, if the device's IP address is not in the same subnet as the PC, an  mark icon will appear before the device model name. At this point, double-click the device's [Model Name] to open the IP information modification dialog box. In the dialog box, you can modify the device's IP address to ensure it is in the same subnet as the client machine. After modification, click the [Apply] button to save the settings, and the device will reconnect to the network using the new IP address.



4.4 Menu - 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.

4.5 Menu - Setting - Center Control Command

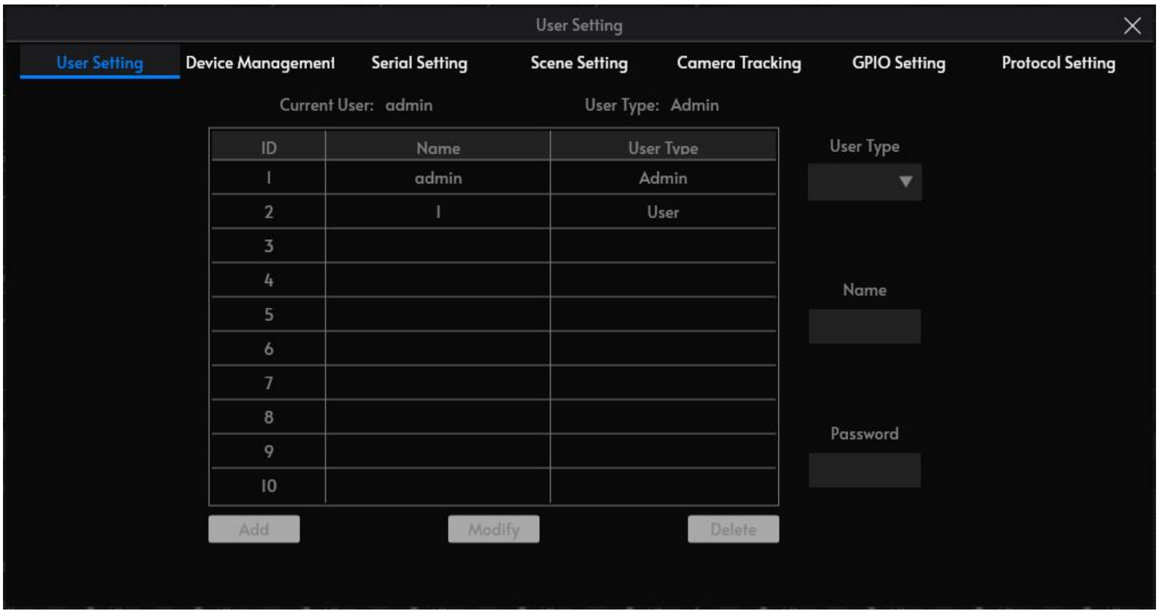
The Central Command Generator is able to convert frequently used operations into a 16-character command code for easy invocation by external devices.

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

4.6 Menu - Setting - Device Setting

4.6.1 User Setting

The user setting is a core component of the system used to centrally manage and maintain user information. It provides functions for displaying, adding, modifying, and deleting user information, with the aim of helping system administrators efficiently manage user permissions and information. With its simple and intuitive interface design and clear operation process, can meet the needs of daily user management, ensuring the security and maintainability of the system.



① **User List - User ID:** A unique identifier automatically generated by the system to distinguish between different users;

② **User Name:** The name used by the user to log in, which is unique.

③ **User Type:** Indicates the user's permission level, divided into two types: "Administrator" and "Normal." Default factory user name and password: admin/123456.

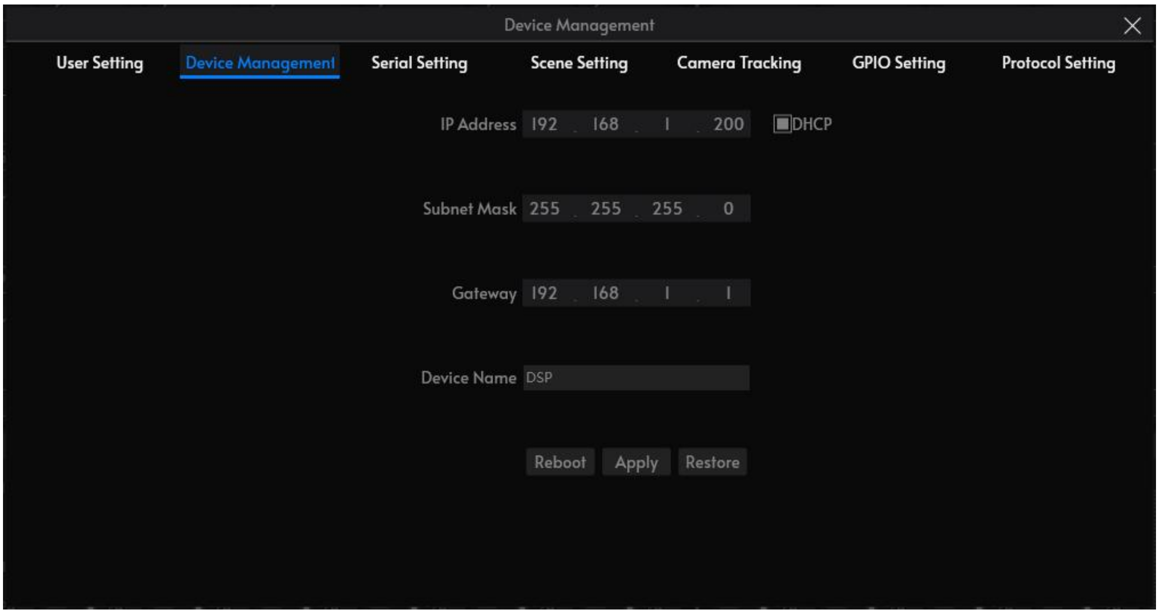
④ **Add User:** Select an unused ID number from the list, then click the user type drop-down box to select Add Administrator or Regular User, enter the user name and password information, click the "Add" button. The system will verify the validity of the entered information (e.g., whether the user name is unique, whether the password meets requirements, etc.). If the verification passes, the new user information will be added to the user list. If the verification fails, a corresponding error message will be displayed (e.g., "User name already exists" or "Password format is incorrect");

⑤ **Modify User:** Modify User Name/Password: Select the target user in the user list box, enter the new user name (the system will check for uniqueness) or password, and click the "Modify" button to complete the modification. The system will automatically update the user list and display the modification results;

⑥ **Delete User:** Select the user to be deleted in the user list box. Click the "Delete" button, and the system will delete all information for that user and remove them from the user list.

4.6.2 Device Management

The device management is a core tool in the system used to centrally manage and configure device network parameters and status. Through this, users can easily modify device network configuration information, such as IP addresses, gateways, subnet masks, and device names, to meet access requirements in different network environments. In addition, it provides a device soft reboot function to quickly restore the device's operating status, as well as a factory reset function to quickly restore the device to its default configuration when it malfunctions or needs to be initialized.



① **IP Address:** According to network planning, users can enter a new IP address to ensure that the device can communicate correctly within the target network;

② **Subnet Mask:** Configure the subnet mask to determine the scope and division of the network in which the device is located, ensuring the accuracy of network communication;

③ **Gateway:** Set the device's default gateway to enable it to access external networks via the specified router or gateway;

④ **DHCP:** Check DHCP, and the device can automatically obtain a dynamic IP address from the DHCP server;

⑤ **Device Name:** Assign an easily identifiable name to the device to facilitate quick location and management in the device list;

Note: After modifying the above network parameters or device name, click the "Apply" button for the system to save and apply the new configuration. During modification, ensure that the entered parameters comply with network standards to prevent the device from failing to connect to the network normally.

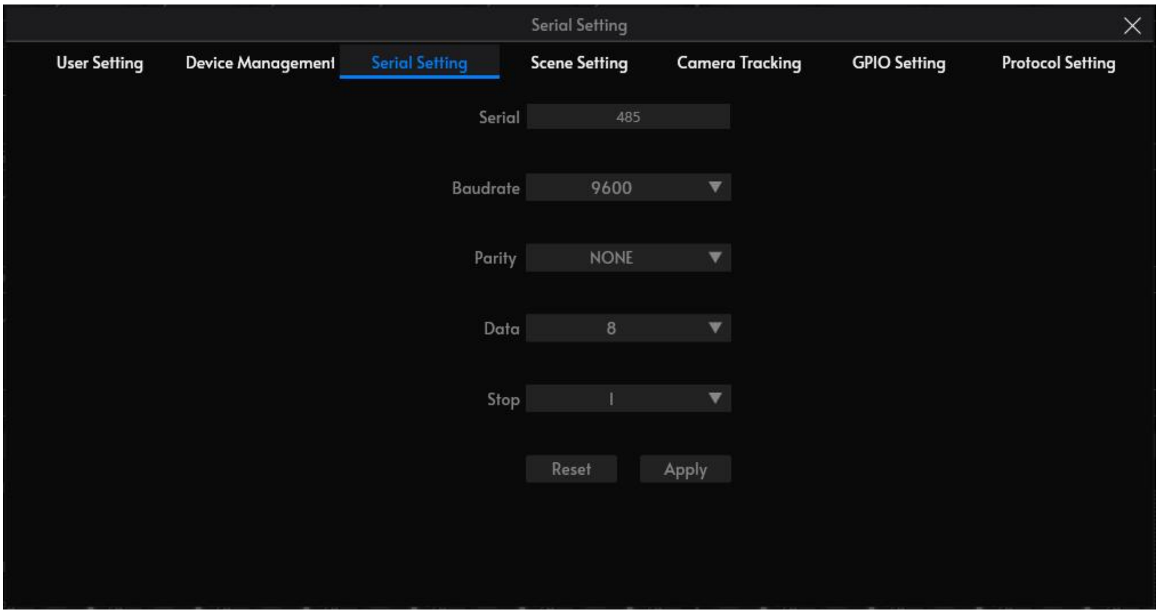
⑥ **Reboot:** The soft reboot function allows users to quickly reboot the device without disconnecting the power supply to resolve temporary faults or performance issues that may occur during device operation. After clicking the "Reboot" button, the device will automatically restart and reload the current configuration, returning to normal operation;

⑦ **Restore:** When the device experiences severe faults or requires configuration initialization, users can use the restore function. This function will clear all current configuration information on the device, including network parameters, user settings, etc., restoring it to the default state at the time of manufacture;

Important Note: After performing a factory reset, all device parameters will be reset to their default values, and previously saved configuration information cannot be recovered. Before executing this operation, please carefully confirm whether a factory reset is necessary and back up important configuration information in advance. Once executed, the device will be unable to retrieve previous parameter settings.

4.6.3 Serial Setting

The serial port setting is designed to provide users with flexible configuration capabilities for serial port communication parameters, supporting the setting of key parameters such as baud rate, parity bit, data bits, serial port type (RS485), and stop bits. Additionally, the module is equipped with a reset button and an application button to enable quick restoration of parameters to their factory default values or confirmation of modified parameters, ensuring efficient and stable serial port communication.



① **Serial Port Type:** RS485: RS485 is a differential, half-duplex (or full-duplex) serial communication standard established by the Electronic Industries Association (EIA). "Differential" means it uses two signal wires (A wire and B wire) to transmit a single signal. The receiving end determines the signal by detecting the voltage difference between these two wires. Communication distances vary (typically up to 1200 meters);

② **Baud Rate:** Setting the baud rate determines the data transmission rate for serial communication. Users can select an appropriate baud rate based on communication requirements and device compatibility. Multiple baud rates are available, including 9600, 19200, 38400, 57600, and 115200. Users can select the desired baud rate via the dropdown menu. The default baud rate is set to 9600;

③ **Parity Bits:** The parity bits is used to verify the integrity of data during serial communication. Users can select an appropriate parity method based on communication protocol requirements. Supported options include: None (No Parity), Odd Parity, and Even Parity. Select the parity bit type via the dropdown menu or radio button. The default setting is None (No Parity);

④ **Data Bits:** Data bits indicate the number of data bits transmitted each time. Users can select the appropriate data bit length based on communication protocol requirements. Available options: 5, 6, 7, or 8 data bits. Select the data bit length via a dropdown menu or radio button. The default setting is 8 data bits;

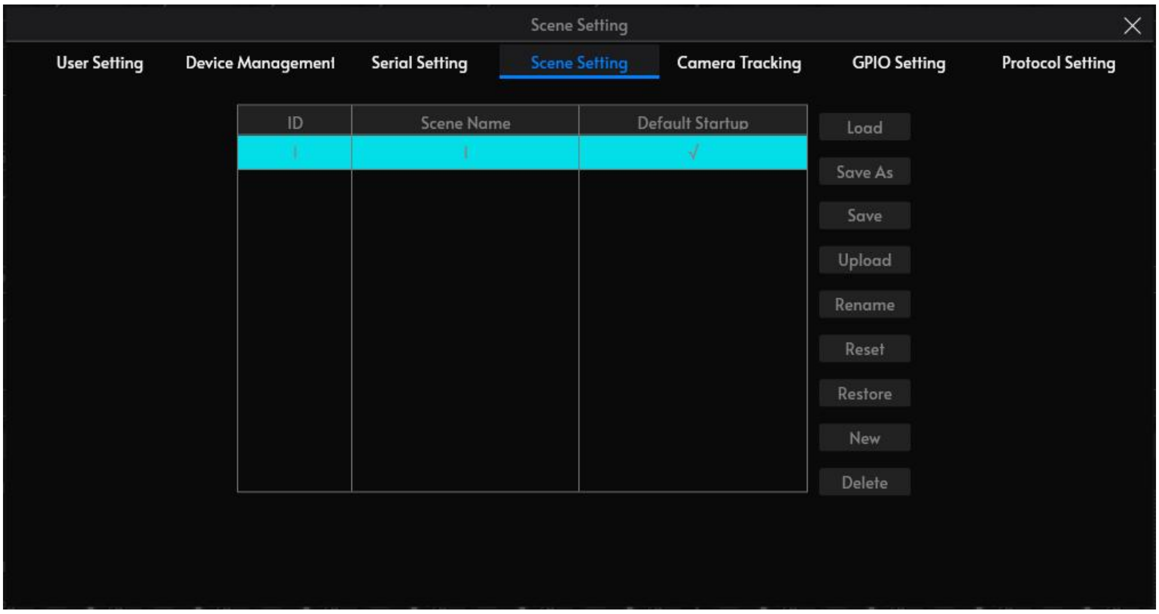
⑤ **Stop Bits:** Stop bits are used to indicate the end of a data transmission. Users can select the appropriate number of stop bits based on communication protocol requirements. Options include 1-bit and 2-bit stop bits. Select the stop bit count via the dropdown menu or radio button. The default stop bit count is 1 bit;

⑥ **Reset:** When users need to quickly restore serial port communication parameters to their factory default settings, they can click the Reset button;

⑦ **Apply:** After modifying serial port communication parameters, users must click the Apply button to confirm the changes and make the new parameters effective.

4.6.4 Scene Setting

The scene setting provides users with a convenient scene management platform that supports scene creation, saving, modification, restoration, deletion, and uploading. Through this module, users can flexibly manage various scene files to meet the needs of different scenarios while ensuring the accuracy and traceability of scene parameters.



① **Load Scene:** Load a saved scene file and restore the scene parameters to their last saved state. After the user selects the target scene file in the scene list interface, clicking the

"Load" button will cause the system to automatically read the parameters from the file and apply them to the current scene;

② **Save As:** Export the scene file stored in the system. After selecting the target scene file in the scene list interface, click the "Save As" button. The system will display a local file selector and prompt the user to enter a new scene name. After entering the name, the system will save the scene file as a new file and store it locally;

③ **Save:** Save the current scene's parameters as a scene file stored on the device for future use. Users click the "Save" button in the scene settings interface, and the system will automatically save the file;

④ **Upload Scene:** Upload a locally stored scene file to the device for use. The user clicks the "Upload" button, and the system displays a local file selector. The user selects the target scene file and clicks the "Open" button. After the upload is complete, the user can load the scene on the device using the corresponding function. Scene files cannot be used across different device systems to avoid causing device malfunctions;

⑤ **Rename Scene:** Modify the name of a saved scene file for better management and identification. In the scene list interface, select the target scene file and click the "Rename" button. The system will display a scene name modification box. Enter the new scene name as needed and click the "Yes" button. The new scene name will be automatically saved;

⑥ **Rest Default:** Rest the current scene's parameters to their factory default settings. Users click the "Rest" button, and the system will restore the default parameters and automatically load the scene;

⑦ **Restore Factory Settings:** Clear all user-created preset scene files, including temporary scenes, and restore the system to its initial state. Users click the "Restore" button, and the system will delete all user-defined scene files and restore the system to its initial state;

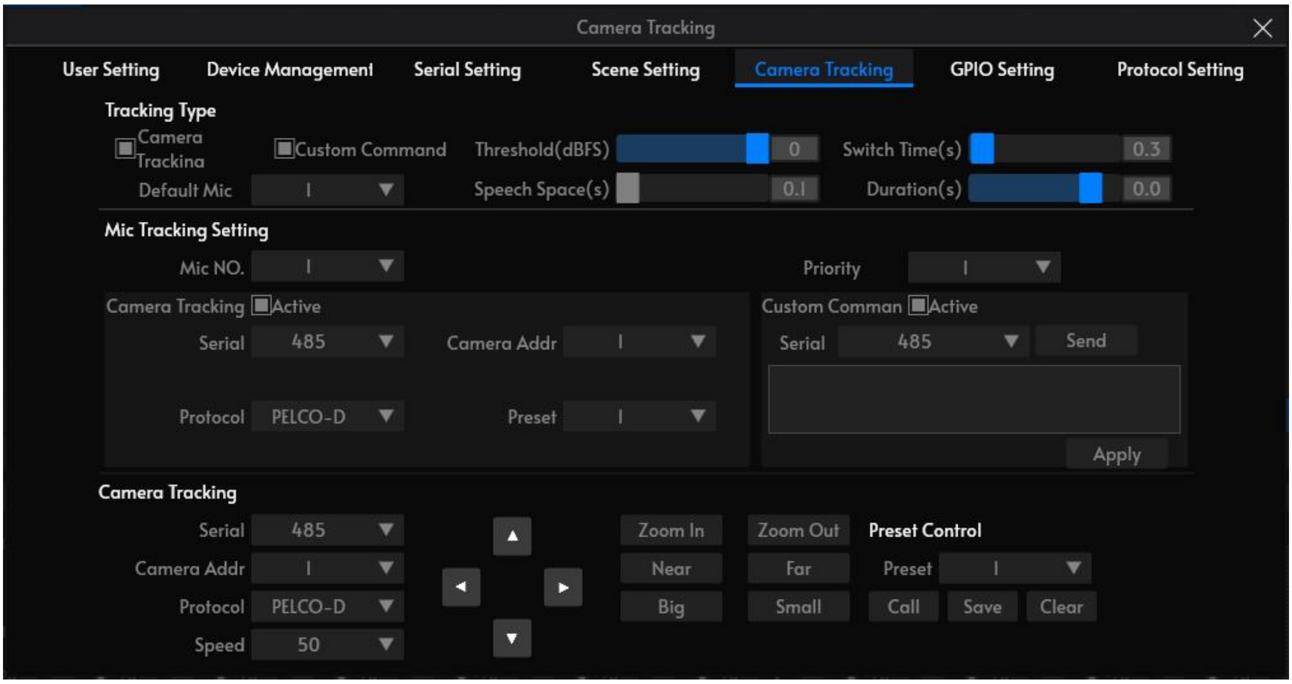
⑧ **New Scene:** Create a new scene file with default settings. When the user clicks the "New" button, the system will prompt the user to enter a scene name. After entering the name, the system will create a new scene file and set its parameters to default values;

⑨ **Delete:** Delete scene files no longer needed by the user to free up storage space. After selecting the target scene file, the user clicks the "Delete" button, and the system will delete the scene file.

4.6.5 Camera Tracking

Camera tracking is an intelligent video surveillance technology that automatically detects and identifies the movement trajectory of target objects to achieve real-time tracking and positioning of targets. It is widely used in many fields such as security surveillance, conference systems, intelligent transportation, and sports events, and can significantly improve surveillance efficiency and automation levels.

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 **Camera Tracking Type:** Check to enable;

- **Camera Tracking:** Control camera movement via channel input signals to automatically track targets;
- **Custom Command:** When channel input signals are triggered, send custom commands to specified ports for more flexible control.

② **Default Microphone:** When there is no input signal from any microphone, the camera will automatically rotate to the position set for the default microphone or send the custom command associated with the default microphone. Microphone numbers with a "#" are virtual numbers used solely for setting the default microphone;

③ **Tracking Threshold:** The system must detect an input signal strength greater than or equal to the tracking threshold to automatically enable tracking functionality. This parameter prevents false tracking caused by weak signals;

④ **Speech Space:** The maximum allowable interruption time for an effective signal. For example, when speaking into the microphone, if the "Speech Space" is set to 3 seconds, any pause during speech lasting no more than 3 seconds is still considered an active signal; if the pause exceeds 3 seconds, the signal is deemed inactive;

⑤ **Switch Time:** The minimum speaking duration required for the camera to switch to a specific active position. For example, when speaking into the microphone, the speaking duration must exceed the "Switch Time" for the channel signal to be considered active, prompting the camera to automatically switch to the designated position. Typically, the "Switch Time" should be greater than the "Speech Space";

⑥ **Duration:** The interval time for sending camera switching commands or custom commands. If set to 0, it indicates special handling, triggering only once;

⑦ **Microphone Number:** The microphone number generally corresponds to the input channel of the device, i.e., the channel number to which the microphone is connected. Microphone numbers with a "#" are virtual numbers, used solely for setting the default microphone;

⑧ **Priority:** The smaller the priority number, the higher the priority level. When multiple microphones have simultaneous signal inputs, the camera will automatically rotate to the preset position corresponding to the microphone with the smaller priority number (higher priority level) or send the corresponding command. If priorities are the same, the signal detected first takes precedence;

⑨ **Camera Tracking Active:** Users can pre-set all microphone parameters, but during actual use, they can select to enable specific microphone settings as needed. Parameters such as preset points, serial port numbers, camera addresses, and protocols are closely related to the actual connection with the camera and must match the camera's actual configuration. Select **Active** to enable;

⑩ **Custom Command Active:** When the matrix detects a microphone input signal (e.g., someone speaking), it can automatically send a custom command to the specified serial port. Users can also pre-set commands, but if the "Active" is not selected, the device will not automatically send the command. However, users can still manually send the command to the specified serial port by clicking the "Send" button;

⑪ **Parameter Saving:** Click the "Apply" button to save the microphone parameters to the device and associate the microphone with the corresponding camera address. The "Enable Microphone Settings" option determines whether the microphone settings take effect when tracking is enabled;

⑫ **Camera Setting:** Camera setting is a configuration interface used to adjust the camera position before tracking begins. After configuration is complete, the relevant parameters are saved to the camera;

⑬ **Serial Port:** Serial port RS485 must correspond to the backplane port connected to the pan-tilt unit;

⑭ **Camera Address:** The camera address must be configured according to the actual camera address;

⑮ **Camera Protocol Type:** The camera protocol type must match the camera model;

⑯ **Preset Control - Preset Point Number:** A identifier assigned by the user to the camera. By adjusting parameters such as up, down, left, right, focal length, and aperture, the specific position and settings of the camera can be defined;

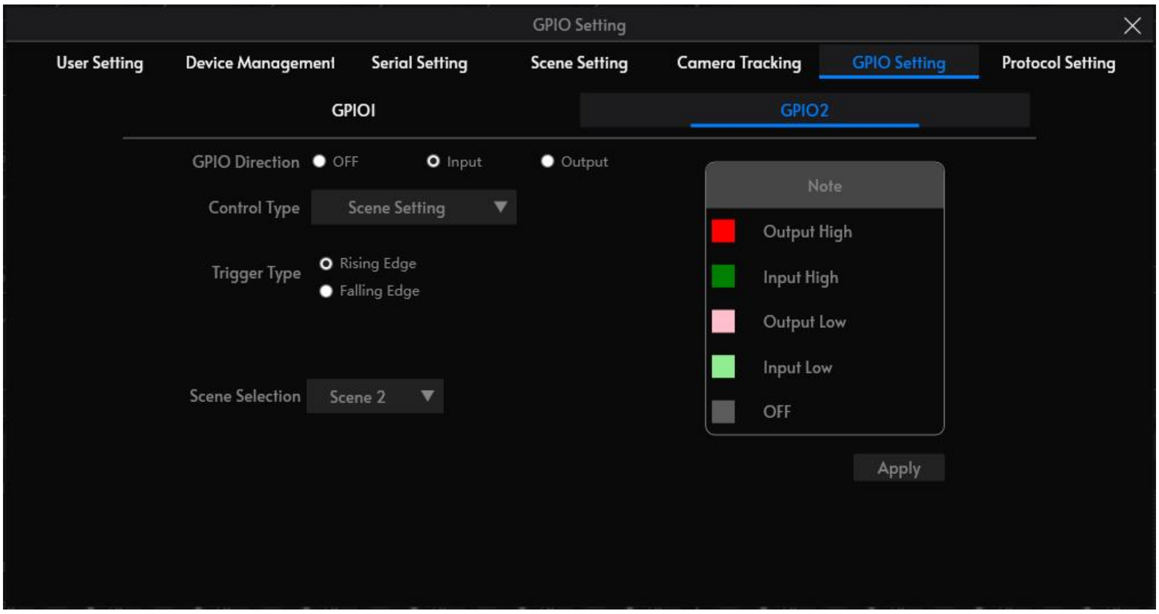
⑰ **Operation Buttons - Save:** Saves the current preset to the camera;

⑱ **Call:** View the camera position saved at the current preset point;

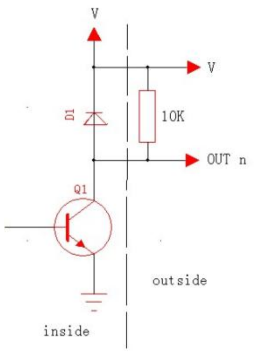
⑲ **Clear:** Deletes the information of the current preset point.

4.6.6 GPIO Setting

GPIO is a programmable interface widely used in embedded systems, microcontrollers, and computer hardware. It allows users to control the input and output signals of hardware devices through software, enabling interaction between hardware and software.

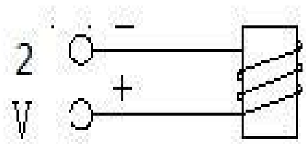


Output Connection 1:



Firstly, 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 a high level 1 according to the change of matrix state, and the level can be used to trigger another GPIO or other devices.

Output Connection 2:



Driver relays (control type): Relays can be used to control alarm devices, etc., with built-in current-continuing diodes.

Trigger Type:

① Rising edge input: The IO port is held high when there is no input;

② Falling edge input: the IO port stays low when there is no input.

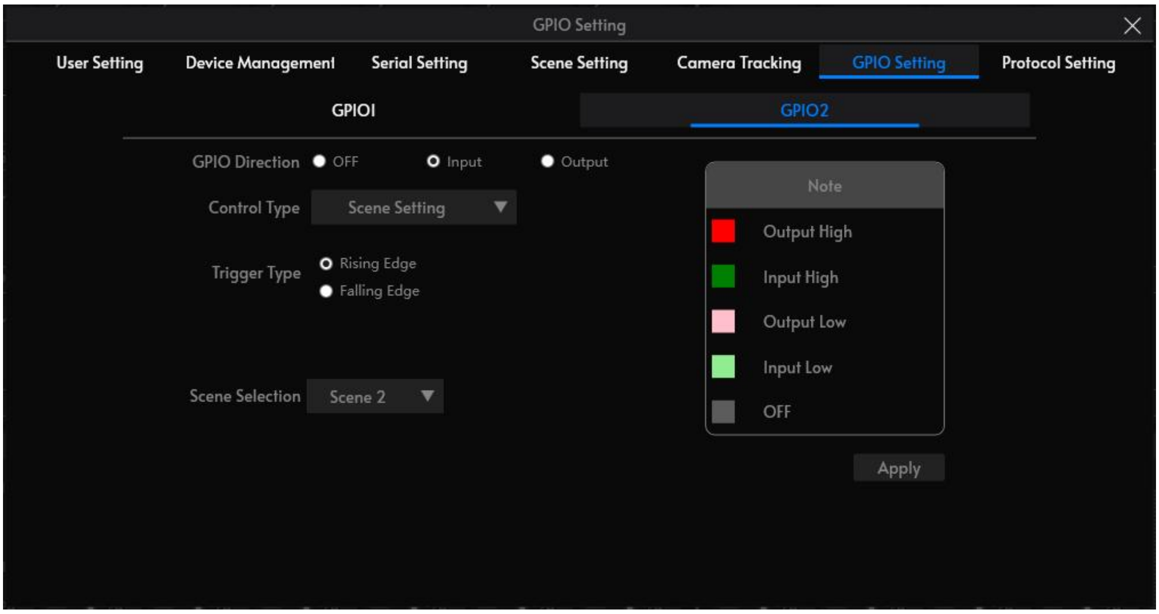
Control Type:

① Inputs: Scene, Mix, Volume, Channel Mute, System Mute, Serial Command Settings;

② Outputs: Scene, Level, Channel Mute, System Mute display.

GPIO Input Example 1

Scene Setting

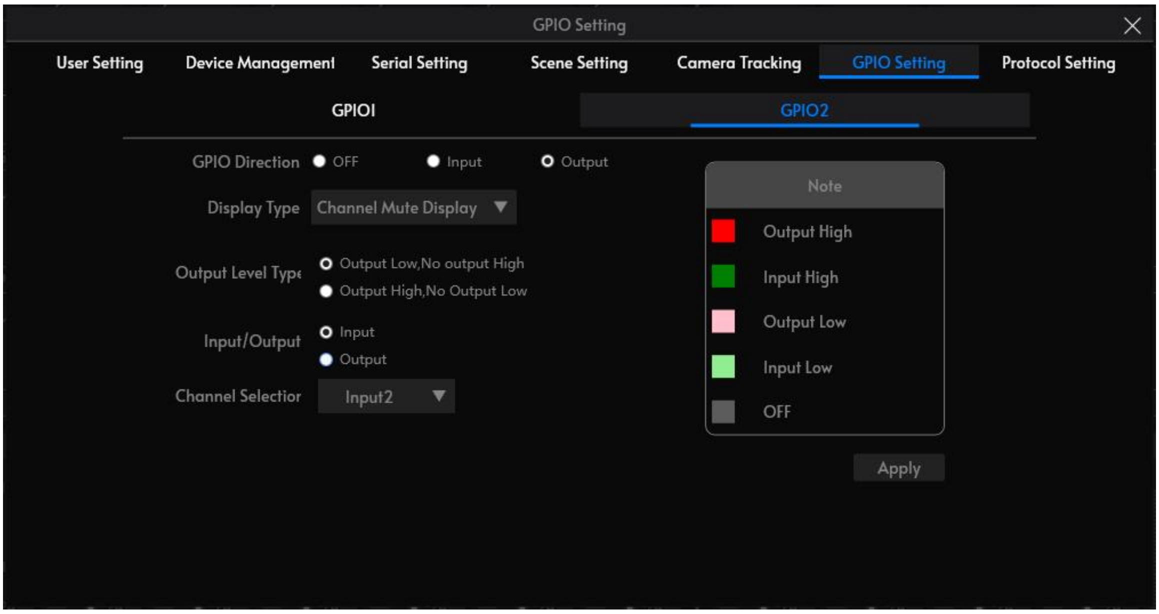


The PC enters the GPIO-2 control window, selects the [Input] direction, the control type [Scene Setting], the trigger type [Rising Edge], the parameter settings are loaded to [Scene 2], and clicks the [Apply] button at the bottom of the window.

When the device hardware GPIO-2 pin level is pulled high from low, the trigger condition is established and the Digital Signal Processor scene preset will automatically switch to scene 2.

GPIO Output Example 2

Channel Mute Display



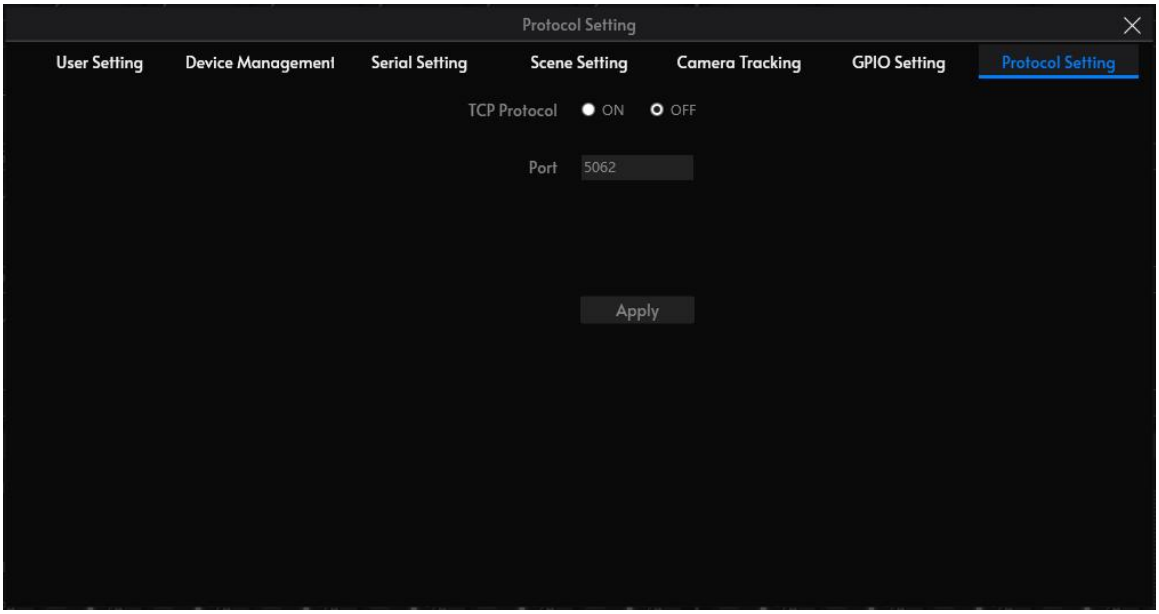
The PC enters the GPIO-2 control window, selects [Output] mode, display type [Channel Mute Display], trigger type [Output Low, No Output High], parameter setting [Inputs] channel selection [Input 2], and clicks the [Apply] button at the bottom of the window.

When the device input IN2 channel is muted, the corresponding GPIO-2 pin output is '1', and when the IN2 channel is unmuted the output is '0'.

4.6.7 Protocol Setting

The device's network transmission protocol defaults to UDP. You may optionally enable TCP protocol and customize its port number. Click "Apply" to applied. The network transmission protocol can be used for software control of this device or for controlling it via central control commands from other central control devices.

- TCP (Transmission Control Protocol) is a connection-oriented, reliable, byte-stream-based transport layer communication protocol;
- UDP (User Datagram Protocol) is a simple, connectionless, unreliable transport layer protocol for transmitting datagrams.



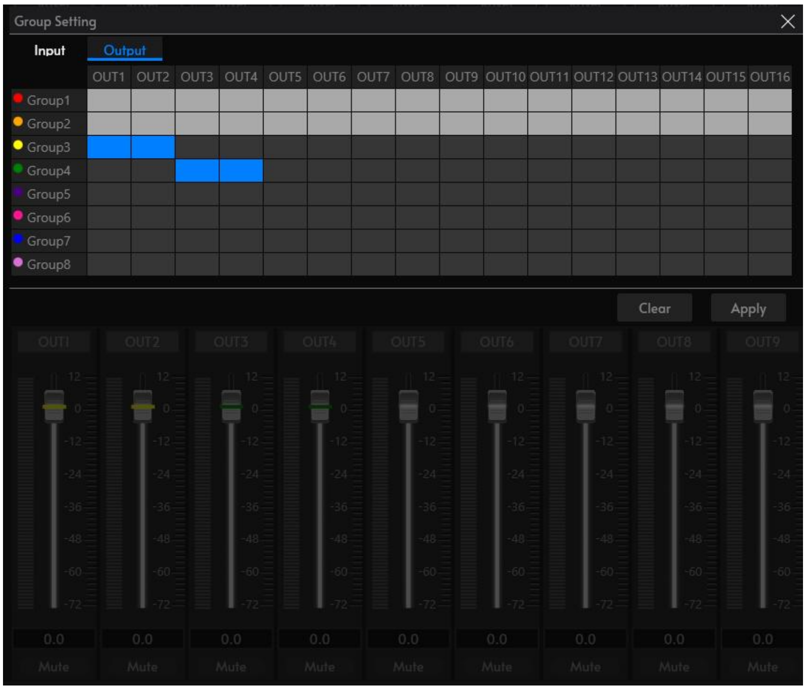
4.7 Menu - Setting - Group Setting

The Group Setting enables synchronized management of fader volume for similar audio channels.

The Group Setting provides 8 independent channel groups with an input locking strategy. Once any input is assigned to a group, that group is immediately locked at the output end, preventing subsequent devices from reassigning it. Once a channel is added to a group, gain or mute parameters within the group are synchronized in real time. When a channel is removed from a group, it retains the last parameter settings before exiting. If multiple channels with different parameters are added to the same group simultaneously, the system calibrates them uniformly based on the minimum value and immediately synchronizes it to all channels within the group.

The system contains dual control modules for input signal grouping and output signal grouping.

4.7.1 Input and Output Configuration



① Input/Output grouping supports up to 8 independent configuration;

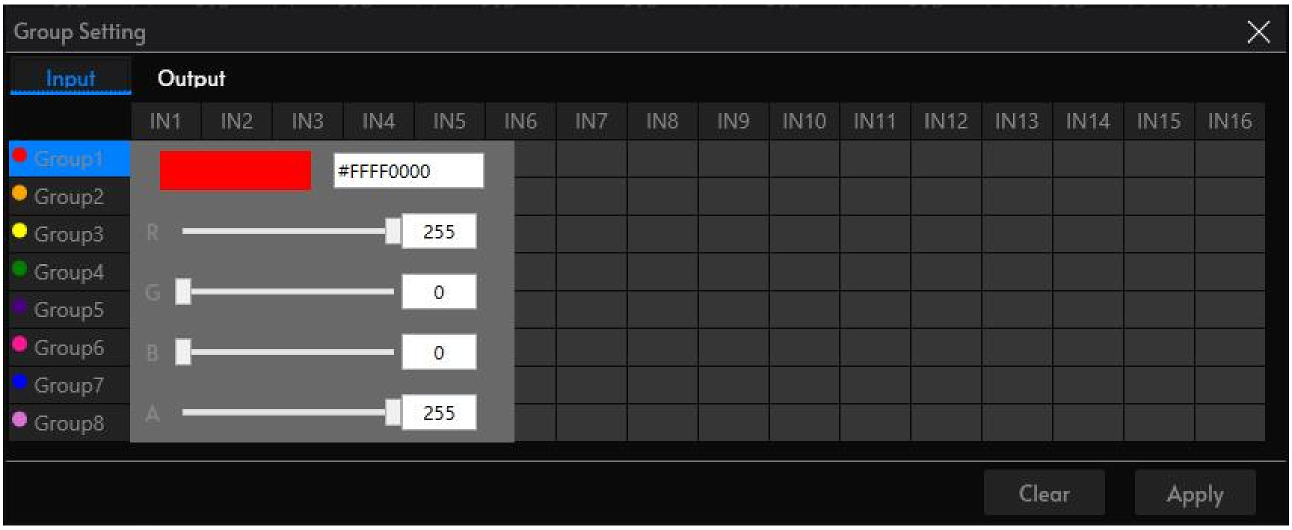
② When you enter the grouping control interface, you must first select the channel signal type (Input/Output);

③ In the Group1 control panel, left-click the target channel to complete the grouping configuration, and then click "Apply" for the grouping to take effect, and the grouping status will be displayed in the fader control interface. --The faders in the same group are automatically synchronized to a uniform logo color. When any fader in the group is adjusted, the other faders in the group will be synchronized. Mute switch button synchronized linkage control.

④ Click "Clear" to cancel all group configurations;

⑤ The group configurations of the output channels follow the same operation logic as the group configurations of the input channels.

4.7.2 Customize Color Setting



Click the "Group" button on the left side of the screen to customize the color setting for the specified group.

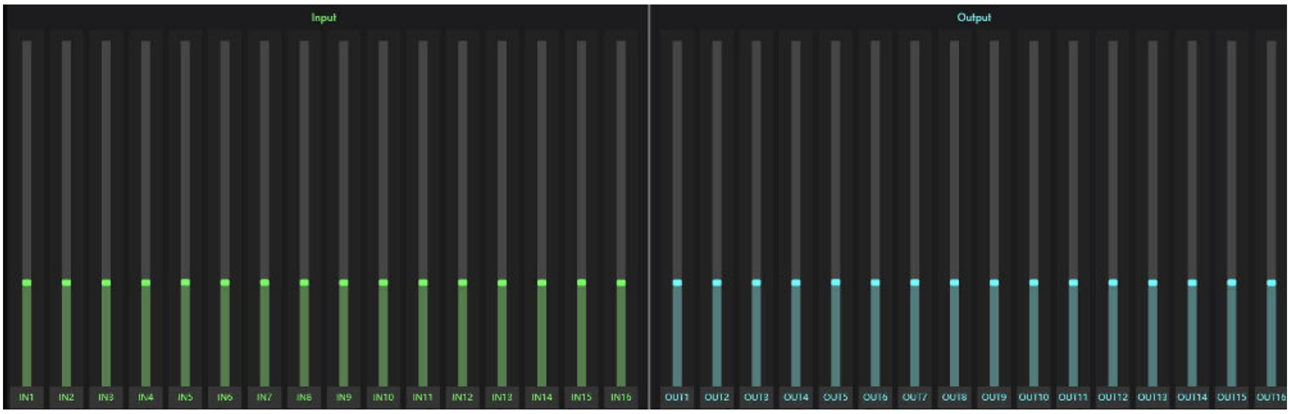
4.8 Menu - View

① **Home:** Show the main interface;

② **Input:** Show input channel components;

③ **Output:** Show Output channel components;

④ **Level:** Show input and output channels Level interface;



⑤ **Auto Mixer:** Show Automixer components;

⑥ **AFC/AEC/ANS:** Show AFC, AEC, ANS components;

⑦ **Matrix Mixer:** Show Matrix Mixer component.

4.9 Menu - Help

① **Content:** View the user manual of the device;

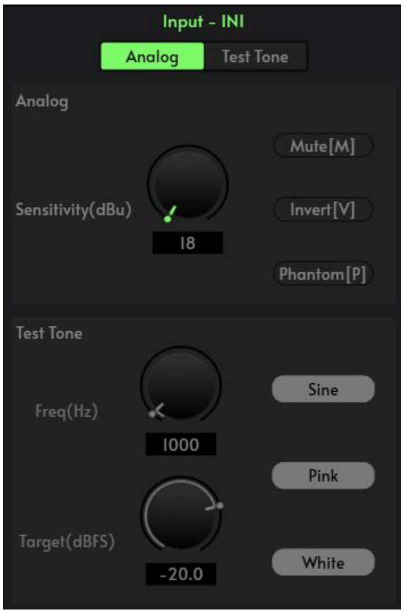
② **Upgrade:** For updating the system software version;

③ **About:** View device software version, control software version, device serial number, and other information.

4.10 Input Component Configuration

4.10.1 Input Setting

The Analog Input component provides line-level input for devices with line-level outputs, and inputs for microphones, instruments, mixing consoles, etc. The Analog Input converts the analog input signal to a processed digital. Connections are made using one three-terminal 3.5mm Phoenix connectors.



① **Model type:** Analog Input or Test Signal;

② **Sensitivity:** Sensitivity determines the gain level of the input signal. Users can optimize the input signal level by adjusting the sensitivity according to actual needs and the characteristics of the audio source. The sensitivity adjustment range is 18 to -33dB, with a step size of 3dB per increment, providing precise adjustment options;

③ **Mute:** Mutes the input signal;

④ **Invert:** Inverts the polarity of the output signal. In audio processing, phase is a critical parameter that determines the starting point and direction of a signal waveform. Through phase inversion, the phase of an audio signal is reversed by 180 degrees. In multi-speaker systems, inconsistent signal phases between different speakers can cause sound cancellation or interference. The phase inversion function allows adjustment of the phase to ensure sound clarity and consistency;

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;

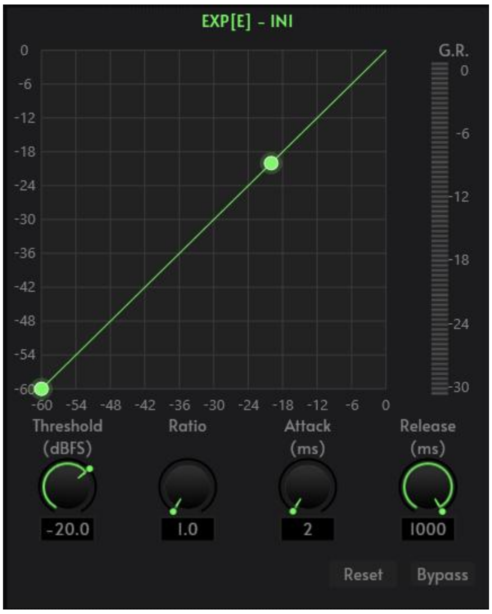
⑥ **Test Tone:** Three test signals are provided (frequency: 20Hz～20KHz; signal range: -80dBFS～0dBFS), including Sine Wave, Pink Noise, and White Noise. These test signals play an important role in audio system calibration and measurement. When the test signals are enabled, the system automatically blocks the analog input signals to ensure the purity and accuracy of the test signals.

- The **Sine Wave** is a pure audio signal with a single, constant frequency, amplitude, and phase, and a smooth, periodic waveform;
- The **Pink Noise** produces random frequencies distributed uniformly by octave throughout the audio spectrum;

- The **White Noise** is a random noise whose power spectral density is constant throughout the frequency domain, that is, all frequencies have the same energy density.

4.10.2 Expander

The purpose of the Expander component is to control the dynamic range of the Output below a set Threshold Level. It adjusts the dynamic characteristics of the signal based on user-defined parameters to enhance audio clarity, reduce background noise, and improve overall sound quality. The core function of the Expander is to dynamically compress the signal or output it unchanged based on the relationship between the input signal level and the set Threshold Level. By flexibly configuring the Expander's various parameters, users can achieve precise control over the audio signal to better adapt to different audio processing requirements and application scenarios.



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. When the input signal falls below the Threshold Level, the Expander compresses the signal at a preset Ratio to increase its dynamic range. When the input signal exceeds the Threshold Level, the signal is output at a 1:1 Ratio, preserving its original dynamic range. The Threshold setting requires adjustment based on the specific application scenario;

Example:

- If the: Threshold Level is -30 dB; Ratio is 2.5; Input level is -40 dB
- Then the Adjusted Output is:
- [(Input Level - Threshold Level) * Ratio] + Threshold Level = Output Level
- {[-40 dB - (-30 dB)] * 2.5} + (-30 dB) = -55 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. Shorter attack time enable the Expander to respond quickly to signal changes, making it suitable for processing rapidly changing audio signals; longer attack time provide smoother transitions, preventing overly abrupt processing effects, and are ideal for gentle signals like vocals or music;

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. Shorter release time enables rapid restoration of signal dynamics, making it suitable for fast-changing audio, but it may induce pumping effects (rapid fluctuations in signal level); longer release time provides smoother transitions and reduces pumping effects, but it may cause the signal restoration process to appear sluggish. Therefore, release time settings should be flexibly adjusted based on audio characteristics and processing requirements;

⑤ **Gain Reduction:** Graphically displays the amount of attenuation between the signal processed by the Expander and the input signal. The Gain Reduction reflects the degree of signal attenuation applied by the Expander;

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

⑦ **Reset:** Resets the parameters to the default.

4.10.3 Parametric Equalizer

The Parametric Equalizer component is a variable equalizer allowing you to individually adjust the Gain, Bandwidth and center Frequency of up to 12 frequency bands. You can also Bypass individual bands. Additionally, you can change any or all of the bands to either a High-shelf or Low-shelf equalizer. By flexibly configuring parameters across various frequency bands, users can achieve frequency adjustments ranging from simple to complex, meeting diverse application needs such as music production, live sound reinforcement, and voice processing to deliver ideal audio results.

① **Filter Type:** Including Parametric equalization, Low-Pass, High-Pass, Low-Shelf, High-Shelf Filter;

- **Parametric Equalization Filter:** Parametric Equalization Filter is an adjustable filter used to precisely boost or attenuate specific frequencies in an audio signal. It optimizes the frequency response of audio by adjusting the center frequency, gain, and bandwidth;
- **Low-Pass Filter:** Allows low-frequency signals to pass through and cut off high-frequency signals based on the set frequency, typically used to remove high-frequency noise or increase low-frequency components;
- **High-Pass Filter:** Allows high-frequency signals to pass through and cut off low-frequency signals based on the set frequency, typically used to remove low-frequency interference or extract high-frequency features;
- **Low-Shelf:** The Low shelf is a gain increased or attenuated for the frequency portion below the set center frequency, typically used to increase the richness of low frequencies or reduce low frequency rumbling;
- **High-Shelf:** The High shelf is a gain increased or attenuated for the frequency portion above the set center frequency, typically used to increase the clarity of high frequencies or reduce the harshness of high frequencies.

② **Frequency:** Sets the center Frequency of an individual band. Frequency is one of the core parameters in audio processing. In audio signals, different frequencies correspond to different sound characteristics. For example, low frequencies are typically associated with heavy drum sounds or bass, mid-frequencies involve vocals and the timbre of most instruments, and high frequencies are related to bright timbres or details. By selecting the appropriate center frequency, users can precisely enhance or attenuate the timbre of specific instruments, optimize the clarity of vocals, or resolve frequency issues in audio;

③ **Gain:** Controls the Gain for an individual frequency band. Users can adjust the gain to increase or decrease the signal strength of specific frequencies. For example, if the sound in a certain frequency range of an audio signal is too loud or too soft, the volume can be balanced by increasing or decreasing the gain. The adjustment range of the gain is usually from negative values (attenuation) to positive values (increase). This is not active when either Low-Pass, High-Pass Type is selected;

④ **Bandwidth (Octave):** Sets the bandwidth of an individual band of the equalizer, from 0。02 octaves to 4.00 octaves (default is 1.00). This is not active when either Low-Pass, High-Pass, Low-Shelf or High-Shelf Type is selected. The size of the bandwidth determines the precision and range of the equalization adjustment. The larger the value, the wider the bandwidth, and the greater the frequency range affected; the smaller the value, the narrower the bandwidth, and the more precise the frequency range of the adjustment. Adjusting bandwidth generally adjusts Q-Factor in an inverse manner;

⑤ **Band Bypass/Active:** 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;

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

⑦ **Reset:** All band filter parameters are restored to the default.

4.10.4 Compressor

The purpose of the Compressor component is to control the dynamic range of the Output above a set Threshold Level, thereby optimizing audio balance and consistency. Compressor are widely used in music production, live sound reinforcement, broadcasting, and voice processing, helping users control audio signal peaks, prevent distortion, and enhance overall audio clarity and audibility by raising the average signal 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. A level below the Threshold Level is not compressed, anything above the Threshold Level attenuation is applied;

Example:

- If the:Threshold Level is -30 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 - (-30 dB)] / 2.5} + (-30 dB) = -22 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):** Attack time is how fast the compressor reacts to a signal crossing the set threshold going up. Short attack time compressors can quickly capture signal peaks, making them suitable for percussion instruments, but if the attack time is too short, it can produce "breathing sounds" and lose naturalness; Long attack times provide smooth transitions, making them suitable for vocals and other gentle signals, preserving more dynamics and details;

④ **Release Time (1～1000):** Release time is how fast the compressor reacts when the signal drops below the threshold and gain is restored to its non-limited level. Fast release time can increase signal loudness, but is prone to suction effects; Slow release time provides a smooth transition and reduces suction effects, but may sound sluggish. Settings should be balanced according to audio characteristics;

⑤ **Output Gain (-24～30):** Controls the Gain of the output, used to compensate for the reduction in signal level caused by compression processing. When an audio signal is compressed, its overall volume decreases. The function of Output Gain is to restore the compressed signal to a volume level close to that before compression by raising the output signal level;

⑧ **Gain Reduction:** Graphically displays the amount of attenuation between the signal processed by the Compressor and the input signal. The Gain Reduction reflects the degree of signal attenuation applied by the Compressor;

⑥ **Output Level:** Graphically displays the level of the output signal;

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

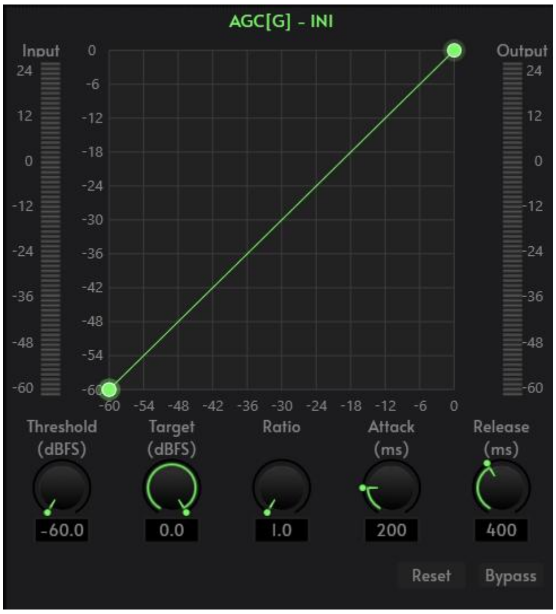
⑧ **Reset:** Resets the parameters to the default.

4.10.5 Automatic Gain Control

The Automatic Gain Control component is a technology that automatically adjusts the gain of an audio signal, the purpose is to control the overall dynamic range of the Output when the Input level changes. The AGC automatically adjusts gain to the target level by real-time detection input levels and dynamically adjusts the Gain to the Target Level by compensating for low Inputs, compressing for high Inputs according to preset parameters.

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

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 Level 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 noise floor 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):** The Level that the AGC wants the output signal to reach. Sets the point from which the Gain is calculated based on the Ratio setting. Input level below the Target Level has a positive Gain applied, input level above the Target Level has a negative Gain (attenuation) applied;

Example 1:

- If the Threshold Level is -40dB, Target Level is -15 dB; Ratio is 2.5; Input level is -30 dB
- Adjusted Output is
- [(Input level - Target Level) / Ratio] + Target Level = Output Level
- {[-30 dB - (-15 dB)] / 2.5} + (-15 dB) = -21 dB

Example 2:

- If the Threshold Level is -40dB, Target Level is -15 dB; Ratio is 2.5; Input level is -5 dB
- Adjusted Output is
- [(Input level - Target Level) / Ratio] + Target Level = Output Level
- {[-5 dB - (-15 dB)] / 2.5} + (-15 dB) = -11 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;

④ **Attack Time (1～1000):** When the input signal level exceeds the Threshold Level, the AGC begins adjusting the gain at a specified rate to reach the Target Level. Shorter attack time can respond quickly to signal changes, but may introduce sudden changes in gain;

⑤ **Release Time (1～1000):** The time required for the applied gain of the AGC to return to zero when the input signal level falls below the threshold level. Longer release time can make gain changes smoother;

⑥ **Input Level:** Graphically displays the level of the input signal;

⑦ **Output Level:** Graphically displays the level of the output signal;

⑧ **Bypass/Active:** Bypass or Active the AGC for the current channel. When the AGC is bypassed, audio is passed through without any change;

⑨ **Reset:** Resets the parameters to the default.

4.10.6 Automixer

The Automixer component has two types: Gain Sharing Automatic Mixer and Gating Automatic Mixer.

Each channel of the Automixer is equipped with an input level meter that displays the current input signal strength in real time. Each channel also features a channel fader, allowing users to flexibly control the channel's level in the mix and its direct output level for precise audio adjustment.

Parameter adjustment steps: When adjusting, prioritize setting the priority level to ensure important microphones (such as the chairman's microphone) have priority access. Then adjust the response time based on the speaker's speech rate and room reverberation to achieve natural transitions. The slope should be adjusted according to the microphone type and background noise level; increase the slope appropriately for boundary microphones or high noise environments.

I. Gain Sharing Automatic Mixer

The Gain Sharing Automatic Mixer component takes a number of audio inputs, allowing one or more of the inputs to pass while attenuating others based on the level.

The Gain Sharing Automatic Mixer is primarily used for multiple live microphones operating in the same room together as a system – for example, in boardrooms, classrooms, churches, and courtrooms. The Automixer controls the live microphones by turning up microphones when someone is talking and turning down microphones that are not used. 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. The gain of each microphone input is calculated as the ratio of its RMS level to the combined RMS levels of all inputs. This ensures unity system gain at all times.

The core purpose of Gain Sharing Automatic Mixer component is to maximize the room's acoustic gain, thereby achieving higher output levels while avoiding feedback. Its sound effect is typically smoother and more natural than Gating Automatic Mixer.

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:** Priority PR can be assigned to each channel based on actual requirements, ranging from 0 (lowest priority) to 10 (highest priority). In the Automixer, channels with higher priority gain greater transmission gain acquisition capability and can reduce the transmission gain acquisition capability of lower-priority channels. In conference scenarios, it is recommended to assign high priority to important speakers (such as the chairperson) to ensure their remarks receive priority protection and are conveyed clearly;

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 **Automixer Type:** Gain Sharing Automatic Mixer or Gating Automatic Mixer;

⑤ **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;

⑥ **Response Time**: Control the time required for the active microphone to achieve full transmission gain, as well as the gain attenuation time for other non-active microphones. Longer setting produces a smoother gain transition, suitable for steady speech; shorter setting results in faster response to speech;

⑦ **Slope:** The slope similar to the ratio of an Expander. A steeper slope increases the gain obtained by active microphones and reduces the gain of inactive microphones; a shallower slope decreases the gain obtained by active microphones and reduces the gain reduction of inactive microphones. Common settings are 2 or nearby values. When using interface microphones, a higher setting is recommended;

⑧ **Max NOM:** Maximum number of microphones. When the number of microphones turned on at the same time exceeds the set value, the newly activated microphone will take over the lowest priority channel that is already turned on. If there are no channels available to take over (such as when all channels have higher priorities), the new microphone will not be able to connect;

⑨ **Bypass/Active:** Bypass or Active the Automixer. When the Automixer is bypassed, audio is passed through without any change;

⑩ **Reset:** Resets the parameters to the default.

II. Gating Automatic Mixer

The Gating Automatic Mixer component is typically used in applications with multiple microphones, and at times, multiple microphones being used at the same time. The microphones in use can be prioritized using three different modes: Automatic, Priority, and First In Last Out or combinations of these. The channels open and close based on Threshold levels set for each channel or manual control. Attenuation controls help control feedback based on the number of microphones open at one time.

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:** Priority PR can be assigned to each channel based on actual requirements, ranging from 0 (lowest priority) to 10 (highest priority). In the Automixer, channels with higher priority gain greater transmission gain acquisition capability and can reduce the transmission gain acquisition capability of lower-priority channels. In conference scenarios, it is recommended to assign high priority to important speakers (such as the chairperson) to ensure their remarks receive priority protection and are conveyed clearly;

③ **Mode:** You can choose one of three modes for the channel;

- **Automatic Mode:** Automatic mode opens the channel when the RMS input level exceeds the Threshold.
- **Priority Mode:** Priority mode opens the channel when the RMS input level exceeds the Threshold and the Priority set for the channel is higher than any other Priority channel currently open.
- **First In Last Out Mode**: When multiple signals are present simultaneously, FIL mode prioritizes the signal that entered the mix bus first. Even if other signals are added later, the signal that entered first will not be immediately removed unless its signal RMS falls below the Threshold;

④ **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;

⑤ **Automixer Type:** Gating Automatic Mixer or Gain Sharing Automatic Mixer;

⑥ **Gain (-20～20):** Adjusts the Mix output channel gain. This applies to all Input channels for the Mix output, but does not affect the individual Output Channels;

⑦ **Depth (-60～0):** The knob sets the amount of attenuation applied to any input channel when the channel's Gate is closed;

⑧ **Response Time (10～10000):** Adjusts the time constant that determines the time it takes for the attenuation to be applied to the output when there is a change in the NOM;

⑨ **Hold Time (0～1000):** The knob the sets minimum time an input stays open once it is opened, or the length of time an input stays open after the RMS input level drops below the Threshold Level. This is to prevent the gate from opening and closing due to momentary pauses in the input;

⑩ **Background Percentage (0 ～ 100%):** This control provides adaptive threshold functionality.This feature allows you to set the microphone Threshold levels fairly low for rapid response, but prevents the gates from opening when the background noise level rises.The background signal is the sum of all the microphone input signals. The Background Percentage knob determines the percentage of background signal used to raise the Threshold levels.Note: Set the Threshold levels when the room is quiet, if the gates open when the room gets noisy, increase the Background Percentage until the gates close;

⑪ **Max NOM:** Maximum number of microphones. When the number of microphones turned on at the same time exceeds the set value, the newly activated microphone will take over the lowest priority channel that is already turned on. If there are no channels available to take over (such as when all channels have higher priorities), the new microphone will not be able to connect;

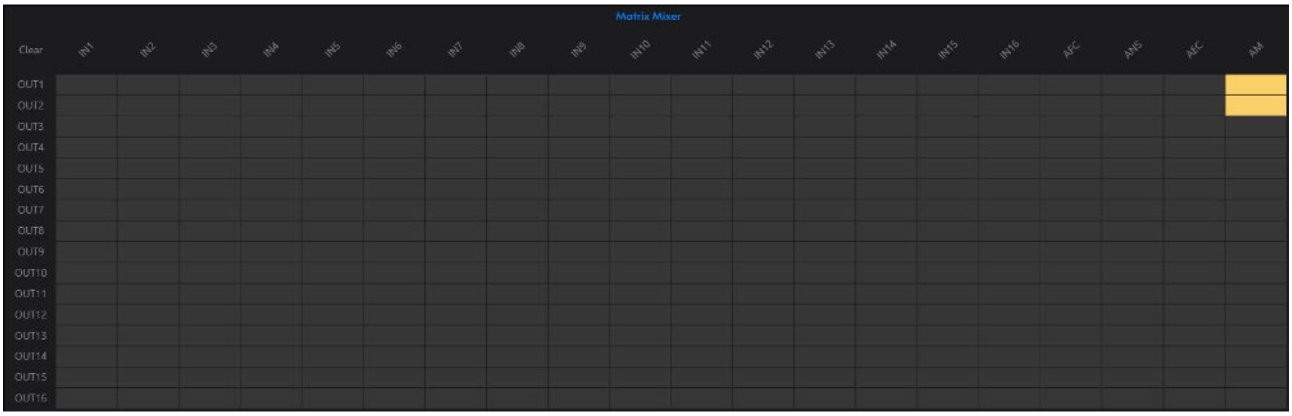
⑫ **Last Mic On:** Leaves the last microphone that was used in the open condition, until another microphone exceeds the Threshold;

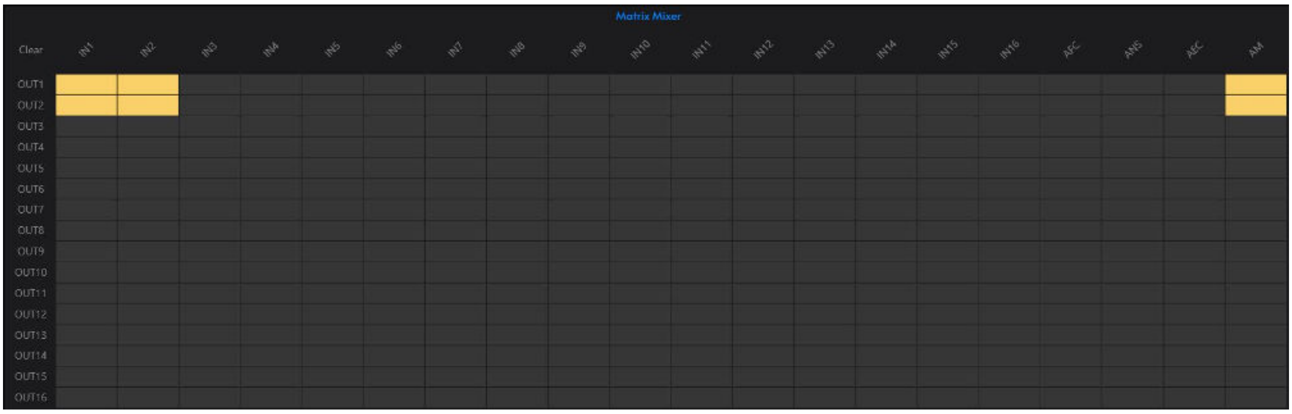
⑬ **ID Gating:** When the ID Gating button is on, will prevent multiple gates from opening if they are fed by the same source such as a talker who is positioned between two microphones;

⑭ **Bypass/Active:** Bypass or Active the Automixer. When the Automixer is bypassed, audio is passed through without any change;

⑮ **Reset:** Resets the parameters to the default.

Example, Automixer and Matrix Mixer association operations





For example, the signals of input channels IN1 and IN2 are automatically mixed and output in output channels OUT1 and OUT2, as shown in the figure above:

① First, open the [Matrix Mixer] configuration;

② Preset the output channel mapping relationship after mixing in [Matrix Mixer] interface, i.e. select the matrix mixing route corresponding to "AM" and output channels OUT1 and OUT2 in the input list, with corresponding channels color displayed;

③ Select the [Auto Mixer] configuration;

④ Check the input channel that needs to be enabled for auto mixing in the input list, i.e. turn on "Auto";

⑤ Configure the input source parameter system according to the signal priority;

⑥ Make fine adjustments to the relevant parameters according to the actual acoustic environment of the scene.

4.10.7 AFC

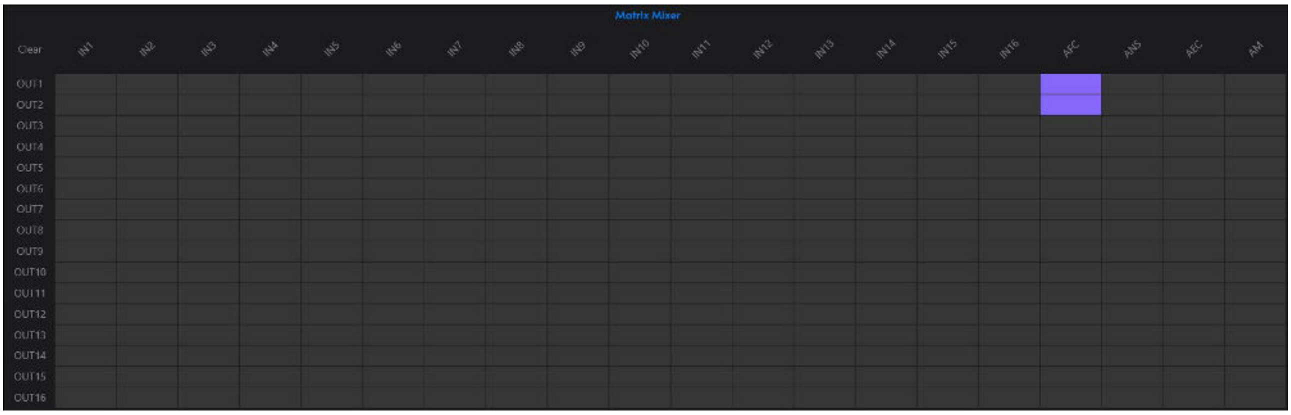
The Acoustic Feedback Canceler component allows you to set feedback suppression based on the room, and automatically suppress feedback. The AFC reduces the gain at the notch frequency while affecting the gain at surrounding frequencies as little as possible.

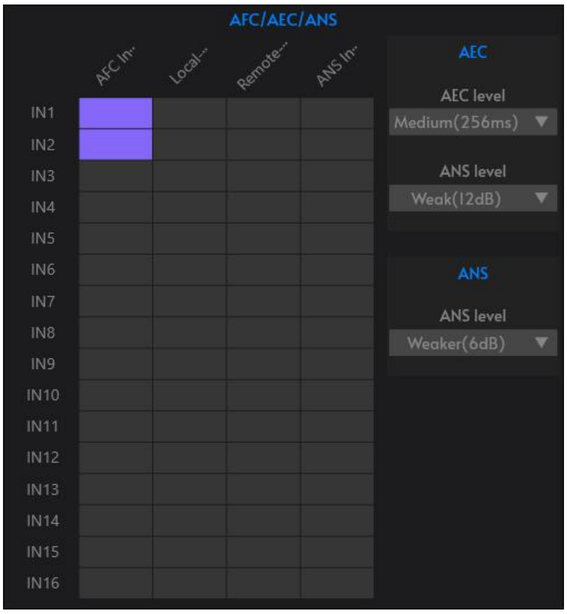
The AFC is commonly used in scenarios such as conference rooms and concerts. Its core functions and features include real-time monitoring of audio input signals, analyzing frequency components that may cause feedback using digital signal processing (DSP) technology, and attenuating or frequency-shifting these components. The AFC can detect feedback frequencies and process them within an extremely short time (0.01 seconds), thereby preventing feedback from occurring. It dynamically adjusts gain based on changes in the audio signal to ensure that feedback is suppressed without compromising audio quality.

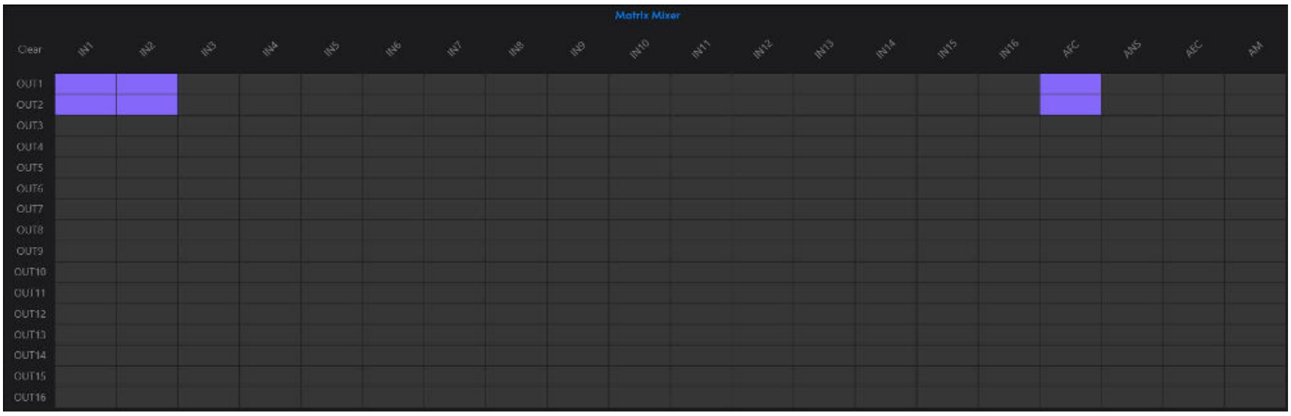
① **AFC Input:** The channel for local Microphone output, i.e. the signal that needs to be processed by the feedback cancellation;

② The AFC/AEC/ANS components feature gain faders that allow independent control of gain for any channel, with a gain range of -72 to 12 dB.

Example, AFC and Matrix Mixer association operations







The signals of input channels IN1 and IN2 will be processed for feedback and output in output channels OUT1 and OUT2, configured as shown in the figure above:

① First, open the [Matrix Mixer] configuration;

② Complete the audio output channel configuration in the [Matrix Mixer] to ensure that the processed feedback signal can be correctly routed to the output channel, i.e., select the points corresponding to the input list "AFC" and the output channels OUT1 and OUT2, with corresponding channels color displayed;

③ Select the [AFC/AEC/ANS] configuration;

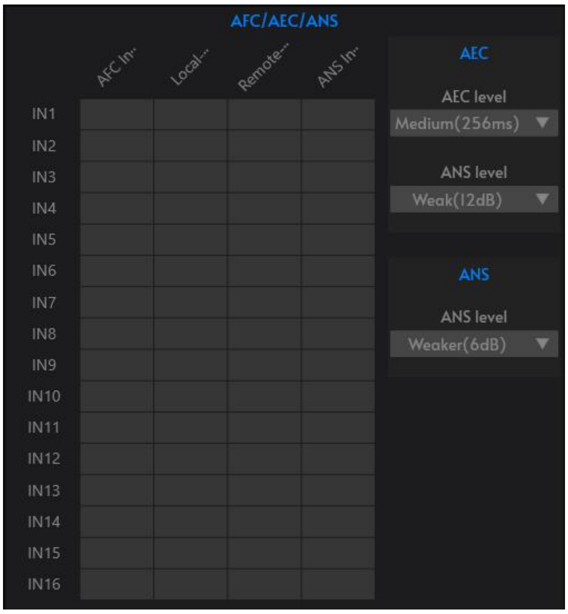
④ Configure the input channels of the audio source to be processed, i.e., the input channels IN1 and IN2;

⑤ Optimize the debugging through the parameter adjustment interface based on the on-site acoustic environment and effect requirements.

4.10.8 AEC

The Acoustic Echo Canceler (AEC) component 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 connected to the local-echo channel of the AEC system. Each channel simultaneously receives the speaker signal containing the remote speaker's voice, referred to as the mixed signal. The remote caller's voice is fed into the remote-echo channel of the AEC system, termed the remote reference signal. The AEC system employs adaptive filters to simulate the actual echo path in the real scenario (i.e., simulating the echo generated after passing through the acoustic environment) by continuously adjusting filter coefficients. It then calculates an echo estimate signal based on the remote reference signal. This echo estimate signal is subtracted from the mixed signal collected from the conference room microphones, thereby eliminating echo. After removing primary echo, a stable double-talk detection method is employed. Even in environments with strong background noise and nonlinear distortion, residual echo and background noise are further eliminated through nonlinear processing, ensuring the purity of the speech signal.



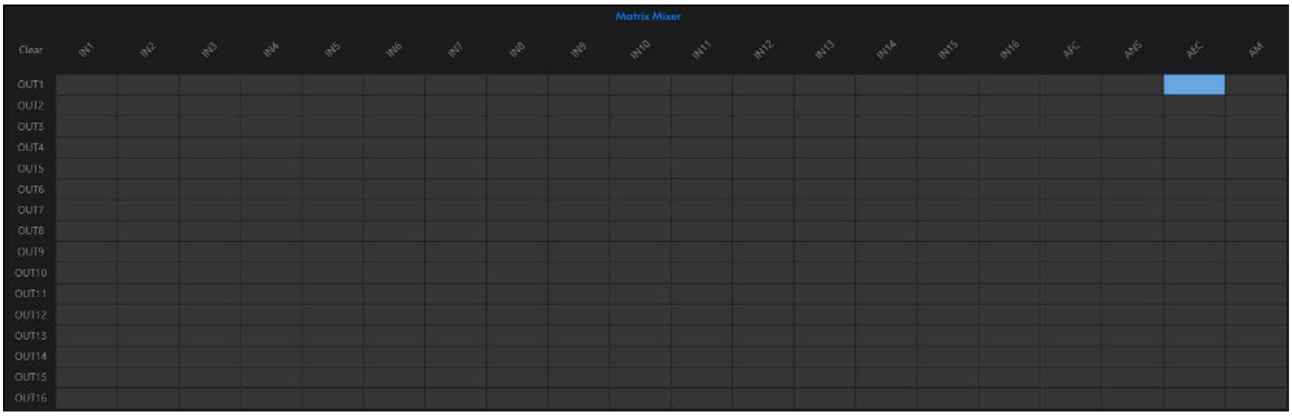
① **Local-AEC:** Local Microphone output channel, i.e. the signal that needs to be processed for AEC;

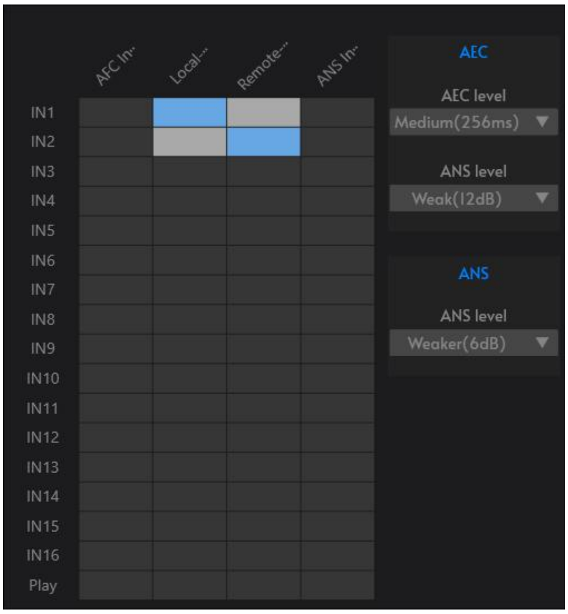
② **Remote-AEC:** The Echo remote input, i.e. the reference signal;

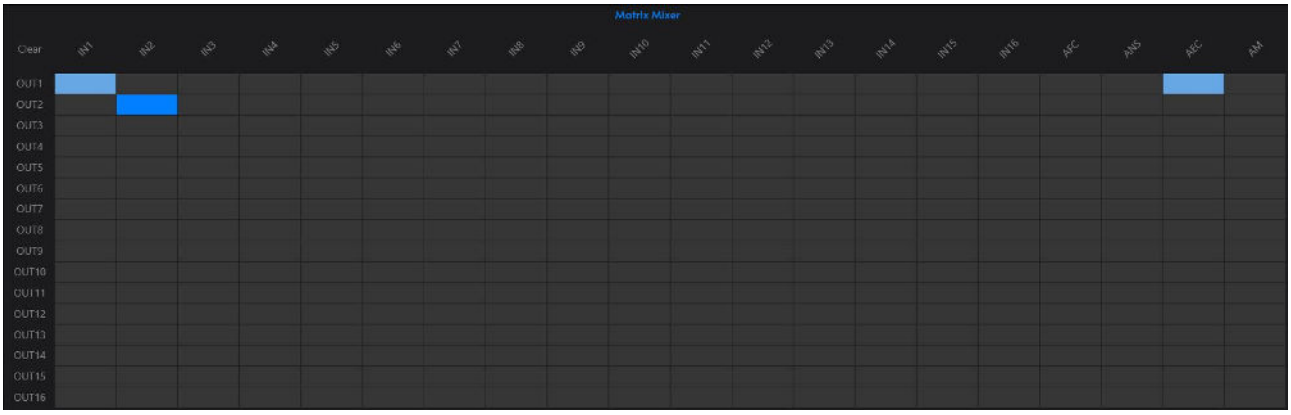
③ **AEC level:** Echo level range [Small room (128ms), Medium room (256ms), Large room (512ms)];

④ **ANS level:** Noise reduction level range (6～30dB).

Example, AEC and Matrix Mixer association operations







The local input signal is IN1 channel, the remote input signal is IN2 channel, the local input signal is output to the remote via OUT1 channel, the remote input signal is output to the local playback via OUT2 channel, the configuration is as described in the above figure:

① First, open the [Matrix Mixer] configuration;

② Complete the configuration of the audio output channels in the Mixing Matrix module to ensure that the processed echo signals can be correctly routed to the output channels, namely Select the input list "AEC" and the matrix mixing route corresponding to the output channel OUT1, with corresponding channels color displayed;

③ Select the [AFC/AEC/ANS] configuration;

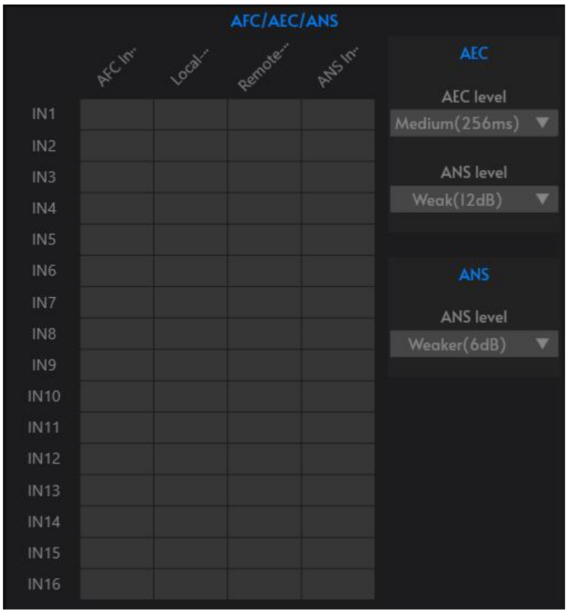
④ Configure the input channels of the audio source to be processed, i.e., the local input channel IN1 and the remote input channel IN2, and the corresponding channels of the local and remote inputs will be displayed in gray to prevent the sound from being reproduced. The corresponding channels will be displayed in gray to prevent the algorithm from being activated abnormally due to checking;

⑤ The remote input signal IN2 is output to the local sound reinforcement through the OUT2 channel, i.e., the matrix mixing route corresponding to the input channel IN2 and the output channel OUT2 will be selected, and the corresponding channel will be displayed in green;

⑥ Optimize the debugging through the parameter adjustment interface based on the on-site acoustic environment and effect requirements.

4.10.9 ANS

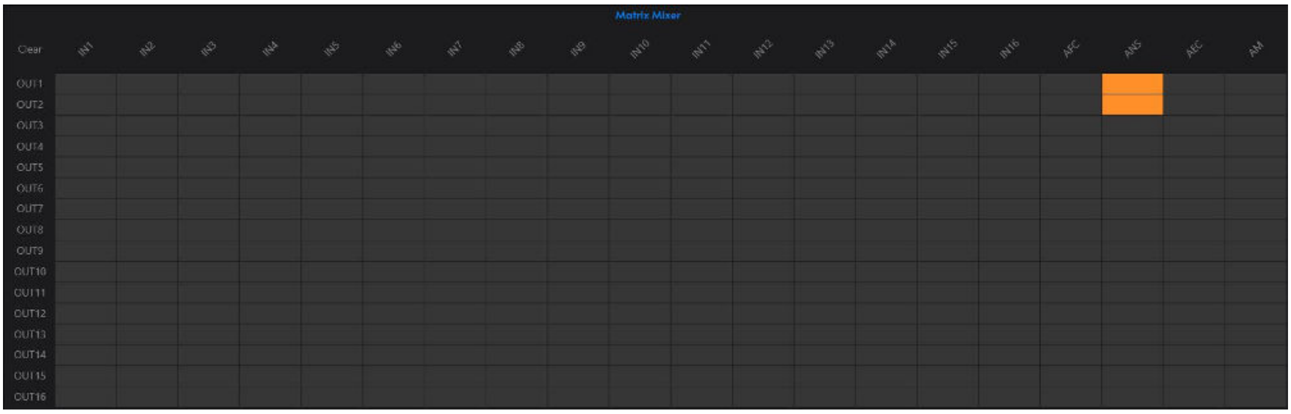
The Adaptive Noise Suppression component aims to remove background noise from audio signals and extract clear speech signals. Based on spectral analysis, it estimates noise spectral characteristics and processes signals using techniques such as spectral subtraction and Wiener filtering. This enables adaptive dynamic adjustment of noise estimation parameters according to varying noise environments, accommodating complex and fluctuating background noise. By distinguishing speech signals from background noise through speech activity detection, it applies noise suppression exclusively to non-speech signals. This technology finds extensive application in speech recognition, voice communication, and related fields. Utilizing a unique post-processing algorithm, the noise suppression component rapidly and accurately tracks environmental noise variations while significantly enhancing speech signal clarity.

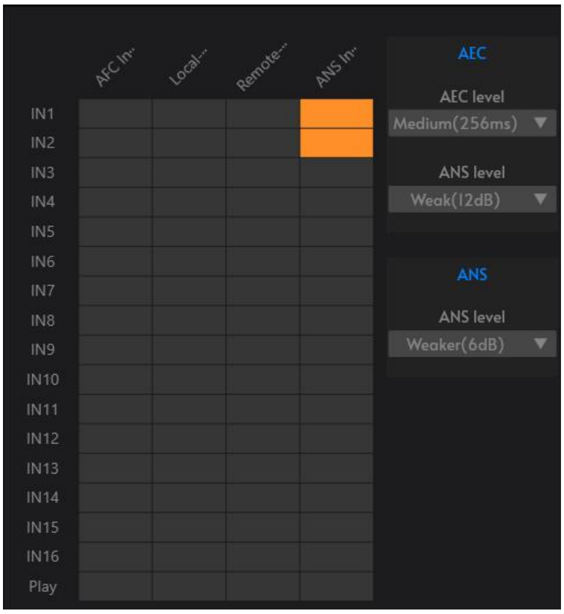


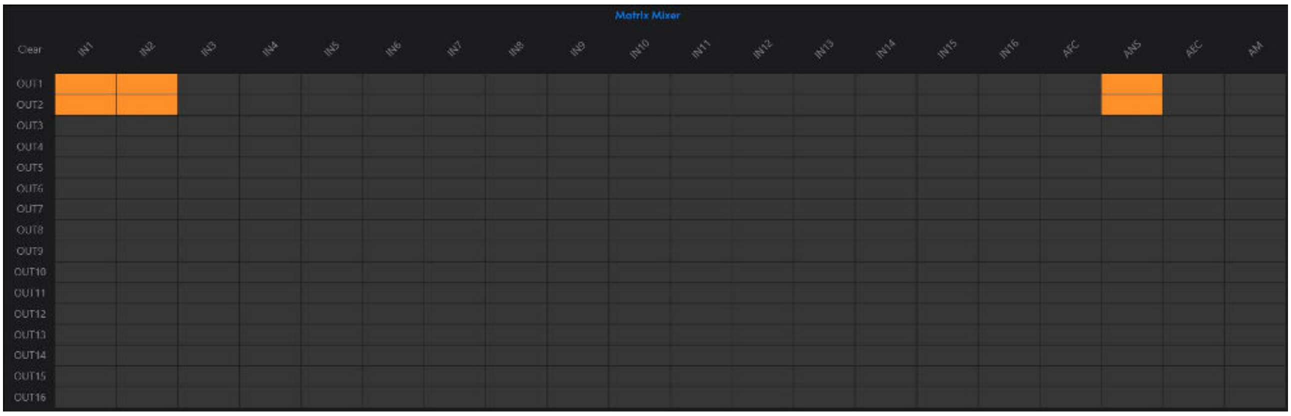
① **ANS Input:** Local Microphone output channel, i.e. the signal that needs noise suppression processing;

② **ANS level:** Range of noise reduction level (6～30dB).

Example, ANS with Matrix Mixer association operations







The signals of input channels IN1 and IN2 will be processed for ANS and output in output channels OUT1 and OUT2, configured as shown in the figure above:

① First, open the [Matrix Mixer] configuration;

② Complete the audio output channel configuration in the [Matrix Mixer] to ensure that the processed noise signal can be correctly routed to the output channel, i.e., select the matrix mixing route corresponding to the input list "ANS" and the output channels OUT1 and OUT2, with corresponding channels color displayed;

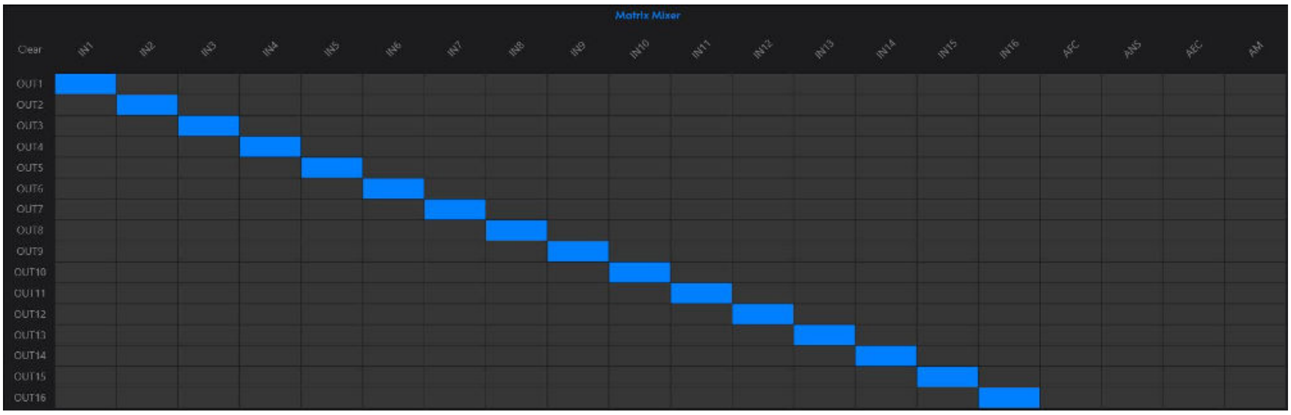
③ Select the [AFC/AEC/ANS] configuration;

④ Configure the input channels of the audio source to be processed, i.e., the input channels IN1 and IN2;

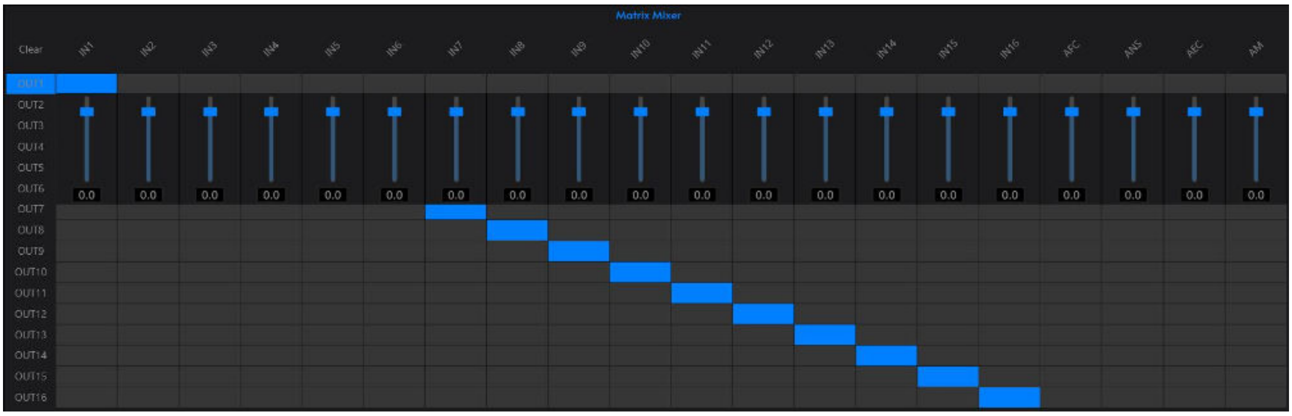
⑤ Optimize the debugging through the parameter adjustment interface based on the on-site acoustic environment and effect requirements.

4.11 Matrix Mixer

The Matrix Mixer component provides signal routing and mixing capabilities to address diverse complex audio processing requirements. Its control logic is clearly organized with input channels arranged horizontally and output channels vertically, enabling flexible adjustment of signal routing and mixing. This facilitates adaptable distribution and blending of input signals, supporting matrix-style full-mix switching that routes signals from any input channel to any output channel.



The Matrix Mixer component's OUT channel features a gain fader that allows independent control of any output channel's gain, with a gain range of -72～12 dB.



4.12 Output Component Configuration

4.12.1 Delay

The Delay component is primarily used to apply time delays to audio signals, enabling various audio effects and optimizing audio system performance. By activating the delay during signal processing and imposing specific time delays on the audio signal, it alters the signal propagation time to achieve specialized sound processing. Users can configure the delay component with fixed delay durations according to their requirements. The delay component supports a delay time range of 0～2000 milliseconds, accommodating diverse application scenarios.

- **Reverb Effect:** By setting an appropriate delay time, the reverb effect simulates the reflection and diffusion of sound in a space, enhancing the sound's spatial and three-dimensional quality, as if you were immersed in a specific acoustic environment.
- **Echo Effect:** By using a delay unit to generate repeated sound signals, natural echoes are simulated, enhancing the layering and depth of the sound.
- **Sound Optimization:** In larger performance venues, delay units can be used to assist with speaker processing. By applying different delays to different speakers, sound is distributed evenly throughout the space, avoiding sound overlap and interference, and optimizing the overall sound field effect.

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

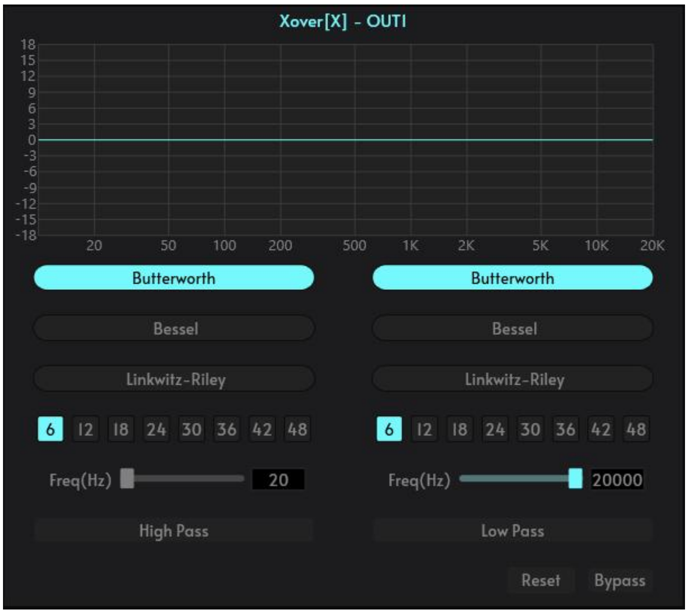
② **Delay distance:** Delay distance range (0～680m). This provides an alternative method for setting delay times in units of distance, ranging from 0 meters to 680 meters. Entering delay times as distances is often more convenient in practical situations;

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.

4.12.2 Xover

The Crossover component 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. Crossover play an indispensable role in fields such as audio processing, sound system design, and professional audio production.



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;

- **Buttworth filter:** A filter with maximum flatness, featuring a frequency response curve that is as flat as possible within the passband with no ripple. Its amplitude response exhibits a moderate transition (roll-off) rate from the passband to the stopband;
- **Bessel Filter:** Exhibits flat amplitude and linear phase (i.e., constant group delay) response within the passband. Its amplitude response features a low transition (roll-off) rate from passband to stopband. Constant group delay ensures minimal waveform distortion by maintaining a linear relationship between phase shift and frequency for all signals within the passband;
- **Linkwitz-Riley Filter:** Composed of two second-order Butterworth filters cascaded together, it exhibits a steep attenuation slope of 24dB per octave while maintaining flat amplitude and phase response within the passband.

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 6dB/Oct, 12dB/Oct, 18dB/Oct, 24dB/Oct, 32dB/Oct, 36dB/Oct, 42dB/Oct, 48dB/Oct;

- **Low slope (Such as 6dB/Oct, 12dB/Oct):** The transition is relatively smooth, suitable for scenarios requiring a gentle transition, but the crossover effect is not clean enough and may result in frequency band overlap;
- **High slope (Such as 24dB/Oct, 48dB/Oct):** The transition is steep, resulting in a clean crossover effect, but may cause sound discontinuity at the frequency band transition points;
- **Common slope:** 24dB/Oct is a commonly used compromise, effectively dividing frequency bands while avoiding overly abrupt transitions.

③ **High Pass Bypass/Active:** Bypass/Active the High Pass filter. **High Pass Filter:** Allows high-frequency signals to pass through and cut off low-frequency signals based on the set center frequency and slope, typically used to remove low-frequency interference or extract high-frequency features;

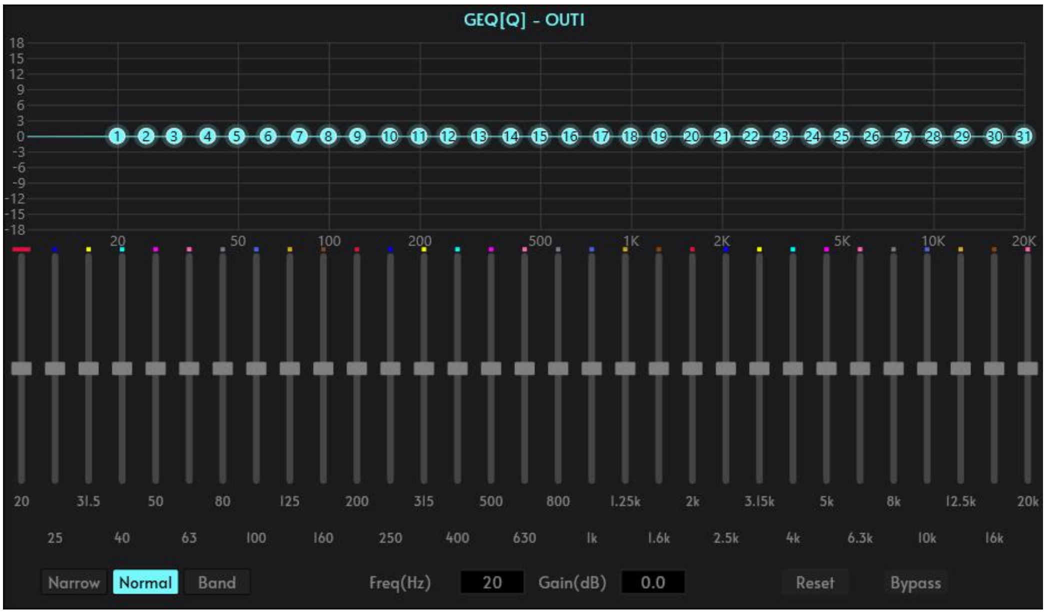
④ **Low Pass Bypass/Active:** Bypass/Active the Low Pass filter. **Low Pass Filter:** Allows low-frequency signals to pass through and cut off high-frequency signals based on the set center frequency and slope, typically used to remove high-frequency noise or increase low-frequency components;

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

⑥ **Reset:** Rests the parameters to the default.

4.12.3 Graphic Equalizer

The Graphic Equalizer component is a common audio processing tool widely used in music production, live sound mixing, home theaters, and professional audio systems. It enables precise adjustment of audio signals through 31 bands with 1/3th octave per band fixed-frequency filters, with gain or attenuation for each frequency band intuitively controlled via sliders. The design of this equalizer draws inspiration from analog-era mixing consoles, with its graphical interface enabling users to quickly and intuitively adjust the audio spectrum.



① **Narrow:** Narrowband equalization filter have a narrow bandwidth and are mainly used for precise adjustment of specific frequencies. They are suitable for removing specific interference frequencies or enhancing a specific tone;

② **Normal:** Normal equalization filter has a moderate bandwidth and is suitable for general audio adjustment scenarios, balancing the details of the tone and the overall effect;

③ **Wide:** Wideband equalization filter has a wide bandwidth and are suitable for adjusting a large frequency range. They are often used to shape or adjust the overall tone;

④ **Frequency:** The center frequency is the center point of the current equalization filter and also the reference frequency for gain adjustment. Adjusting the center frequency can change the filter's range of action;

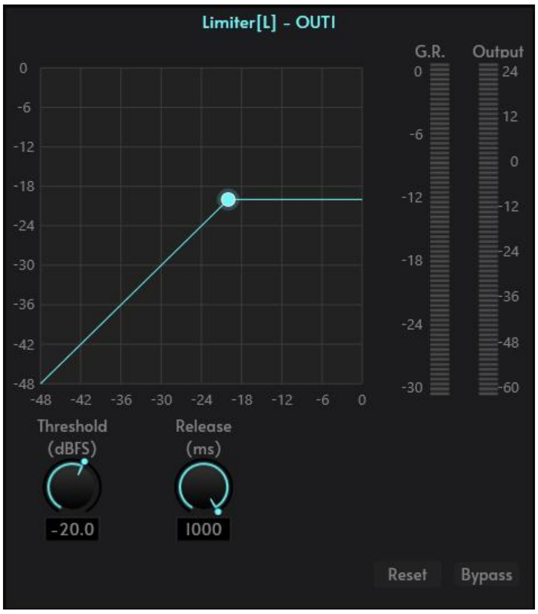
⑤ **Gain:** Controls the output gain of an individual frequency band, positive values indicate an increase in gain for that frequency band, while negative values indicate a decrease in gain for that frequency band;

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

⑦ **Rest:** Rests all the band gains to the default.

4.12.4 Limiter

The Limiter component serve as a vital dynamic range control tool in audio processing. Their primary function is to limit the output level to the Threshold Level, prevent signal overload and transient interference while ensuring stable and consistent audio output. When the input signal exceeds the Threshold, the Limiter automatically reduces the signal's gain, thereby avoiding clipping distortion caused by signal overload.



① **Threshold (-48～0):** Sets the level at which the Limiter has an effect, and the level at which the output is held;

② **Release Time (1～1000):** The release time determines how long it takes for the input signal to return to its original dynamics from maximum attenuation. It dictates how long the limiter continues to attenuate the signal after it falls below the Threshold Level. Shorter release time enables rapid signal dynamics release, suitable for fast-changing audio but prone to Pumping effect (rapid fluctuations in signal level). Longer release time provides smoother transitions and reduces Pumping effect, though it may make the signal release process appear sluggish. Settings should be balanced according to the audio characteristics;

③ **Gain Reduction:** Graphically displays the amount of attenuation applied to the Channel, Gain Reduction reflects the degree to which the limiter attenuates the signal. For example, if the input signal exceeds the threshold by 3dB, the limiter may attenuate the signal by 3dB, resulting in a compression of 3dB;

④ **Bypass/Active:** Bypass or Active the Limiter for the current channel. When the Limiter is bypassed, audio is passed through without any change;

⑤ **Reset:** Resets the parameters to the default.

4.12.5 Output Setting

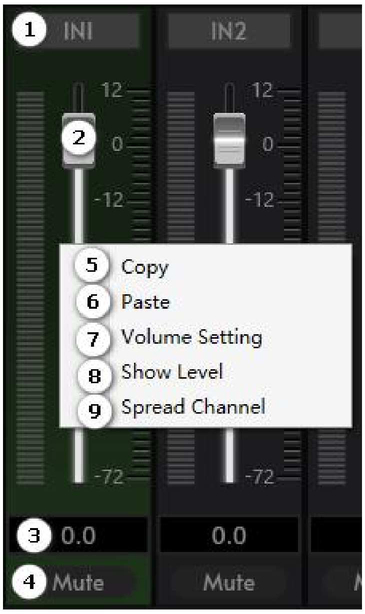
The Analog Output component provides one channel of line-level output for device. The Analog Output converts the processed digital signal to analog. Connections are made using one three-terminal 3.5mm Phoenix connectors. Inverse and Mute are two crucial functions within the output section, playing a vital role in audio processing and live sound mixing.

① **Invert**: Inverts the polarity of the output signal. In audio processing. Phase is a critical parameter that determines the starting point and direction of a signal waveform. Through phase inversion, the phase of an audio signal is reversed by 180 degrees. Application Scenarios Resolving Phase Issues In multi-speaker systems, phase differences between signals from different speakers can cause sound cancellation or interference. By utilizing channel phase inversion functionality, phase adjustments can be made to ensure sound clarity and consistency;

② **Mute:** Mutes the output signal.

4.13 Other Functions

4.13.1 Input Channel Control



The channel right-click to access more functions.

① Customizable channel name;

② **Fader:** Mouse pointer to the gain fader, the channel gain fader adjustment method is based on the step fine-tuning of the up and down arrow keys in steps of 0.1 dB;

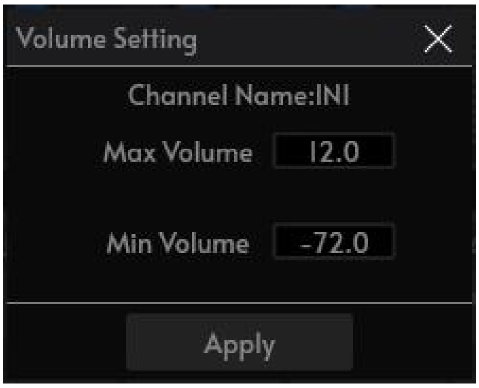
③ Can manually enter the volume within the range of -72 to 12dB. **Note: Press [Enter] to apply the setting**;

④ **Mute:** Mutes the current channel;

⑤ **Copy:** Copy current channel information;

⑥ **Paste:** Paste the copied channel information into the current channel;

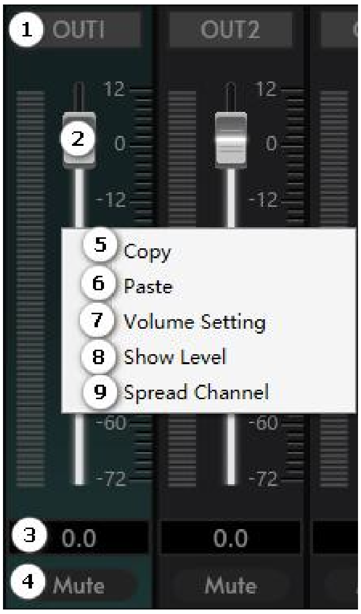
⑦ **Volume Setting:** Right-click the channel to directly access the volume fader constraint range settings. **Note:** The control code layer holds the highest priority for volume fader constraint permissions, and its settings will override the parameter range configured via the channel right-click;



⑧ **Show Level:** Display the current channel level value;

⑨ **Spread Channel:** Spread all input channels, the parameter panel is instantly displayed globally, facilitating centralized and batch parameter adjustments.

4.13.2 Output Channel Control



The channel right-click to access more functions.

① Customizable channel name;

② Mouse pointer to the gain fader, the channel gain fader adjustment method is based on the step fine-tuning of the up and down arrow keys in steps of 0.1 dB;

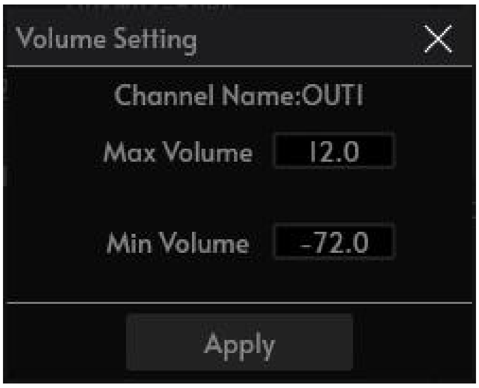
③ Can manually enter the volume within the range of -72 to 12dB. **Note: Press [Enter] to apply the setting**;

④ **Mute:** Mutes the current channel;

⑤ **Copy:** Copy current channel information;

⑥ **Paste:** Paste the copied channel information into the current channel;

⑦ **Volume Setting:** Right-click the channel to directly access the volume fader constraint range settings. **Note:** The control code layer holds the highest priority for volume fader constraint permissions, and its settings will override the parameter range configured via the channel right-click;

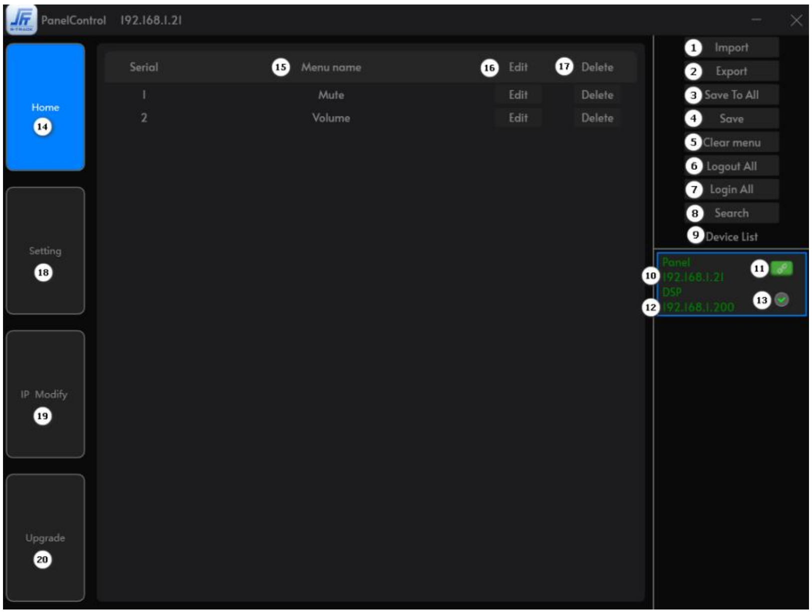


⑧ **Show Level:** Display the current channel level value;

⑨ **Spread Channel:** Spread all output channels, the parameter panel is instantly displayed globally, facilitating centralized and batch parameter adjustments.

Chapter 5 P1 Plus Configuration User Guide

5.1 Menu - Setting - P1 Plus Panel Control Main Interface



① Import panel configuration information into the software;

② Export panel configuration information to the panel;

③ Save configurations of all panels in the device list to the control panel;

④ Save configurations of specified panels in the device list to the control panel, with a maximum of 99 menu items;

⑤ Clear all menu items;

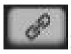
⑥ Log out all connected control panels;

⑦ Log in all connected control panels;

⑧ Search for online control panels;

⑨ List of online control panels;

⑩ Display names and IP addresses of online control panels;

⑪ Control Panel Connection: : Connected; : Disconnected;

⑫ Display the names and IP addresses of DSPs bound to control panels;

⑬ Display DSP connection status: : Connected; : Disconnected;

⑭ Home interface;

⑮ Added menu items;

⑯ Edit menu item information;

⑰ Delete menu item;

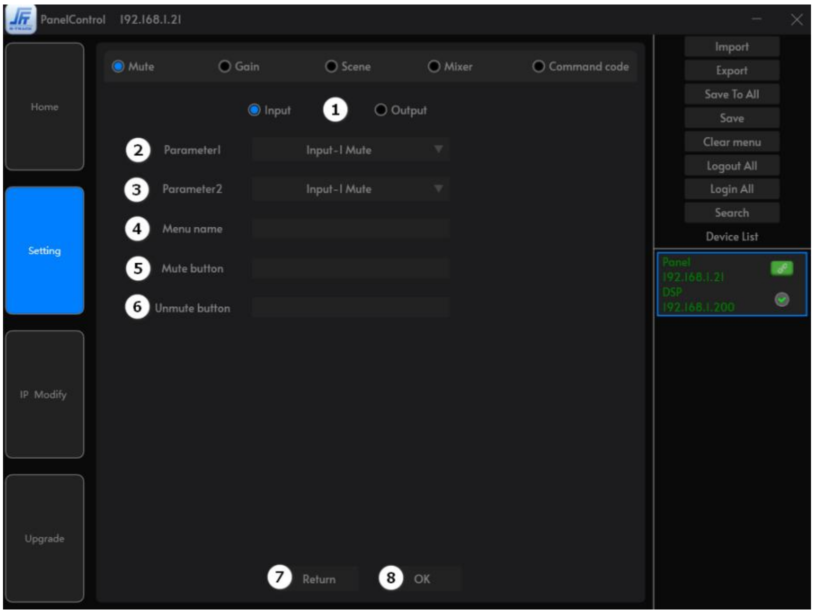
⑱ Setting interface

⑲ IP address modification interface;

⑳ Upgrade interface.

5.2 Setting

5.2.1 Mute



① Select input or output channel configuration parameters;

② Start channel;

③ End channel;

④ Menu name displayed on the Home and control panel;

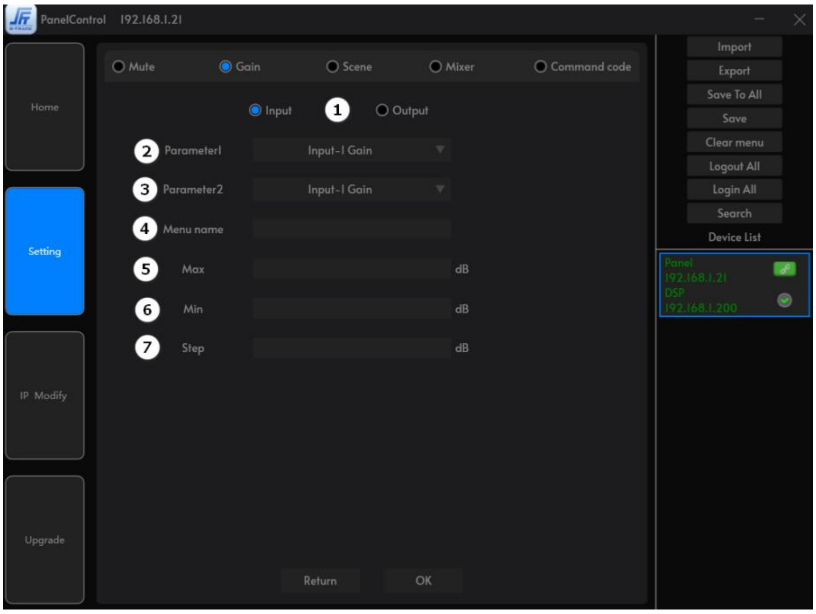
⑤ Mute button name displays on the control panel;

⑥ Unmute button name displays on the control panel;

⑦ Return to Home;

⑧ Confirm.

5.2.2 Gain



① Select input or output channel configuration parameters;

② Start channel;

③ End channel;

④ Menu name displayed on the Home and control panel;

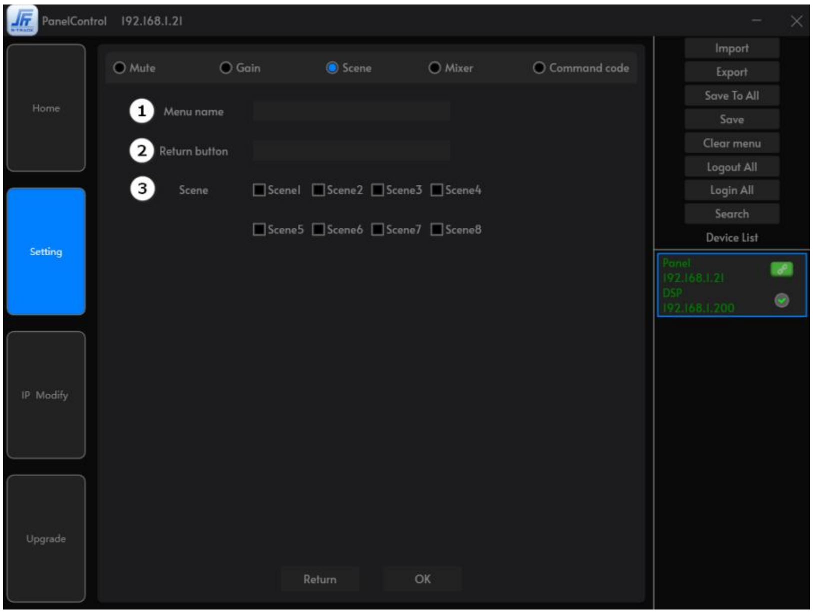
⑤ Set the maximum gain;

⑥ Set the minimum gain;

⑦ Set the step length for adjusting gain;

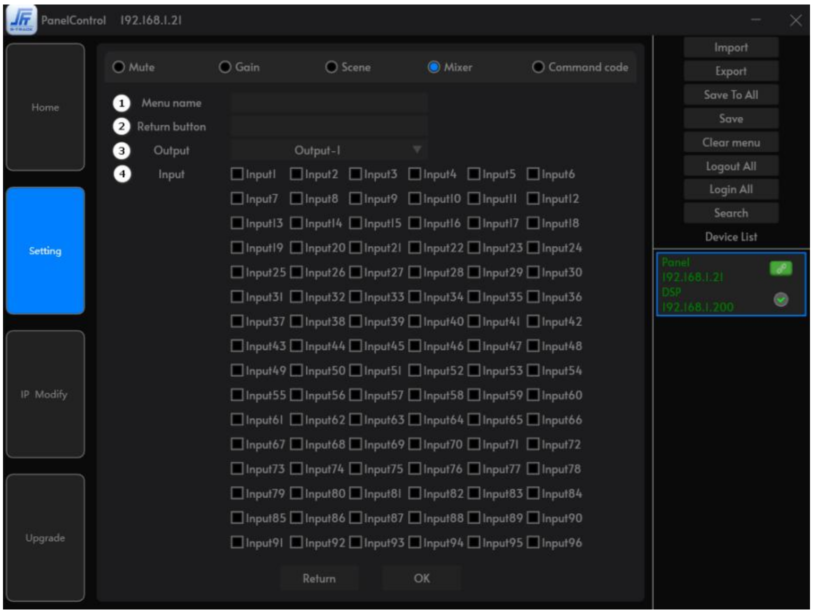
Note: The DSP maximum is 12dB, minimum is -72dB, and step length ≥0.1dB.

5.2.3 Scene



1. Menu name displayed on the Home and control panel;
2. Return button name displays on the control panel;
3. Select the scene for control switching.

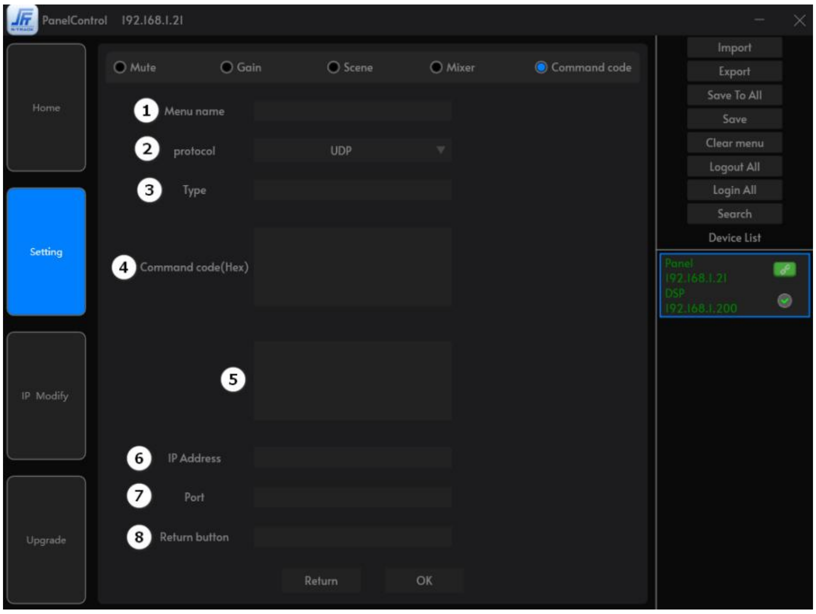
5.2.4 Mixer



1. Menu name displayed on the Home and control panel;
2. Return button name displays on the control panel;
3. Select Output channel;

④ Select Input channel;

5.2.5 Command code



① Menu name displayed on the Home and control panel;

② Central control command protocol;

③ Central control command type;

④ Hexadecimal central control command code input field;

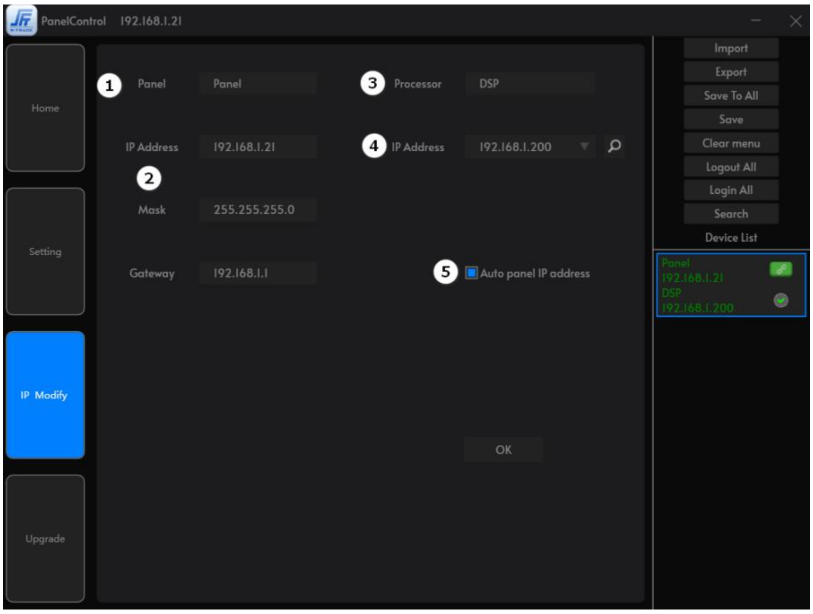
⑤ ASCII central control command code input field. Note: Entering ASCII code will display hexadecimal in ④;

⑥ Controlled device IP address;

⑦ Controlled device IP address port number;

⑧ Return button name displays on the control panel;

5.2.6 IP Modify



① Customizable control panel name;

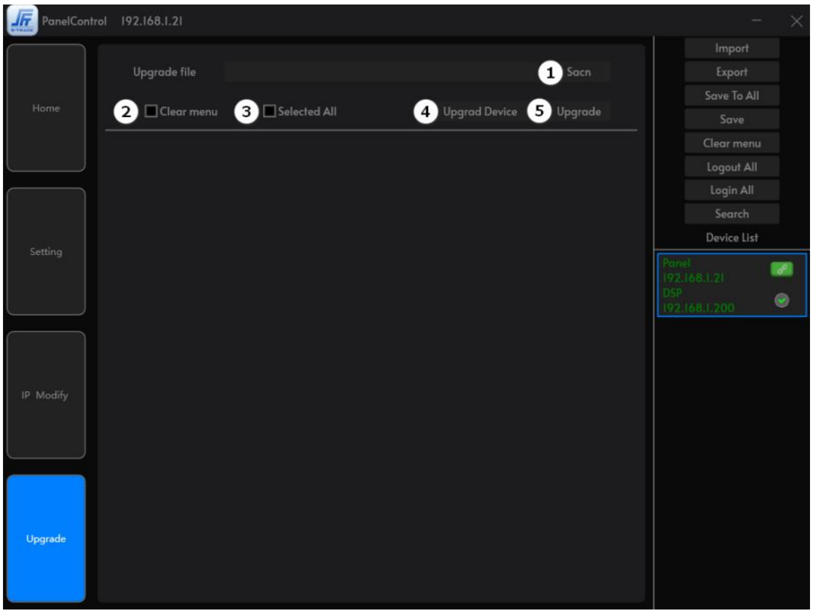
② Control panel IP address, subnet mask and gateway;

③ Name of the bound signal processor;

④ IP address of the bound signal processor, which can be searched and selected via a dropdown menu;

⑤ Configure the control panel to automatically obtain an IP address from the DHCP server.

5.2.7 Upgrade



① Open the upgrade file;

② Clear menu items when upgrading the control panel;

③ Select all control panel upgrades;

④ Upgrade the control panel firmware;

⑤ Upgrade the Panel Control software.

Chapter 6 FAQ

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: Unplug all GPIO connection cables and reboot the device.

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

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.

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.

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. 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 check DHCP).

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.

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.

8. RS485 center control command does not work

First, check whether the connection is normal, the "+" of the center control host connects to the "+" of the device, the "-" of the center control host connects to the "-" of the device. ", the central control host and device ground interconnection; second, check the software configuration of the device interface items: baud rate, start bit, stop bit and other settings with the central control host interface configuration is consistent. If the problem is not solved, please contact the manufacturer.

Chapter 7 Packing List

Device	Power Adapter	Quick Guide	12 pin Phoenix Connector	6 pin Phoenix Connector	Small Screwdriver
1PCS	1PCS	1PCS	-	1PCS	1PCS

Chapter 8 Specification

Category	**Parameter Item**	**Parameter Description**
Peripherals	Input Interfaces	16 Analog
	Output Interfaces	16 Analog
	Control Interfaces	1 RJ45 interface, 1 RS485 interface, 2 GPIO control 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 component: Preamplifier, Signal Generator, Expander, 5-band Parametric 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 component: Delay, Crossover, 31-band Graphic Equalizer, Limiter. 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	≤0.004% @1kHz, +4dBu
	Channel Isolation	104dB@1kHz
	Input Range	≤+18dBu (A-Weighting)
	Crossover	Three types of high and low pass filters: Butterworth, Bessel and Linkwitz-Riley
	Equalizer	Parametric Equalizer: Frequency: 20～20kHz, Gain: -15～+15dB, Bandwidth: 0.02～4 Graphic Equalizer: Frequency: 20～20kHz, Gain: -15～+15dB
	Maximum Output Level	18dBu
	Maximum Input Level	18dBu
	Analog/Digital Dynamic Range	114dB
	Digital/Analog Dynamic Range	120dB
	Equivalent Input Noise	≤-120dBu
General specification	Operating Voltage	Power adapter input: AC 100V～240V, 50Hz/60Hz; Output: DC 12V/2A.
	Maximum Power	24W

	Operating Temperature and Humidity	0℃～40℃, 10%～90%RH, No condensation
	Chassis	1U
	Product Dimensions (L×W×H)	482.4mm×210.5mm×44mm
	Net Weight	2.5kg
	Package Dimensions (L×W×H)	590mm×430mm×110mm
	Package Weight	3kg

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.

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