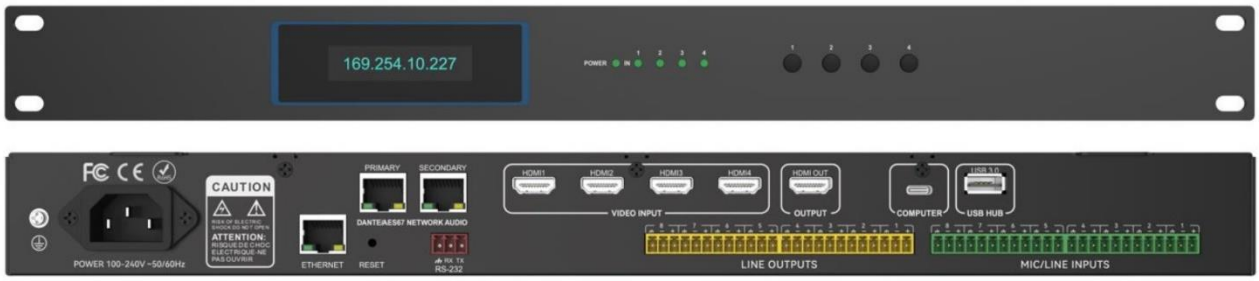
APROC-8841DV

Easy Conference Incorporate Solution



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1. Technology Overview

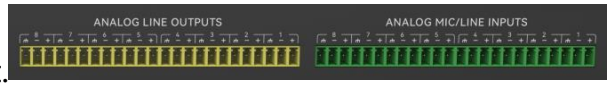
1.1 Introduction To Technology

The smart and simple meeting system DSP family incorporates several core technical features that enable audio engineers to enhance their job performance. Remote audio hardware, based on DSP, is routed and processed by a computer. This manual primarily introduces the technology required to achieve this objective.

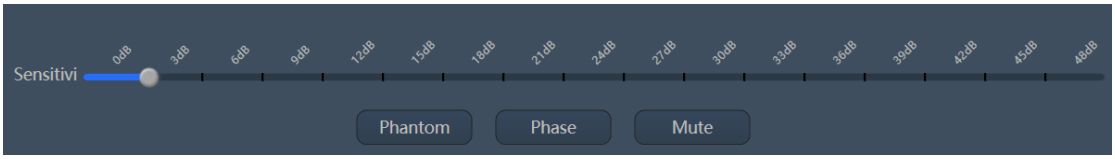
DSP Controller is a Windows-based application that configures and controls DSP hardware. The DSP Controller has 16 built-in presets, each of which can flexibly design processing modules and sequences according to the requirements of designers. After the design is completed, it only needs to be saved and can be used forever.DSP Controller. The sequence and parameters of the built-in processing modules of the DSP Controller conform to most application scenarios and can be easily used without any changes.

DSP Controller is a full-featured application that includes editable processing modules and Settings for parameter adjustment and peripheral accessories such as RS232, RS485, drag panel configuration, etc. Most interesting is the user interface, which allows engineers to create custom interfaces that can be edited by integrators and operated by technicians in the field or non-technical end users. Advanced security features give end users access to only the controls allowed by the engineer or system designer.

1.2 Audio Input Section

The DSP can have up to 16 fixed analog audio inputs, which are connected via a detachable balanced Phoenix connector. Analog input part can Line and Mic level 0/3/6/9/12/15/18/21/24/27/30/33/36/39/42/45/48 is a line level signal. +48VDC phantom power is available per input.

The front-end amplification gain and phantom power supply are easily controlled via the DSP Controller.



A/D Technical indicators

Sampling rate: 48kHz

THD+N:≤0.001%@4dBu（A weigh）

Dynamic range：113dB

Audio format:：24Bit MSB TDM

1.3 Audio Output Section

The first stage of the analog output part is the D/A converter (DAC). The DSP uses an advanced 24-bit 256X sampling converter. As with A/D converters, the multibit architecture allows for a wider dynamic range, but with the same excellent distortion characteristics as conventional unit-to-analog converters.

D/A Technical indicators

Sampling rate: 48kHz

THD+N:≤0.001%@4dBu（A weigh）

Dynamic range：113dB

Audio format:：24Bit MSB TDM

1.4 Float Point DSP

DSP devices utilize SHARC DSPs, which feature 32-bit and 40-bit floating-point processing. This processing capability is comparable to 40-bit floating-point processing found in other devices. Floating-point processing offers significant advantages in terms of sound quality and ease of use.

Limitations of Fixed-Point Processing

The primary issue with fixed-point processing is the risk of clipping distortion, which can occur with significant gain changes or data loss. For example, when a 24-bit fixed-point processor processes a 24-bit audio signal, if certain conditions lead to an audio signal attenuation of 42 dB, the overall dynamic range of the new audio signal will be greatly reduced. The accuracy of the 24-bit fixed-point processor may not adequately capture this reduced dynamic range. In such cases, the extra bits may be rendered unusable for the next large dynamic range event. As a result, the audio signal after attenuation may be worse than before, which contradicts our expectations. With gain attenuation, 7 bits can be lost permanently.

Moreover, clipping distortion becomes a significant concern when a large gain is applied to a signal that is already close to 0 dBFS. The signal will be clipped at 0 dBFS, causing audio distortion. Even if the signal level is subsequently reduced below 0 dBFS through post-adjustment, clipping has already occurred, leaving persistent distortion. While fixed-point processing may attempt to create a dynamic margin above 0 dBFS, it inevitably sacrifices some bits to achieve this. For instance, establishing a 12 dB (2-bit) dynamic margin means the entire system must accommodate this margin, making effective 24-bit fixed-point processing more difficult and costly.

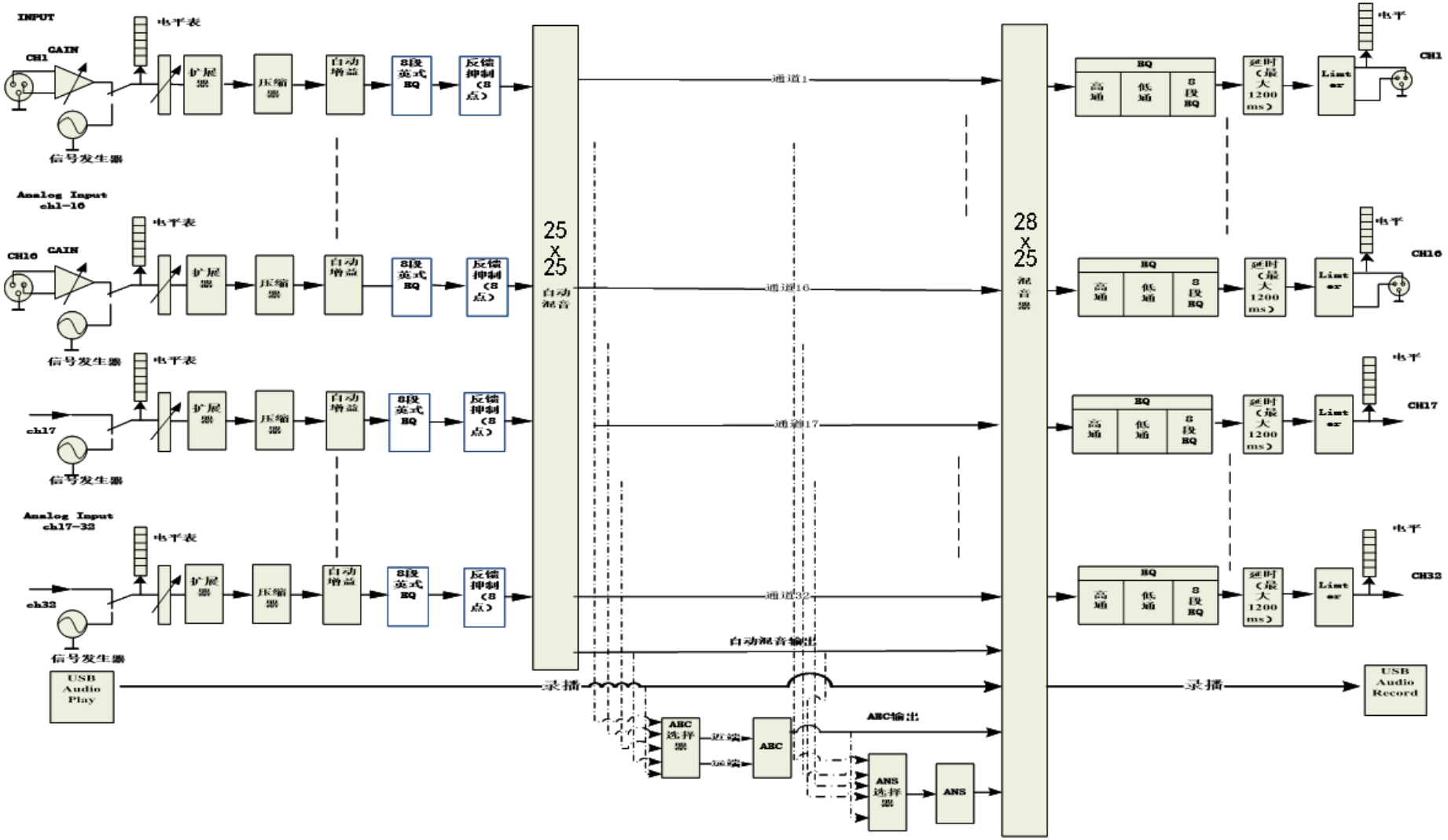
Advantages of Floating-Point Processing

In contrast, floating-point processing uniformly allocates all available bits to the signal, regardless of its level. Floating point uses some bits as exponents to represent the signal level, while the remaining bits store signal values independently of that level. Consequently, whether the signal is -200 dB or exceeds 0 dBFS, the saved signal maintains optimized accuracy with no clipping distortion. SHARC DSPs provide both 32-bit and 40-bit precision processing, allocating 25 bits to the stored signal, irrespective of its level. This ensures that the accuracy of low-level signals is significantly better than that of 24-bit fixed-point processing. With 40-bit precision processing, signals can be stored with up to 33 bits of accuracy.

Practical Significance

What does floating-point processing mean for users in practical terms? It eliminates concerns over gain levels between multiple modules. If one module reduces the signal level by 50 dB and another subsequently restores it, there will be no data loss. In a fixed-point system, users must check the signal level before sending it to the A/D converter, as all analog-to-digital converters are fixed-point. In DSP systems, if clipping is detected before sending the signal to the analog-to-digital converter, it can be immediately corrected at the output. Conversely, in a fixed-point system, users must investigate each processing module to identify the source of clipping.

1.5 Audio Signal processing flow



2 Hardware

2.1 Safety

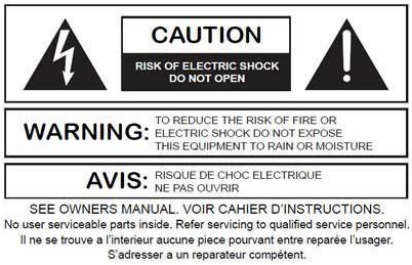
(Safety)

Critical safety note

1. Read these instructions.
2. Save the instructions properly.
3. Pay attention to all warning messages.
4. Follow all instructions.
5. Do not use the equipment near water. The equipment should not be exposed to water droplets or splashes, and ensure that there are no objects containing liquids near the equipment, such as vases.
6. Use only dry cloth to clean the equipment.
7. Do not block the ventilation opening. Install only according to the manufacturer's instructions.
8. Do not install any heat sources, such as radiators, heat registers, furnaces, or other devices that generate heat (including amplifiers).
9. Use a protective grounding connection to connect this device to a power outlet. Do not use polarized or grounded plugs. A polarized plug has two blades, one of which is wider than the other. A grounded plug has two blades and a third grounding terminal. Wide blades or a third grounding terminal provide safety for users. If the provided plug does not match the power socket, please contact an electrician to replace the old socket.
10. Protect the power cord from being stepped on or squeezed, especially at the connections between plugs, sockets, and wires and equipment.
11. Use only accessories/fittings specified by the manufacturer.
12. Only use hand carts, tripods, stands, or tables designated by the manufacturer or sold together with the equipment and instruments. When using a handcart, be careful when moving the handcart/device combination to avoid injury due to tipping over.

During thunderstorms or when not in use for a long time, please unplug the device.

14. Find qualified warranty personnel to handle all repair issues. Repair is necessary when the device is damaged in any way, such as when the power cord or plug cord is damaged, liquid spills or objects fall into the device, the device is exposed to rain or moisture, incorrect operation, or the device falls off.



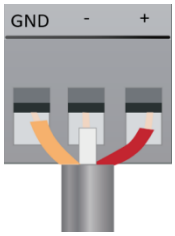
The lightning symbol with an arrow symbol inside an equilateral triangle is designed to make users aware of the uninsulated "dangerous voltage" inside the product casing, which can cause electric shock to the human body. The exclamation mark inside the equilateral triangle is intended to make users aware of the importance of the operation and maintenance (repair) instructions in the accompanying literature of the product.

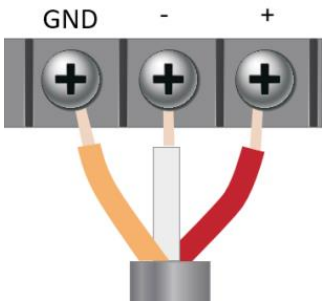
WARNING To prevent electric shock, do not use polarity plugs, sockets, or other outlets provided on devices with extension cords, unless the pointed end cannot be fully inserted.2.2 Audio Wiring Reference

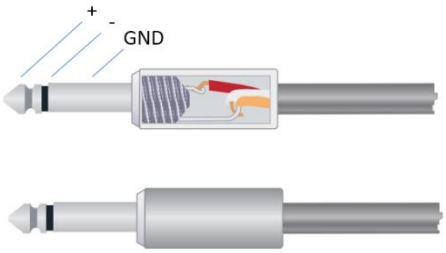
Balanced connection

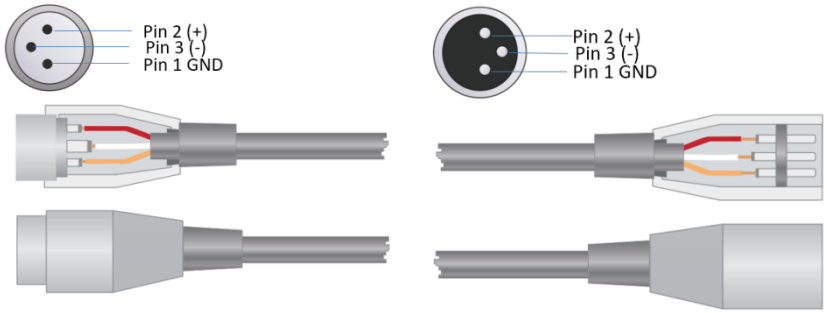
Any of these interfaces may appear on both sides of the balanced connection.

Note: For an XLR interface, the female connector is connected to the output, while the male connector is connected to the input.



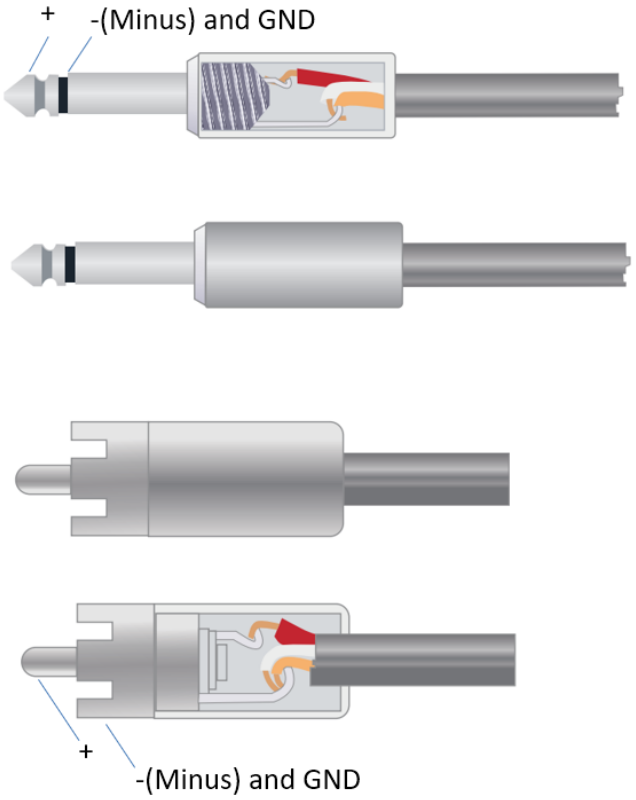






Unbalanced connection

The RCA interface and 1/4 inch TS interface are unbalanced interfaces, equipped with a shielded multi stranded wire and can be placed at both ends of the unbalanced connection.



2.3 Specifications

PROCESSOR	ADI SHARC 21489(x2)
SAMPLING RATE/QUANTIZATION BIT	48K/24bit
INPUT GAIN	0/3/6/9/12/15/18/21/24/27/30/33/36/39/42/45/48 dBu
PHANTOM POWER	+48V/10mA max
FREQUENCY RSPONSE（20~20KHz)	±0.3dB
MAX LEVEL	+18dBu
THD+N:	≤0.001%@4dBu（A 计权）
INPUT DYNAMIC RANGE	113dB
OUTPUT DYNAMIC RANGE	113dB
CHANNEL ISOLATION @1kHz	108dB
INPUT IMPEDANCE(BALANCED CONNECTION)	5.4KΩ
OUTPUT IMPEDANCE(BALANCED CONNECTION)	600Ω
LATENCY	<3ms
WORKING POWER SUPPLY	AC110~240V 5Hz-60Hz
DIMENSIONS（WxDxH)	482 x 260 x 45mm
WEIGHT	4KG

2.4 Mechanical Data

Required space:

1U (width, depth, and height: 18.91 "x 9.5" x 1.72 "/48.02 cm x 24.13 cm x 4.37 cm). Depth does not include reserved space for joints.

At least 3 inches of extra space should be reserved for the connection on the rear panel. The reserved depth depends on the wire and connection method used.

Electrical properties:

110-240 VAC， 50/60 Hz， 40 W maximum universal input power supply.

Improve air circulation

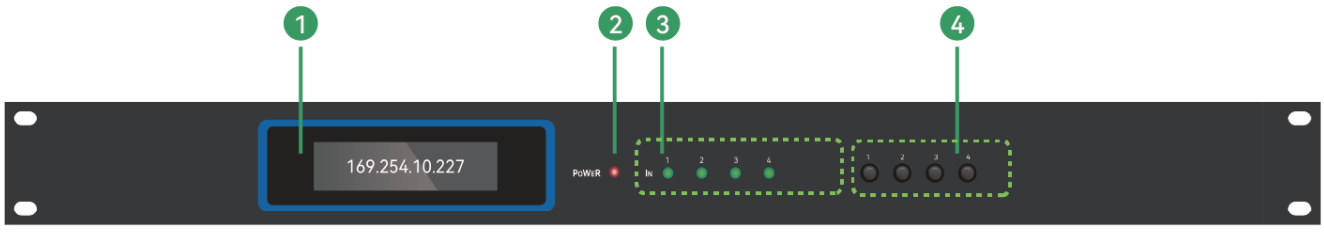
The recommended maximum operating environment temperature is 30 ℃/86 ℉.

Ensure that there is no obstruction on both sides of the device (at least 5.08 cm, 2 inches of clearance should be reserved). Do not cover the device's heat dissipation port with items such as newspapers, tablecloths, and curtains.

Shipping weight:

8.8 lbs. (4 kg)

2.5 Front Panel



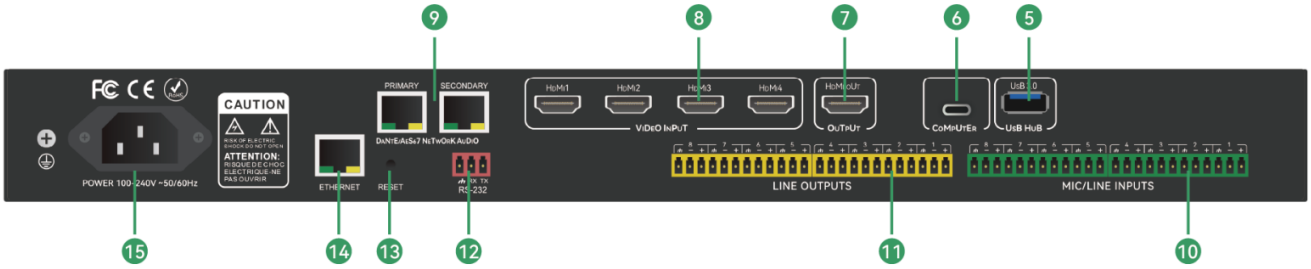
1、OLED display screen: displays device IP and firmware information;

2、POWER : Device power status indicator light;

3、IN indicator light: video source switching indicator light;

4、1,2,3,4 buttons on the right side: video source switch, four video sources can be switched to the output through the button;

2.6 Rear Panel



5、USB interface: docking station

6、Type-C interface: connects to OPS host or computer, can achieve audio and video signal transmission, supports soft video conferencing.

7、HDMI interface: Connect output devices such as display screens (camera image monitoring) to the OUTPUTS area.

8、HDMI interface: Connect the INPUTS area to the camera signal source.

9、AES67/Dante network interface: used to connect to AES67/Dante audio networks.

10. INPUT analog signal input interface: can connect to devices such as microphones and DVDs.

11. OUTPUT analog signal output interface: can be connected to devices such as power amplifiers and active speakers.

12. RS232 interface: Connect to control terminals or central control devices.

13. Factory reset switch: Press and hold for 5 seconds while powered on to restore factory settings.

14. Ethernet network control interface: By connecting to this network port, client computers can debug and monitor devices.

15. POWER power interface: Connect to 110V-240V AC power supply, and start automatically when powered on.

3 Software

3.1 Software Installation

A Windows PC with a 1 GHz or higher processor and:

Windows 7 or higher version.

1 GB of free storage space.

1024 x 768 resolution.

24 or higher colors.

2 GB or higher memory.

Network (Ethernet) interface.

CAT5 cable or existing Ethernet network

The audio processor is equipped with built-in control software, which does not a CD. Simply access the IP address of the audio processor to quickly download it. Enter the device IP address in the browser address bar to access the audio processor, find the download link, and download the installation software locally to complete the installation. The default IP address of the device at the factory is 169.254.10.227 subnet mask 255.255.0.0. Please add the address of this network segment in the PC first for normal device access. After the device is started, enter the address in the browser address bar “ http://169.254.10.227/ ” .

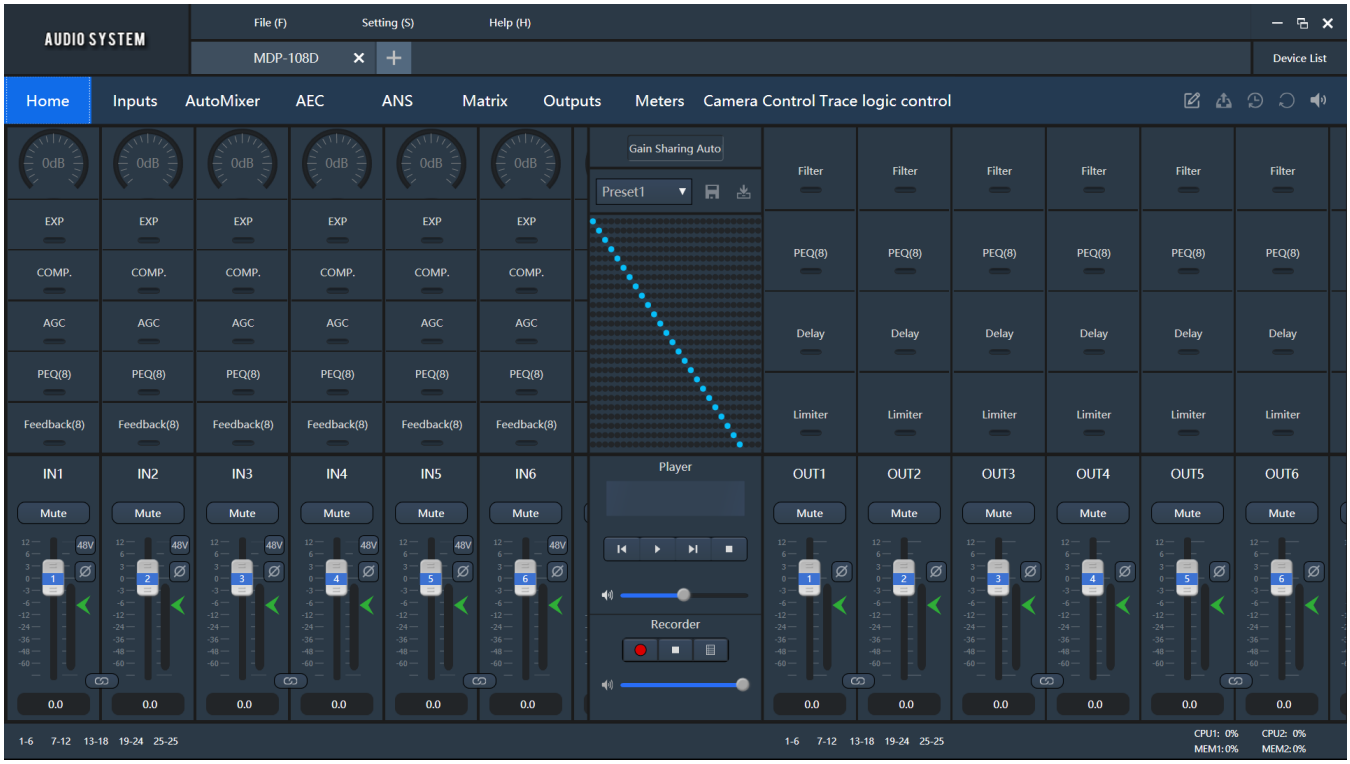


Before installing the PC software, please ensure that Microsoft has been installed on the PC Net Framework is a newer version.

Attention: If downloading and installing the software fails, please try using IE or Google Chrome, or enter browser settings, cancel the browser download pop-up prompt and try again.

3.2 Using the Software

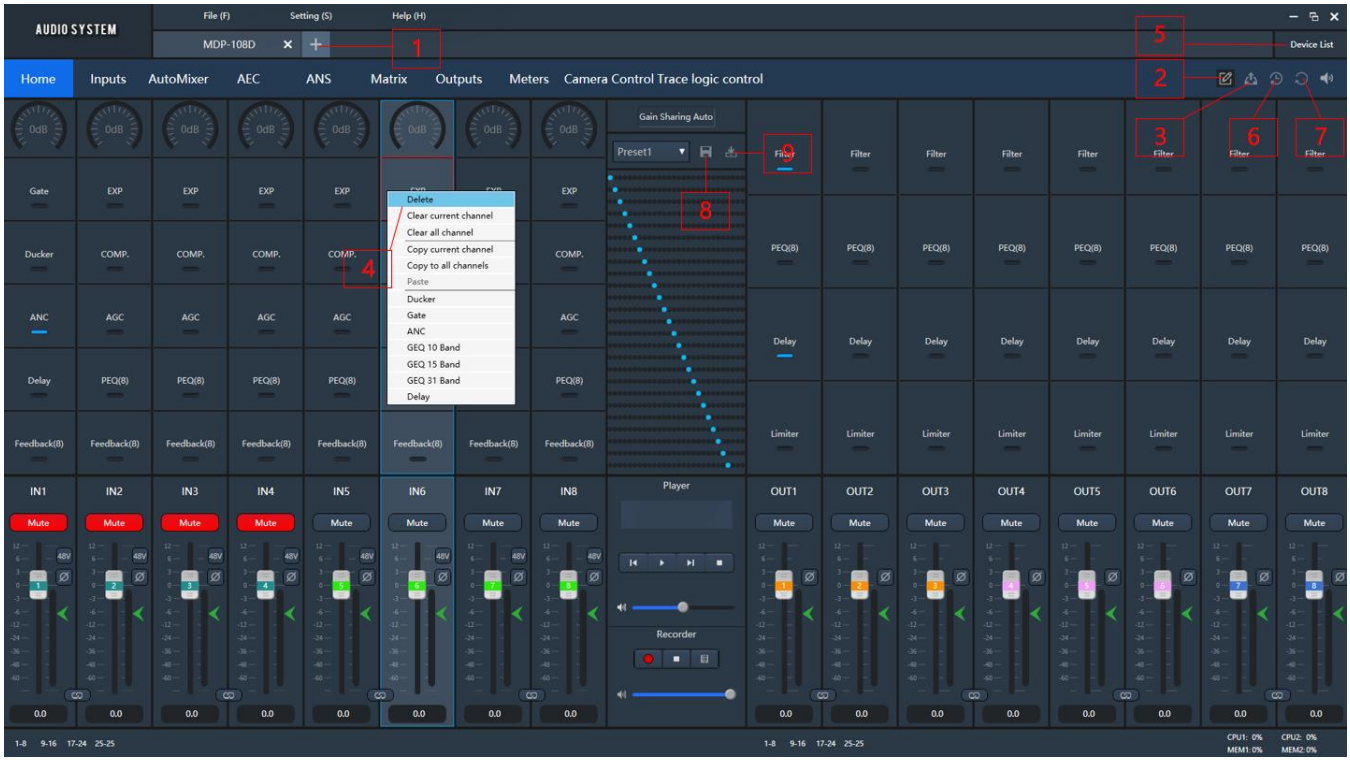
After opening the software, the main interface is displayed:



Click on the settings Device List button in the upper right corner of the main interface to automatically search for all processors on the network. Users can connect to the specified processor as needed, and the connection indicator light will flash when connected online. A processor can support up to 8 users to connect and control online at the same time.

3.3 Custom edit processing module

Click the edit button, right-click on the input or output channel processor module, and select the edit dialog box. You can replace the current processing module, delete, copy, and perform other operations. After editing, click upload to the host. Attention: When the CPU display exceeds 100, it will turn red. At this point, the resources cannot be uploaded to the host due to exceeding the limit and need to be re edited.



(Configurable software)

1、Add multiple identical devices or devices of different models, supporting up to 8 devices for simultaneous online editing.

2、DSP module editing function, which can replace each input or output DSP module.

3、Upload the edited program file to the device.

4、After selecting the edit DSP module function, right-click on the DSP module that needs to be replaced, and all replaceable DSPs will appear.

5、Device list, search for online devices that can be connected.

6、Restore the default settings of the scene, and the device will clear all current preset settings.

7、Restore factory settings, the device will clear all settings.

8、Save all current settings to the specified preset.

9、Load preset, recall specified preset.

At the same time, we also support: up to 8 devices can simultaneously edit online, configure storage (upload to device), restore presets, factory restore, save presets, load presets, and other functions.

3.4 Audio Module Parameters

There are two ways to adjust module parameters. The first method is to click directly on the input or output channel module to enter the parameter interface of that module; The second method is to right-click on the module to bring up the module's configuration interface. The following module parameters are explained in the first way.

3.4.1 Input Srouce



Sensitivity: Microphone gain, available in 17 levels of 0/3/6/9/12/15/18/21/24/27/30/33/36/39/42/45/48 dBu.

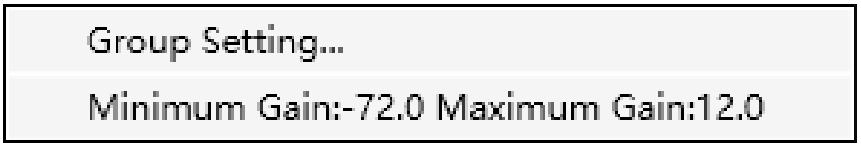
Phantom power supply: Feed the external condenser microphone, click this button when needed. Do not turn on the line input or when no power is needed to prevent damage to external devices.

Sine wave: Dragging the frequency can generate a sine wave of a specified frequency (20-20kHz). The output level can be adjusted as needed, in dBFS units. Use a pusher to adjust or click on the text input box to specify a numerical value.

White noise: White noise contain equal energy on each frequency component. Observing it on a constant bandwidth spectrometer, it exhibit a flat spectrum. At this point, frequency adjustment is ineffective and the voltage level is available.

Pink noise: The frequency component power of pink noise is mainly distributed in the mid to low frequency range, and it decreases at a rate of 3dB/Oct in the spectrum. At this time, frequency adjustment is ineffective and the voltage level is available.

In addition, right-click on each pusher on the main interface to see the following menu settings.

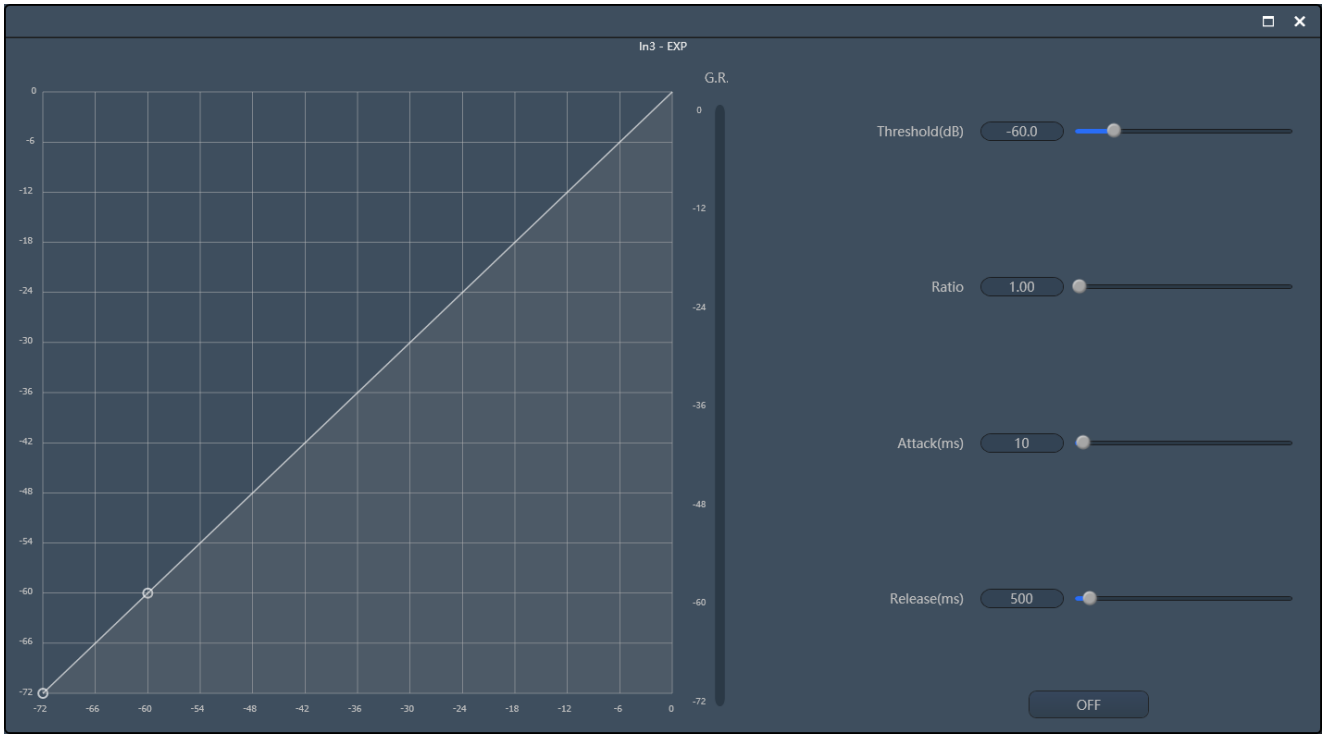


Group settings: Quickly open the group settings interface.

Minimum and maximum gain: Limit the maximum and minimum gain values of the channel. After debugging, if you do not want external changes to affect the stability of the system, you can set the maximum gain.

3.4.2 Expander

Expander is theoretically opposite to compressor, as it can expand the dynamic range of the signal. The most fundamental difference between these two devices is that the compressor works on signals above the threshold, while the expander works on signals below the threshold. Expanders can reduce the level of small signals. As shown in Figure 3.2, when the expansion ratio is 1:2, input signals below the threshold of 20dB will produce output signals below the threshold of 40dB. From the situation shown in the figure, the signal below the threshold will stretch downwards, resulting in a lower level. When using a 1:20 expansion ratio, the transmission characteristics of the expander look like a noise gate. In fact, the noise gate is an expander that uses a large expansion ratio.



The extender has the following control parameters:

Threshold: The signal must exceed this level to open the expander (allowing the signal to pass through). Actually, it is usually set to the level of environmental noise.

Ratio: The slope below the threshold point on the gain curve. Start approaching the door when the ratio is set high.

Attack time: The duration of the input signal, above the threshold, and the time required to open the extender. A faster opening time allows for faster transient opening of the expander.

Release time: The time required for the gain to return to a value below the threshold after the input signal drops below the threshold.

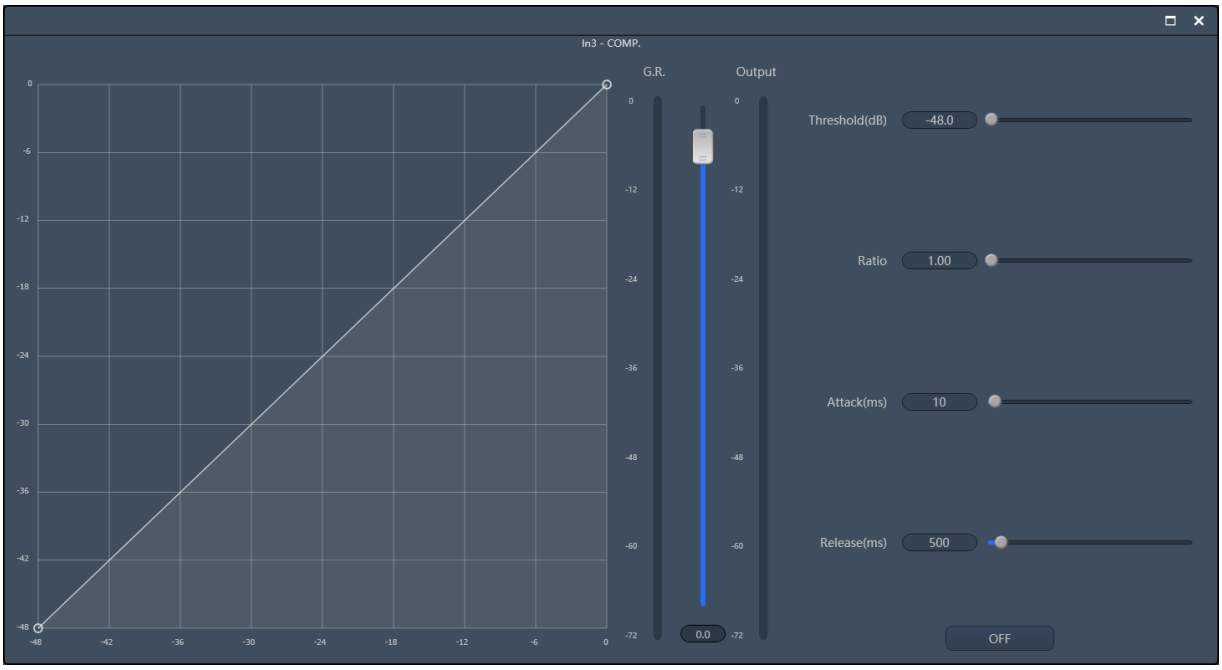
Whether it is attack time or releasing time, their function is only to reduce the rate of change in gain attenuation. The speed at which the gain increases from -40dB to 0dB is controlled by the attack time and slows down, while the speed at which the gain decays from 0dB to -40dB is controlled by the release time and slows down. The attack time or release time is independent of the threshold setting. If the signal undergoes high and low changes below the threshold, the attack time and release time will also have an impact on the gain attenuation. Once the signal level rises above the threshold, the gain attenuation generated by the expander will decrease at a rate controlled by the attack time. When the gain attenuation decreases to 0dB, the

expander stops expanding. Subsequently, when the signal drops below the threshold again, the expander starts again and the release time begins to take effect.

3.4.3 Compresser Limiters

compressor

The compressor reduces the dynamic range of signals above the user set threshold, while the signal level below this threshold remains unchanged. The compressor has the following control parameters:



Threshold: When the signal level exceeds the threshold, the compressor/limiter begins to decrease the gain. Any signal that exceeds the threshold will be considered an overshoot signal and its level will be reduced under normal circumstances. The larger the range of signals exceeding the threshold, the more the level is attenuated.

Ratio: compression ratio. The ratio determines the degree to which the overshoot signal decays towards the threshold level. The smaller the compression ratio, the easier it is for the signal to be higher than the threshold. Once the signal exceeds the threshold, the compression ratio parameter determines the ratio of the input signal change to the output signal change. For example, when the compression ratio is 2:1 and the input signal exceeds the threshold by 2dB, the portion of the output signal that exceeds the threshold only changes by 1dB. A compression ratio of 1:1 indicates that the compressor has not proportionally attenuated the signal. The adjustable range of compression ratio is 1-20.

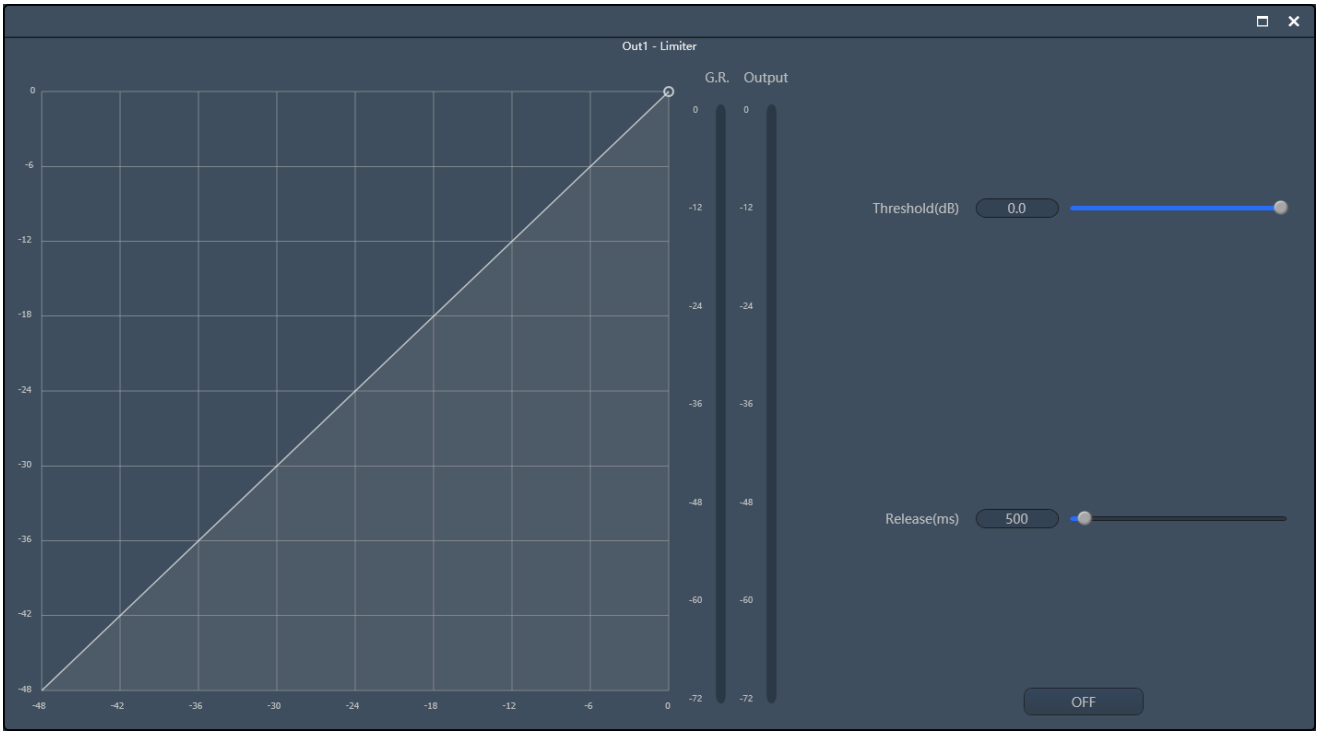
Attack time and release time: To preserve the natural vibration sensation, it is usually desirable for the initial part of the voltage level to pass through the compressor without being affected (or only slightly affected). To achieve this goal, it is necessary to slow down the reaction time of the compressor. Similarly, if there is a significant rapid attenuation and recovery of signal gain, a suction effect will occur. The attack time and release time of the compressor are to avoid this situation from happening. The establishment time can determine the speed of gain attenuation, while the release time can determine the speed of gain recovery.

Output gain: also known as gain compensation multiplier. If the compressor significantly reduces the signal level, it may be necessary to increase the output gain to maintain the volume level. This lifting operation has the same lifting amount for all parts of the signal, regardless of the setting of other parameters of the compressor.

G. R. and output level meter: G R. Indicate the compression capacity of the compressor; Output represents the output level of the signal passing through the compressor module. The compression amount is displayed on an inverted level meter. If the input signal is -6dB, the threshold is set to -30dB, and the ratio is 2:1, then the compression amount is 12dB. G.R. level meter

H. Indicate at -12dB position, output indicates at -18dB position.

limiter



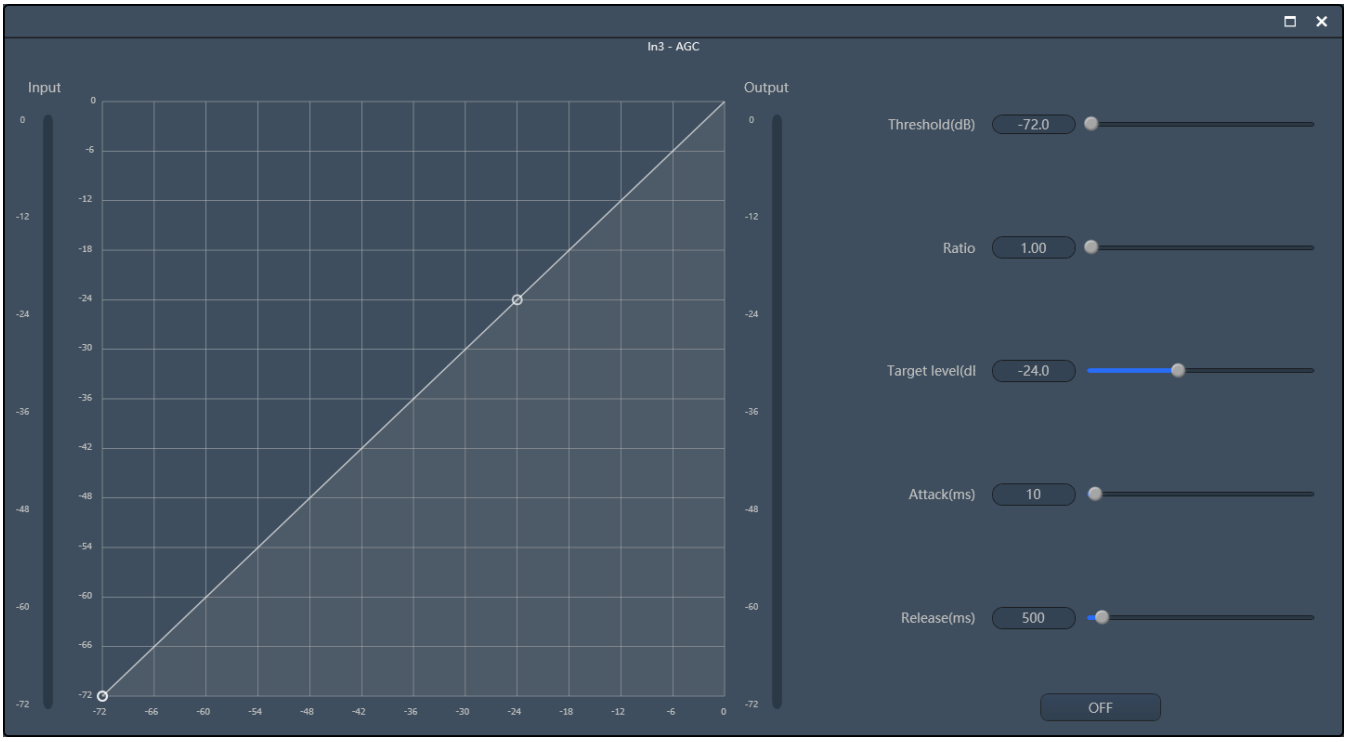
The one primary task: to ensure that the signal does not exceed the threshold level under any circumstances. By adjusting the control parameters of the compressor, its working mode can be made very similar to that of the limiter. The core of the limiter's working principle is that it truly relates to the signal content below the threshold level, as well as how the gain attenuation starts to occur before the signal experiences overshoot. The restriction period is completed through two processing stages. In the first stage, only slight restrictions are applied, but overshoot signals are not processed. In the second stage, if signals experience overshoot, they will attenuate in a very severe manner.

The limiter only provides two parameters, **threshold** and **release time.** For signal processing, occasional clipping should be resolved through the limiter, while frequent clipping usually requires attenuation of the signal level.

3.4.4 Automatic Gain Control

Automatic gain control (AGC) is a special case of a compressor, with a threshold set at a very low level, moderate to slow attack time, long release time, and low ratio. Its purpose is to raise signals with uncertain levels to a target level while maintaining dynamics. Most automatic gain controls include some form of silent detection to prevent gain attenuation loss during periods of silence. This is the only function that distinguishes automatic gain control from regular compressors/limiters.

Using automatic gain control to normalize the voltage level of CD players playing background music, foreground music, or waiting music, in order to eliminate some variations in the level of paging microphones.



Automatic gain control includes the following controls and switches:

Threshold: When the signal level is below this threshold, the input/output ratio is 1:1. When the signal level exceeds this threshold, the input/output ratio varies with the ratio control setting. Set this threshold to be just above the background noise level of the input signal.

Ratio: The ratio between the change in input signal level above the threshold and the change in output signal level.

Target threshold: The required output signal level. If the signal is above this threshold, the controller will compress the signal according to the ratio.

Attack time: Control the response time above the threshold level.

Release time: Control the level response time below the threshold signal

3.4.5 PEQ (Parametric Equalizer)

The main function of an equalizer is to correct frequency ranges that are overly emphasized or missing, regardless of whether these frequency ranges are wide or narrow. In addition, equalizers can also help us narrow or broaden the frequency range, or change the size of

certain components in their spectrum. In simple terms, an equalizer can alter the timbre of a signal.

The equalizer has the following control parameters:

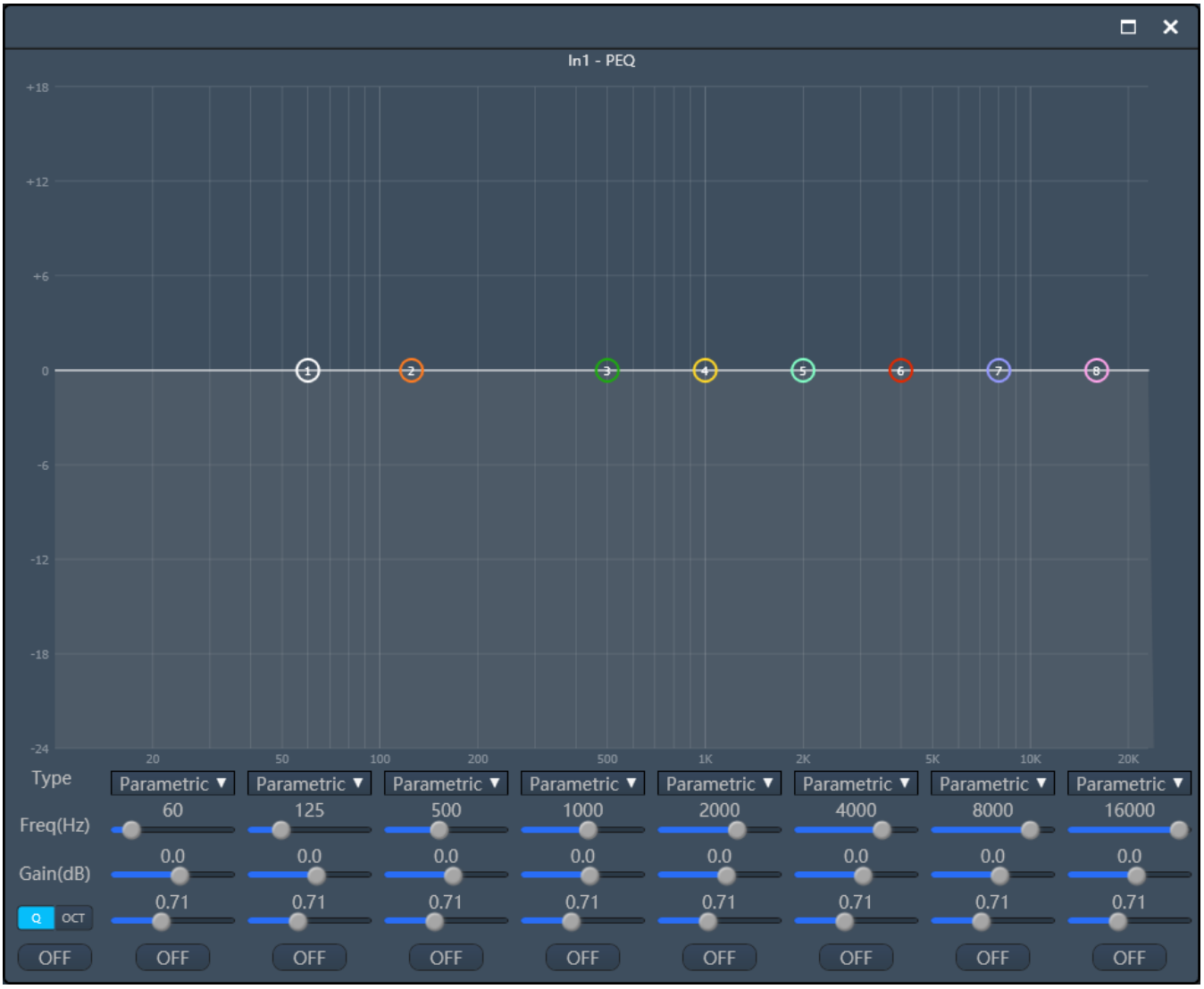


Image 3.6 Equalizer

Type: Default parameter equalization, optional high and low frame filters and high and low pass filters. Each type of filter has different forms and can accomplish different functions.

High&Low pass filter: The reference frequency of the pass through filter is called the cutoff frequency. The pass through filter allows the frequency components on one side of the cutoff frequency to completely pass through the filter, while continuously attenuating the frequency components on the other side of the cutoff frequency. Among them, a high pass filter can pass frequency components above the cutoff frequency and filter out frequency components below the cutoff frequency. Low pass filters, on the other hand, allow frequency components below

the cutoff frequency to pass through while filtering out frequency components above the cutoff frequency.

High&Low shelf filter: The meaning of elevated filter is to increase or decrease the gain of the frequency part above the set frequency. A low frame filter is used to increase or decrease the gain of the frequency portion below the set frequency. The set frequency is not the 3dB cut-off frequency, but is located at the center point of the falling or rising edge of the filter. The Q value affects peak value and has a corresponding mathematical relationship with peak value.

Frequency (Hz): The center frequency of the filter.

Gain (dB): The decibel value of gain increase or attenuation at the center frequency position.

Q: The quality factor of the filter. The adjustable range of Q value is 0.02~50;

When the type is parametric equilibrium, the Q value represents the width of the bell shaped frequency response curve between the cutoff frequencies on both sides.

When the type is high-low frame or high-low pass, if Q>0.707, there will be some peak phenomenon in the filter response. If Q<0.707, the slope will be flatter and the timing of roll off will be advanced.

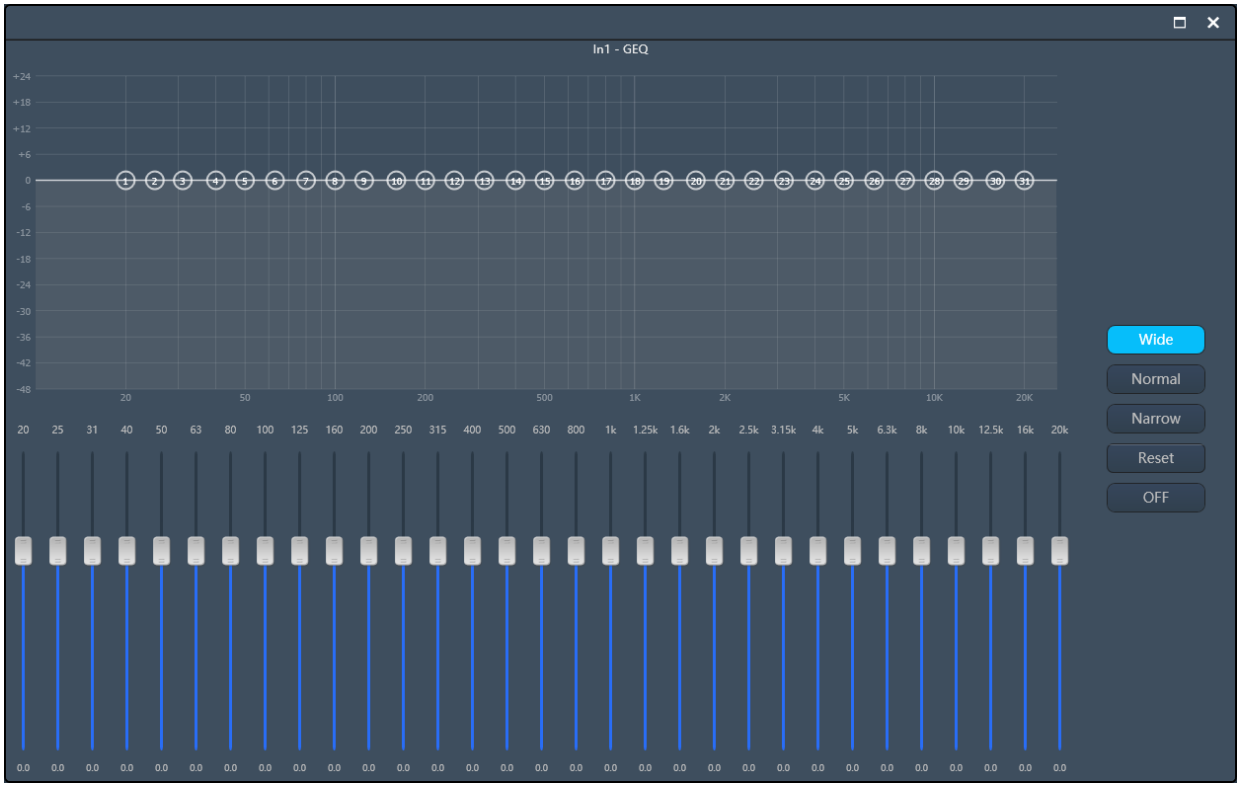
There is a switch under each equalizer segment, indicating whether to turn on or off the segment. When turned off, the parameter settings for each segment do not work. The equalizer has a main switch that indicates whether to enable or disable the module.

Parameter equilibrium has 5, 8, and 12 segments to choose from.

3.4.6 GEQ

Using constant Q-value technology, each frequency point is equipped with a push-pull potentiometer, which ensures that the bandwidth of the filter remains unchanged regardless of whether a certain frequency is increased or decreased. The commonly used professional graphic equalizer divides the 20Hz~20kHz signal into 10, 15, 27, and 31 segments for adjustment.

There are currently 10, 15, and 31 segments to choose from for the graphic equalizer.



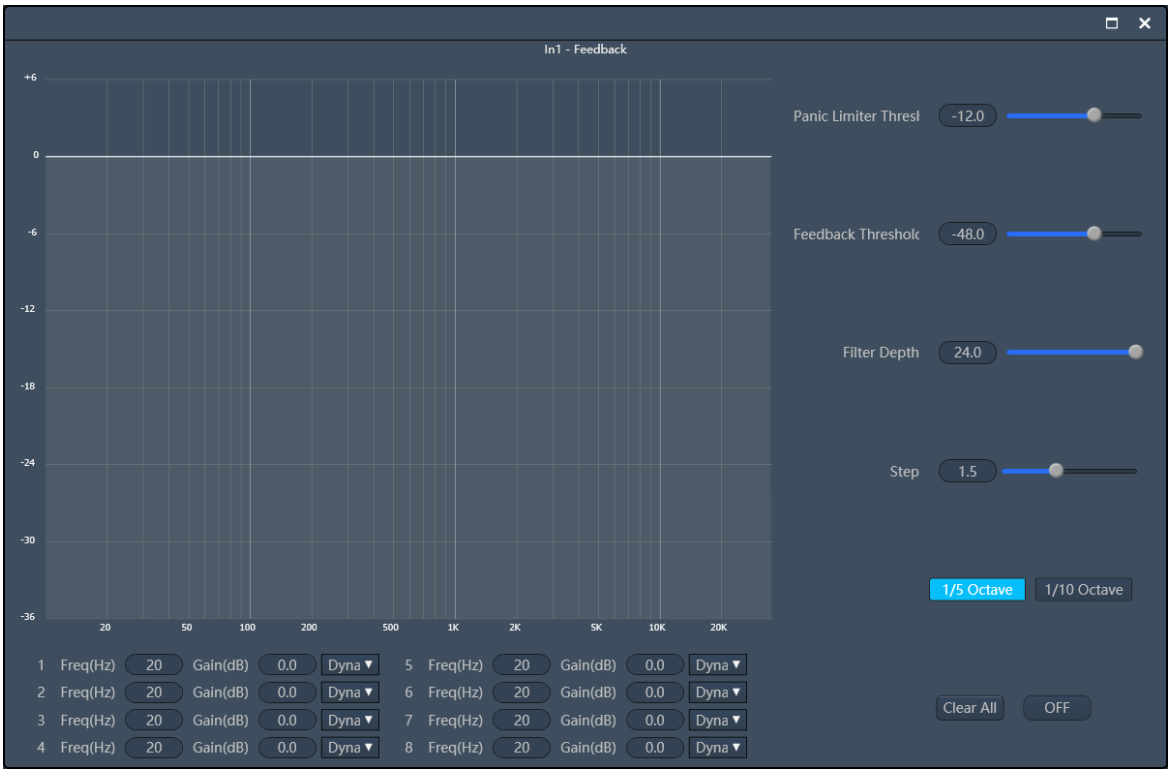
3.4.7 FeedBack

When using feedback suppression modules, they should always be combined with good system design and engineering practice, rather than replacing. Traditional methods such as limiting the number of microphones that can be turned on, minimizing the distance from the sound source to the microphone, locating the microphone and speaker to obtain minimal feedback, and balancing the room to obtain a flat response should still be used. Afterwards, feedback suppressors can be used to obtain additional gain. Feedback suppressors do not magically solve system design flaws or increase sound transmission gain beyond the physical limitations of the system.

The feedback suppression module automatically detects and suppresses acoustic feedback in the audio system. The module distinguishes between feedback and expected audio based on the characteristics of the signal. When feedback is detected at a certain frequency, a notch filter is automatically added at the feedback frequency point to attenuate it. When first added, the notch filter only attenuates slightly. If the feedback still exists, the notch filter will continue to attenuate according to the set parameters until the feedback disappears or reaches the maximum value set by the parameters. Multiple user parameters can be used to precisely fine tune the effectiveness of the module.

After the ringing output, the filters can be locked to prevent them from changing during the performance. The filter settings can be copied to a dedicated notch filter module (such as an equalizer). Eight filters are automatically cycled and set as automatic filters. In this way, filters that are only temporarily needed can be removed.

Each channel has a feedback suppression module. Use the mouse to drag the input module to find the feedback suppression module or click the shortcut key on the right to quickly enter the feedback suppression module. To enable the feedback suppression module, click the enable button to automatically detect feedback points and use narrowband filters to eliminate them. Each feedback suppression module has 8, 12, and 16 narrowband filters to choose from.



The adjustable parameters of the feedback suppression module include:

Panic threshold: This parameter tells the module that "anything above this level is definitely feedback." When the signal level is above the feedback threshold, several situations occur: a) the output gain is temporarily attenuated to control the speed of feedback establishment, b) the output level is limited to prevent loss of control, c) the filter sensitivity increases to detect feedback faster. Once the output level drops below the threshold, the gain recovers and the sensitivity returns to normal. This value refers to the peak level of the full-scale digital signal. When this value is set to 0, it is equivalent to turning off this feature.

Feedback threshold: This reference tells the module that "anything below this level is definitely not feedback." This can prevent the module from detecting feedback in soft music segments or due to low-level humming.

Filter depth: Set the maximum attenuation that a single filter can achieve. A shallower setting may prevent the filter notch from causing too much damage to the signal, and may also lead to worse feedback control, especially in large narrow resonance systems.

Bandwidth: Available in 1/10 and 1/5 Oct, with a constant Q value, the filter will not widen with increasing depth. It is recommended to set the bandwidth to 1/5 Oct when the filter bank is used up in a speech environment and feedback often occurs, as it has a wider bandwidth and a larger impact range.

Step size: The speed at which the gain decreases.

Automatic notch filter: Each notch filter has two modes, "automatic" and "manual", set to automatic. When the filter participates in the filter cycle and detects new feedback when 8 filters are used up, the module will search for the "automatic" filter setting and use it to suppress the new feedback. When set to manual, the gain of the notch filter can be manually set and the frequency point can be adjusted. If you need to save these feedback parameters, please click the Save Presets button.

Clear: Click this button to instantly clear all filters. It will clear the feedback points that were previously checked and suppressed, which is usually done when re debugging the feedback module.

Feedback suppressors can be used as tools for system debugging to find feedback frequency points or as preventive measures during normal operation. If you want to achieve high system sound transmission gain and feedback suppression effect, it is recommended to follow the following steps for debugging:

(a) Reduce the system gain and use the clear button to reset all filter parameters

(b) Set the parameter values for the feedback suppression module. Simultaneously reducing the panic threshold to lower the level of feedback occurrence

(c) Turn on all microphones and slowly increase the system gain until feedback occurs. Stop adding when feedback occurs

(d) Wait for the feedback suppression module to take action. After the feedback disappears, continue to increase the gain

(e) Repeat the operation until the system reaches the desired gain or all filters have been allocated

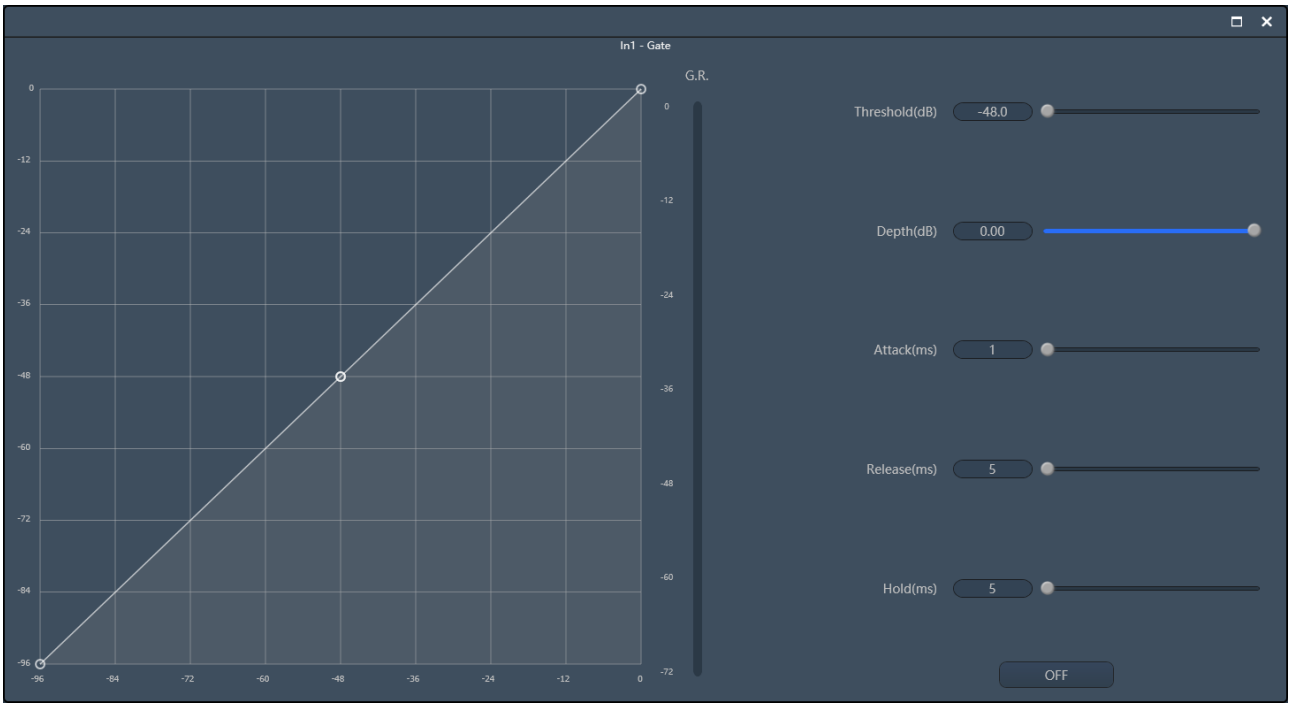
(f) Change the panic threshold to just above the maximum level of the expected non feedback signal

At this point, if necessary, each filter can be set to a fixed mode, or the dynamic state can be saved to handle feedback that may occur during the performance. Another possible approach is to copy the filter to a notch filter module (such as an equalizer). This allows for the addition of more filtering capabilities.

If the devices includes speakers, it is recommended to use a compressor/limiter module for additional protection. Setting appropriate limiters will ensure that the speaker is not damaged, even if all notch filters are used up or feedback suppressors cannot control feedback, such as when the system gain is too high.

3.4.8 Gate

Noise gate: The main purpose of a noise gate is to attenuate signals below a threshold, and this attenuated signal is usually noise.



Threshold: signals exceeding the threshold pass through, signals below the threshold activate and attenuate

Depth: Attenuation amount, which determines how much signal below the threshold needs to be attenuated.

Attack time: Start time refers to the speed at which the noise door opens.

Release time: The release time is opposite to the start time and refers to the speed at which the noise door closes.

Hold time: The hold time is such that when the signal is below the door closing threshold, under normal circumstances, the signal will immediately begin to decay. If you set a Hold time of 10ms at this time, the noise gate will maintain its original output for 10ms and only begin to decay after 10ms.

3.4.9 Ducker

Ducker: When the level value of one channel exceeds the specified threshold, the level of the other channel will be attenuated, which is the ducker effect.

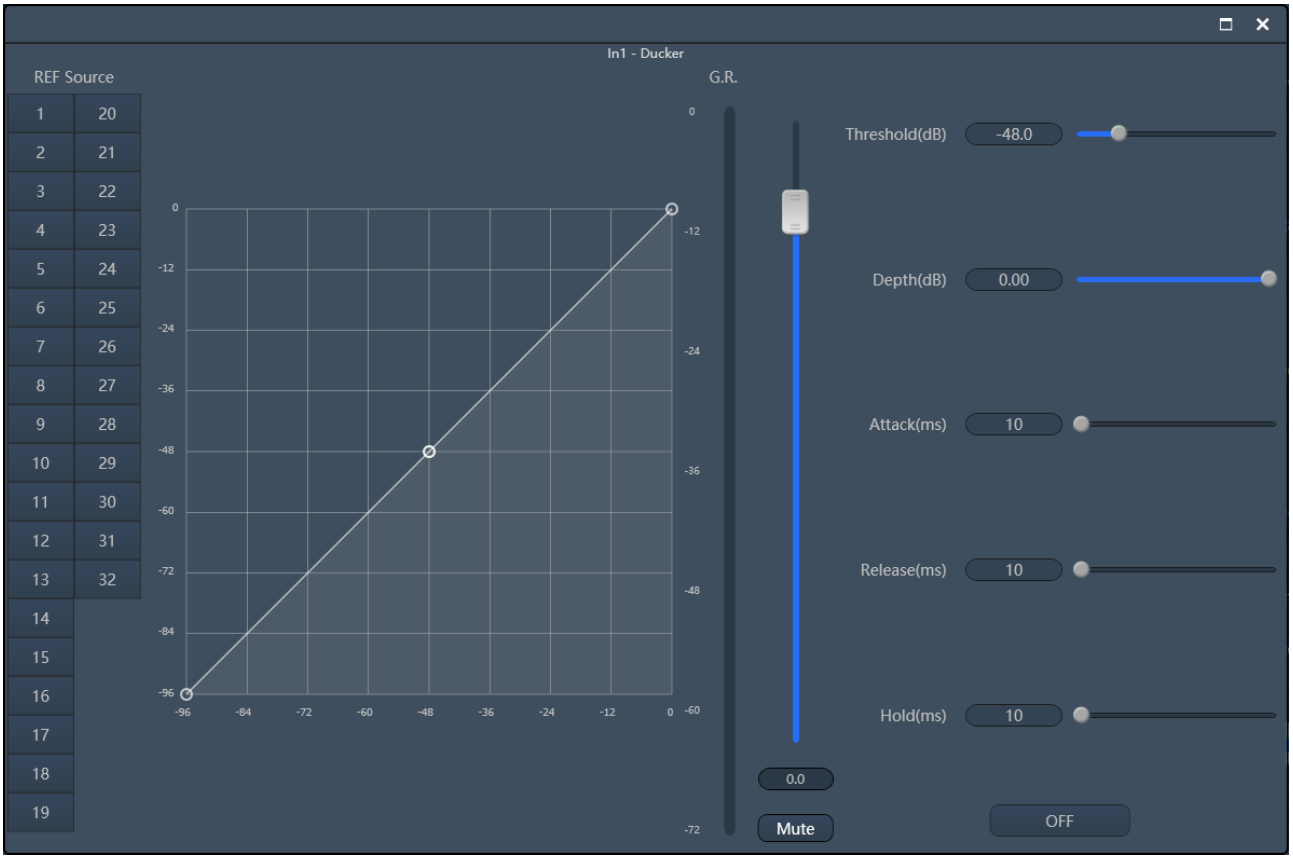
Threshold: The reference signal begins to decay when it is above the threshold, and recovers when it is below the threshold.

Depth: The amount of signal reduction caused by evasion.

Attack time: the time when the reference signal starts to attenuate the signal of the dodged channel when it is higher than the threshold value.

Release time: when the reference signal is lower than the threshold value, the dodged signal will recover to the original signal size.

Holding time: Holding time refers to how long the dodge on the dodged channel needs to be maintained after the control signal is below the threshold.



3.4.10 Automatic Noisegain Compensator (ANC)

Noise gain compensator: Automatically adjusts the output volume based on environmental noise induction and processing.

Maximum gain: The maximum amount that can be adjusted.

Minimum gain: The minimum amount that can be adjusted.

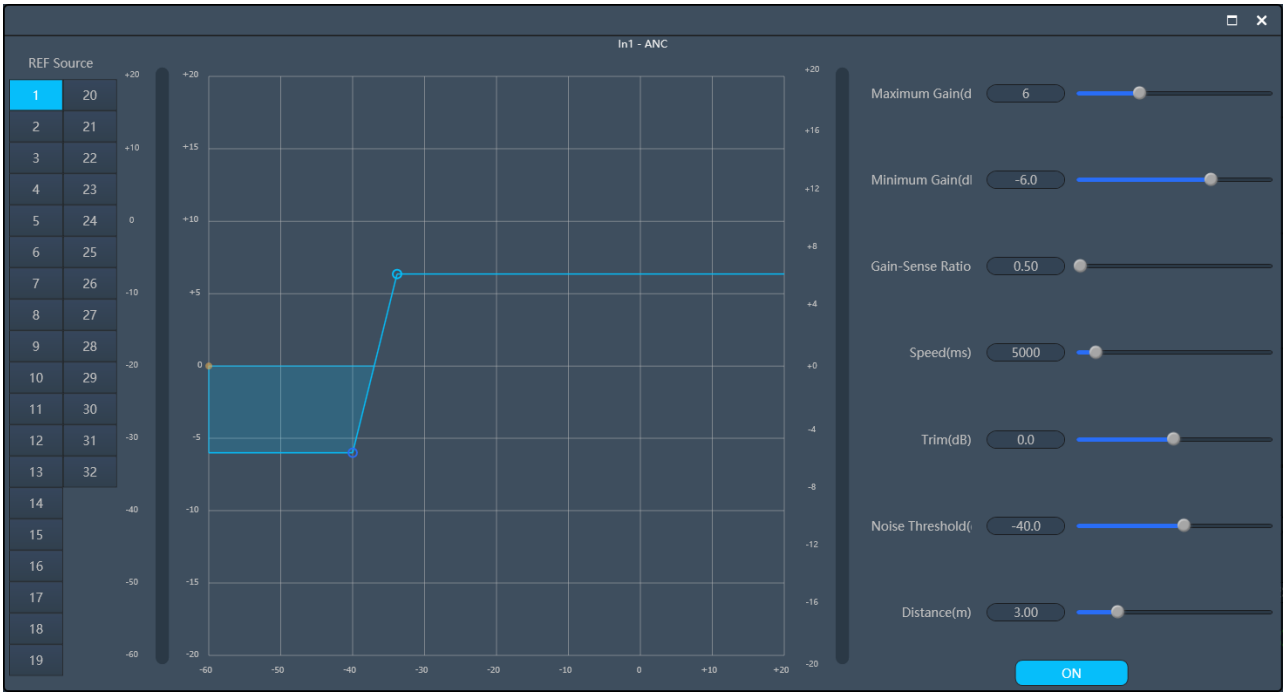
Gain induction ratio: the ratio of increase or decrease.

Speed: The speed of increase or decrease.

Trim: Digital Gain.

Noise threshold: If it exceeds the threshold for increasing gain, it will attenuate if it is less than the threshold.

Distance: the distance between the reference signal and the local signal



3.4.11 AutoMixer

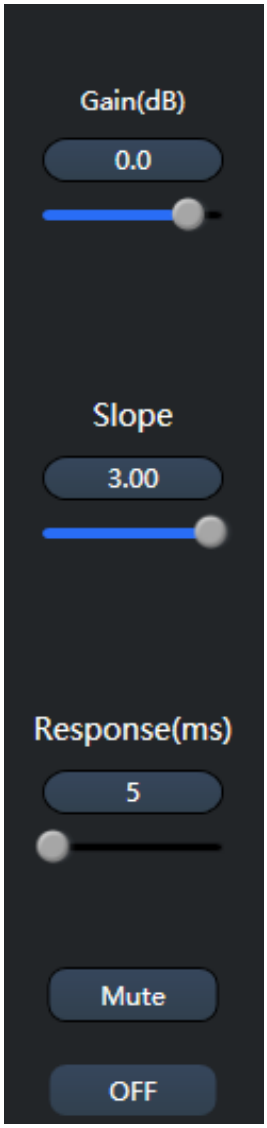
In a conference room, if multiple microphones are turned on to the same gain level and only one person is speaking, the result may not be very clear, and other microphones will pick up room noise, reverberation, etc. When these signals are mixed with normal microphone signals, the quality of the mixed audio output will be greatly reduced, and the entire sound amplification system is prone to whistling and cannot obtain sufficient sound transmission gain. To solve this problem, it is necessary to turn off other microphones that are not currently in use. The automatic mixer can complete this closing process and has a much faster response speed than manual operation.

The processor is equipped with a gain sharing automatic mixer that supports up to 32 channels of audio signal input. Each channel in the automatic mixing matrix has a direct output that is not affected by automatic gain and channel faders, but only by channel muting. Channels suitable for fixed volume, such as background music channels, need to be maintained at a fixed level without being controlled by automatic mixing; For example, if it is necessary to keep the chairman's microphone in a normally open state and its gain is not affected by automatic mixing, the output of that channel can be directly adjusted in the output matrix routing. At this point, the automatic mixing button of the channel can also be turned off, its gain will not be adjusted, and the signal level on that channel will not affect the gain on other channels.

The automatic mixing module has two sets of control parameters: main control parameters and channel control parameters.



(1) Main control parameters



The bottom button turns on or off the automatic mixing function

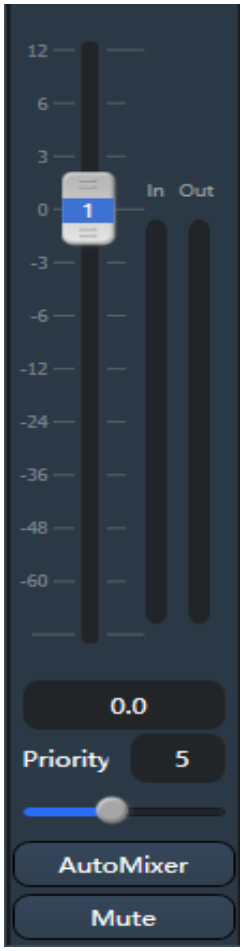
Gain: Control the main output volume of automatic mixing

Slope: Slope control affects the attenuation of lower levels. When the slope is higher, channels with lower levels will also experience more attenuation. The slope control working mode is similar to the ratio control working mode on the extender. Suggest setting the value to 2.0 or a value around 2.0. If set to 1.0, the effect is equivalent to turning off automatic mixing for all channels; Setting it to 3.0 may result in larger gain adjustments, which may lead to unnatural effects. The larger the set value, the more channels are opened, and the overall attenuation is also greater. When the slope is set to 2.0, ideal gain sharing can be achieved, which is the preferred value in use.

Response time: Faster time ensures that the beginning of the words spoken will not be cut off. When the time is slow, the operation is smoother. Practice has shown that the best results are achieved when the response time is between 100ms and 1000ms. The design of automatic gain is to make the speed of opening the microphone much faster than the speed of closing the microphone, so even a 100ms response usually does not deduct the initial words spoken. If set to a slower time of a few seconds, the automatic mixer response time will have a longer hold time,

and the previous active channel will save its open state within a few seconds.

(2) Channel control parameters



Automatic mixing: Each channel has an automatic mixing on/off button, and the channels that need to participate in automatic mixing need to turn on this button. It can also be turned off, as this channel does not participate in automatic mixing.

Mute: After automatic gain, both channel mute and pusher can reduce the level gain of other channels even if one channel has been muted if its level is high.

Gain: Adjusting the gain multiplier can increase/decrease the proportion of volume in automatic mixing.

Priority: Setting priority can overpower high priority channels over low priority, thereby affecting the automatic mixing algorithm. The parameter range is 0-10, and the higher the value, the higher the priority.

Channel mute and fader are both adjusted after automatic gain, and any adjustments to these two parameters will not affect the automatic mixing operation. For example, even if a channel is muted, if the level of that channel is high, the level gain of other channels can still be reduced. To mute a signal and prevent its impact on automatic mixing, it is necessary to turn on mute and turn off automatic mixing. The mute button on each channel will mute that channel in the mix and also output the mute directly. The channel pusher also controls the mixing level and direct output level of the channel. Click on the text box and enter a dB value to precisely control the channel level.

Priority control allows high priority channels to override low priority channels, thereby affecting the automatic mixing algorithm. This control can take values from 0 (lowest priority)

to 10 (highest priority), with a default value of 5 (standard priority). You can adjust the priority by using the slider or by clicking the edit box to enter a specified priority between 0-10. Increasing this value increases the priority.

If two channels have the same signal level, the channel with higher priority will receive higher automatic gain. If there is a one unit difference in priority between two channels, the channel with the higher priority will receive an additional 2dB (assuming the slope of both channels is set to 2.0) of automatic mixing gain. For example, if the priority of channel one is set to 6 and the priority of channel two is set to 3, and the input levels of the two channels are the same, channel one will receive an additional 6dB of mixing gain compared to channel one. During use, it should be noted that the slope setting of the main control parameter can also affect the mixing gain difference caused by the priority weight of the channel. If the slope is set to 3.0, a priority unit difference between channels will result in a gain difference of 4dB. If the priority of all channels is the same, then keep all settings at the default level 5.

Note: In some settings, extra caution is required when using extreme priority levels between channels, such as 0 and 10. If a channel with very high priority is recognizing signals such as background noise from the speaker, it may mask channels with lower priority, even if the channel with very high priority is not in use. The higher the slope, the more serious the problem. If you encounter this issue during installation and debugging, you can consider adding a noise gate or expander between the automatic mixers on the highest priority channel, while setting the threshold to a level where the threshold or expander will not be opened by background noise or speaker recognition.

In addition, a gating type automatic mixer can be added to choose from:



The gating automixer is developed based on the noise gate. Each channel has a noise gate, which is either open or closed.

Gain: The overall output gain of an automatic mixer.

Hold time: Hold function can adjust the length of time the channel remains open after the speaker stops speaking. This function ensures that brief pauses between words and sentences during the speaker's speech will not cause the channel to close.

Off gain: When the sound cannot pass through the threshold, the sound signal will be added to the set value of off gain.

Sensitivity: Sensitivity control, when the adaptive noise threshold of the channel is exceeded by a certain amount, the channel must be turned on. If the value of this setting is increased, the channel will easily conduct even in very quiet noise.

NOM Atten: The NOM attenuation multiplier can affect the magnitude of attenuation for all channels when more microphones are turned on. This parameter reflects the attenuation when two microphones are turned on, as well as the additional attenuation that increases with each doubling of the number of microphones turned on. For example, when set to 3dB, opening two microphones will attenuate 3dB, opening four microphones will attenuate 6dB, and opening eight microphones will attenuate 9dB. Similarly, doubling the number of microphone openings will increase the output attenuation value by 3dB.

NOM Limit: Set the maximum number of microphones that can be turned on.

Noise threshold: If it exceeds this value, open the channel; if it is less than this value, close it.

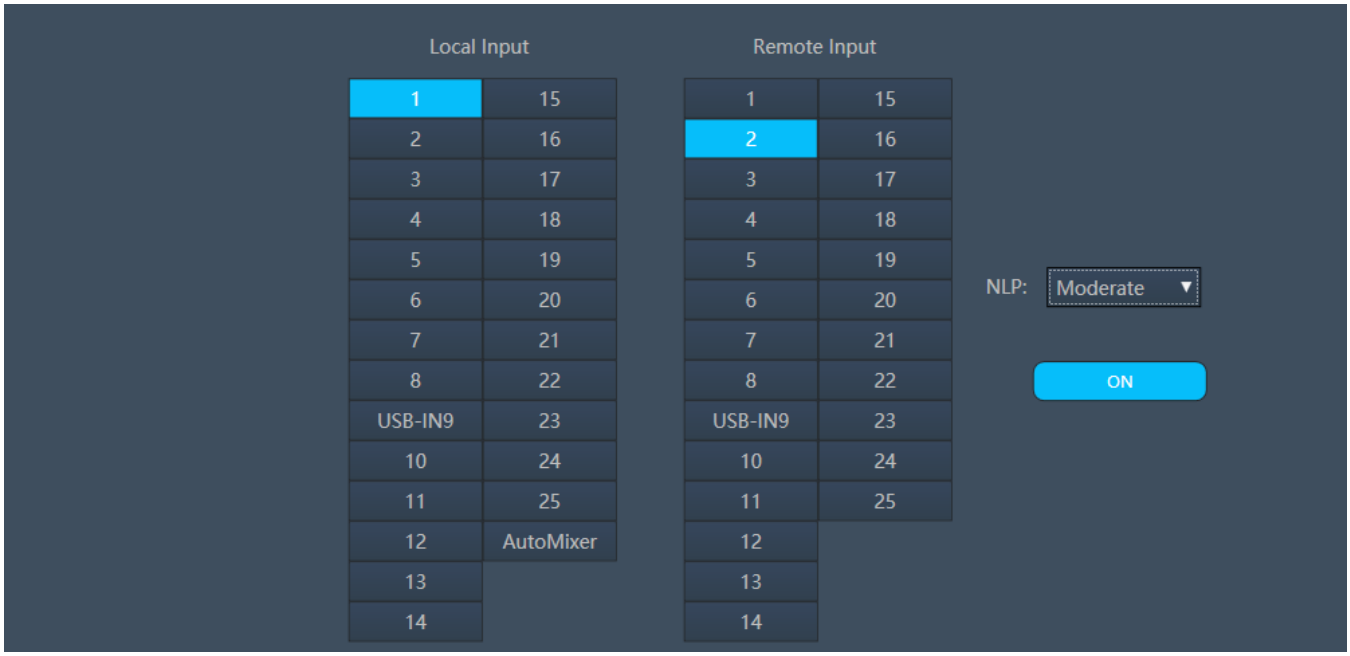
3.4.12 Acoustic Echo Cancellation (AEC)

Acoustic Echo Cancellation (AEC) is a digital audio signal processing technique used in audio and video conference calls when a conversation occurs between a local conference room participant and one or more speakers at a certain distance. The AEC program increases the clarity of remote speakers' speech by eliminating acoustic echoes generated in the local room.

The echo cancellation module applied in remote communication can facilitate local amplification of remote voice signals and attenuate the interference of acoustic echoes. Its basic working principle is to simulate the echo channel, estimate the possible echo formed by the remote signal, and then subtract this estimated signal from the input signal of the microphone, so that the input speech signal no longer contains echo, in order to achieve the purpose of echo cancellation.

There is only one echo cancellation module in the DSP Controller, and two local input and remote input mixers are pre-set to achieve multi-channel signal participation in echo cancellation, as shown in the figure. There is a parameter that can be adjusted:

Nonlinear programming（NLP）：Conservative, Moderate ,Aggressive. These three optional types select the suppression level of echo.



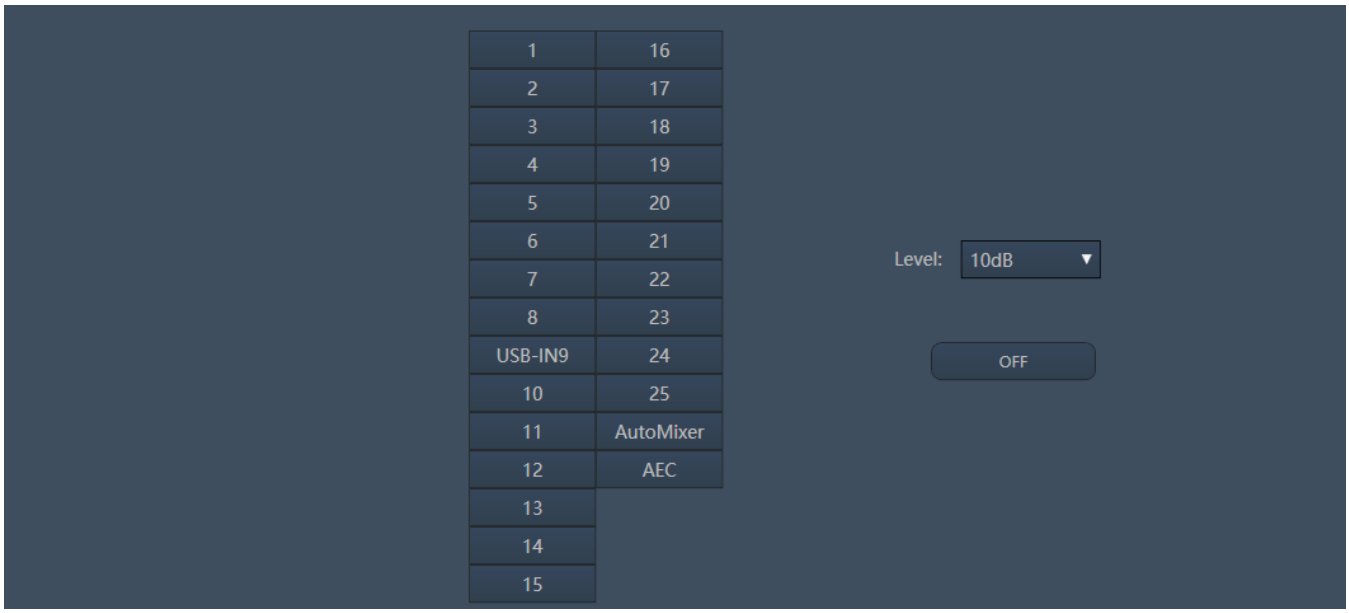
Note: The echo cancellation module needs to be used in conjunction with the matrix module to set signal routing.

3.4.13 Automatic Noise Supression（ANS)

The Automatic Noise Supression module can effectively remove the non human voice. Distinguish between human and non-human voices, and treat non-human voices as noise. A segment of audio containing both human voice and noise is processed by this module, theoretically leaving only human voice.

There is only one Automatic Noise Supression module in the DSP Controller, and a multi-channel mixer is preset to enable multi-channel signals to participate in noise cancellation, as shown in the

figure.

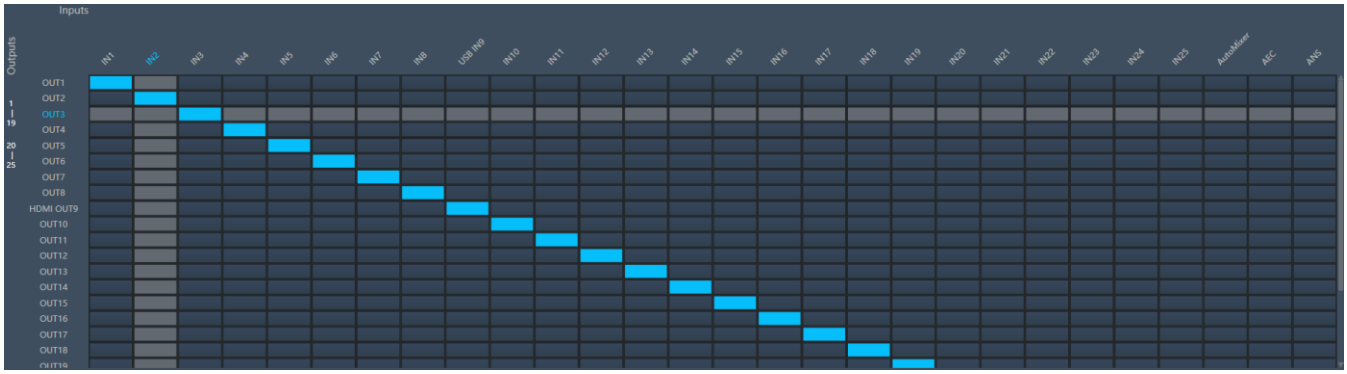




Suppression level: There are four types available: 6, 10, 15, and 18dB. The meaning of dB is how much noise is suppressed and reduced by dB. The larger the value, the greater the damage to speech, which cannot be avoided.

3.4.14 Matrix

The matrix has dual operation functions of routing and mixing. Horizontal represents input channels, vertical represents output channels, default one-to-one input/output, as shown in the figure. If you need to mix the sound from input channel 1 and input channel 2 into output channel 1, simply dot both horizontal 1 and 2 on output channel 1. If input 1 and input 2 participate in automatic mixing, the output is not affected by automatic mixing. Similarly, after the automatic mixing, echo cancellation, and Automatic Noise Supression modules are set, the matrix needs to be set to obtain the correct signal routing relationship.



3.4.15 HighPass&LowPass Filter

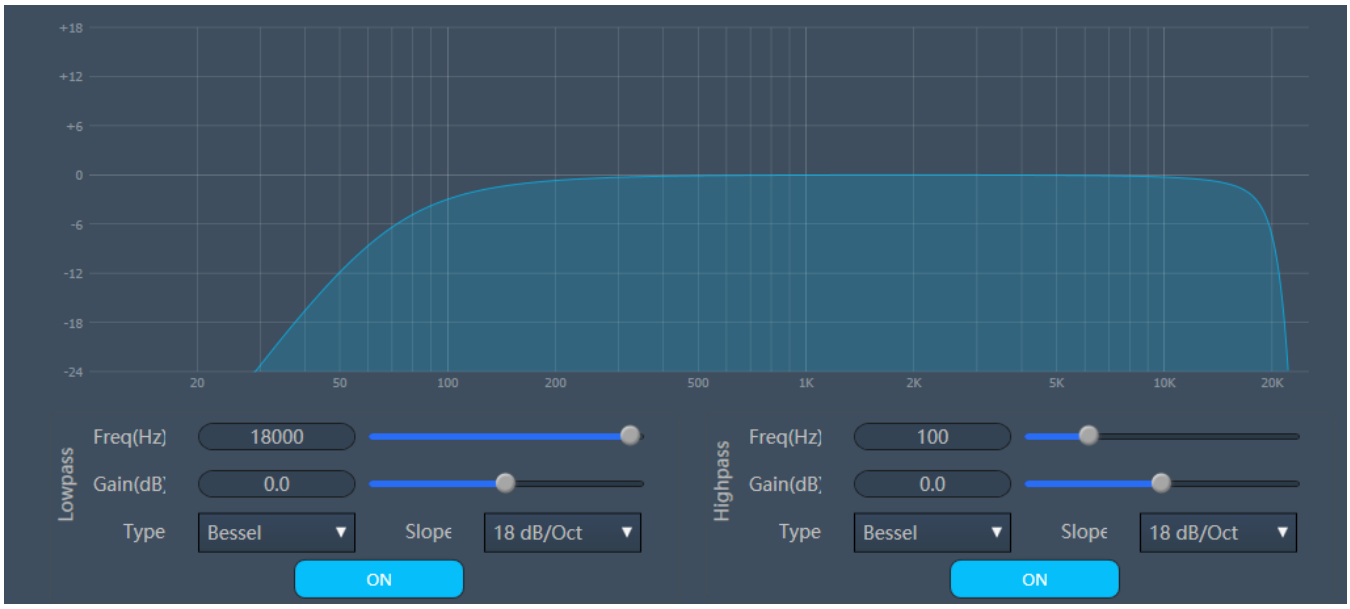
Each output channel provides a high-low pass module, consisting of a high pass filter and a low-pass filter. Each filter has the following four parameters:

Frequency: The cut-off frequency of the filter is defined at -3dB for Bessel and Butterworth, and at -6dB for Linkwitz Riley.

Gain: The gain setting affects the full frequency range enhancement or attenuation of the signal.

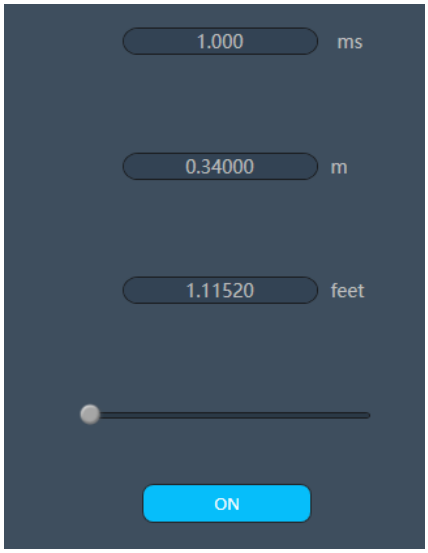
Type: Choose the filter type, which includes Bessel, Butterworth, and Linkwitz Riley. Butterworth has the smoothest passband.

Slope: The attenuation size of the transition band of the filter can be selected from eight options: 6, 12, 18, 24, 30, 36, 42, and 48dB/Oct. For example, 24dB/Oct represents the amplitude attenuation of 24dB for every octave difference in frequency in the transition band.



If you want to use a high pass or low-pass module, click the OFF/ON at the bottom.

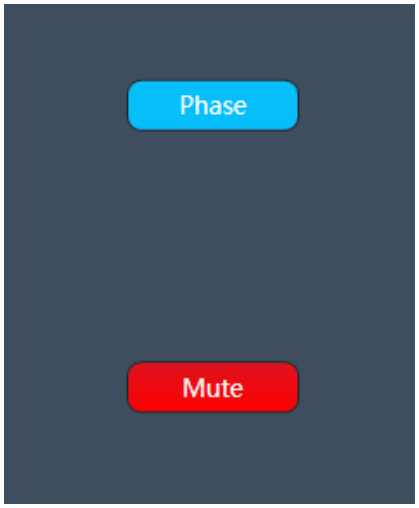
3.4.16 Delay



Activate button: Activate the specified delay module in the module, insert it into the audio signal path, and add a fixed delay time to the signal.

Milliseconds: Set the delay time of the delay timer. The value range is 1-1200 milliseconds. Both meters and feet are converted units in milliseconds.

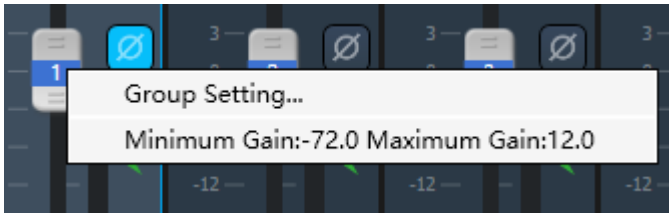
3.4.17 Output



Reverse phase: The audio signal phase is reversed by 180 °.

Mute: Set mute/non mute.

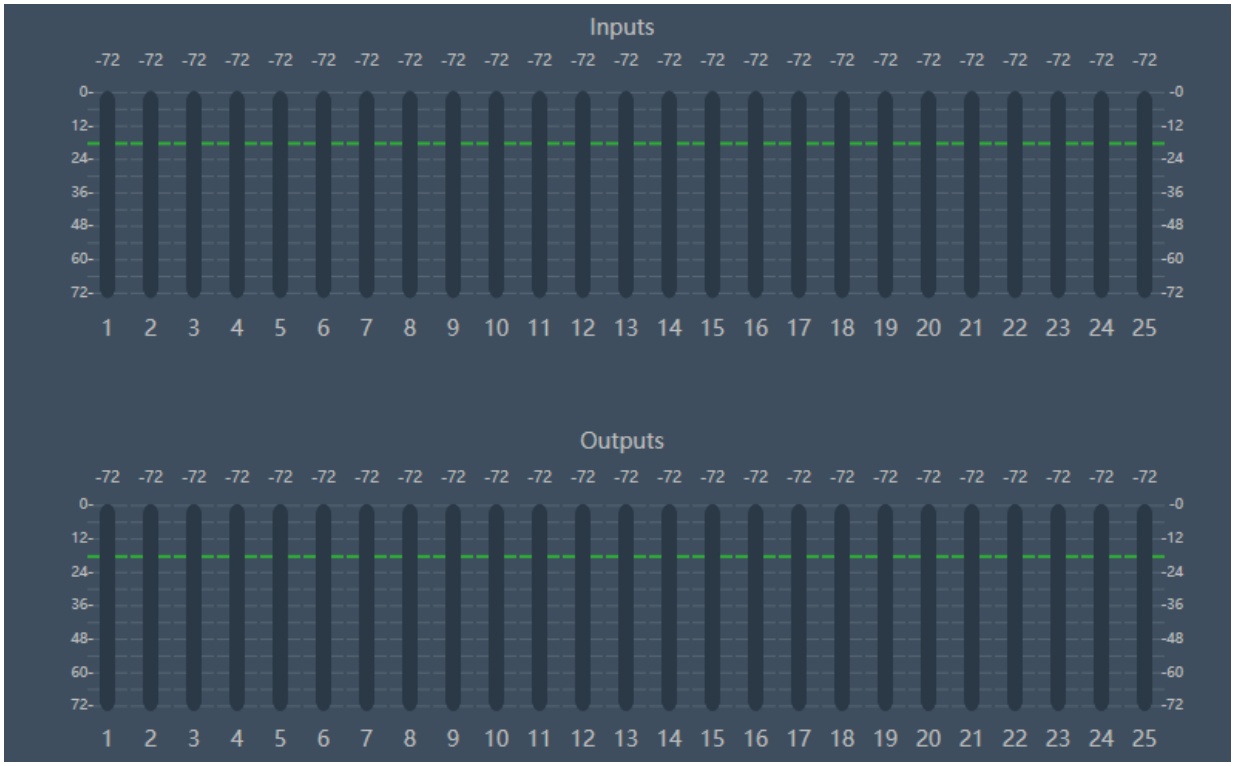
On the output channel, there are also some menus that can be set by right clicking on the same input channel. It can be set according to the needs.



3.4.18 Meter

The main function of a digital level meter is to monitor and analyze the strength of audio signals in real-time, ensuring the performance and quality of audio equipment. By using a digital level meter, one can intuitively see the level signal size of the audio signal in the channel, thereby determining whether the gain structure of the audio system is correct, whether there is signal distortion or overload, and whether the performance of the audio system meets the expected standards.

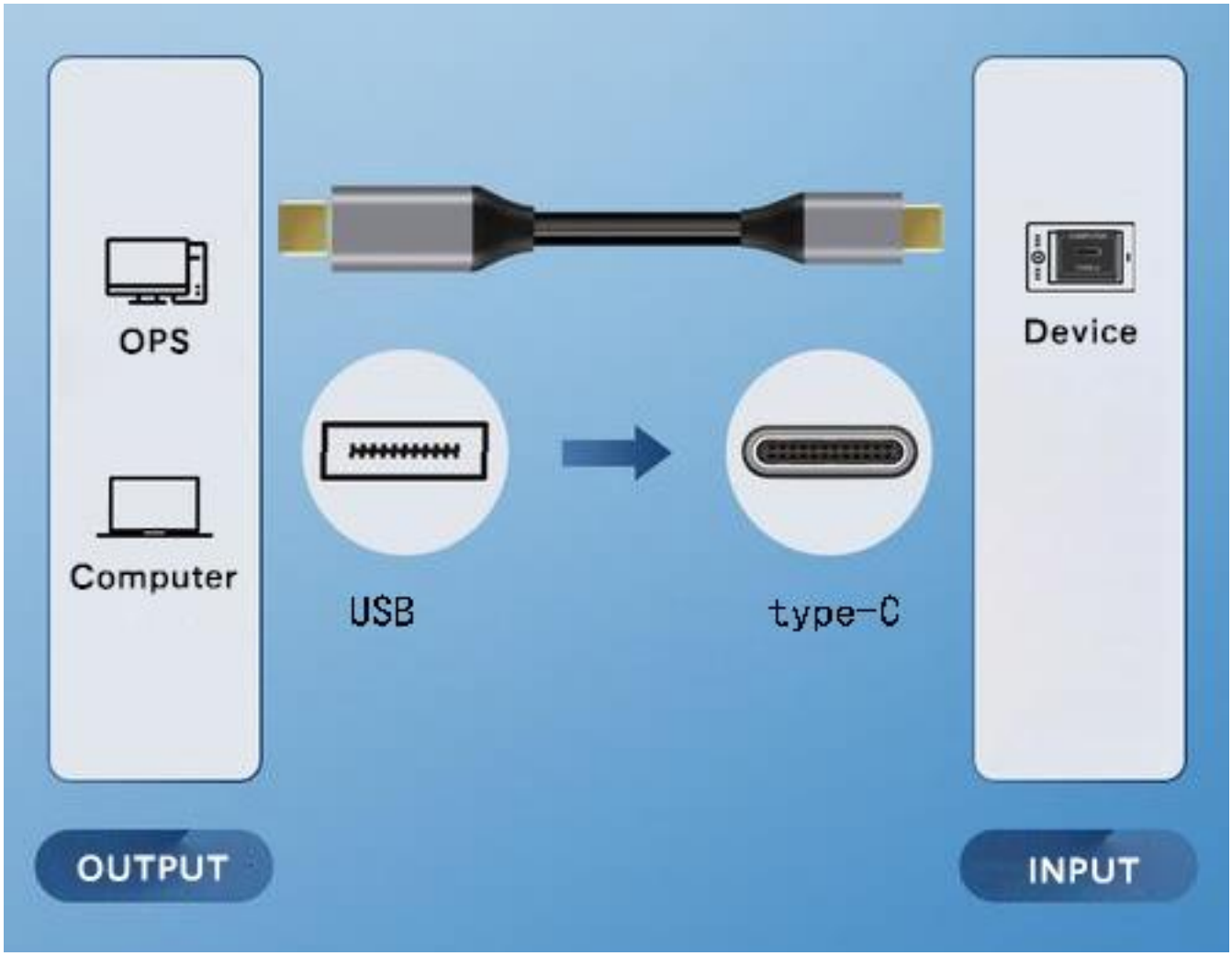
Level meter measurement unit: Digital level dBFS (maximum value is 0 dBFS) Note: The green scale position is the voltage level standard value (0 dBu)



3.4.19 USB Soundcard

There are two purposes of using a USB sound card: one is to achieve recording and playback; The second is remote conferencing on the PC end. The USB sound has undergone echo cancellation and noise cancellation modules, making it easy to connect to remote meetings. The USB playback of the software interface is only used for recording and playback functions.

Remote Meeting Instructions

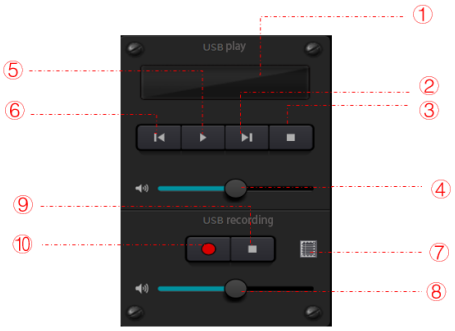


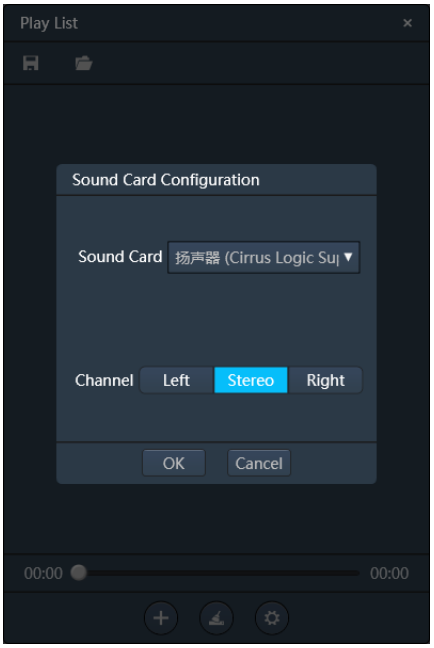
① Schematic diagram of remote video conference link on PC side, USB sound has passed through the echo cancellation and noise cancellation modules of Huiyi host, making it easy to connect to remote conference.

USB sound card settings

The USB sound card can be easily connected to the DSP and OPS host through a dual head Type-C interface USB cable. When connected for the first time, the computer will pop up to discover new hardware and automatically install drivers. After installation, a new USB Soundcard device will appear in the computer sound card list, and it will be automatically selected as the default sound card in the sound card settings.







① Playback information, double-click to enter the playlist

② Next

③ Pause

④ Playback volume

⑤ Play

⑥ Previous

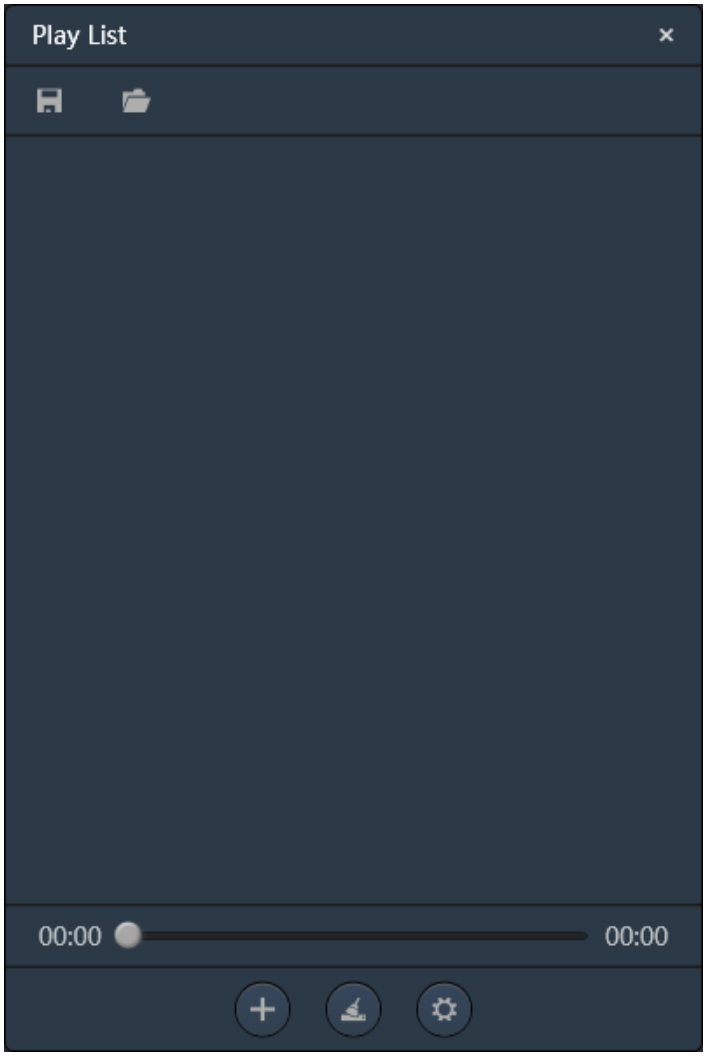
⑦ Recording List

⑧ Recording volume

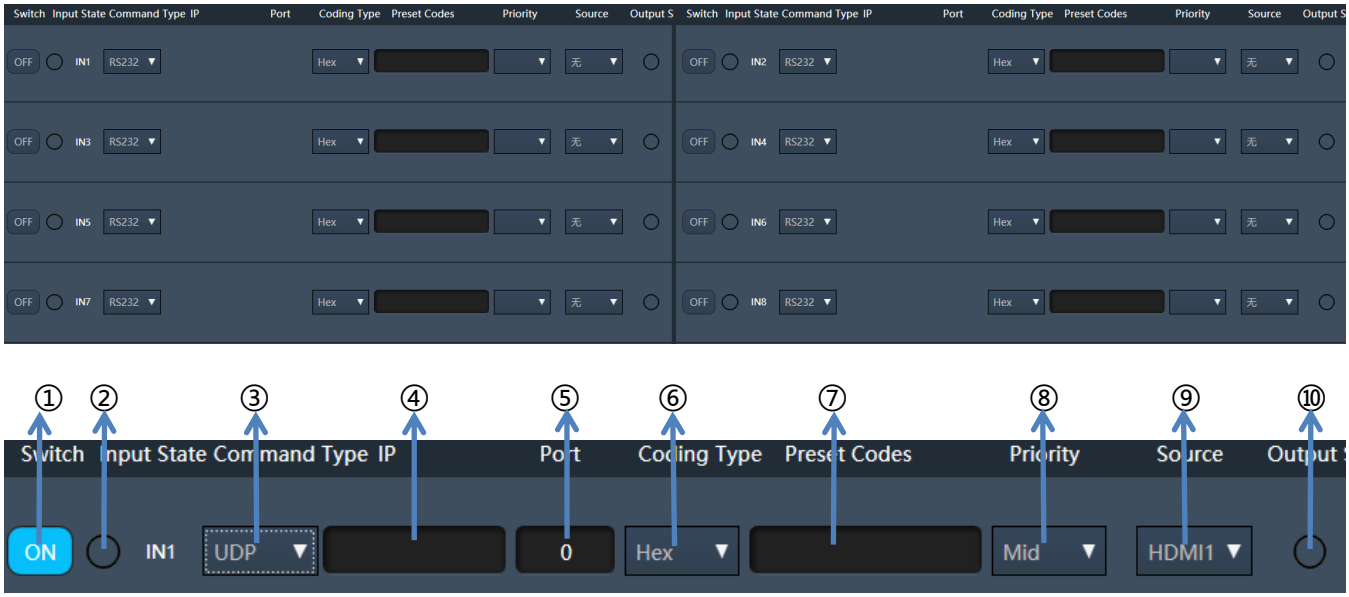
⑨ Stop recording

⑩ Start recording

Playlists allow you to manipulate song files or save songs as a list, which can be opened directly the next time you use them. Click the "+" at the bottom of the playlist to open the folder and select the song to play,  Clear Song List,  Enter the sound card settings interface



3.4.20 CameraTracking



①. Video tracking switch status

②. Input channel audio signal status (flashing status can trigger video tracking)

③. Communication protocol (processor and camera)

④. Camera network IP address

⑤. Camera communication port number

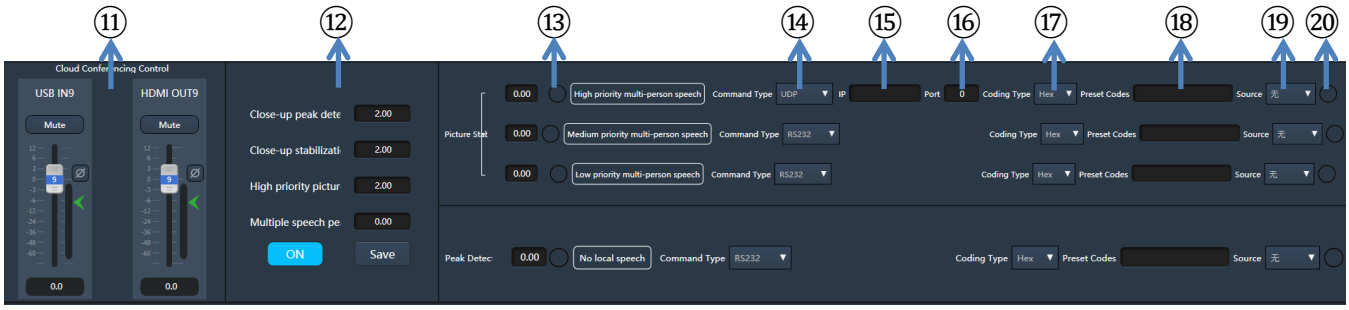
⑥. Control code type

⑦. Camera control code (code for calling preset bits)

⑧. Priority (high, medium, low, three gears)

⑨. High definition input source (up to 4 high-definition signal sources from cameras connected to corresponding HDMI interfaces)

⑩. Control signal output status.



⑪. HDMI input/output channel audio signal control

⑫. Several reference coefficients for camera tracking (close-up peak detection coefficient, close-up image stability coefficient, high priority image coefficient, multi person speech peak detection coefficient)

⑬. Input channel audio signal status (flashing status can trigger video tracking)

⑭. Communication protocol (processor and camera)

⑮. Camera network IP address

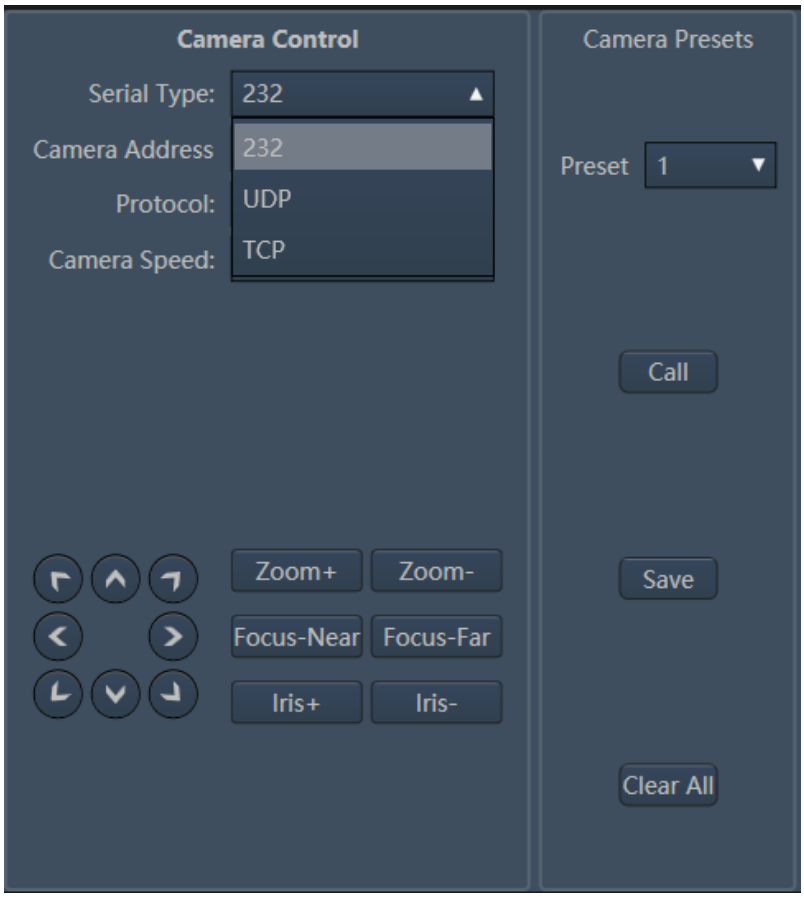
⑯. Camera communication port number

⑰. Control code type

⑱. Camera control code (code for calling preset bits)

⑲. High definition input source (up to 4 high-definition signal sources from cameras connected to corresponding HDMI interfaces)

⑳. Control signal output status.



The camera page can call, save, and clear preset points according to your needs.

Communication methods: UDP/TCP/232, etc. Communication through UDP and TCP requires the IP address and port number of the camera, as well as the protocol and camera model. If communicating through the 232 serial port, please pay attention to adjusting the baud rate of the processor's 232 serial port to be consistent with the baud rate of the camera, and the connection method should correspond to the port on the back board connected to the gimbal;

The last preset point number is the identification defined by the user for the camera, and then adjust the parameters such as up, down, left, right, focal length, aperture, etc

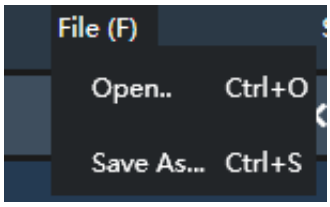
The number will define the position and settings of the camera;

Finally, click "Save" to save the parameters, "Clear" to delete the information of the current pre made point, and "Call" to use

View the camera position saved at the current preset point.

3.5 Setting Menu

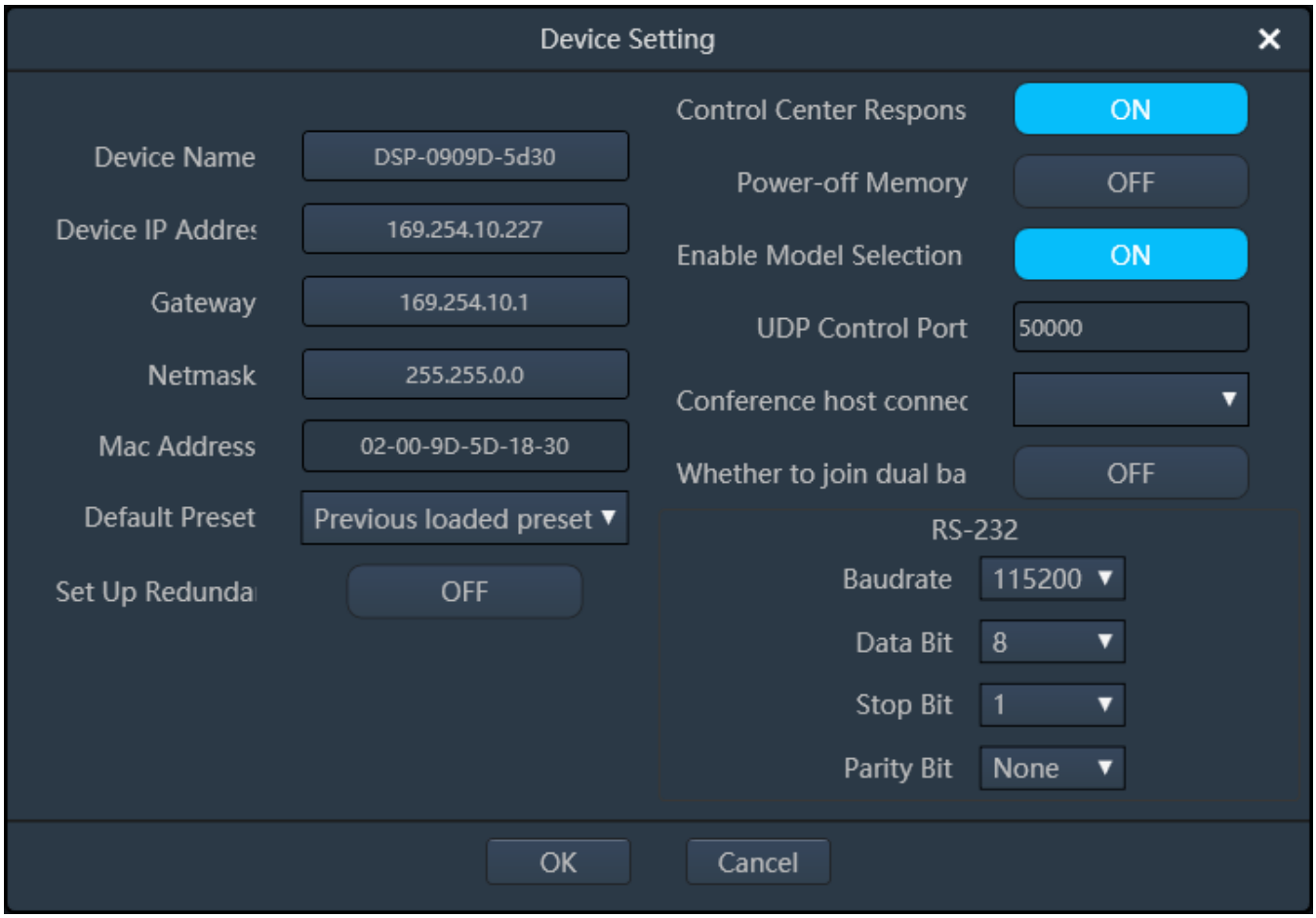
3.5.1 File Menu



In offline mode, click to open the pop-up file dialog box and select to open an existing preset file (suffix: *. xxxdsp). You can also right-click on the preset file and open it using the DSP.exe application.

"Save As " saves the preset settings on the application to the local hard drive for easy copying and storage.

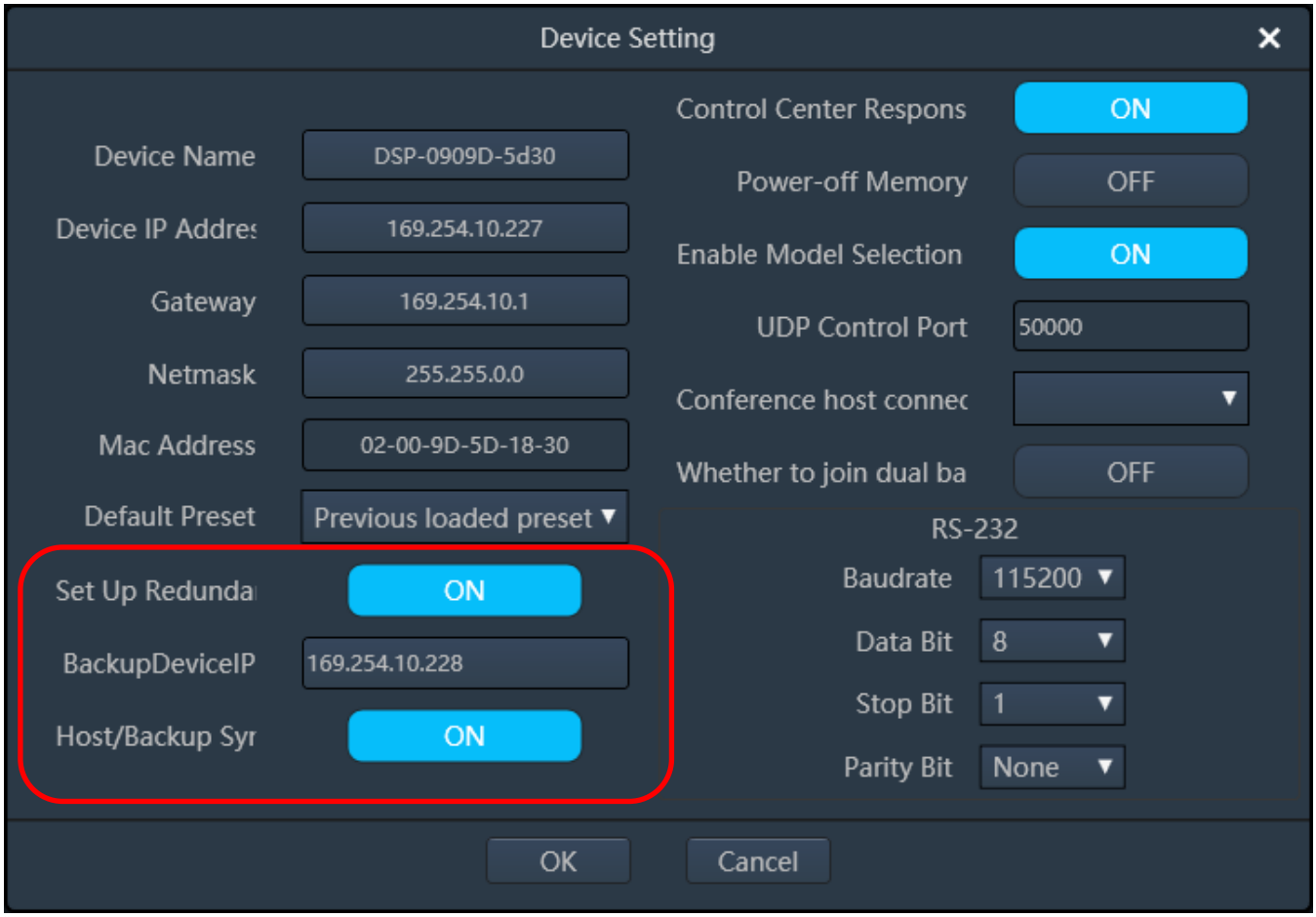
3.5.2 Device Setting

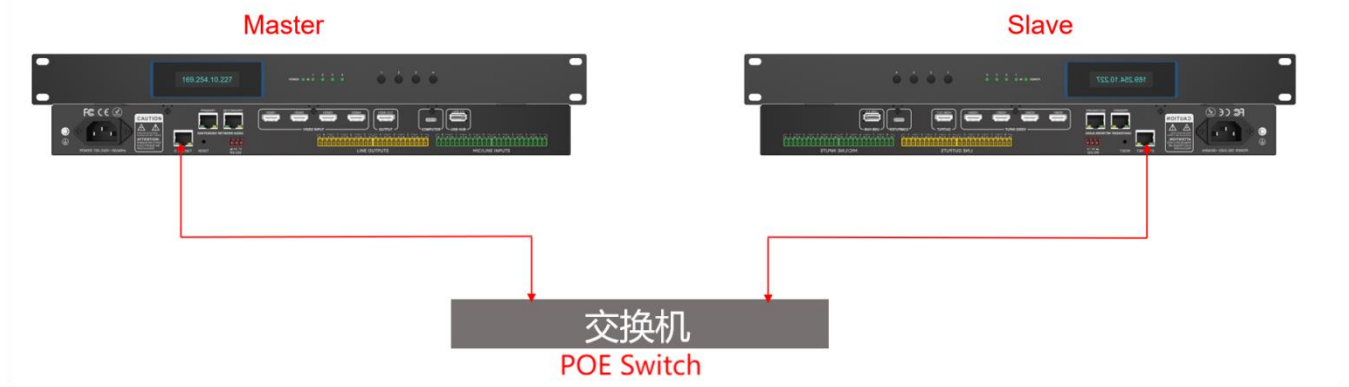


Set device name, network address, dual machine hot backup, and serial port baud rate information. The maximum length of the device name is 16 characters and 5 Chinese characters.

Default Startup: Two startup preset modes can be selected. One is to specify any one of the 16 presets as the startup preset, which will be used every time the device is turned on. The second option is to select the preset that was last loaded and used before power failure as the preset for the next boot up.

Dual machine hot backup:

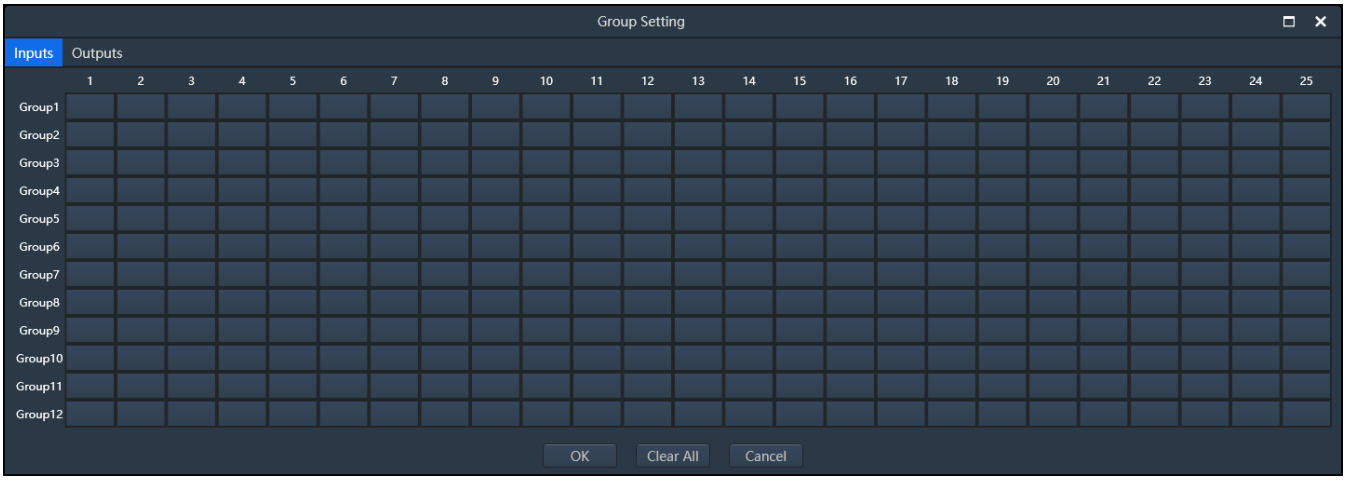




Click on 'Set as Host' and enter the IP address of the backup device to complete the setup .

3.5.3 Group Setting

The grouping interface is divided into two tags: input and output, and grouping can be set under each tag. A channel can only participate in one group. Under the same group, their channel volume and mute adjustments are synchronized. The biggest difference from Link functionality is that other module parameters are not synchronized.



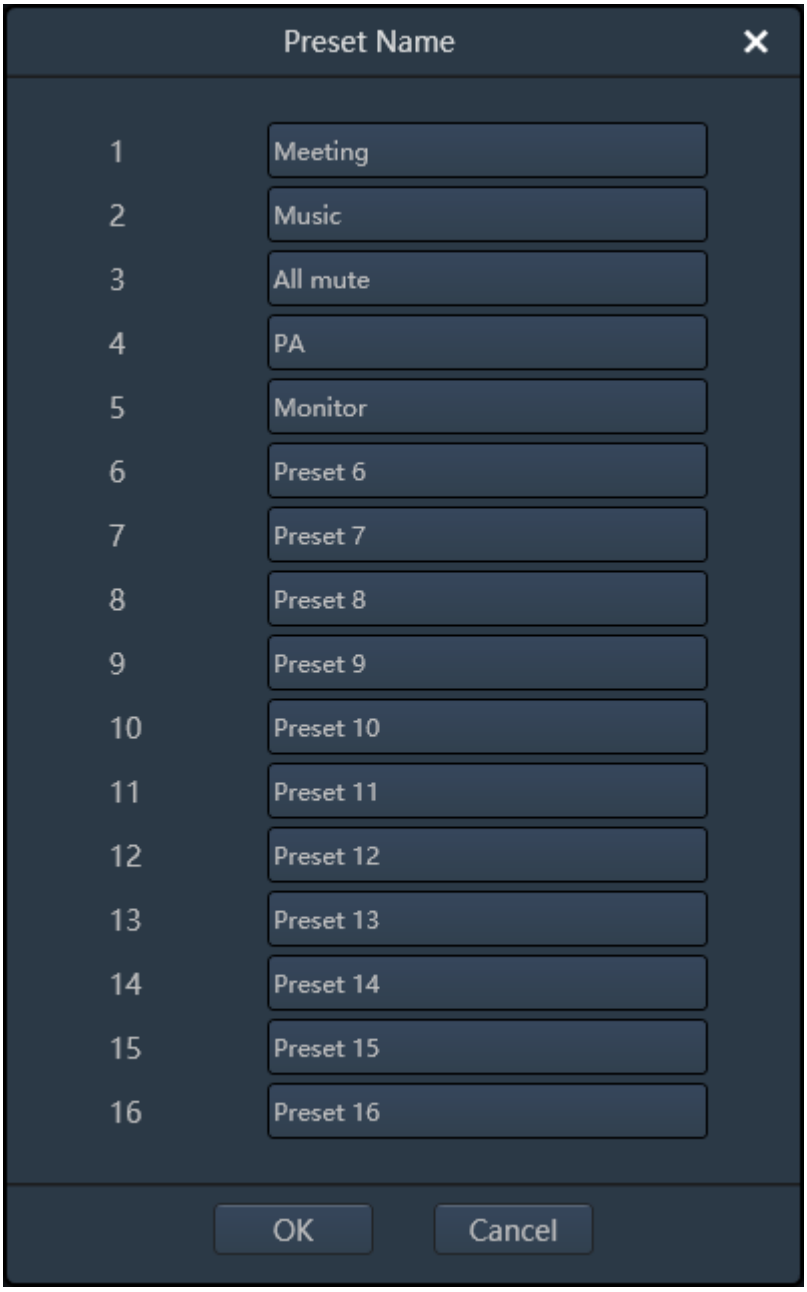
Each group can choose 1- the maximum number of channels for the device. The maximum number of channels for the device depends on the model you purchased. The channel is set as a group, which will be distinguished by a color on the main interface.



The relationship between grouping and LINK: When a channel is grouped, it will not participate in LINK. This means that the priority of the group is higher than LINK. The difference between grouping and LINK is that grouping can only control channel gain and mute, while LINK links all parameters on that channel.

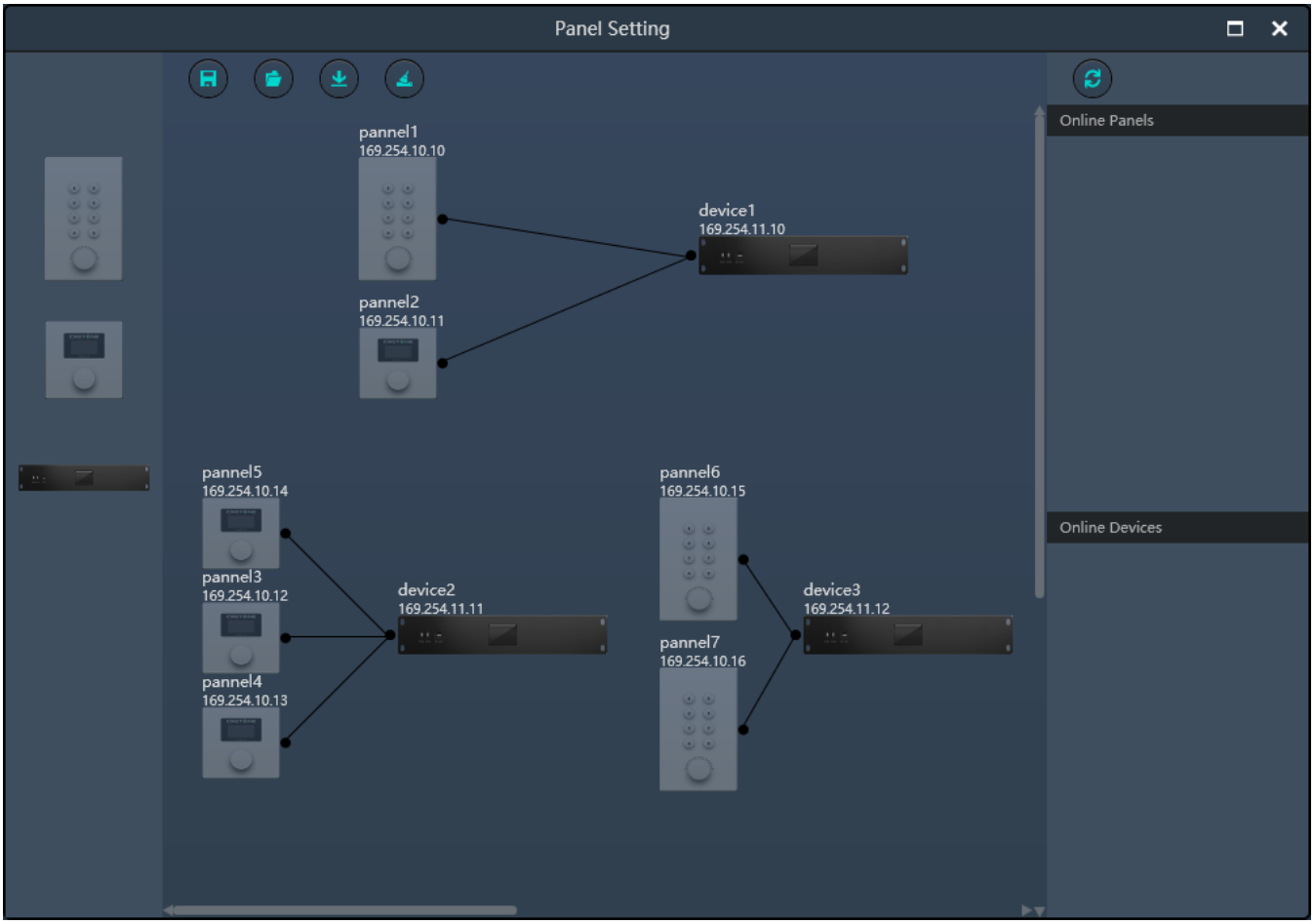
3.5.4 Preset Name

Support name editing for 16 preset settings for easy scene retrieval.



3.5.5 Panel Setting

The panel settings include two panel types: button version and OLED screen version. Connect multiple physical panels to DSP devices through the panel settings interface, and then achieve panel control of DSP devices through simple panel settings.



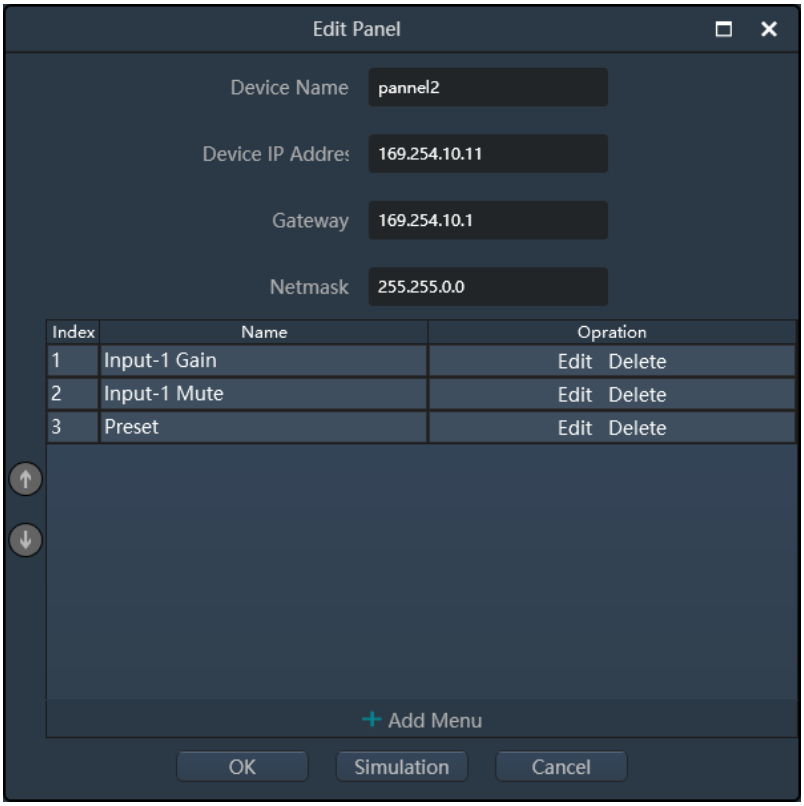
Offline device: suitable for offline editing status. The debugging engineer first configures the panel parameters locally and then downloads them to the online panel. Of course, you can also directly edit the online panel by dragging it out from the online panel column to the panel design area and double clicking to edit.

Note that there is a small circle on both the panel and the device. Click on the circle to draw a line and select the destination device, thus establishing a connection between the two devices.

Double click the panel in the design area to enter the panel configuration interface. The configurations of the two panels will be described below. After the configuration is completed, click the download icon on the toolbar to download the panel configuration to the hardware.

OLED Panel:

The OLED panel consists of a 1.3-inch OLED screen and a knob. The OLED screen display strategy is graded according to the menu, with three types of menus: volume, buttons, and presets. In the panel design area, double-click an OLED panel to enter its detailed settings.



Click on 'Add Menu', a menu selection box will pop up, select the corresponding menu item, and confirm. After completing the software menu configuration, click the download icon on the toolbar to download the configuration to the panel hardware.

Panel operation steps:

On the main interface, the displayed panel name and IP address can be switched between menus by turning the knob left or right.

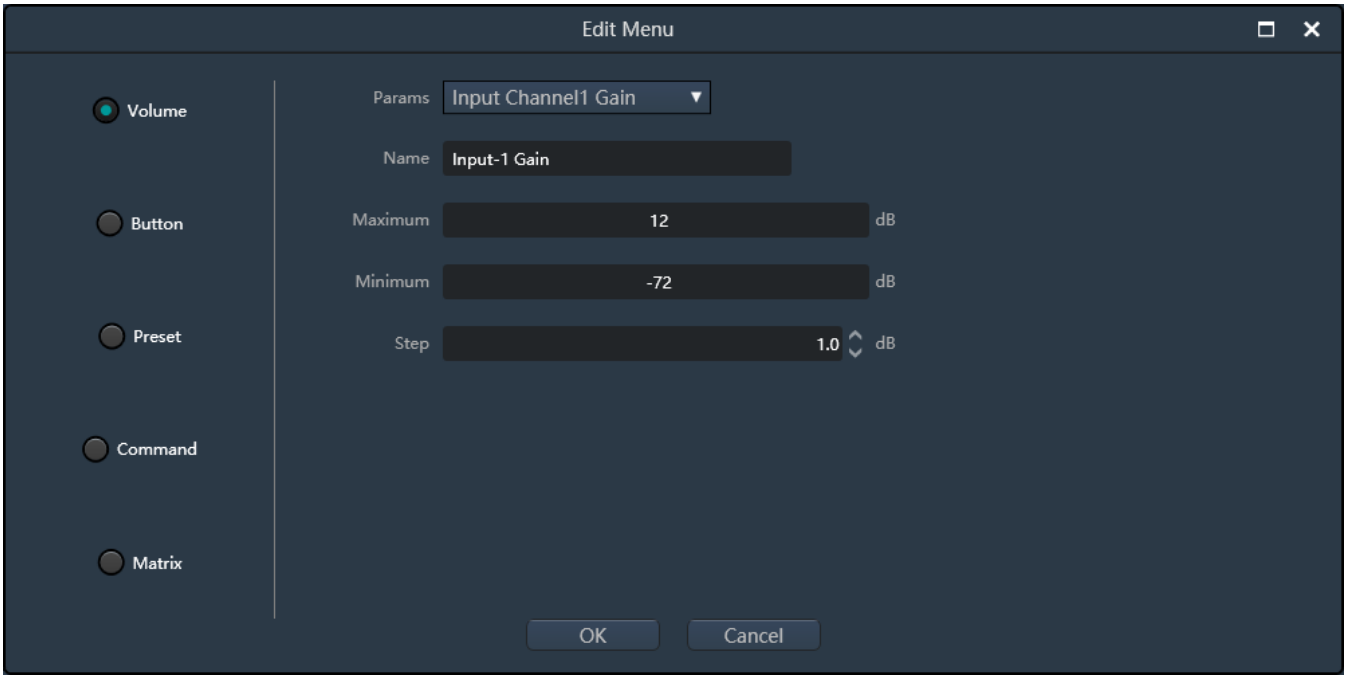
2. Press the button on the knob, and the second line of the menu interface will start flashing, indicating that you have entered editing mode.

3. Use the knob to turn left or right to change the value.

4. Press the button on the knob again to exit editing mode and return to menu mode.

Key panel:

The button panel has a total of 8 buttons and one knob. The knob is used for gain adjustment, and 8 buttons can be programmed to achieve different functions. The button functions can be divided into four types: volume adjustment, mute, preset, and command. Dragging an item on the specified key in the ribbon completes the programming of that key.



Panel operation light indication:

The light constantly on: indicates that the button is configured with a mute function.

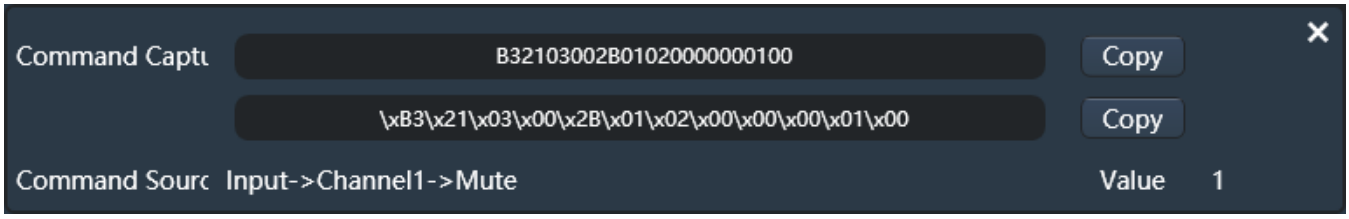
The light flashing: indicates that the button is configured with a gain function, and the configuration knob can be used together to adjust the gain of the channel. The 13 lights around the knob represent the gain, which changes on and off according to the gain size. All off represents a gain of -72 dB, and all on represents 12 dB.

Instantly lit: The button light illuminates instantly when pressed, indicating that the button is configured for preset or command functions.

Command function: Command data comes from central control commands. Refer to Section 5.

3.5.6 Help Menu

（1）Control command



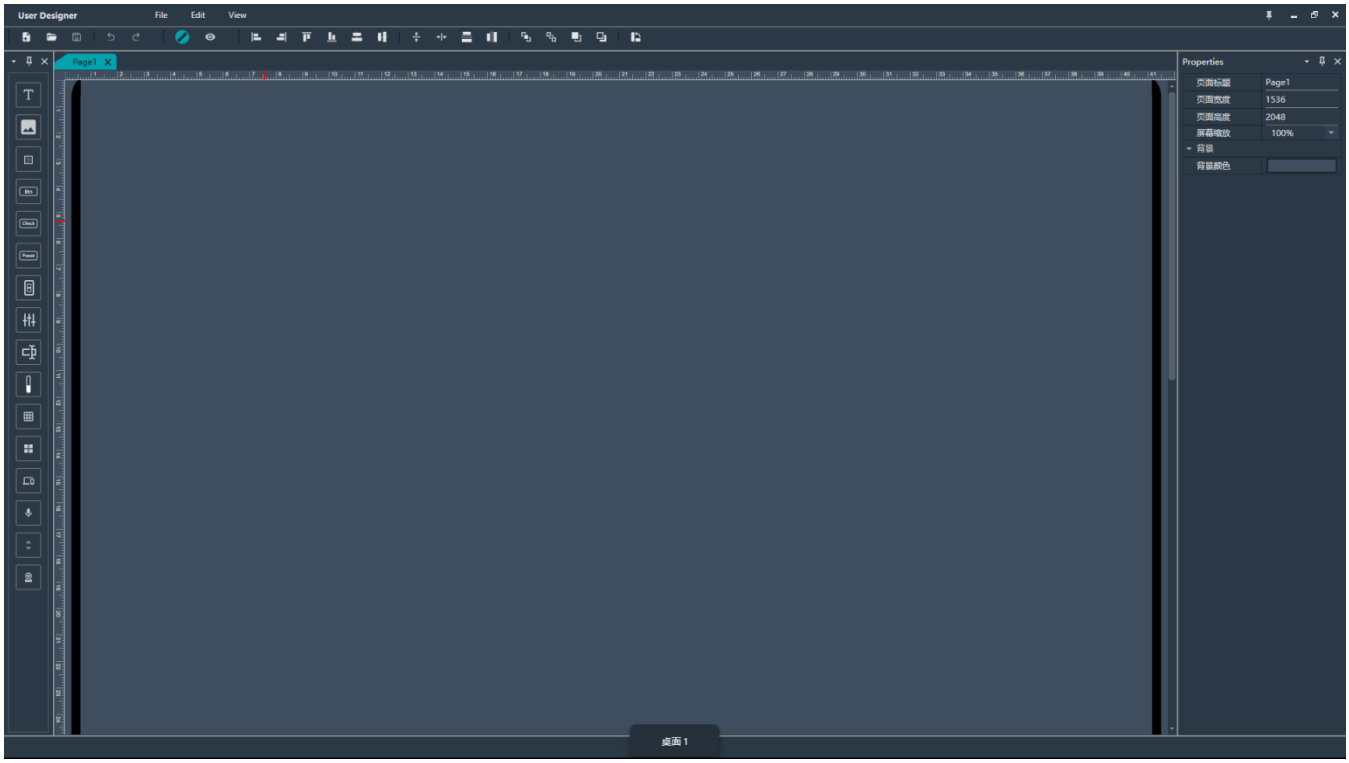
Open the central control command window, click on the parameter to be controlled on the interface, and the central control command window will instantly display the current command. Copy the command and send it to the device using UDP or RS232.

(2) About

Regarding device version number, technical support contact information, copyright information, etc.

3.5.7 User Interface

User interface, this feature allows engineers to create custom interfaces that can be edited by integrators and operated by on-site technicians or end users who do not understand technology. Advanced security features allow end users to access only the controls allowed by the engineering vendor or system designer.

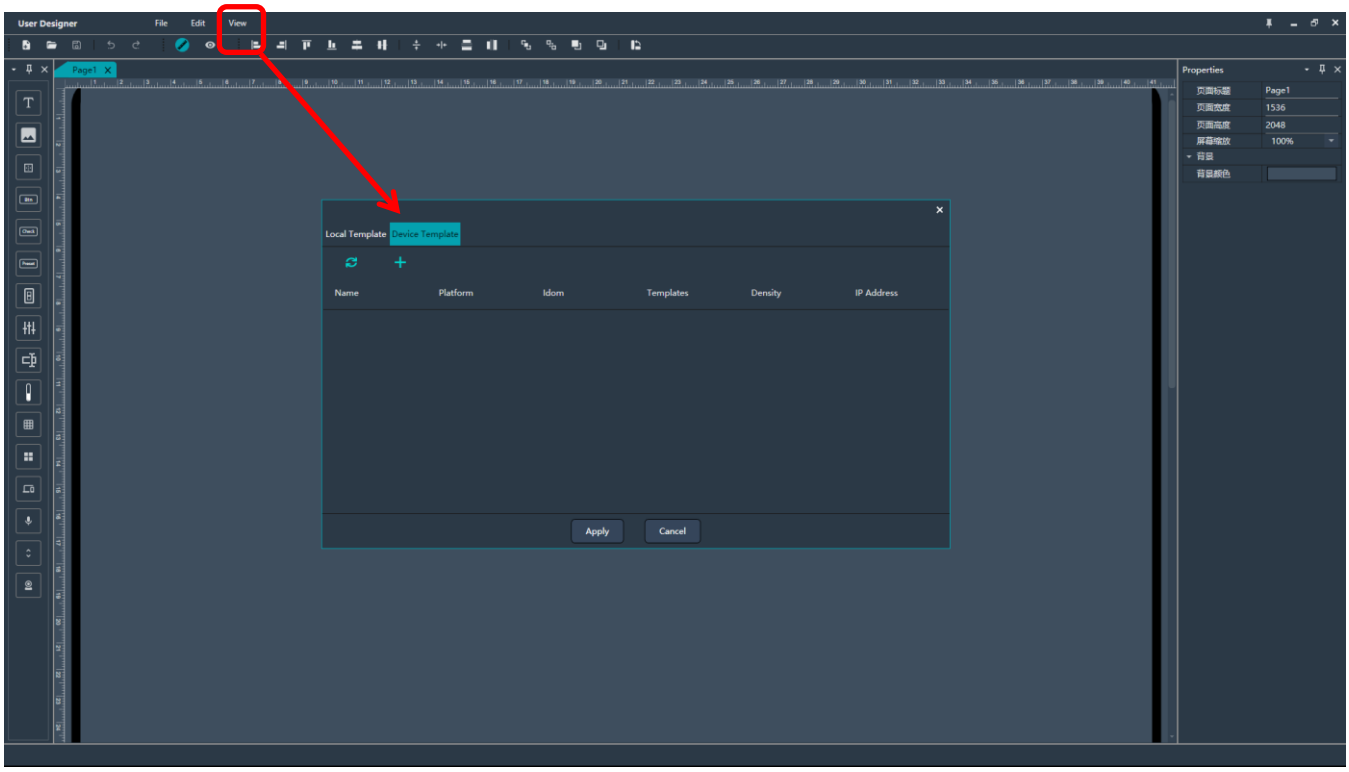


3.5.8 Mobile device usage

Online template selection

After installing the software on the mobile end, be careful not to turn off the screen and keep the mobile and PC ends on the same LAN.

Please inquire with the device supplier for the download address of the mobile software.

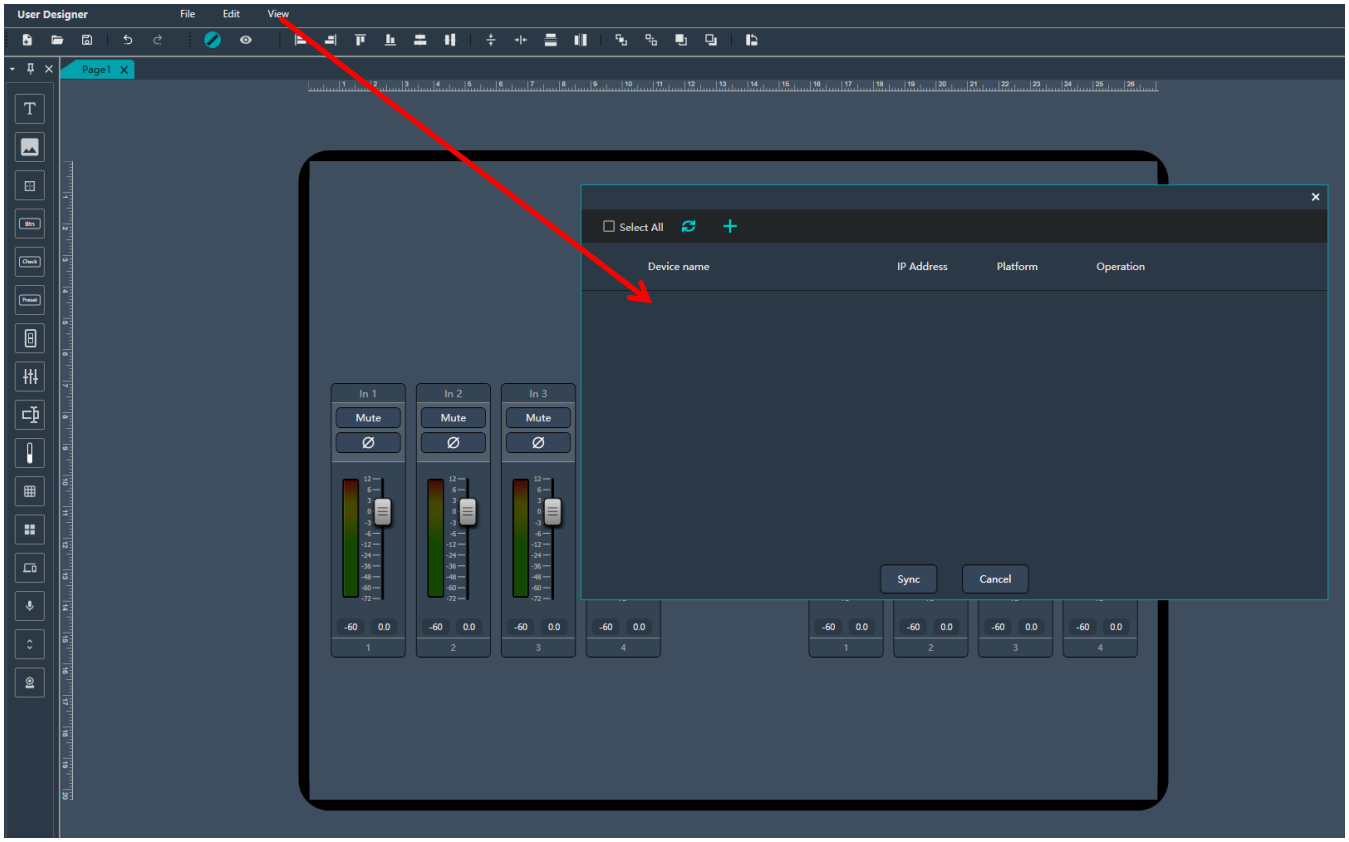


Edit interface

Drag the left control to change the corresponding properties



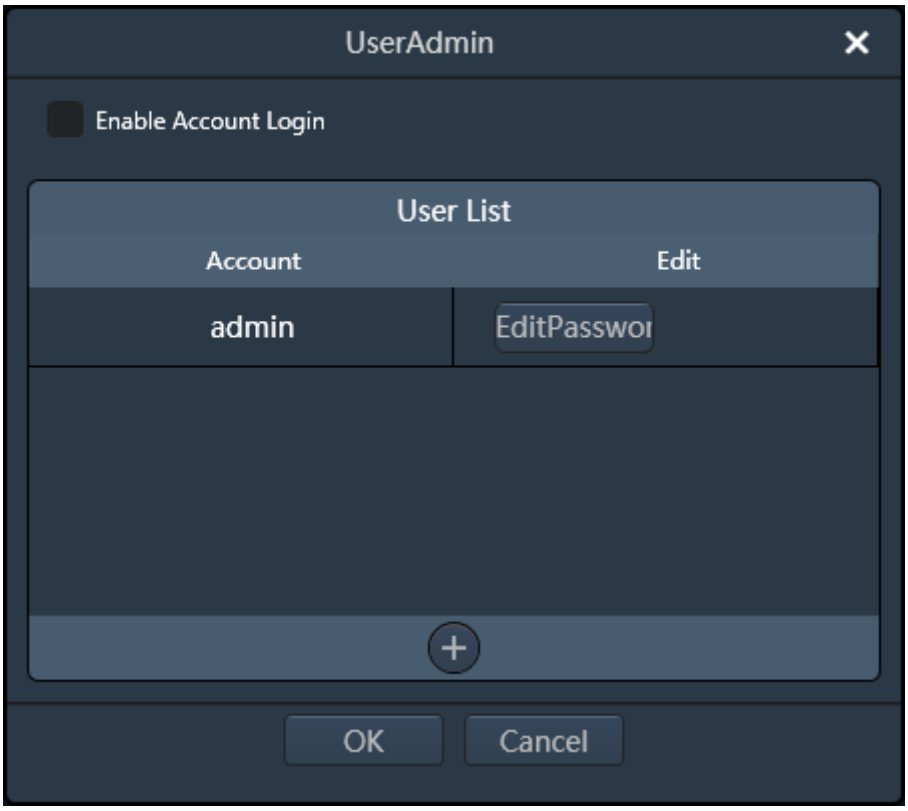
Upload to mobile devices (synchronize data)



Click to synchronize data and upload it to the mobile device. If you encounter upload timeout or cannot find the

mobile device, you can close the software and reopen it.

3.5.9 User Management



Log in with an account - click the "+" sign to add a new user (secondary account), and there is only one main account that can be added or deleted as desired;

After enabling account login, accessing the device requires entering the corresponding username and password.

After enabling user management login, the software can be locked.

Important: When creating a new user or setting a password, please remember the password you have set

4 Control

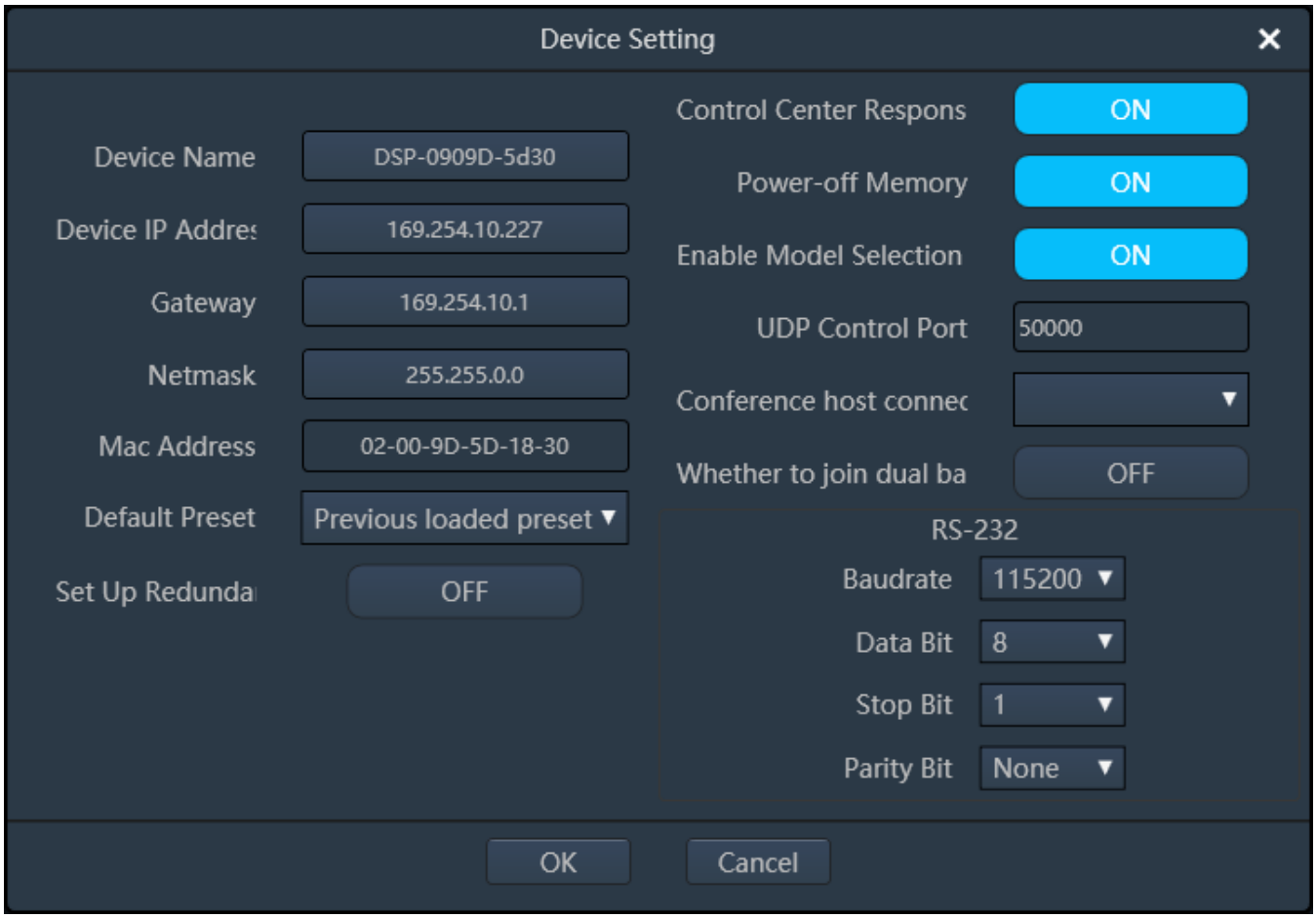
4.1 External Control Programer

External control programming supports both UDP and RS232, and the control protocol covers all control parameters of the processor, including parameter control, parameter acquisition, and preset calling.

For UDP control, the default port is 50000, which can be set in the "Device Settings" section of the upper computer software.

Using RS232 control, the default baud rate is 115200, data bit 8, stop bit 1, and no checksum. It can also be set in the "Device Settings" section. **When sending via RS232, it is recommended to maintain an interval of at least 150 milliseconds between messages.**

If the control device needs to return a command, please turn on the central control reply switch in "Device Settings".



4.2 ASCII Control Command

Preface:

Note:

1、 Since the starting position of the channel is 0, it can be understood that channels 0-3 correspond to both IN1-4 software display channels, and 0-3 is only an example of the actual number of channels based on the device model

2、 In the function on/off settings, 1 is on and 0 is off; For example, the last 1 of set: output # mute # 0-3 # 1 represents "mute enabled".

4.2.1 Input volume command control and get

set:input#gain#0-3#1

(Setting: Input # gain # channel number # to 1dB)

get:input#gain#0-3 --> get:input#gain#0-3#1#1#1#1

(Get: Input # gain # channel number) Return information Example: get: input # gain # 0-3 # 1 # 1 # 1 # 1 (Input 1-4 channels volume is 1/1/1/1 in sequence)

4.2.2 Output volume command control and get

set:output#gain#0-3#1

get:output#gain#0-3 --> get:output#gain#0-3#1#1#1#1

4.2.3 Phantom power command control and get

set:input#phant#0-3#1

get:input#phant#0-3 --> get:input#phant#0-3#1#1#1#1

4.2.4 Input mute command control and get

set:input#mute#0-3#1

get:input#mute#0-3 --> get:input#mute#0-3#1#1#1#1

4.2.5 Output mute command control and get

set:output#mute#0-3#1

get:output#mute#0-3 -> get:output#mute#0-3#1#1#1#1

4.2.6 Sensitivity command control and get

set:input#sens#0-3#1**(Representing 3db, representing the second gear)**

get:input#sens#0-3 -> get:input#sens#0-3#1#1#1#1

4.2.7 Matrix control acquisition: multiple inputs control a single output, one input controls multiple outputs

set:mixer#switch#0#0-3#1（Set input 1 to output 1 to 4 with routing enabled）

set:mixer#switch#0-3#0#1（Set input 1 to 4 to output 1 with routing enabled）

set:mixer#gain#0-3#0#1（Set input 1 to 4 to output 1 with a routing gain of 1db）

get:mixer#switch#0-3#0(0-3:input,0:output) -> get:mixer#switch#0-3#0#1#0#1#1

4.2.8 Scene recalling and saving

scene:toggle#3(Scene recall, PC displays as Scene 4)

scene:save#3 (save scene)

4.2.9 Input level meter command get

get:input#level#0-3（Channel Start - Channel End） -> get:input#level#0-3（Channel Start - Channel End） #-105.4#-102.5#-105.2#-104.8(dbfs)

4.2.10 Output level meter command get

get:output#level#0-3（Channel Start - Channel End） -> get:output#level#0-3（Channel Start - Channel End）#-56.0#-40.8#-43.6#-46.4

4.2.11 System mute command control and get

set:sysctl#mute#1(system mute enabled)

get:sysctl#mute -> get:sysctl#mute#1

4.2.12 Set and retrieve channel names for input and output

set:input#name#0#1

get:input#name#0-3 -> get:input#name#0-3#IN1#IN2#IN3#IN4

4.2.13 Control and get of input-output inversion

set:input#phase#0-3#1

set:output#phase#0-3#1

get:input#phase#0-3 -> get:input#phase#0-3#1#1#1#1

get:output#phase#0-3 -> get:output#phase#0-3#1#1#1#1

4.2.14 Control and get of input-output step size

set:input#step#0-3#10

set:output#step#0-3#10

4.2.15 Input/output link control and get

set:input#link#0-3#1

set:output#link#0-3#1

get:input#link#0-3 -> get:input#link#0-3#1#1#1#1

get:output#link#0-3 -> get:output#link#0-3#1#1#1#1

4.2.16 Control and get of signal generator

set:input#type#0-3#1

get:input#type#0-3 -> get:input#type#0-3#1#1#1#1

4.2.17 Restore factory settings control

set:refactory

4.2.18 Scene reset control

set:rescene

4.2.19 Get any scene name set

set:scene#name#0-3#pre1(PC only supports UTF-8 encoding, sending Chinese ASCII code will result in garbled characters)

get:scene#name#0-3 (Scene number:0-15) -> get:scene#name#0-3#pre1#pre1#pre1#pre1

4.2.20 module name:input,output|mixer

item 名:

(input)mute,gain,sens,phant,type,freq,name,phase,step,link,level

(output)mute,gain,name,step,link,level

(mixer)switch,gain

(scene)toggle,save,name

(sysctl)mute

(rescene)

(refactory)

4.2.21 Set instruction format：

Set: Module Name # Item Name # Start Channel - End Channel # Parameter Value

for example

set:input#mute#0-3#0/1

4.2.22 Get Instruction Format

Get: module name # item name # start channel - end channel

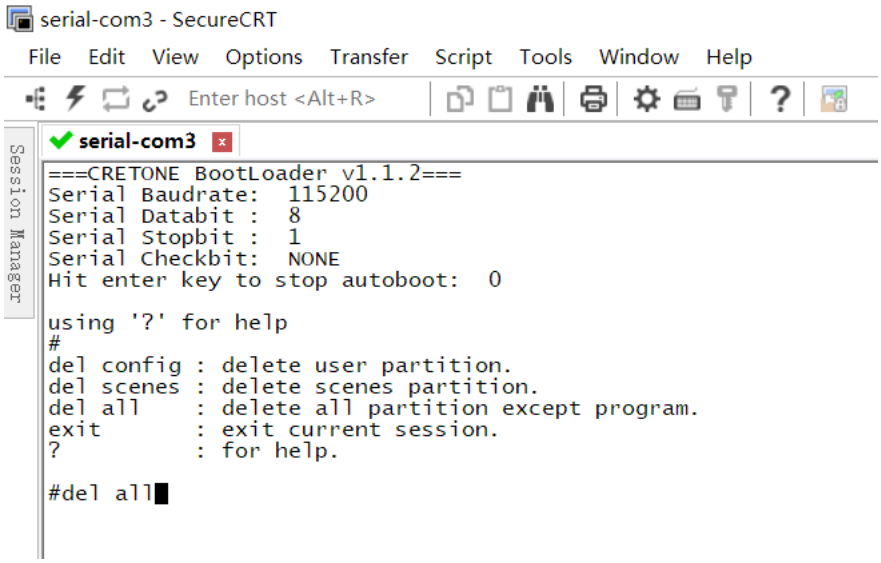
for example

get:input#mute#0-3

5 Frequently Asked Question （FAQ）

1. How to restore factory settings?

Connect to the computer via RS232 and run serial software (recommended to use SecureCRT). The default baud rate of the serial port is 115200, with 8 data bits, no parity check, and 1 stop bit. After connecting the SecureCRT to the serial port, long press the enter key on the terminal interface to restart the machine and enter the bootloader boot dialog box.



Command Explanation:

Del config: Delete configuration information, such as IP addresses and other network configurations. After deletion, the device will be restored to its default IP: 169.254.227.

Del sects: Delete presets. All 16 presets of the DSP device have been restored to their default values.

Del all: Delete all partitions except for the program.

Note: Some SecureCRT installations may not display a response. Please check "Local echo" in Options ->Session Options.

