Dante 8-Channel Audio DSP

User Manual

500601

94-001083-A/SE-001083-A

Table of Contents

1. Safety Precautions

To ensure the best performance from the product, please read all instructions carefully before using the device. Save this manual for future reference.

- Follow basic safety precautions to reduce the risk of fire, electrical shock, and injury.
- Do not dismantle the housing or modify the module. It may result in electrical shock or burns.
- Do not open or remove the housing of the device as you may be exposed to dangerous voltage or other hazards.
- To prevent fire or shock hazard, do not expose the unit to rain, moisture and do not install this product near water. Keep the product away from liquids.
- Spillage into the housing may result in fire, electrical shock, or equipment damage. If an object or liquid falls or spills on the housing, unplug the module immediately.
- Do not use liquid or aerosol cleaners to clean this unit. Always unplug the power to the device before cleaning.
- Using supplies or parts not meeting the product specifications may cause damage, deterioration or malfunction.
- Refer all servicing to qualified service personnel.
- Install the device in a place with adequate ventilation to avoid damage caused by overheat.
- Unplug the power when left unused for a long period of time.
- Information on disposal of devices: do not burn or mix with general household waste, please treat them as normal electrical waste.

2. Introduction

The Dante 8-Channel Audio DSP (Model 500601) is a product developed for the transmission, routing, and processing of audio signals in Dante networks. It can perform high-quality, low-delay audio transmission, supports 8-way input and output balanced analog channels, and can be powered via a 110-220VAC power receptacle. This processor supports 8-way analog audio signal balance input and output and 8-way Dante digital audio signal input and output. It also features a USB audio card that supports one channel of audio input and one channel of audio output. The device has one of the best DSP algorithms in the industry that includes options such as AEC, AFC, ANC, Ducker, Compressor, Limiter, Mixer and Graphic Equalization among others. In addition, the product provides PC version control software to monitor and operate all the functions on the DSP in an easy and intuitive way. Comprehensive API Commands are available for system automatization. Supports multiple sets of scene presets.

3. Features

- Built-in Dante module
- Features 8 balanced audio inputs and 8 balanced audio outputs
- Features a USB sound card
- The input supports 48VDC phantom power supply
- DSP processing functions include gain adjustment, stage parameter equalization, compressor, ducker, mixer, graphic equalization, limiter, setting, etc
- Provides software for the Windows platform to manage all audio operations
- API provided for RS232, RS485 and TCP/IP control
- Supports AES67 RTP audio transport
- Supports GPIO for additional interface expansion

4. Package Contents

- One (1) Dante 8-Channel Audio DSP
- One (1) Power Cable
- One (1) 3-pin Phoenix Connector
- Five (5) 12-pin Phoenix Connector
- One (1) Small Screwdriver
- One (1) User manual (available via download)

5. Specifications

Specification	
Analog Audio	8 balanced inputs + 8 balanced outputs
Dante Audio	8 inputs + 8 outputs
USB Sound Card	1 input + 1 output
Analog Maximum Gain	51dB
Quantization Bits	24bit
Sample Rate	48k
Phantom Power	48VDC
Frequency Response	20Hz~20KHz±0.25dB
THD+N	≤0.003% @ 1KHz, +4dBu
Digital/Analog Dynamic Range	114dB
Analog/Digital Dynamic Range	120dB
Input Impedance	20kΩ Balanced，10kΩ Unbalanced
Output Impedance	100Ω Balanced，50Ω Unbalanced
Maximum Input Level	18dBu，Balanced
Maximum Output Level	18dBu，Balanced
Input Common Mode Rejection	84dB @ +24dBu @60Hz
Channel Isolation	>104dB
Number of GPIO	8 ports, can be configure as input and output
Number of RS485 serial ports	1
Number of RS232 Serial Ports	1
Number of USB sound cards	1
Operating Temperature	0℃ - 40℃
Working Power Supply	110-220VAC, 50/60Hz
Warranty	2 years
Order Information	500601　Dante 8-Channel Audio DSP (UPC: 627699006016)

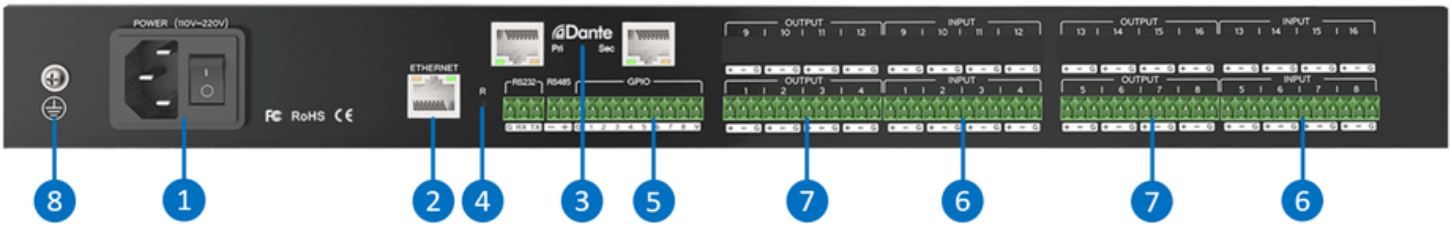
6. Interface description

Front Panel

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

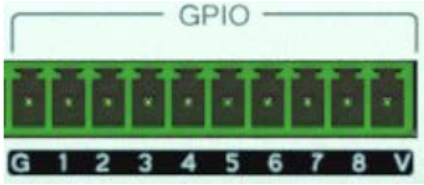
② SYS: System operation indicator, the indicator will flash once per second to indicate that the system is operating normally;

③ USB: USB audio card, support U disk recording and playback, or USB audio card, please refer to the actual device.



① POWER: Power supply interface, connect AC 100V-240V power supply, build-in switch control processor power;
② ETHERNET: Network control interface, through the connection of this network port, the client computer can control and monitor the device;
③ Dante: Dante network audio transmission interface, equipped with primary and secondary dual network interface, can be used for redundant backup of Dante network signals;
④ RESET: Reset button, long press to restore factory settings and reboot the processor;
⑤ RS485+RS232+GPIO: Connect to the control terminal or central control devices;
⑥ INPUT: Analog input interface, can be connected to mixer, microphone, PC and other devices;
⑦ OUTPUT: Analog output interface, can be connected to the amplifier, active speakers and other devices;
⑧ Ground screw: Used to ground the chassis, protect against leakage, electrostatic discharge and other measures.

GPIO



G: Ground
1~8: 8 GPIO ports, Port 2-8 are freely configurable input or output, port 1 supports only input.
V: The default GPIO power output pin(5VDC to 12VDC).

Note: ● Output high; ● Input high; ● Output low; ● Input low; ● Disable

Output Port: Each output port is OC output, the default reference voltage 5VDC and the reference voltage can also be external is less than 15V, take the default reference voltage, the total output ports is 200mA:

Input Port: Port input level > 2VDC will be recognized as high level; When there is no external power supply, the highest port input level is 5.5VDC; when there is an external power supply of 12VDC, the highest port input level is 12.5VDC.Input level< 0.3VDC is recognized as low.

Use of GPIO:

1. **Output**: When a parameter matrix of internal changes (such as mute), cause the change of the output level of a GPIO output pins, to drive an external circuit.
   Audio matrix internal change -> GPIO pin level change -> external driving circuit.
2. **Input**: GPIO is an input pin level changes due to the external circuit is changed, thus a parameter matrix changes.
   The external circuit state changes -> GPIO pin level change -> audio matrix internal change.

7. Software Operation

The installation software source files are embedded in the Digital Signal Processor device. Download the software simply by entering the device's factory default IP address (default IP: 192.168.1.200) information in the URL address bar of the browser, after will be able to navigate to the download interface, according to the content of the web interface information click on the download option. Additionally, ensure that the client PC has installed the NET Framework 3.5 or above for Windows system.

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

After the device started input address in webpage http://192.168.1.200

Control: Control channel parameters

Scene: Load a scene

Software: Download

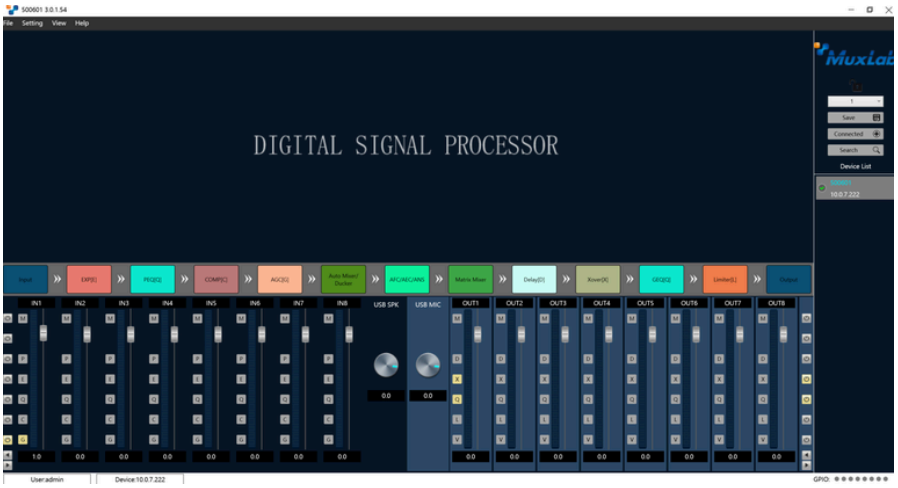
7.1 Login

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

7.2 IP Address Modification

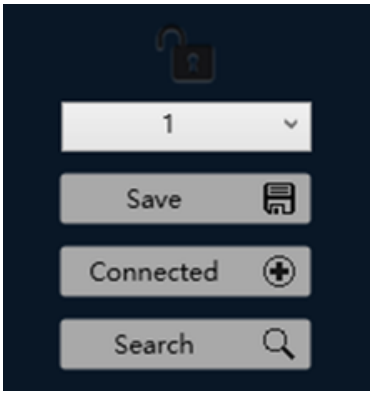
When the IP information of the device is not in the same network segment as the client, an 'exclamation mark' will appear in front of the model name of the device in the list of settings, at this time, you only need to double-click on the [model name] to bring up the IP information modification [dialogue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

7.3 Main Interface



① Menu Bar;
② Button Function Area;
③ Input...Output: Audio Processing Modules;
④ IN1...IN16: These are the analog input channels;
⑤ OUT1...OUT16: These are the analog output channels.

7.3.1 Button Function Area



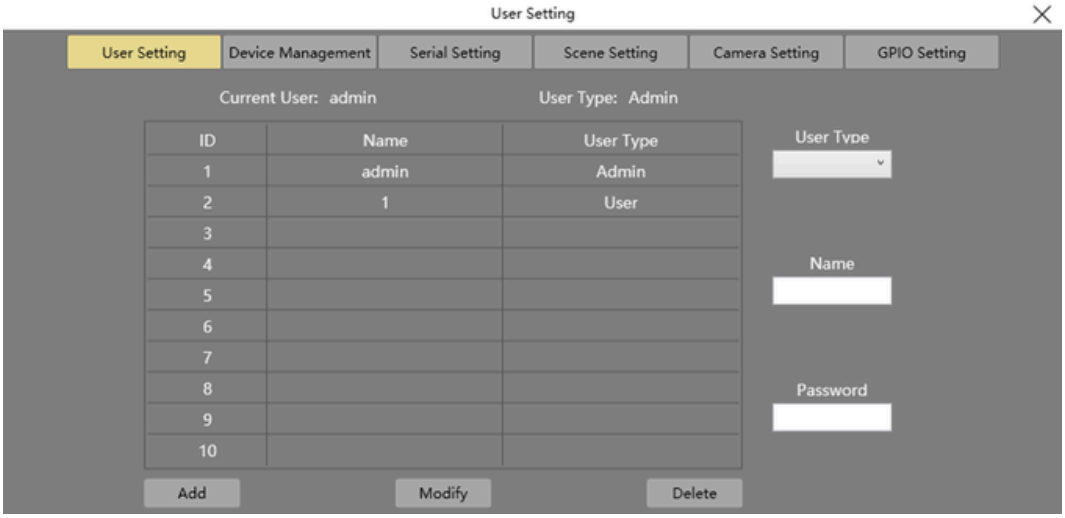
① Button: Device Search Button: Click to search for available devices and display device IP;
② Button: Device Connection Button: The IP of the device is known, and you can connect directly by entering the IP address, user name and password in the pop-up box;
③ Button: Save Scene button: saves (overwrites) the parameter changes to the selected scene;

7.3.2 File

① New: Create a new scene, the parameters are factory configured and only available offline;
② Open: Open the locally saved scene;
③ Save as: Save the current configuration (i.e. scene) as a file locally;
④ Exit: Close the software.

7.3.3 Setting - Device Setting

I: User Setting



① Initial user name of the device: admin, password: 123456, administrator can add, delete, modify all user information, ordinary users can only modify personal information;
② Add a user: Select an empty line in the left list, and enter the new user's information in the right user name and password edit box (should be empty), click "Add" button to add a new user;
③ Modify user: First select the user you want to modify in the user list, the user name and password edit box will display the information of the currently selected user, enter the new information and click the "Modify" button;
④ Delete User: Select the line in the user list to be deleted, click "Delete" button to delete the user.

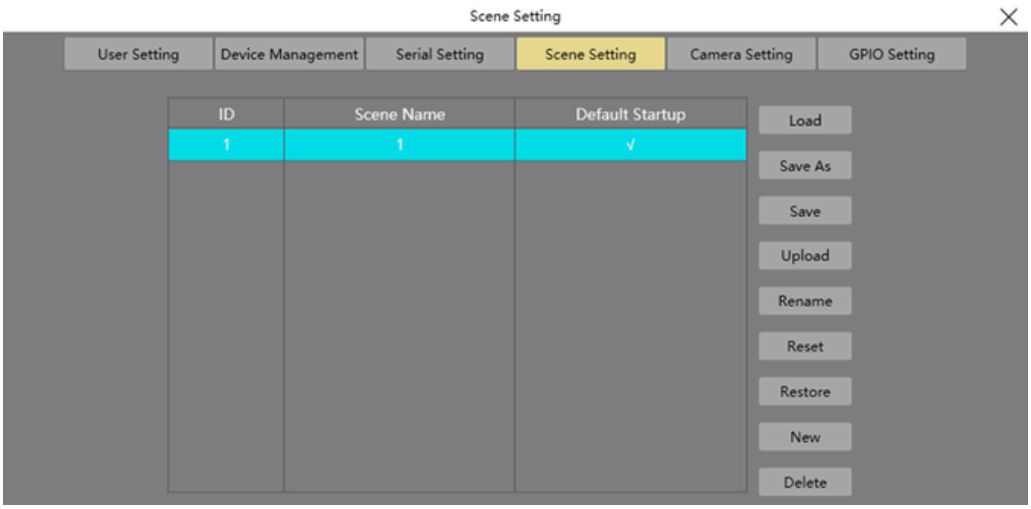
II: Device Management

View and modify the network address information and device name of the device, enter the network settings window, type IP address, subnet mask, gateway in the corresponding input box, enter the name of the device in the corresponding position, click the [Apply] button to complete the modification, [Reboot] can be in the window to control the device to soft reboot, click the [Restore] can be restored to the factory settings of the device.

III: Serial Setting

Serial window can view or modify the RS232/RS485 serial port number baud rate, Parity, data bit, and stop bit settings, when the settings are complete, click on the [Apply] button to modify the current device's serial port information, if you need to restore to the initial default value, click on the [Reset] button.

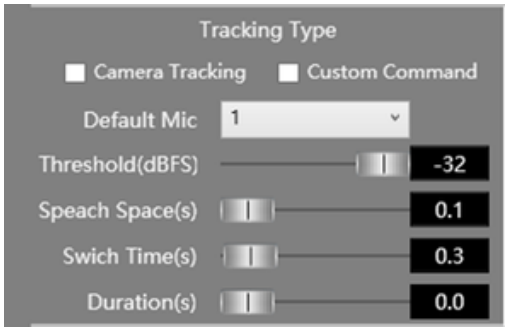
IV: Scene Setting



① Load Scene: Enable the currently selected scene, usually used for scene replacement;
② Save As: Save the selected scene locally;
③ Save: Save the currently running parameters to the selected scene;
④ Upload: Upload the scene from PC and overwrite the selected scene;
⑤ Rename: Modify the name of the selected scene;
⑥ New: Create a new scene file, you can customize the new scene, support up to 100 groups of scenes;
⑦ Delete: Select any scene in the list and delete it;
⑧ Reset: Restore the currently selected scene to the factory default state;
⑨ Restore: Restore all scene configurations to the default configuration and clear all new scenes, only retain the factory default 8 groups of scene files, please use with caution.

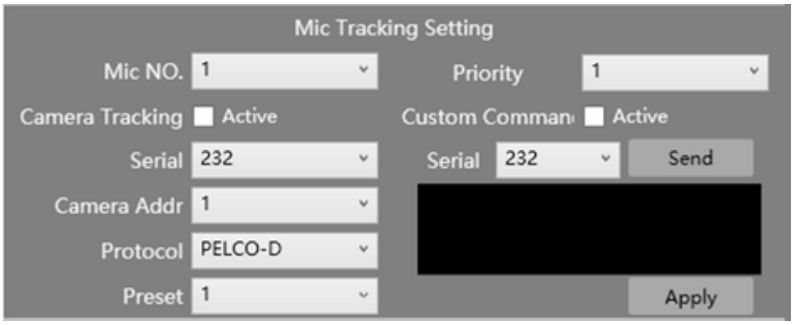
V: Camera Tracking setting

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



1. Tracking type: There are camera tracking and custom commands. Camera tracking is used to control camera rotation by channel input signal; custom command sending is used to send corresponding custom commands to the corresponding port by channel input signal;
2. Default: When there is no input from all microphones, rotate the camera to the position set by the default Mic or send the associated command defined by the default Mic. The one with # sign indicates the virtual number, which can only be used to set the default microphone;
3. Threshold: Means the detected input signal must be greater than or equal to the Tracking Threshold, and the system automatically enables the tracking parameter;
4. Speech space: Maximum intermittent time for a valid signal. If you use the microphone to speak, set the response time to 3 seconds, the signal is still regarded as continuously valid within 3S pause in the middle of the speech, and the signal is regarded as invalid if it exceeds 3S;

5. Switch time: The shortest speaking time required for the camera to switch to a valid position. If you use the microphone to speak, the length of speech must be greater than the "switching time", the channel signal is considered valid, and then the camera will automatically turn to the set position. Usually the "switching time" is greater than the "reaction time";
6. Duration: The interval time between sending camera switching commands or custom commands, such as 0 means special treatment, only triggered once;

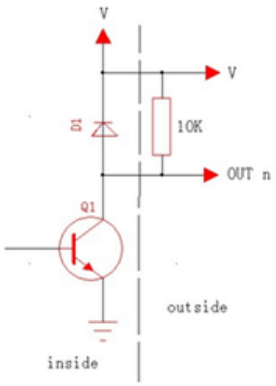


7. The microphone number generally corresponds to the input channel of the device, i.e. it is the channel number to which the microphone is connected. The microphone number with # is a virtual number, which can only be used to set the default microphone;
8. The smaller the priority number is, the higher the priority level is. When the priority level is the same, it will be processed in accordance with the triggering priority order; for example, if two microphones are speaking at the same time, the camera will automatically rotate to the preset position corresponding to the microphone with the small priority number (i.e., the high priority level) or send the command corresponding to the microphone with the small priority number (i.e., the high priority level); however, if the two microphones are with the same priority level, the signal that is checked first will prevail;
9. Enable this Mic setting: you can set all the microphone parameters in full in advance, but when you use it, only some of them will be enabled according to the actual situation;
10. Preset points, serial port numbers, camera addresses, protocols and camera-related, must correspond to the actual connection of the camera;
11. Custom Command means that when the microphone of the matrix checks the input signal (usually when someone speaks), it will automatically send the corresponding command to the defined serial port, and secondly, you can also pre-set the command, but do not check "Enable Custom Command", the device will not send it automatically, but you can still click the "Send" button, and the command in the input box will be sent to the specified serial port at any time;
12. Click on "Save" to save the parameters to the device, so that the microphone for the channel is now associated with the corresponding camera address. Then use the "Enable Microphone Settings" option to determine whether the microphone settings are valid when tracking is enabled;

13. Camera Setting is a camera control interface, generally configure the camera position before tracking starts, and finally the parameters of this part will be saved on the camera;
14. Firstly, serial port setting, there are 2 serial ports (232, 485), which correspond to the back panel port that the PTZ is connected to;
15. Next is the camera address and protocol type, please refer to the actual address of the camera for the camera address, and the protocol is related to the camera model;
16. Lastly the preset position number is the user-defined identification for the camera, and then the adjustment of the up, down, left, right, focal length, aperture and other parameters will define the camera's position and settings;
17. Finally, click "Save" to save the parameters to the camera, "Clear" is to delete the information of the current preset, "Recall" is used to view the camera saved by the current preset. "Clear" is to delete the information of the current preset point, and "Recall" is used to view the camera position saved by the current preset.

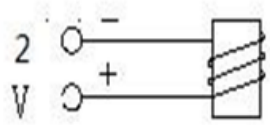
VI: GPIO Setting

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 protecting 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.

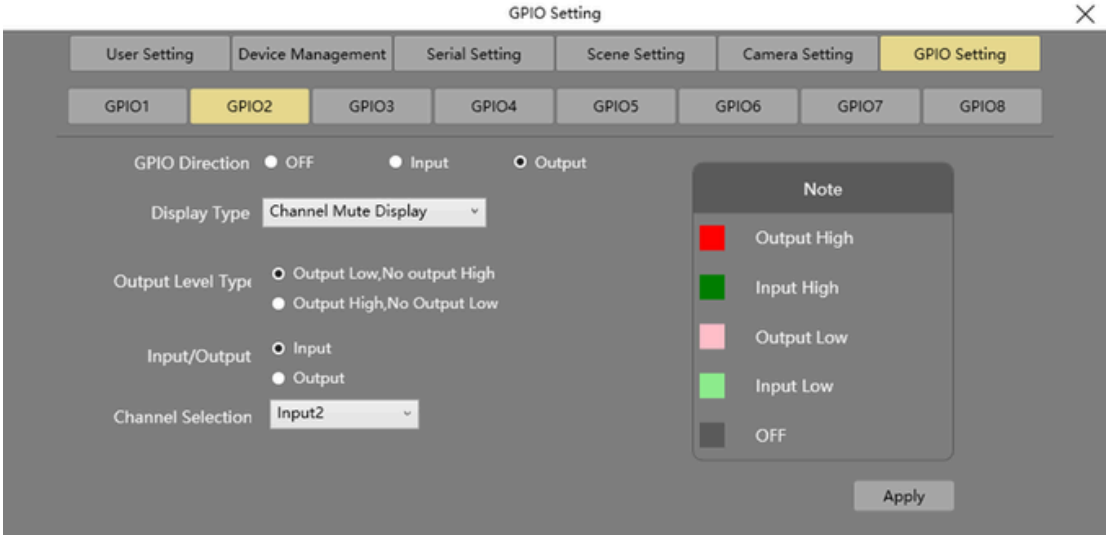
GPIO Input Example 1

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 Input 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'.

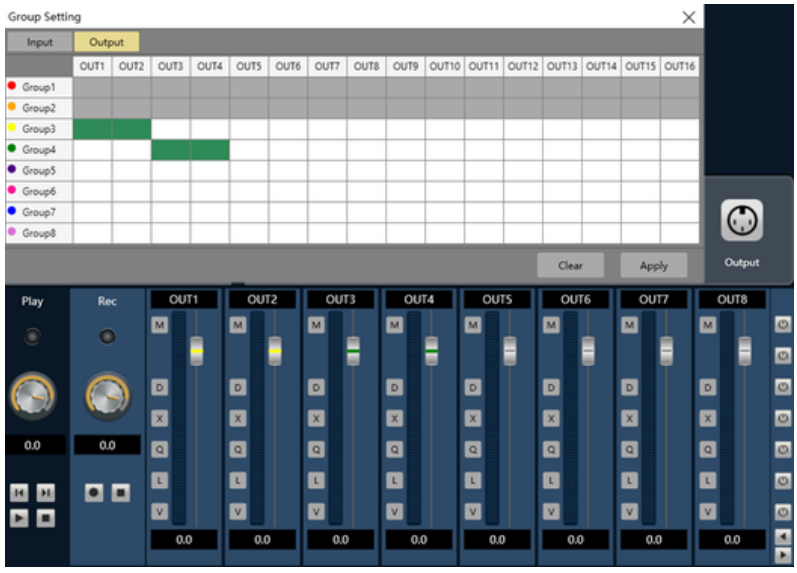
VII: Center Control Command

The Central Command Generator is able to convert frequently used operations into a 16-character command code for easy utilization by external devices.
Control command types: Scene, Input, Output, Mixer, Parametric Equalizer, Graphic Equalizer, Expander, Compressor, Automixer, Delay, Crossover, Limiter, AEC, ANS.

7.3.4 Setting - Group Setting

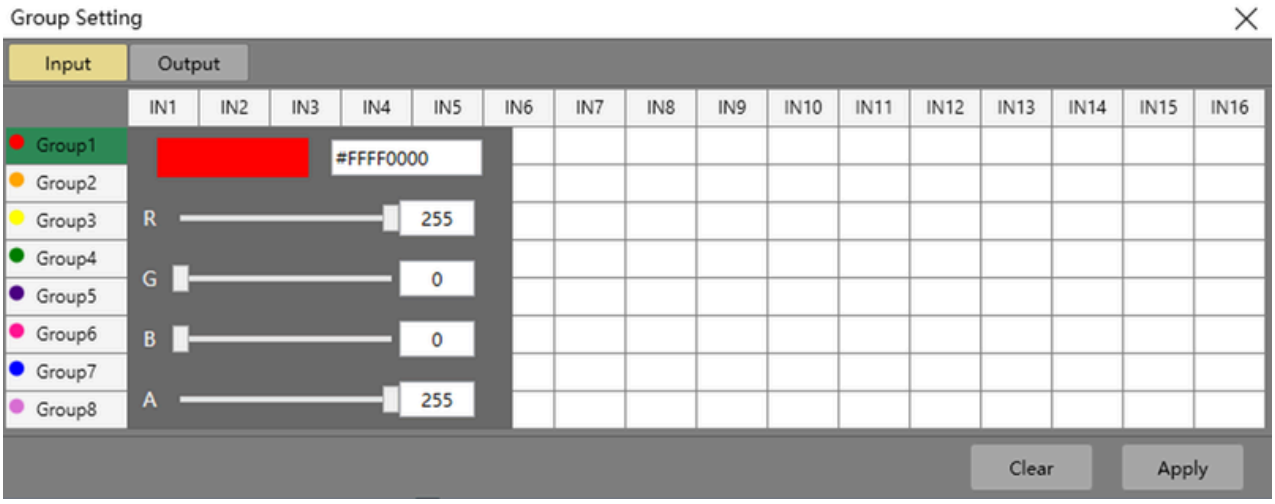
The Group Setting enables synchronized management of fader volume for similar audio channels.
The system contains dual control modules for input signal grouping and output signal grouping.

I: Input and Output Configuration



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

II: Customize Color Setting



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

7.3.5 View

① Open All: All function module will be display;
② Open Input: All input function module will be display;
③ Open Output: All output function module will be display.

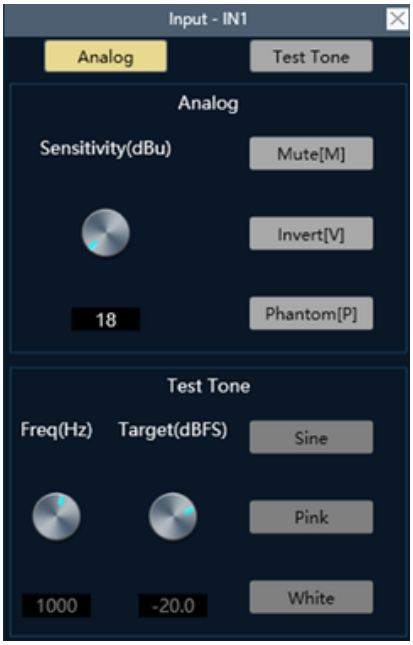
7.3.6 Help

① Help: View the embedded user documentation of the device;
② Upgrade: For updating the system with a new firmware;
③ About: Display software version information.

7.4 Input Configuration

7.4.1 Input Setting

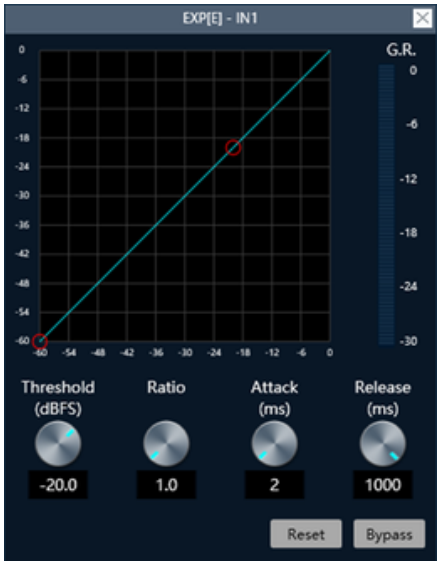
The Analog Input provides line-level input for devices with line-level outputs, and inputs for microphones. The Analog Input converts the analog input signal to a processed digital. Connections are made using one three-terminal 3.5mm Phoenix connectors.



① Model type: Analog input or test signal;
② Sensitivity: Analog signals can be adjusted by adjusting the sensitivity of the input can be selected, from 24 ~ -27dBu, step 3dBu a grade;
③ Mute: Mutes the input signal;
④ Invert: Inverts the polarity of the output signal;
⑤ Phantom: Toggle turning on and off phantom power (+48VDC) to the condenser microphone. Please do not turn on the Line Input or non-condenser microphone to prevent burning;
⑥ Test Tone: Including Sine, Pink Noise, White Noise, the system will automatically block the Analog input signal when the Test Signal is enabled.

7.4.2 Expander

The Expander is to increase the dynamic range of the input according to the user's needs: when the input signal is less than the "threshold", the Expander will compress the input signal according to the set "ratio", the output level = Threshold - (Threshold - Input Level)/Ratio; when the input signal is greater than the "threshold", it will be output at 1:1, the output level = Input Level. When the input signal is greater than the "Threshold", the output is 1:1 and the output level = input level.



① Threshold (-60~20): Sets the point from which the attenuation is calculated based on the Ratio setting. This is where the Expander starts working. Assuming only one Input, a level below the Threshold Level is attenuated, anything above the Threshold Level is not attenuated;
Example:
- If the: Threshold Level is -15 dB; Ratio is 2.5; Input level is -25 dB
- Then the Adjusted Output is:
- [(Input Level - Threshold Level) * Ratio] + Threshold Level = Output Level
- {[-25 dB - (-15 dB)] * 2.5} + (-15 dB) = -40 dB.

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

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

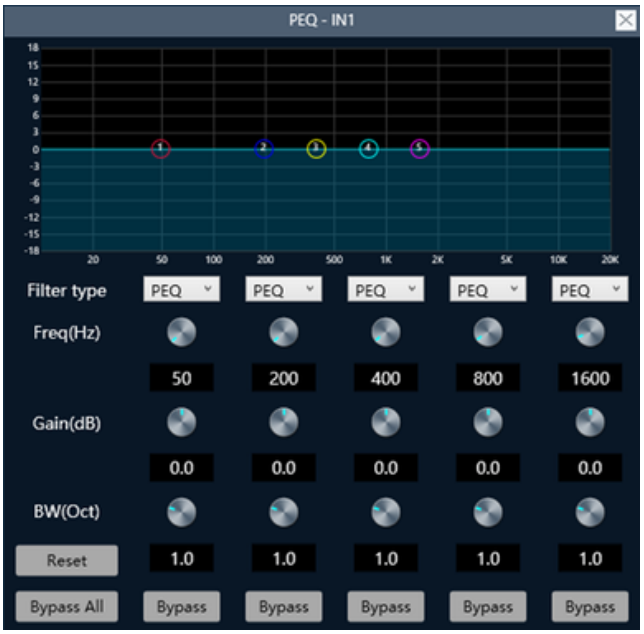
④ Release Time (1~1000): The time required for the input signal level to return from the extended state to the original non-extended state;

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

⑥ Reset: Resets the parameters to the default.

7.4.3 Input Equalizer

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



① Filter Type: Including low-pass, high-pass, low-shelf, high-shelf type
1) Low-pass: Cut-off frequency point of low-pass filtering;
2) High-pass: The cutoff frequency of high-pass filtering;
3) Low Shelf: Quantitative boosting or attenuation of all frequencies below the cutoff frequency;
4)High shelf: Quantitative boost or attenuation for all frequencies above the cutoff frequency.

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

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

④ 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;

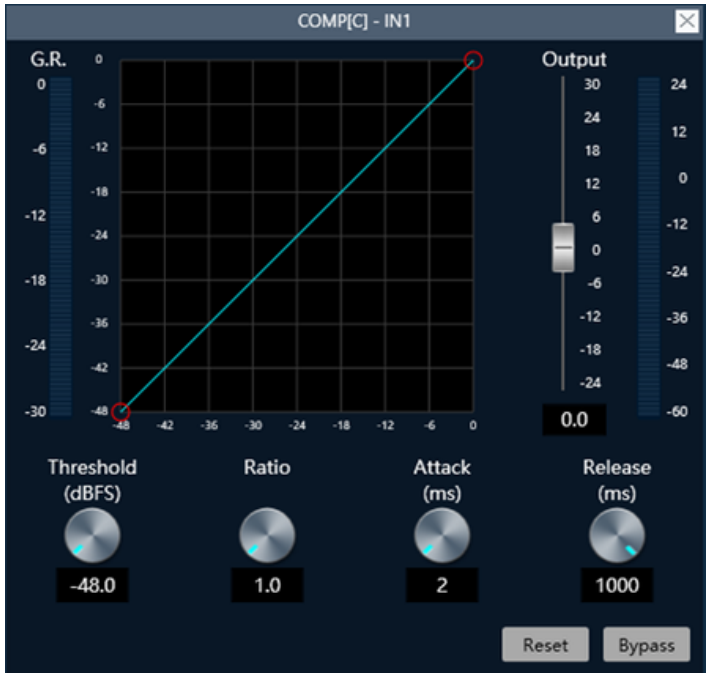
⑤ Frequency: Sets the center Frequency of an individual band;;

⑥ Gain: Controls the Gain for an individual frequency band;

⑦ Bandwidth (Octave): Sets the bandwidth of an individual band of the equalizer, from 0 octaves to 4.00 octaves (default is 1.00). This is not active when either Low-shelf or High-shelf Type is selected. Adjusting bandwidth generally adjusts Q-Factor in an inverse manner.

7.4.4 Compressor

The purpose of the Compressor is to control the dynamic range of the Output above a set Threshold Level.
The Compressor can be adjusted from unity (1:1) with the Input, to an almost flat (20:1 - very little amplitude variation) Output



① Threshold (-48~0): Sets the level where compression begins. This is the point from which the amount of attenuation is calculated based on the Ratio setting. Assuming only one Input, a level below the Threshold Level is not compressed, anything above the Threshold Level attenuation is applied;
Example:
- If the: Threshold Level is -15 dB; Ratio is 2.5; Input level is 10 dB
- Then the Adjusted Output is:
- [(Input Level - Threshold Level) / Ratio] + Threshold Level = Output Level
- {[10 dB - (-15 dB)] / 2.5} + (-15 dB) = -5 dB.

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

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

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

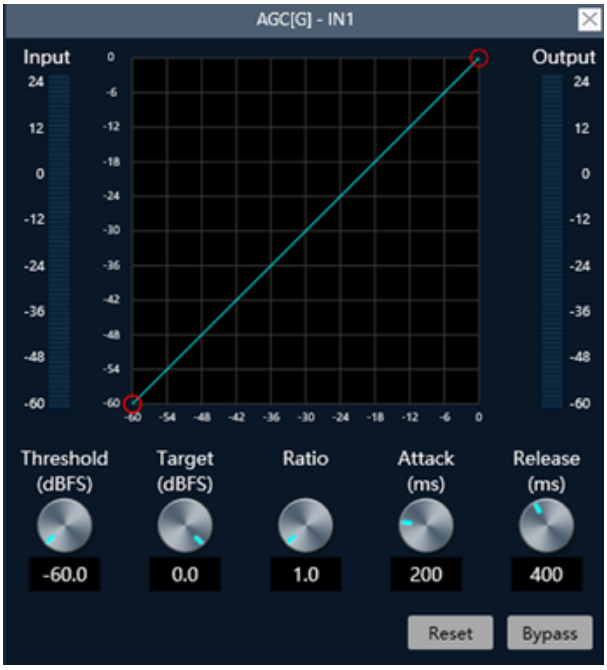
⑤ Output Gain (-24~30): Controls the Gain of the output;

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

⑦ Reset: Resets the parameters to the default.

7.4.5 Automatic Gain Control

The purpose of the Automatic Gain Control is to control the overall dynamic range of the Output when the Input level changes.
The AGC can be adjusted from unity (1:1) with the Input, to an almost flat (20:1 - very little amplitude variation) Output. The AGC automatically adjusts the Gain to the Target Level by compensating for low Inputs, compressing for high Inputs.
Typical use: For example, when the user speaks in front of the microphone, the distance between the mouth and the microphone will be far away and close to each other, which will cause the output volume to go up and down, or even feel that the speech is intermittent. AGC is to set the threshold value for the input signal below the threshold in accordance with the ratio of 1:1 output, for the level above the threshold is in accordance with the ratio of direct enhancement, set the target level, the sound signal can be stable output.



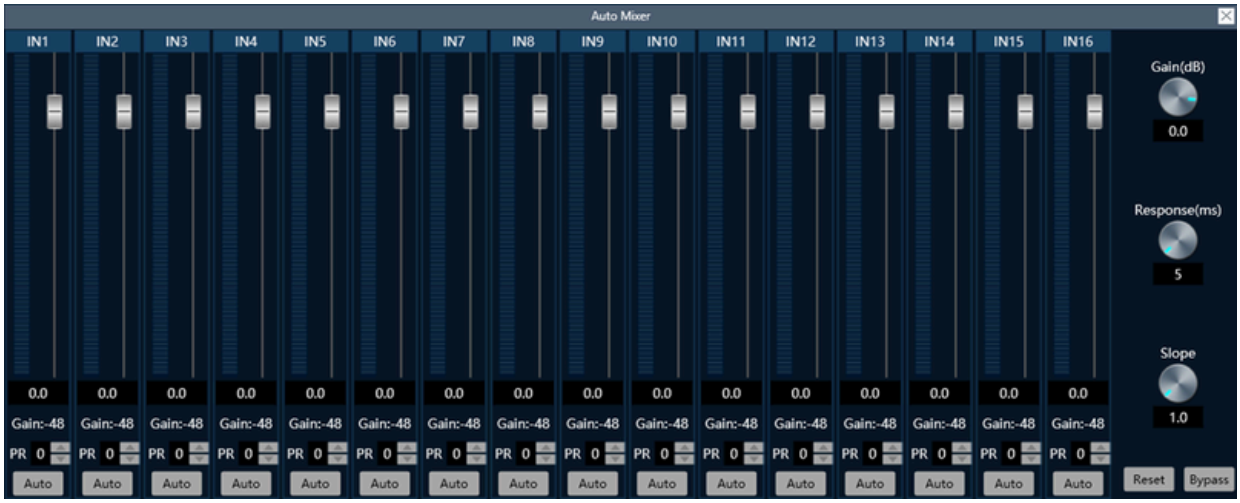
① Threshold (-60~0): The level at which the AGC Component becomes active. This should be set at a level so that the anticipated noise floor does not activate the AGC. When an Input exceeds the Threshold Level, the Gain for that input is adjusted to calculated level based on the Ratio and the Target Level;
② Target Level (-60~0): Sets the point from which the Gain is calculated based on the Ratio setting. Assuming only one Input, a level below the Target Level has a positive Gain applied, a level above the Target Level has a negative Gain (attenuation) applied;

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

③ Ratio (1~20): The ratio between the Input and the Output as measured from the Target Level. The closer the Ratio is to 20, the closer the Output will be to the Target Level, which also means smaller dynamic changes in the Output level. When the Input is below the Target Level and the Gain applied by the Ratio setting is greater than the Maximum Gain setting, the Output is clipped per the Maximum Gain setting;
④ Attack Time (1~1000): Controls the response time of signals above the threshold level;
⑤ Release Time (1~1000): Controls the response time of signals below the threshold level;
⑥ Bypass/Active: Bypass or Active the AGC for the current channel. When the AGC is bypassed, audio is passed through without any change;
⑦ Reset: Resets the parameters to the default.

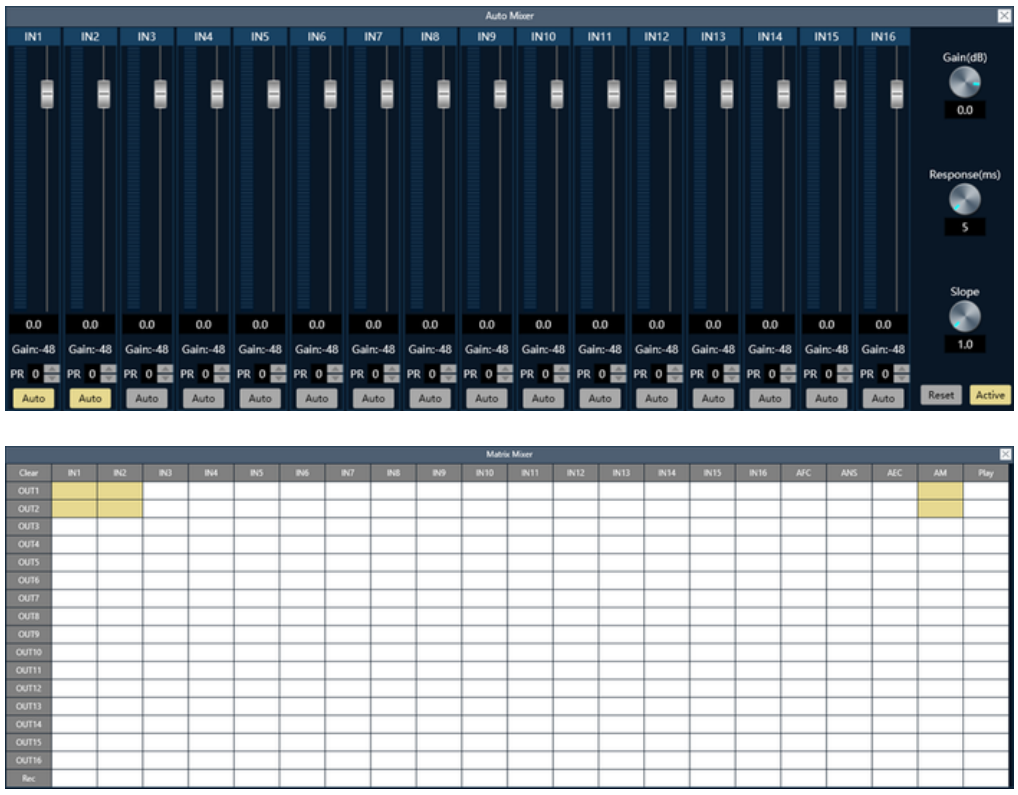
7.4.6 Automixer

Automixer is a mixer optimized for solving the problem of multiple live microphones operating together as a system – as found in boardrooms, classrooms, churches, courtrooms, etc. The Automixer controls the live microphones by turning up microphones when someone is talking and turning down microphones that are not used; thus, it is a voice-activated, real-time process without an operator. The Automixer controls the additive effect of multiple microphones being on at the same time and adapts to changing noise floor conditions, thereby maintaining the natural ambience of the room.
The Automixer is implemented as a gain-sharing type. The gain of each microphone input is calculated as the ratio of its RMS level to the combined RMS levels of all inputs. This insures unity system gain at all times.
Consider a typical conference room scene with ten participants, each with a microphone, if ten microphones are switched on at the same time and only one person is speaking as a result, then the output will definitely not be ideal as the other nine microphones pick up room acoustics, reverberation, etc., which will reduce the output of the entire system.
Each channel of the Automixer has an input, gain level meter and an auto gain, channel fader, priority, and channel mute. Channel Controls Each channel has an "Auto" button that is pressed to add the channel to the Automixer. Channel Mute and Fader are both Auto Gain types. To mute a signal and prevent it from going into the Automixer, turn Mute on and off. The channel fader controls the mix level and direct output level of the channel.



① Priority Gain (-72~12): Adjusts the gain or attenuation applied to a particular Input channel. For example, you can boost the gain of a mic being used by a soft spoken person and attenuate the gain of a louder person so they can be heard equally in the room;
② Priority PR: Selectable Priority that adds the Priority Gain to the automix function of the corresponding input – this gives the corresponding input more share of the available gain, but does not change the level of the actual input. The higher the value the more gain acquisition capability and the higher the priority, able to reduce the ability of channels with a low priority level to acquire transmission gain. This control defines the priority level with a value between 0 (lowest priority) and 10 (highest priority), and the default value is 5 (standard priority). If all channels have equal priority, set the priority of all channels to 5;
③ Auto: Each channel has an automix Enable/Disable button, which should be turned on for channels that need to participate in automix. It can also be turned off so that the channel does not participate in the automix;
④ Gain (-72~12): Adjusts the Mix output channel gain. This applies to all Input channels on the Mix output, but does not affect the individual output channels;
⑤ Slope: The slope control affects the attenuation of lower levels. At higher slopes, channels with lower levels are also attenuated more. Ratio similar to Expander, the larger the slope, the more the speaking microphone acquires the transmission gain, the more the non-speaking microphone attenuates the transmission gain; the smaller the slope, the less the speaking microphone acquires the transmission gain, and the less other non-speaking microphone attenuates;
⑥ Response Time: A master Response control adjusts the speed at which the mixer changes its automatic gain. The time for a microphone to acquire all the transmission gain when it is speaking or the time for other non-speaking microphones to attenuate the transmission gain. The longer the setting, the longer it takes for the speaking microphone to acquire the full transmission gain and the longer it takes for the other non-speaking microphones to attenuate the transmission gain. The shorter the setting, the opposite is true;
⑦ Bypass/Active: Bypass or Active the Automixer. When the Automixer is bypassed, audio is passed through without any change;
⑧ Reset: Resets the parameters to the default.

Example, Automixer and Matrix Mixer association operations

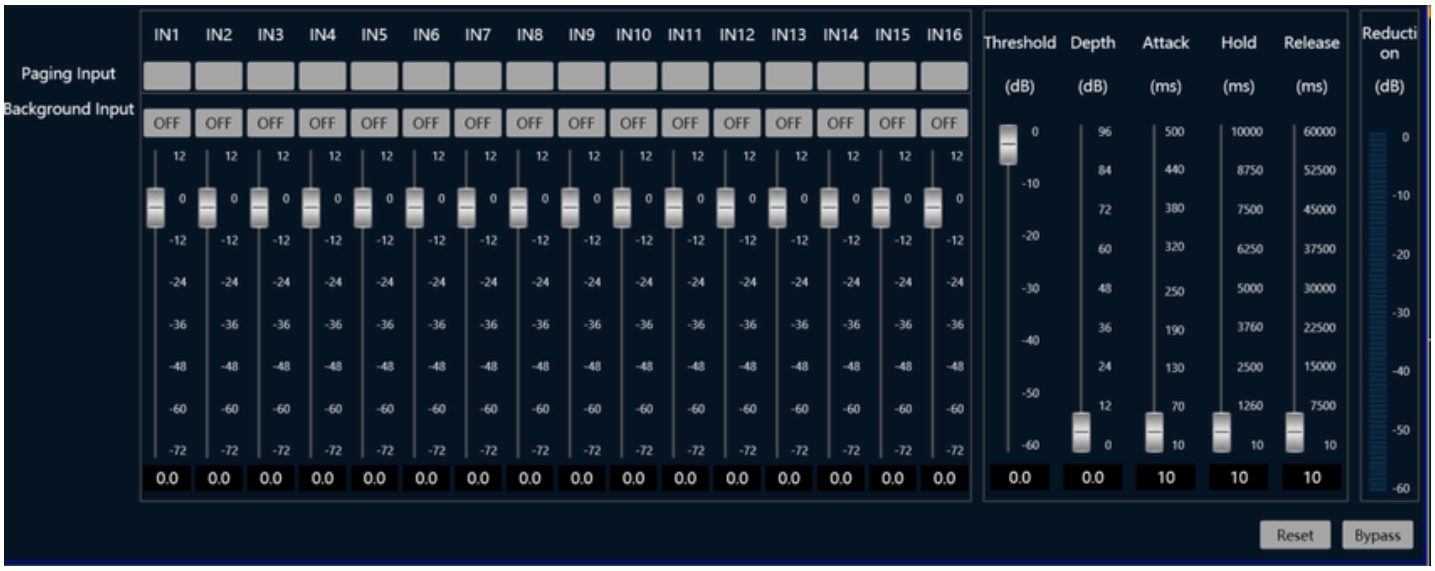


For example, the signals of input channels IN1 and IN2 are automatically mixed and output in output channels OUT1 and OUT2, as shown in the figure above:

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, and the corresponding channels are shown in teal color;
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.

7.4.7 Ducker

When the level value of one track exceeds the specified threshold, the level of the other track is attenuated, this is the dodging effect, the dodger is attenuating the level value above the threshold.



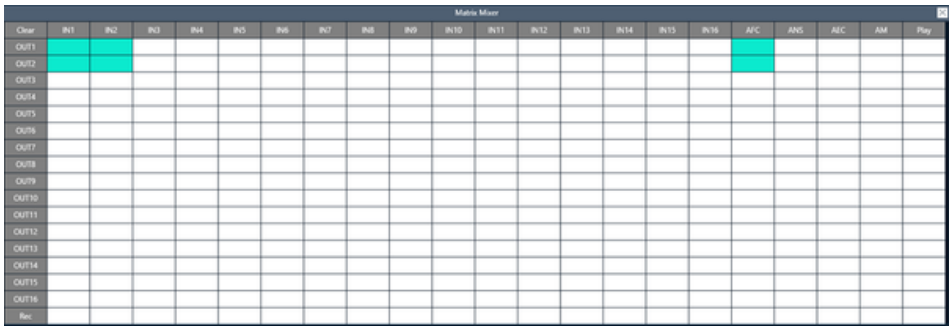
① Call input: Control signal input channel;
② Background input: controlled signal input channel;
③ Threshold: block out the controlled signal when the control signal level is higher than the threshold value;
④ Depth: controlled signal attenuation value;
⑤ Start Time: the controlled signal volume decay speed after the control signal level magnitude is higher than the threshold value;
⑥ Holding time: algorithm holding time when there is no signal input to the control signal input channel;
⑦ Release time: after the control signal level size is lower than the threshold value, the controlled signal volume recovery speed;
⑧ Attenuation: attenuation level value.
⑨ Local Output: Select the channel for local output of the background input; the call input needs to be mixed to the local output channel in the matrix mix;
⑩ Enable/Pass-through: enable or disable the dodger;
11 Reset: restores the parameter to the default value.

7.4.8 AFC

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

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

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:

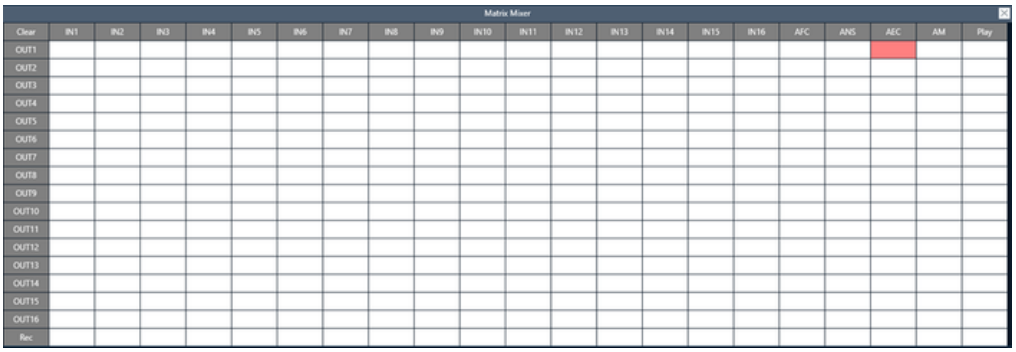
① First, open the [Matrix Mixer] configuration;
② Complete the audio output channel configuration in the [Matrix Mixer] to ensure that the processed feedback signal can be correctly routed to the output channel, i.e., select the points corresponding to the input list "AFC" and the output channels OUT1 and OUT2, and the corresponding channels will be shown in cyan;
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.

7.4.9 AEC

The Acoustic Echo Canceler (AEC) is used in conference rooms (Near-End) and other installations where people call in from remote locations. The remote caller's (Far-End caller's) voice is broadcast over loudspeakers in the conference room. The sound is picked up by microphones in the conference room and echoed back to the Far-End caller. The purpose of the AEC is to eliminate these echoes while at the same time allowing the Far-End caller to hear clearly, what people in the room are saying.
Each microphone in the conference room is plugged into one channel of the AEC. Each channel also receives the loudspeaker signal that carries the remote talker's voice. This is called the AEC reference signal. To remove the echoes, the AEC subtracts a filtered version of the reference signal from the microphone signal.

① Local Input: Local Mic output channel, i.e. the signal that needs to be processed for AEC;
② Remote Input: 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, the corresponding channel will be displayed in red;
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.

7.4.10 ANS

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

① ANS Input: Local Mic output channel, i.e. the signal that needs noise suppression processing;
② ANS level: range of noise reduction level (6~30dB).

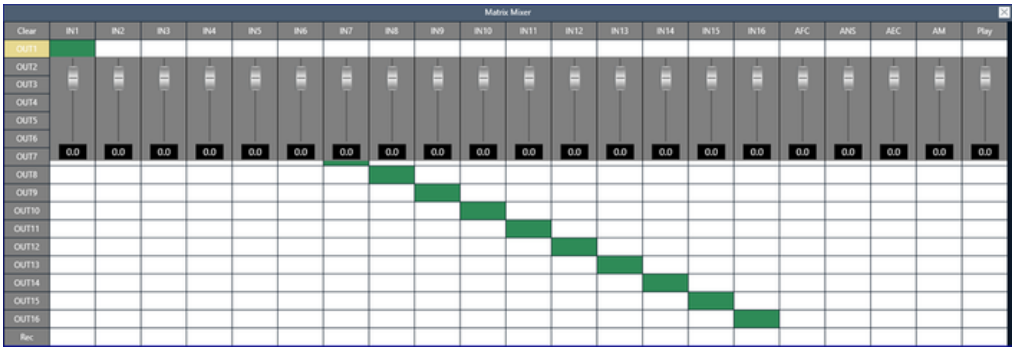
Example, ANS with Matrix Mixer association operations

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, and the corresponding channels are displayed in lime green;
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.

7.5 Matrix Mixer

Matrix both signal routing and mixing double multiplexing function, the control logic is horizontal for the input channel, vertical for the output channel, the matrix initialization state is (as shown in the figure above: one-to-one) input and output.



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

7.6 Output Configuration

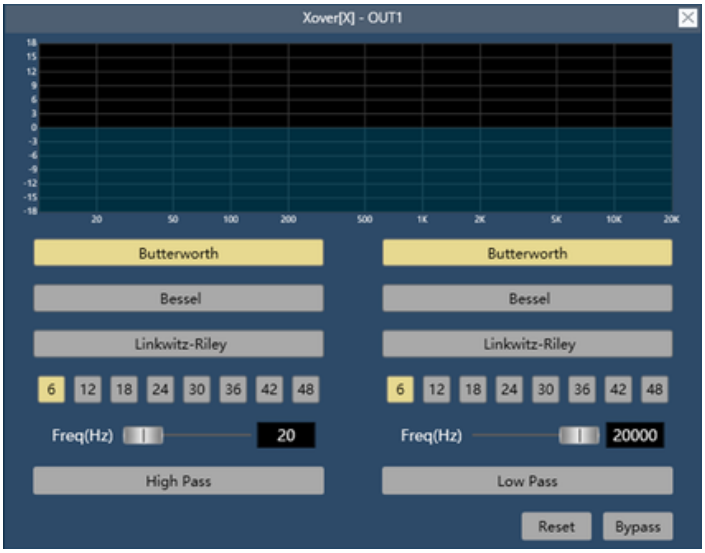
7.6.1 Delay

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

① Delay time: Delay time range (0~2000ms);
② Delay distance: Delay distance range (0~680m);
③ Bypass/Active: Bypass or Active the Delay for the current channel. When the Delay is bypassed, audio is passed through without any change;
④ Reset: Rests the parameters to the default.

7.6.2 XOVER

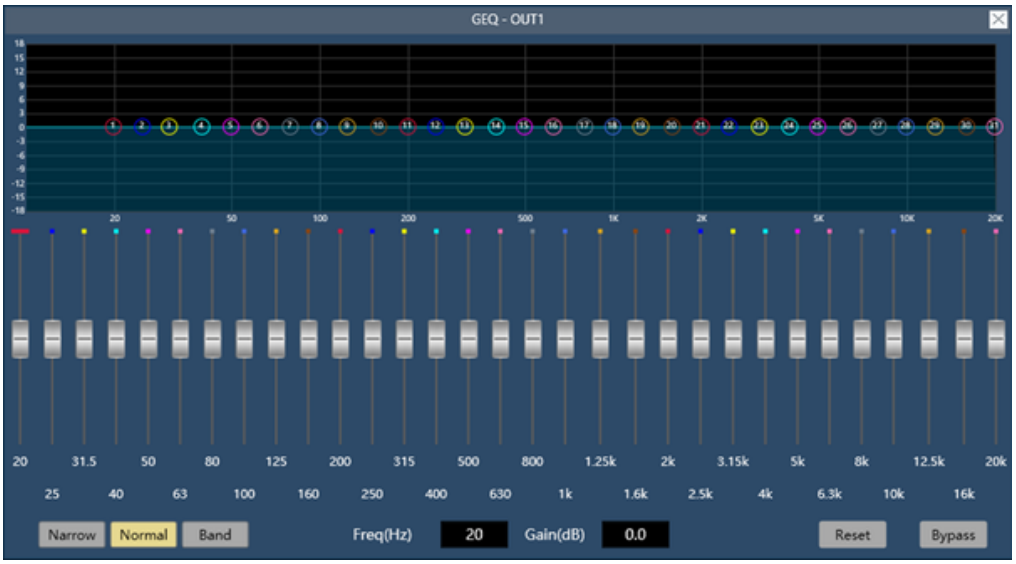
The XOVER divides the audio input signals into 3 frequency bands: low-pass, band-pass and high-pass, you can set the Slope rates and filter types (Butterworth, Linkwitz-Riley, Bessel) for each filter in each band.



① Filter Type: The Type setting includes: Butterworth, Linkwitz-Riley, Bessel filters. You can select any combination of two of these filters for a band's high-pass and low-pass frequencies;
② Slope: Determines the rate of change of attenuation at the high-pass and low-pass frequencies of the band. The slope setting establishes the crossover region between two adjacent bands. The Slope includes 6, 12, 18, 24, 32, 36, 42, 48;
③ High Pass frequency: High Pass filter cutoff frequency;
④ Low Pass frequency: Low Pass filter cutoff frequency;
⑤ High Pass Bypass/Active: Bypass/Active the High Pass filter;
⑥ Low Pass Bypass/Active: Bypass/Active the Low Pass filter;
⑦ Bypass/Active: Bypass or Active the XOVER for the current channel. When the XOVER is bypassed, audio is passed through without any change;
⑧ Reset: Rests the parameters to the default.

7.6.3 Graphic Equalizer

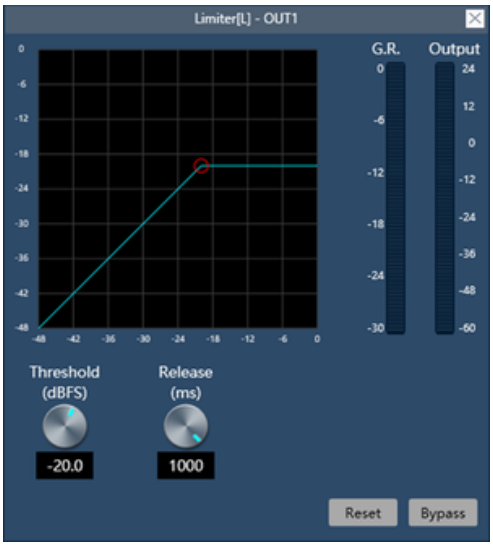
The Equalizer is mainly used to modify over-emphasised or missing frequency ranges. Whether the frequency range is narrowed or widened, the Equalizer can help to repair the narrowed frequencies or widen the frequency range to achieve the ideal signal tone.
The Graphic Equalizer provides from 31 bands with 1/3th octave per band. Each band can be adjusted from -18dB to +18 dB.



① Narrow: Narrowband equalization filter; Normal: Normal equalization filter; Wide: Wideband equalization filter;
② Freq.: Frequency indication of the current equalization filter;
③ Gain: Gain indication or control of the current equalization filter;
④ Rest: Rests all the band gains to the default;
⑤ Bypass All/Active All: Bypass or Active the Graphic Equalizer for the current channel. When the Graphic Equalizer is bypassed, audio is passed through without any change.

7.6.4 Limiter

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



① Bypass/Active: Bypass or Active the Limiter for the current channel. When the Limiter is bypassed, audio is passed through without any change;
② Threshold (-48~0): Sets the level at which the Limiter has an effect, and the level at which the output is held;
③ Release Time (1~1000): When the input signal falls below the Threshold Level, the sound channel is not turned off immediately, but is delayed for the Release Time. During this time, the sound channel stays on as long as there is a signal above the Threshold Level.
④ G.R.: Graphically displays the amount of attenuation applied to the Channel;
⑤ Reset: Resets the parameters to the default.

7.6.5 Output Setting

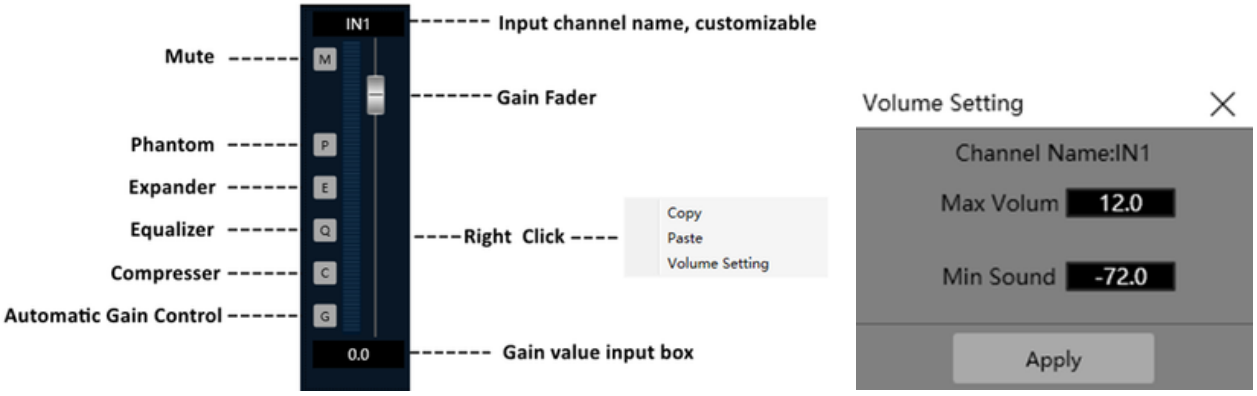
The Analog Output provides one channel of line-level output for device. The Analog Output converts the processed digital signal to analog. Connections are made using one three-terminal 3.5mm Phoenix connectors.

① Mute: Mutes the output signal;
② Invert: Inverts the polarity of the output signal.

7.7 Other Functions

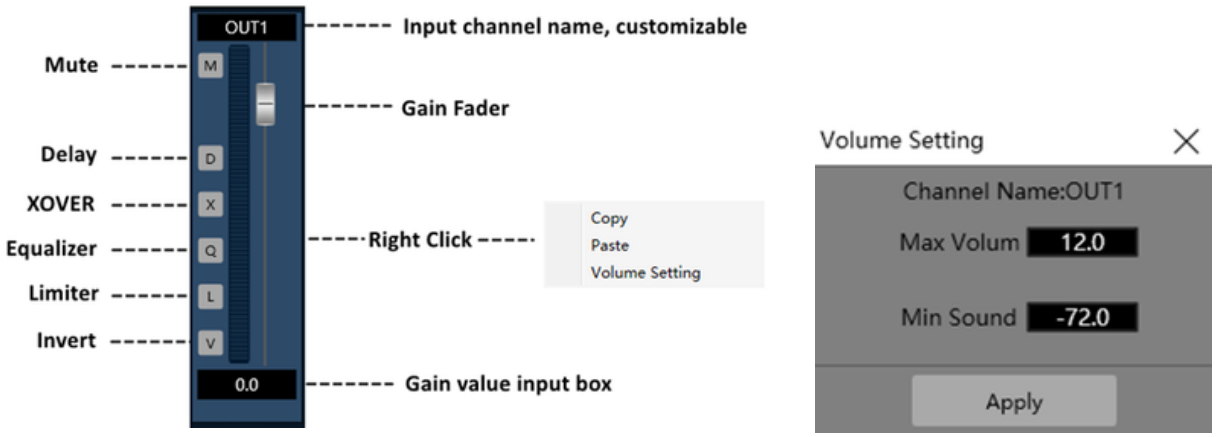
7.7.1 Channel Control

I. Input Channel Control and Shortcut



① 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;
② The channel right-click directly invokes the volume fader constraint range setting function. Note: The constraint authority of the volume fader at the center control code level has the highest priority, and its setting will override the parameter range set by the channel right-click.

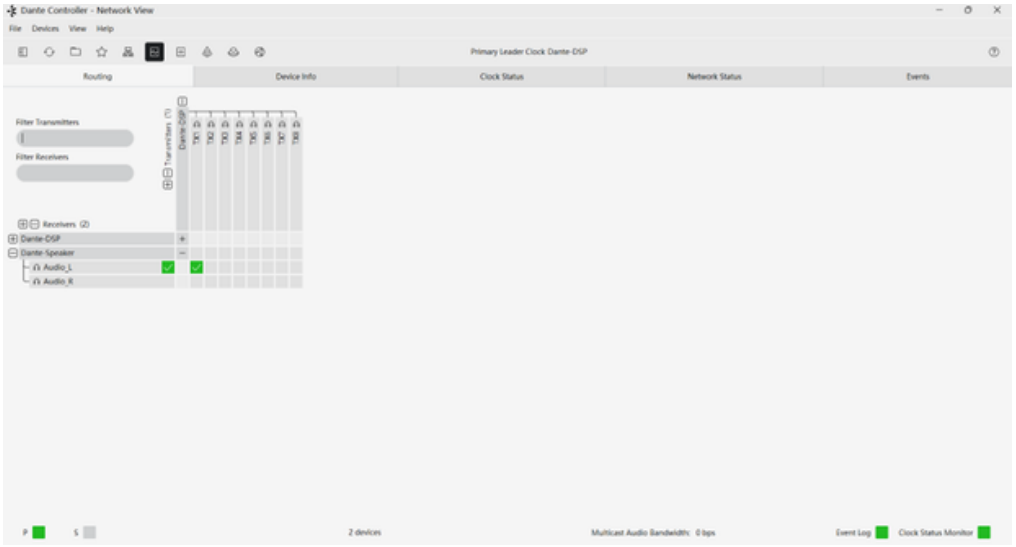
II. Output Channel Control and Shortcut



① 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;
② The channel right-click directly invokes the volume fader constraint range setting function. Note: The constraint authority of the volume fader at the center control code level has the highest priority, and its setting will override the parameter range set by the channel right-click.

8. Dante Network Audio Routing

In a Dante audio network, the Dante Controller software is required to set up the routing of the various signals accessing the processor. It can realize 1-to-1, 1-to-N mapping operation from input to output within Dante network.



The software "Dante Controller" is free to download from the company of Audinate (the owner of Dante technology).To install the software on the computer, please visit the link:
https://www.getdante.com/products/software-essentials/dante-controller.

And the "User Guide" of "Dante Controller" is available on the Audinate website:
https://dev.audinate.com/GA/dante-controller/userguide/webhelp/content/front_page.htm.

Note:
1. Dante can not run in the Wi-Fi connection environment, is dependent on a reliable and secure wired network environment to transmit perfect audio;
2. Dante Controller software corresponds to the platform of Windows 7, Windows 10, Windows 11, macOS, please select the appropriate software version according to your system platform.

9. FAQ

9.1 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.

9.2 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.

9.3 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. Fourth, check that the audio signals are routed correctly in the Dante Controller. If the problem is not solved, please contact the distributor or manufacturer.

9.4 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.

9.5 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.

9.6 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.

9.7 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.

9.8 RS232 center control command does not work

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

9.9 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.

2321 Cohen | St-Laurent , H4R 2N7 | Québec, Canada Tel: 514-905-0588 | Fax: 514-905-0589 | Toll free: 1-877-689-5228 info@muxlab.com | salesteam@muxlab.com | www.muxlab.com