PRESTEL AP 3.0

Audio Processor Software Manual

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1 Overview

This software is an audio system control software, mainly for the open architecture audio processor series of products configuration, device programming, controlling and so on. The interface is simple and intuitive, which makes it easy for users to set and recall various parameters and presets. The graphical control software is one of the cores of this system. Configuration with signal chain via software installed on the computer is clear and understandable.

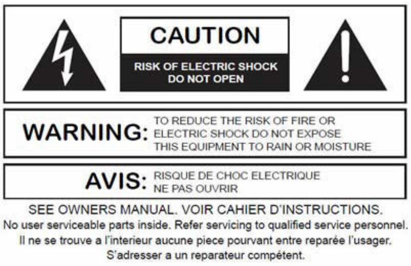
When using this software, the user can complete the entire audio system setup. Through a simple mouse operation, module parameters and presets can be stored, making it easy for users to get started. At the same time, the user interface software also supports wireless touch screen control, providing more convenient wireless control. The manual has also been written to provide detailed operating instructions to help users better understand the functions and uses of the software, thereby improving your satisfaction and experience.

2 Hardware

2.1 Safety

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.

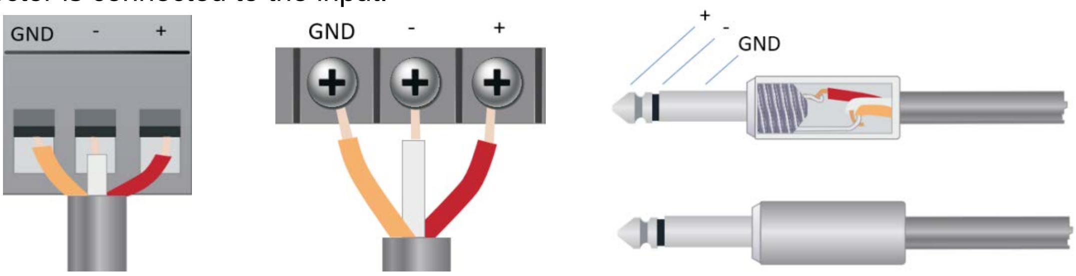
12. Only use hand carts, tripods, stands, or tables designated by the manufacturer or sold together with the equipment and instruments. When using a handcart, be careful when moving the handcart/device combination to avoid injury due to tipping over.
13. During thunderstorms or when not in use for a 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.

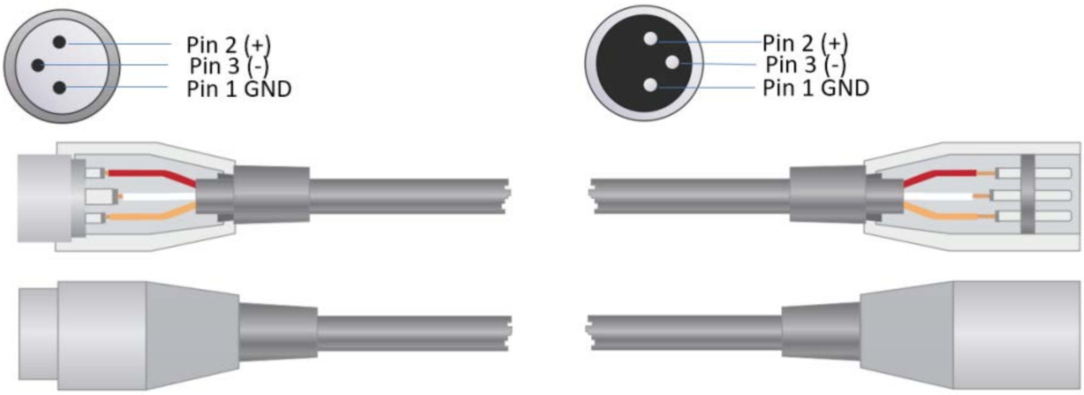


The lightning symbol with an arrow symbol inside an equilateral triangle is designed to make users aware of the uninsulated "dangerous voltage" inside the product casing, which can cause electric shock to the human body. The exclamation mark inside the equilateral triangle is intended to make users aware 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

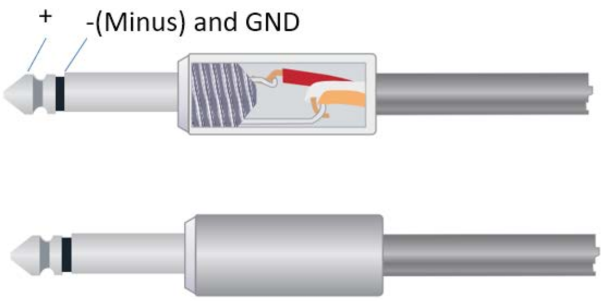
Balanced connection
Any of these interfaces may appear on both sides of the balanced connection.
Note: For an XLR interface, the female connector is connected to the output, while the male connector is connected to the input.

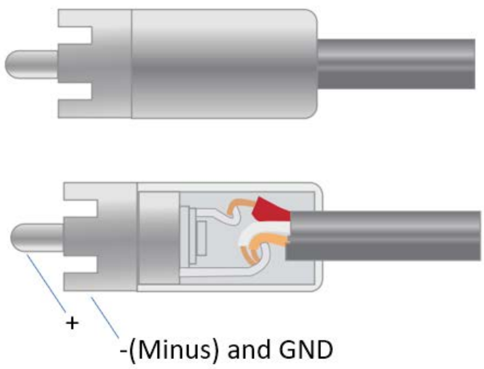




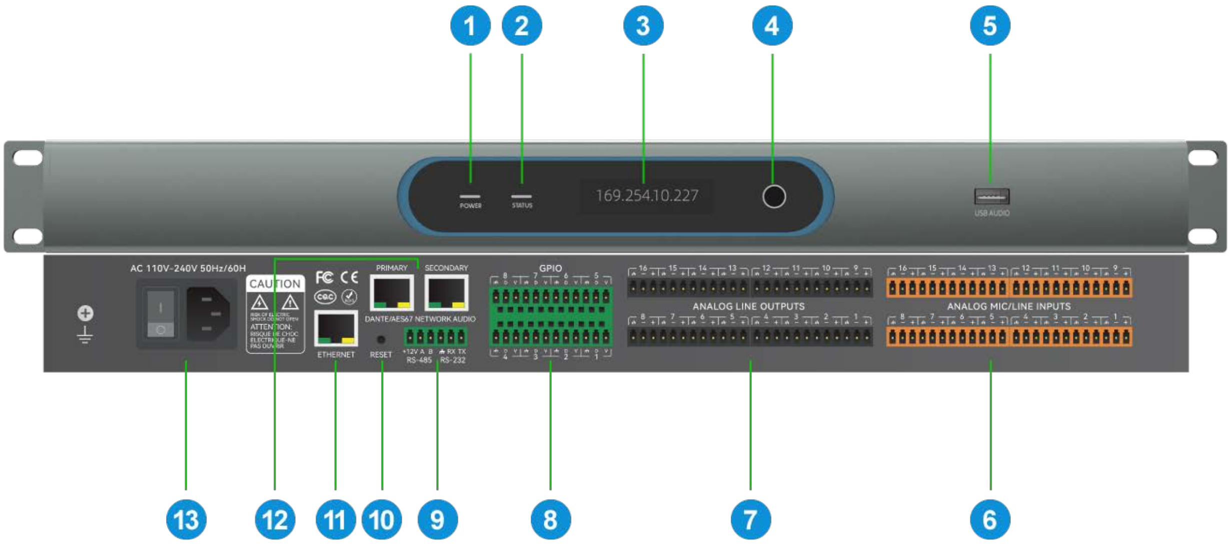
Unbalanced connection

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 Interface



1 - Power Indicator: constant green light for normal.

2 - Operation Status Indicator: constant green light for normal.

3 - OLED Display: display device status information, IP address, overview information, etc.

4 - Wake Up / Navigation Button: this transient button can switch the system dashboard display.

5 -USB-A sound card: provide 2x2 audio, used to connect to the PC for remote meetings or for recording and playback.

6 - Analog Mic / line Input: Up to16-channel balanced analog audio inputs, independent preamp, phantom power.

7 - Analog Line Output: Up to 16-channel balanced analog outputs, you can use the software to independently control the level and the mute.

8 - GPIO/Logic: Used to connect the control terminal or central control equipment.

9-RS-232: Used for the third-party device to control this equipment or for the equipment to control the third-party device.

10 - RESET: Restore factory settings, device data, default IP169.254.10.227.

11 - Ethernet Port: used to connect the software for programming, management and control of equipment

12 - Dante Ports: redundant 1000 Base-T Ethernet port, providing 128 (64x64) channels of Dante network audio channels

13- Power Supply: detachable IEC cable power supply (100-240VAC, 50-60Hz, 60W maximum)

3 Operation Environment

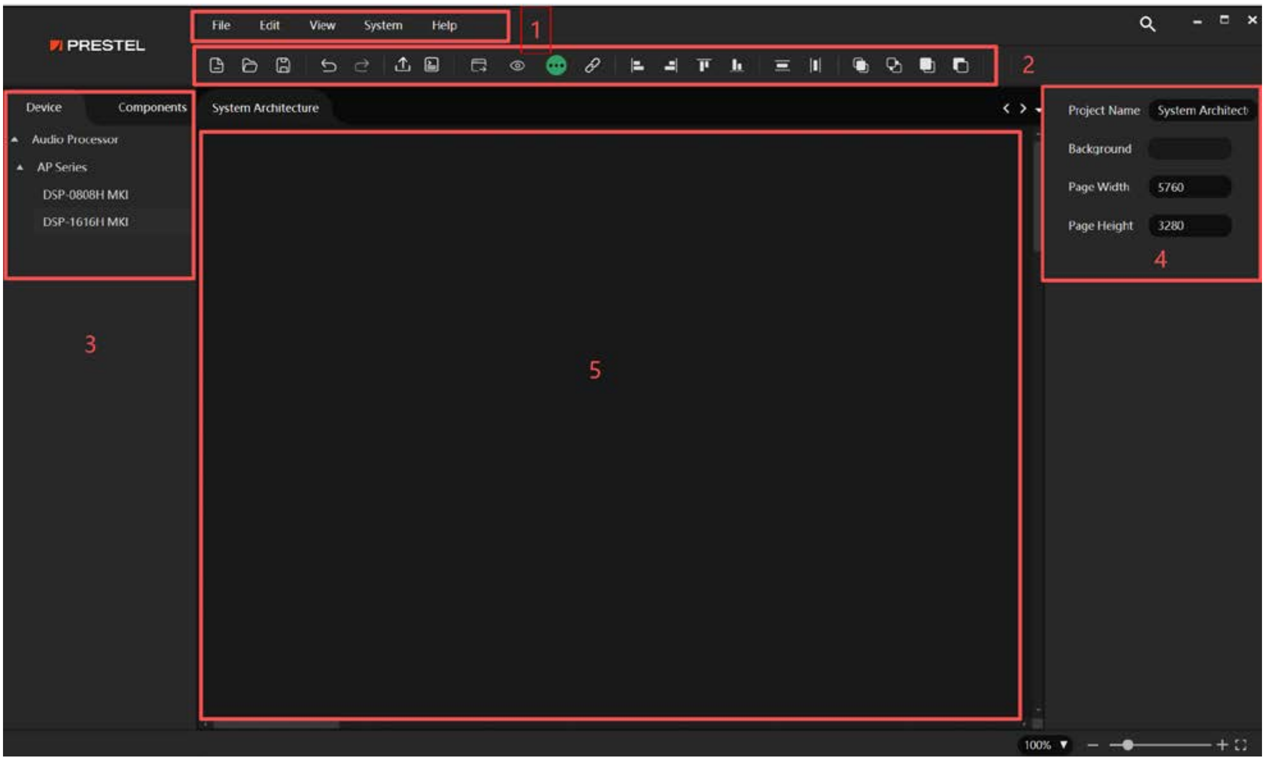
3.1 Hardware Requirement

- Intel Celeron Processor J1800 and higher
- 4G or more of RAM;
- 64G or more of ROM.

3.2 Software Requirement

- Windows 7 or higher;
- Install .NET Framework and higher.

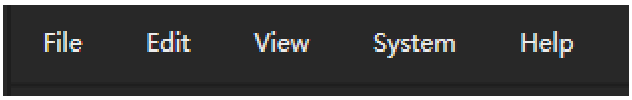
4 Software Overview



After opening the software, you can find out the screen is composed of 5 parts:

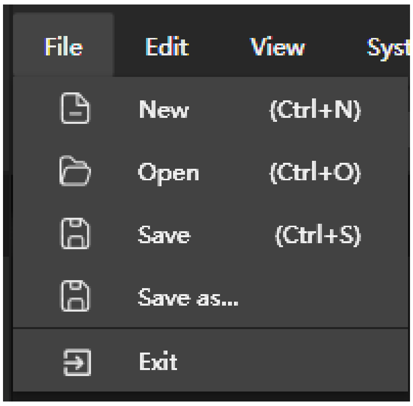
1 Menu Bar; 2 Tool Bar; 3 Device/Components Window; 4 Properties Window; 5 Design Area.

4.1 Menu Bar



There are 5 menus in the menu bar, “File”; “Edit”; “View”; “System”; “Help”.

(1) File



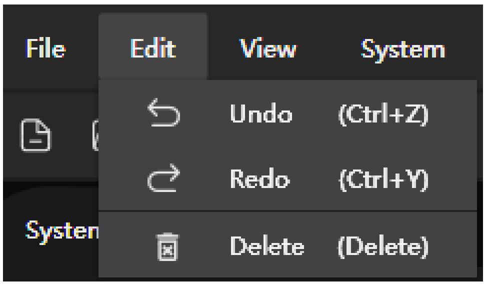
New: Clicking on it creates a new project and prompts if the current unsaved project needs to be saved;

Open: Click to select a project from the path to open；

Save: Click to save the project to the default path;

Save as...: Click to save the project to the specified path.

(2) Edit

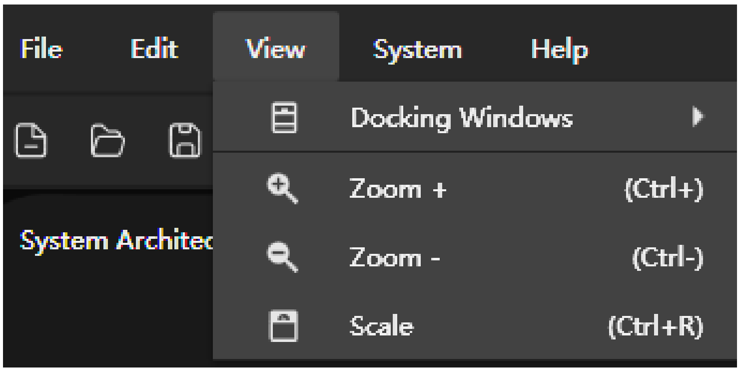


Undo: To undo the previous step; use the hot key ”Ctrl+Z”;

Redo: Reverses the previous Undo operation; use the hot key ”Ctrl+Y”;

Delete: Deletes the selected components; use the hot key ”Delete”.

(3) View



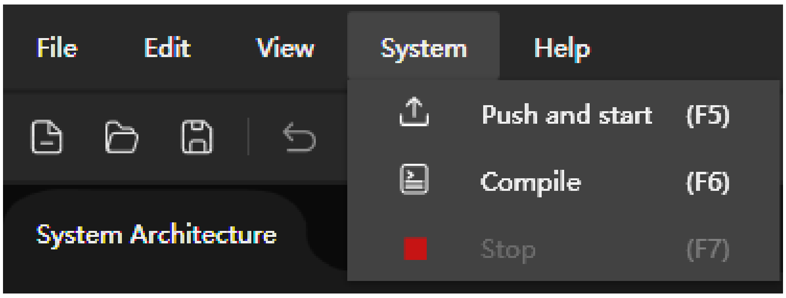
Docking Windows：Select the docking window that need to be displayed in the screen;

Zoom +：Zoom in the design area (up to 400%) with the hot key “Ctrl” & “+” or scroll up mouse wheel with “Ctrl”;

Zoom -：Zoom out the design area (min. 50%) with the hot key “Ctrl” & “-” or scroll down mouse wheel with “Ctrl”;

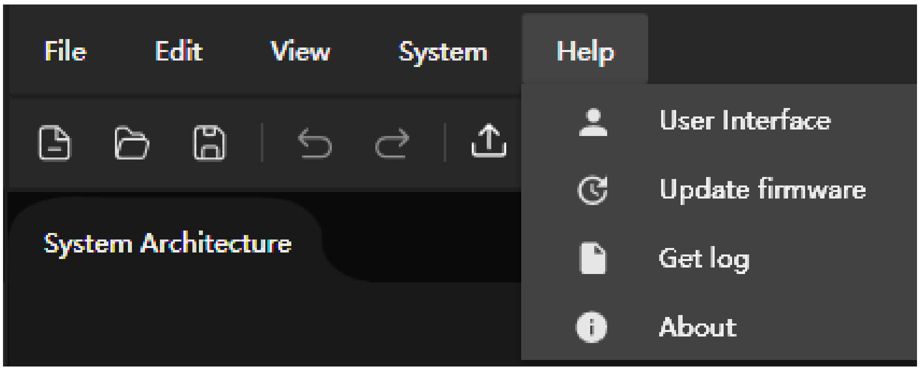
Scale：Return the normal size of design area view (100%) ; hot key “Ctrl +R”.

(4) System



Push and start: Upload the current designed program to the device and run in online edit mode; hot key “F5”;
Compile: Compile the current designed program and entering the offline edit mode; hot key “F6”;
Stop: Disconnect with the device from online edit mode or quit compiling from offline edit mode; hot key “F7”.

(5) Help



User Interface: Customized user interface with control programming inside;
Get log: Select the corresponding device on Project page for getting the log records of the device;
About: Displays software information; version.

4.2 Tool Bar

(1) New project: Clicking on it creates a new project and prompts if the current unsaved project needs to be saved;

(2) Open project: Click to select a project from the path to open；

(3) Save project: Click to save the project to the default path;

(4) Undo: To undo the previous step; use the hot key ”Ctrl+Z”;

(5) Redo： Reverses the previous Undo operation; use the hot key ”Ctrl+Y”;

(6) Push and start: Upload the current designed program to the device and run in online edit mode; hot key “F5”;

(7) Compile: Compile the current designed program and entering the offline edit mode; hot key “F6”;

(8) Output Window: Display the program output window on the screen;

(9) Aerial View Window： Display the aerial view window on the screen;

(10) Signal Path: When turned on; selecting a wire will display the signal path that is associated with it from the beginning to end;

(11) Auto Connect: Selecting multiple modules and clicking on Auto-Connect will automatically connect the channels one by one;

(12) Align Left: Align the selected modules to the left;

(13) Align Right: Align the selected modules to the right;

(14) Align Top: Align the selected modules to the top;

(15) Align Bottom: Align the selected modules to the bottom;

(16) Space Evenly Vertically: Make selected modules vertical spacing equal;

(17) Space Evenly Horizontally: Make selected modules horizontal spacing equal;

(18) Bring Forward： Move the selected module up one layer;

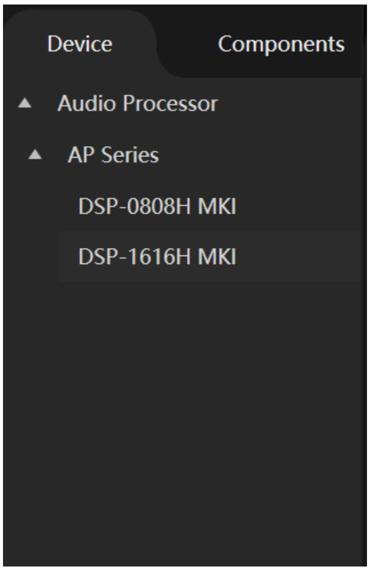
(19) Send Backward： Move the selected module down one layer;

(20) Bring to Front: Move the selected module to the top layer;

(21) Send to Back: Move the selected module to the bottom layer;

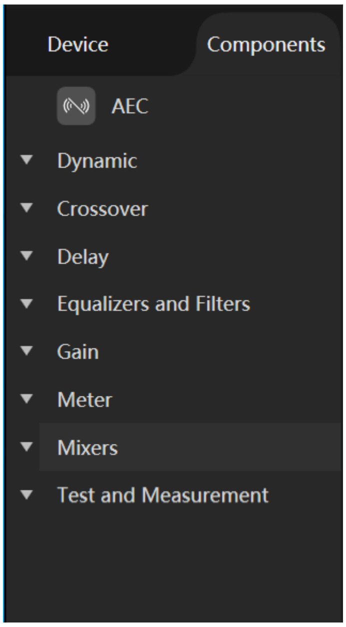
(22) Text: Adding text that can be used for notes.

4.3 Device Window



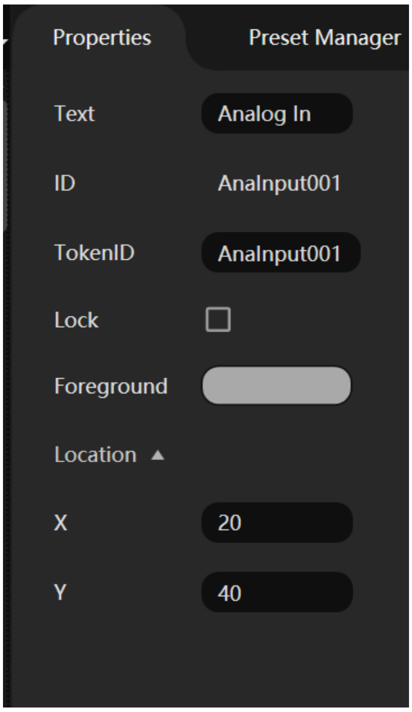
All the available models of devices will be shown in the Device column. When connecting a device or programming; you need to drag the corresponding model of the device to the design area.

4.4 Components Window



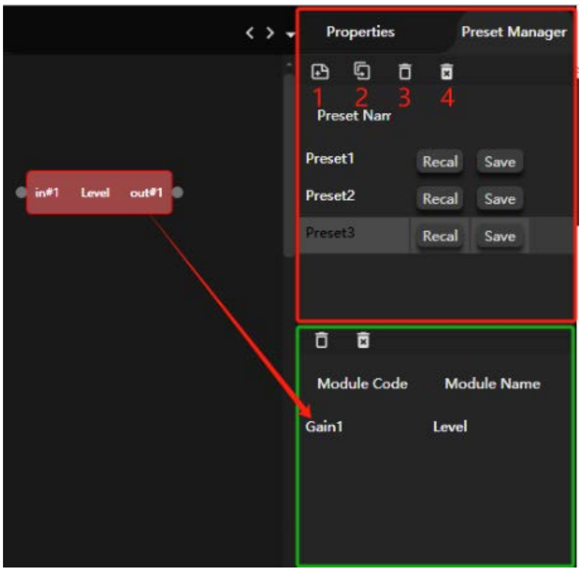
The Components column will show all the DSP processing modules supported by the unit. We can choose according to the actual scenario and the use of demand; drag and drop the module into the design area to perform editing and configuration of signal processing flow.

4.5 Properties Window



Attributes of the module can be edited in the Properties field after selecting one. Such as changing the module's global setting; ID; text color; position. And locking the module from being moved.

4.6 Preset Manager Window



(1) Adding new preset

(2) Copy selected preset

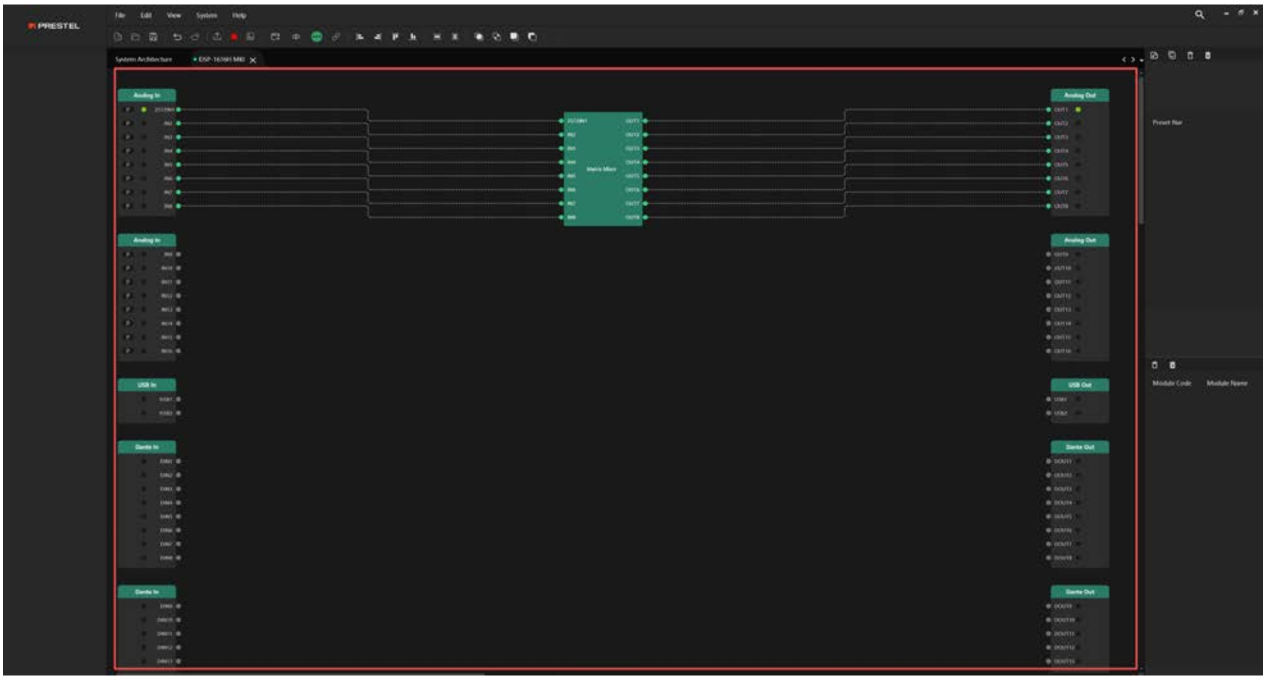
(3) Delete selected preset

(4) Clear all presets

The red area is the preset management area where preset name can be edited.

The green area is preset configuration area. After selecting a preset; all modules that have been saved in the preset will be shown. To save a module’s current content to the preset; you need to drag a module from the Design Area into this window. Notice that once a module is already added; you won’t be able to cover its saved content unless delete it from the preset.

4.7 Design Area



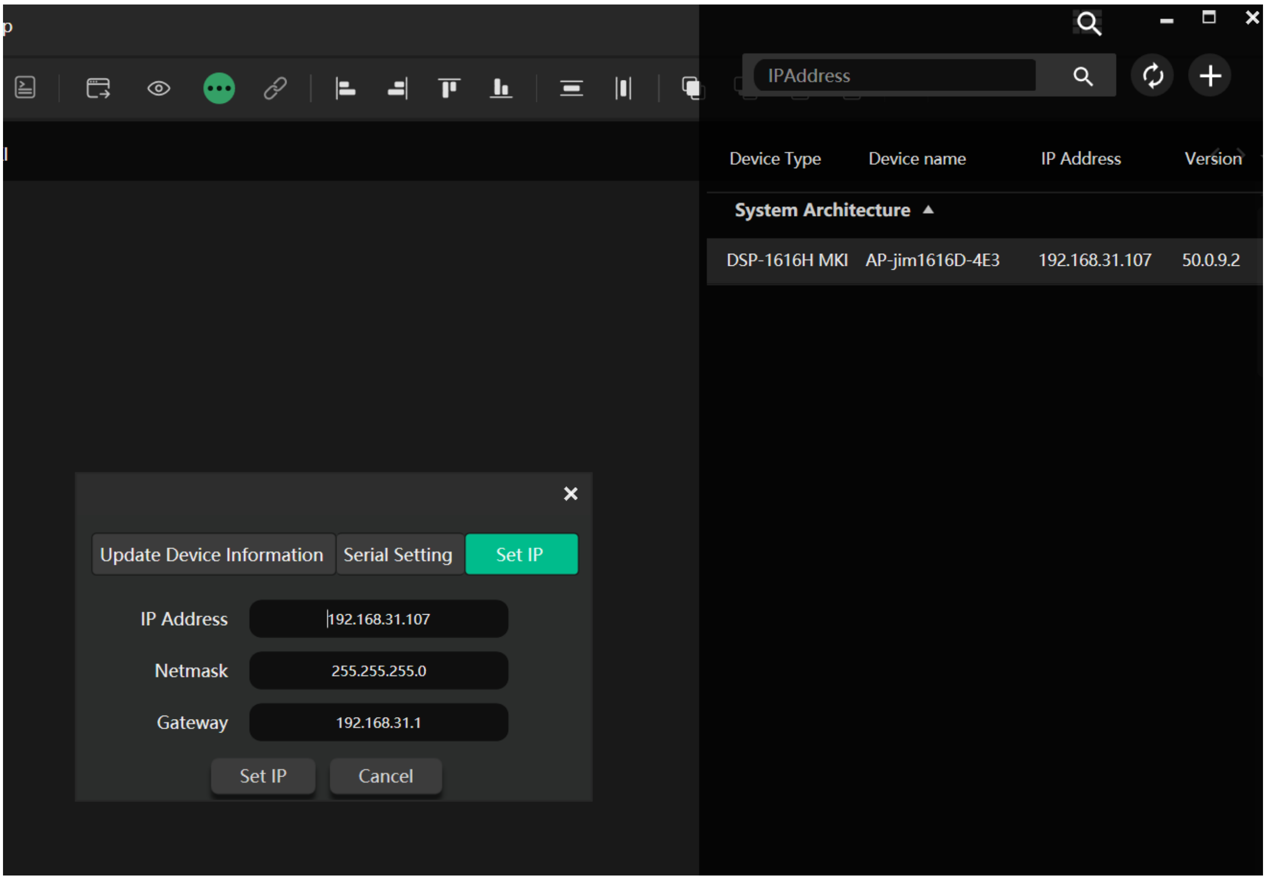
The Design Area is the project editing area where all DSP modules are edited; configured; and wired.

5 Software Instruction

5.1 Discovery of Hardware

In the upper right corner of the of the software; click the Device Discovery (small magnifying glass icon); all the recognizable devices in the network will pop up on the right side. And the Device Discovery list will show the device type; device name; IP address and firmware version of all known devices in the network.

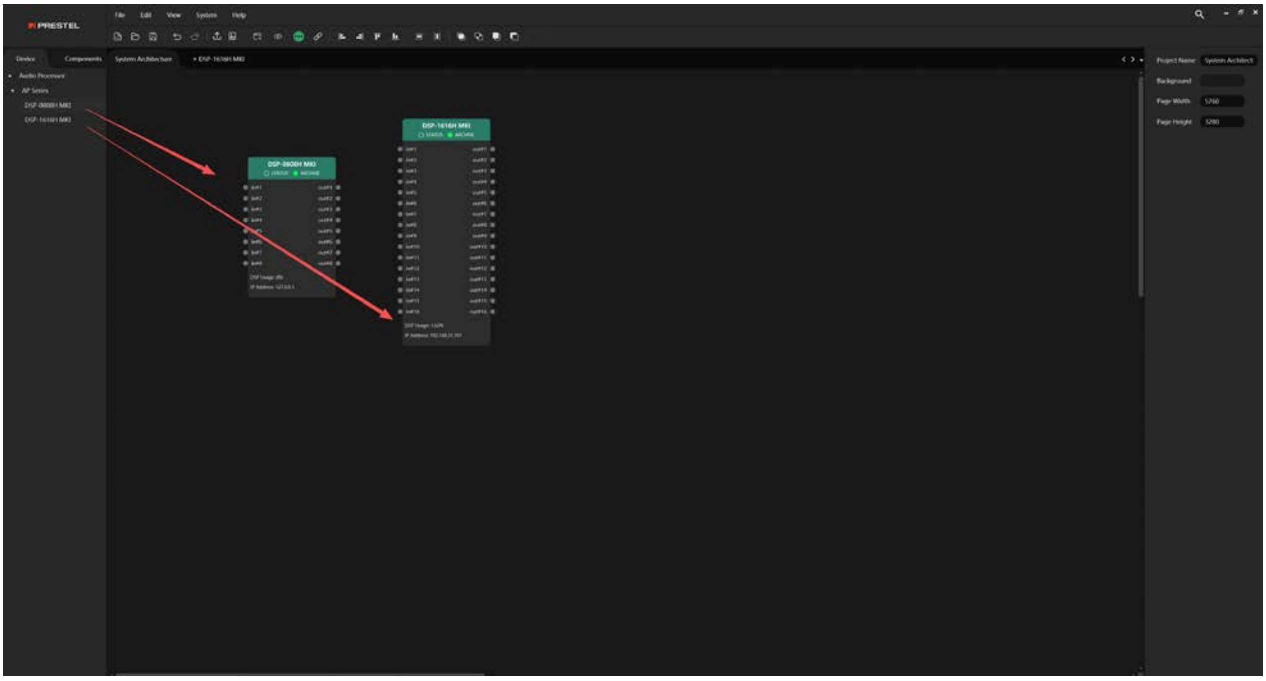
Double-click a device in the device list; in the pop-up box; you can modify the basic information of the device and the serial port settings; as shown in the following figure:



Note: As long as the devices are on the same LAN; you will find them on the device discovery list even if they are on different subnets. However; if you want to connect to the device; your PC must be on the same subnet as the device.

5.2 Create a DSP to the Design

In the System Architecture Window; you can add a DSP. Drag and drop the target device from the Device Window to the center of the screen; as shown in the following figure:



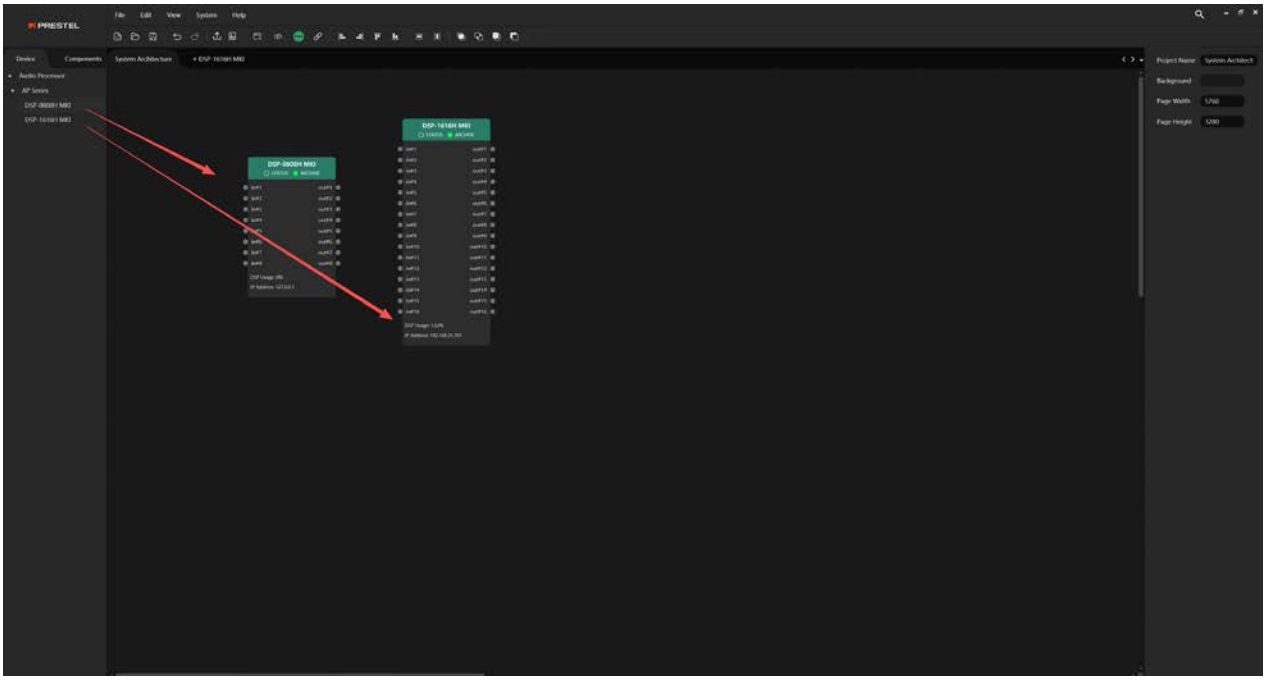
Left click on the created device and the Properties Window on the right side of the screen displays the properties configuration of the created device module. Then you are able to set the following properties:

1. IP Address：You need to fill in the IP of the target device in the Device Discovery list；

Note: Incorrect configuration of this item will result in the program not being uploaded to the corresponding hardware properly.

2. Text：Display name of the DSP.

Design and manage multiple DSPs under a same project file is supported. Editing or controlling each DSP only requires double-clicking on the device in the System Architecture Window to enter the its program design page for that DSP; as shown below:

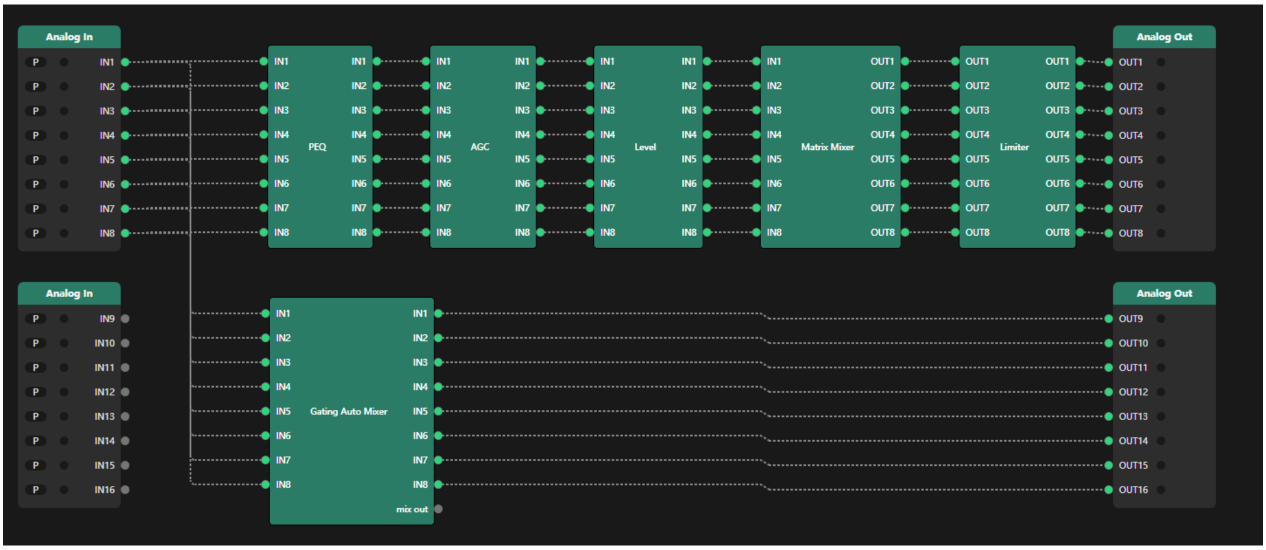


5.3 Program Design

5.3.1 Wiring

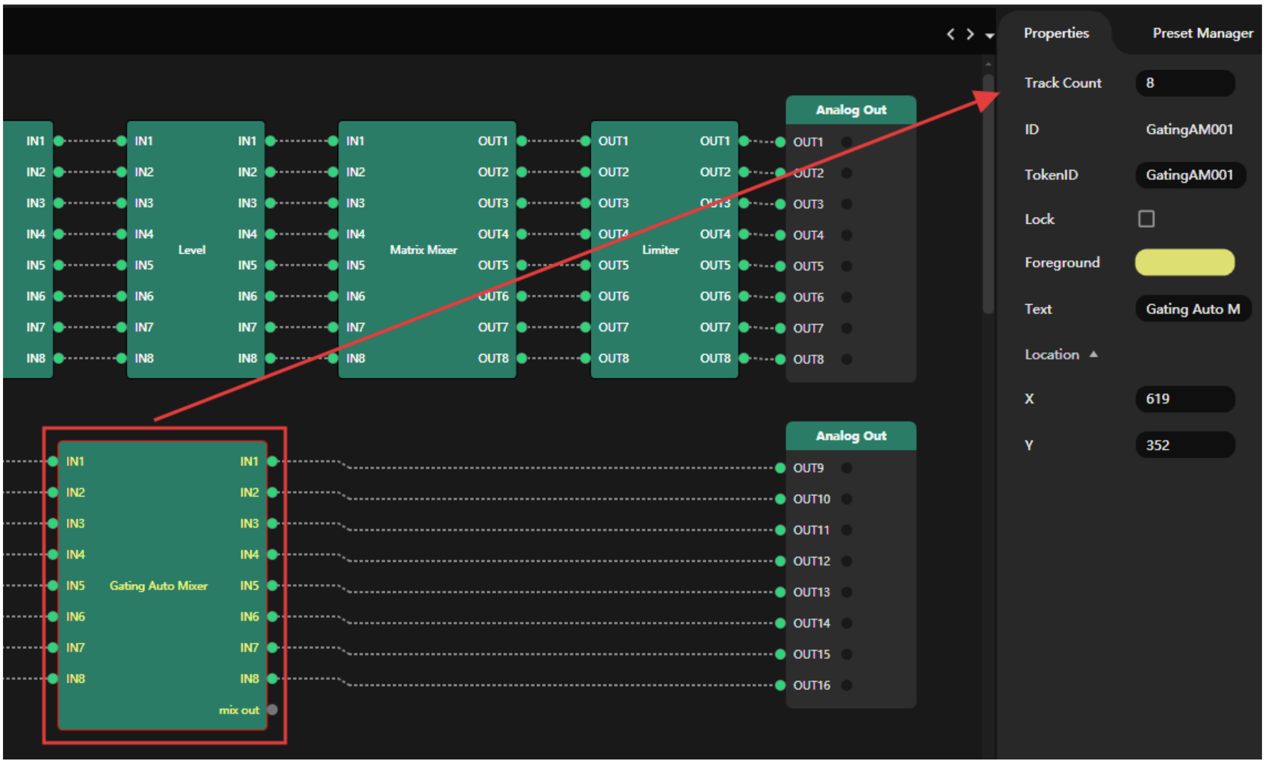
To implement the programming of the DSP; modules are connected by means of intuitive module pin-outs wiring. The output pin of one module are connected to the input pin of another module by drawing a wire between them. Input pin and output pin that connected with a wire means they will share the same signal; as shown in the figure below.

All you need to do is click on a pin and drag it to the other pin. Multiple pins can also be box-selected and dragged for multi-channel wiring.



5.3.2 Modifying Module Properties

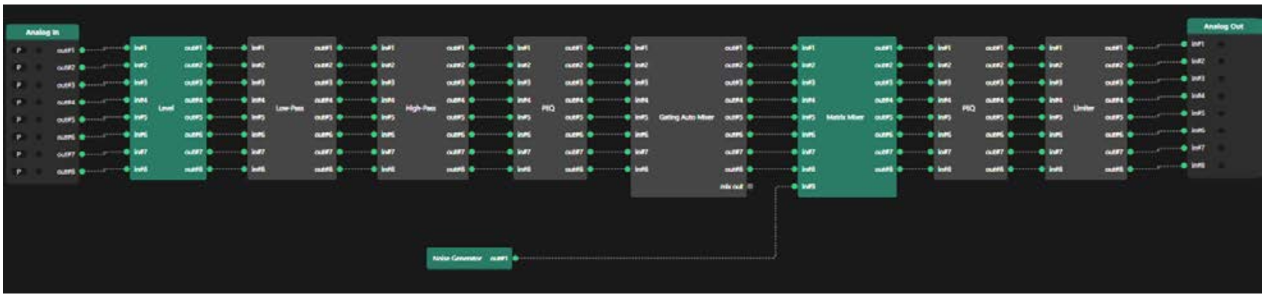
Clicking on a module will display the module's properties on the right side of the screen. You can make targeted changes to it. For example: Track Count; Text Color and other properties (as shown below).



5.3.3 Module Applications

There are many kinds of DSP modules to choose from. After double clicking into the Device Window page; the left side of the screen shows all available modules of the device. According to your demand of the system; you can drag and drop the relevant module from Components Window to complete the routing; control and processing of audio signals.

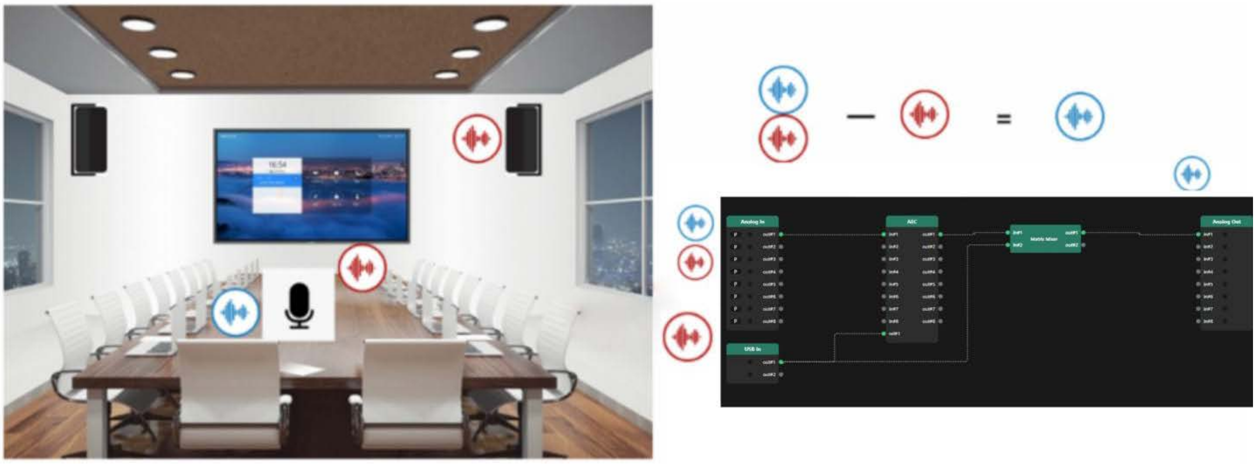
Common modules include Level Control; Meter; High-Pass and Low-Pass Filters; PEQ (Parametric Equalizer); NHS (Notch-filter-based Howling Suppression); Delay; Route; Matrix Mixer; and more. As shown below:



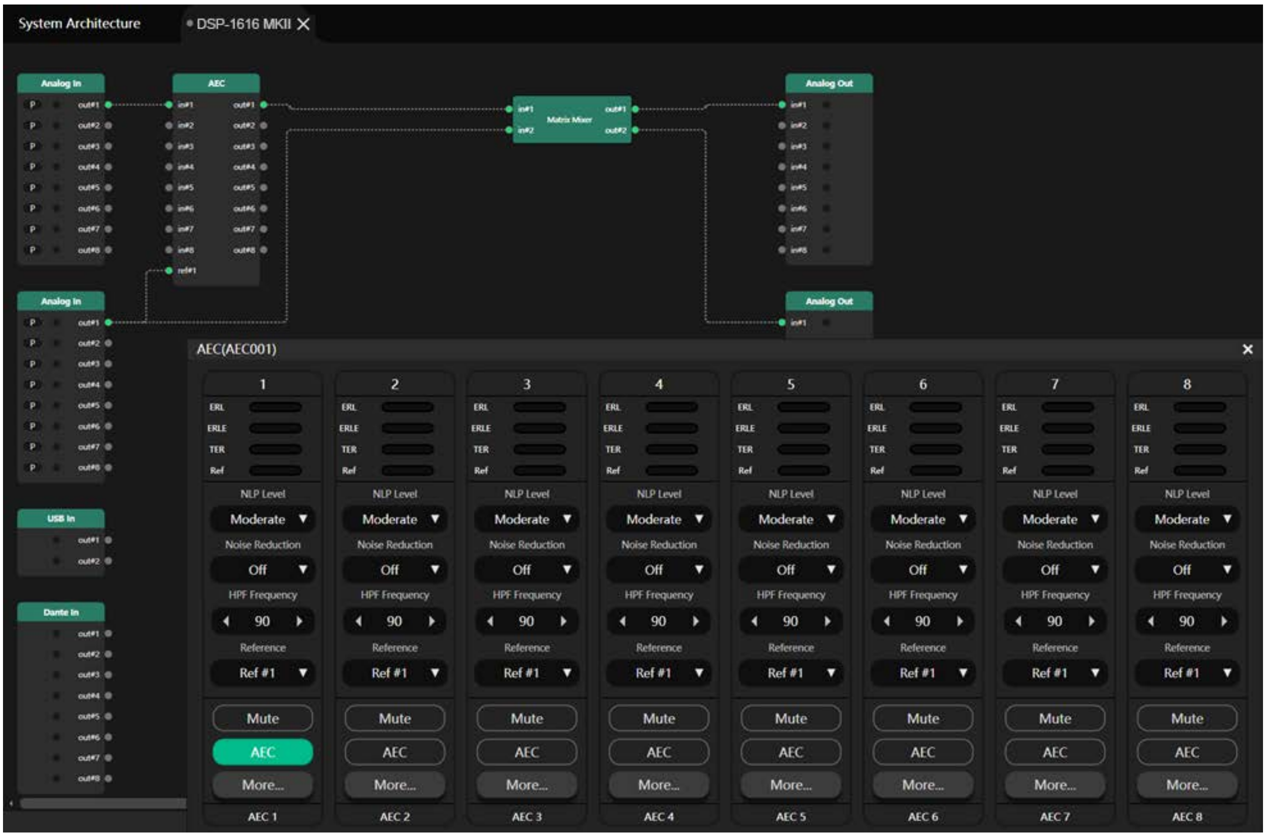
5.3.4 AEC Application

AEC (Acoustic Echo Cancellation) is a technique used to eliminate audio echoes. In duplex audio communication; due to the reflection of sound caused by sound reinforcement; there may be an acoustic echo phenomenon. When far-end speaks; they will hear their words coming back again from us; which affects the conference speech. AEC technology can improve the quality and stability of audio communications by using specialized algorithms and signal processing to detect and eliminate acoustic echoes in real time.

In audio systems; AEC algorithm is mainly used in scenarios such as conference rooms; voice communication; and remote education. In these scenarios; duplex audio communication is always required. And echoes can cause interference and noise to such audio communication. The DSP has a built-in high-performance dedicated chip to run the AEC algorithm; which detects and eliminates echoes in real time; making audio communication more efficient and stable.

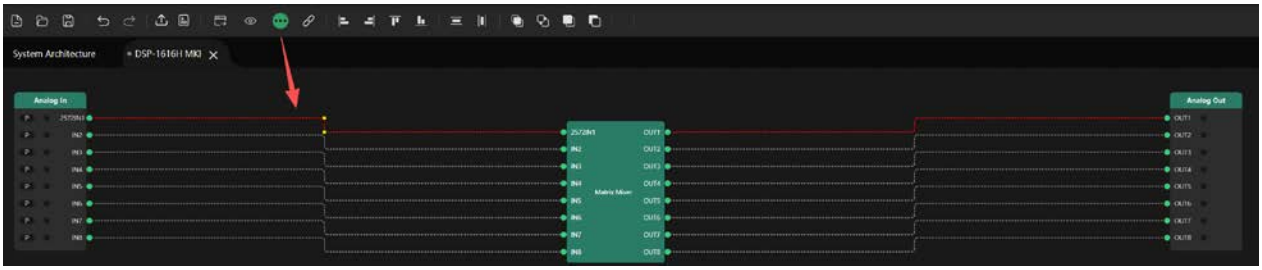


Each AEC channel can have a corresponding reference input call “AEC Ref” (where the reference signal will be removed from that channel). Connect the remote signal source to the corresponding channel of the reference to complete the AEC connection. Then turn on AEC and it will start to work in the real time (as shown in the figure below).



5.3.5 Locate Signal Path

In complex audio systems; quickly locating the transmission path of audio signals to find the complete path of the signal from input to output can facilitate troubleshooting and maintenance. Click on the “Signal Path” icon in the Tool Bar; and then select the wire you want to view. Then the signal transmission path can be shown in red; as shown in the following figure:

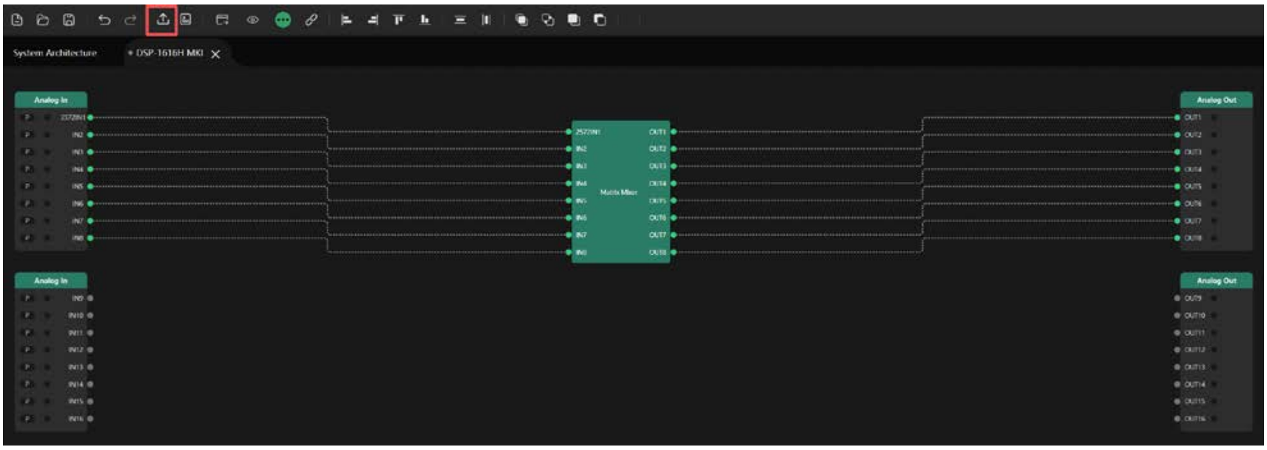


5.3.6 Compile

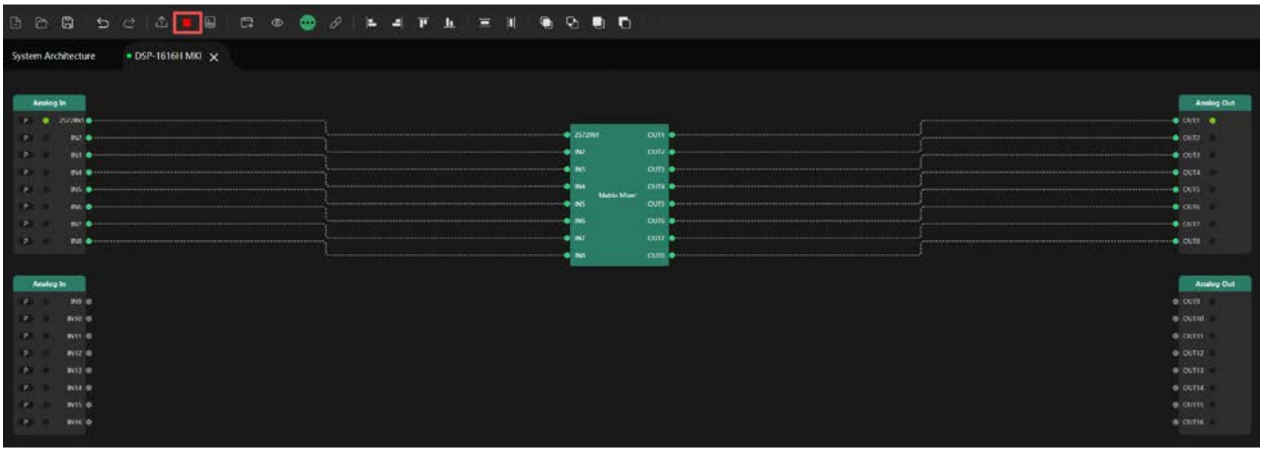
After finish your wiring; you can compile offline to set the parameters of the processing module. Click the Compile button in the Tool Bar (as shown in the figure below) or press “F6” on the keyboard to enter this mode.

5.3.7 Upload and Run the Program/ Stop the Program

If you want to make the program work after editing; then you need to upload it into the DSP. Click the Upload button in the Tool Bar (shown below) or press “F5” key on your keyboard.



After the program is uploaded successfully; the Tool Bar will change from Edit Status to Run Status. At this time; modules and wires are not allowed to be moved. You need to press “F7” or click on the Stop button to quit real-time running.

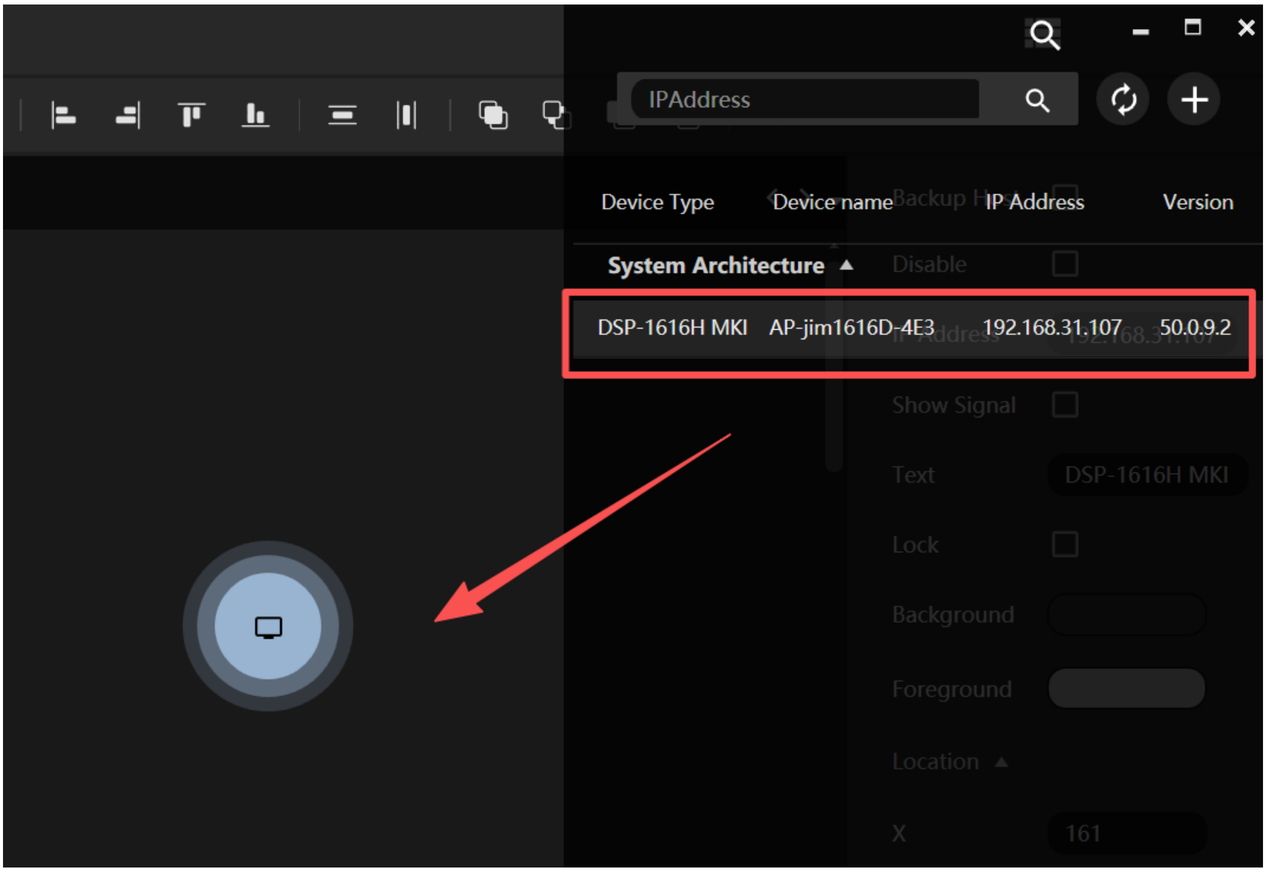


Note: The “DSP Usage” at the bottom left of the software window cannot exceed 100% when creating a program. If this value is exceeded; the software and the corresponding devices will not run normally.

Note: The IP Address setting of the DSP in your program must match target hardware. Otherwise; the uploading process will fail and a dialog box will pop up.

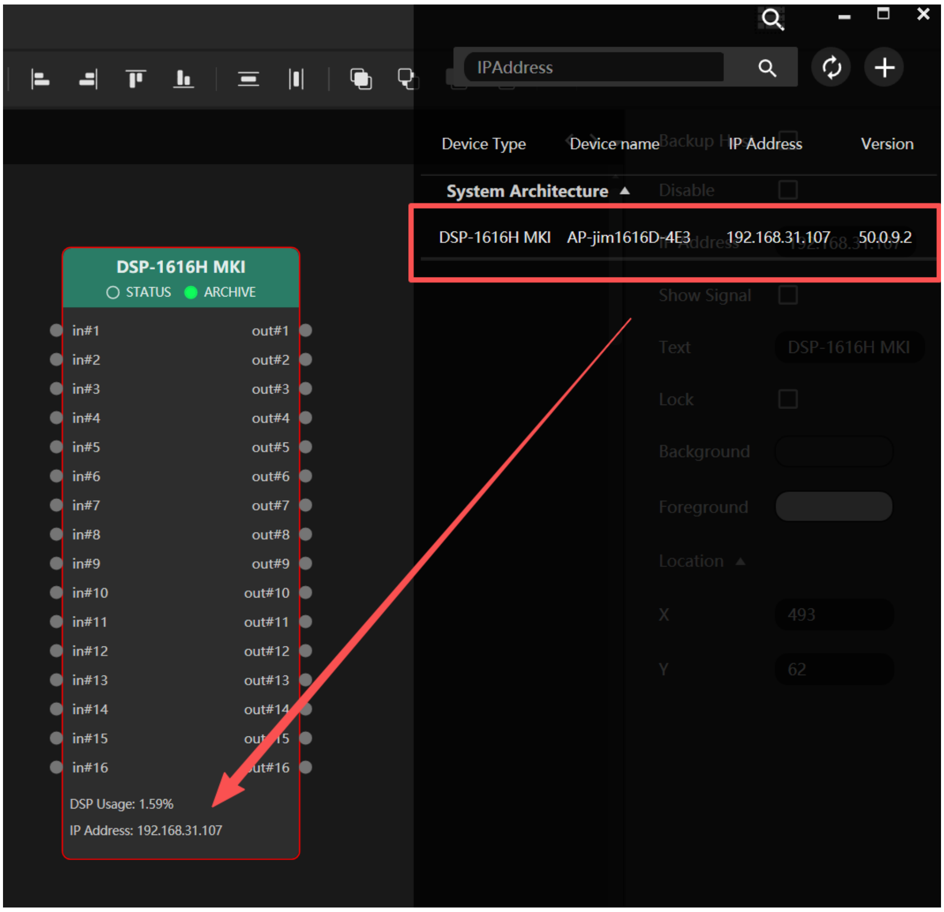
5.3.8 Download and Run the Program

Users can also view; control; and modify the archived program that already pushed in the DSP. First of all; you need to download the program file in the DSP found in the current network. Keep your screen on System Architecture page; then select the target DSP from the right side of the Device Discovery list and drag it to blank area. As shown in the figure below.



Note: Your PC must be on the same subnet as the target DSP to download its program file.

If the download is successful; the software will be in Run Status. The target DSP module will appear in System Architecture page and the status light on it will turn green and you are able to control it in real time.



Note: If your target DSP has modified the IP address without re-uploading the program; the downloaded program file will still keep the IP address when it was pushed. In this situation; the status light on the device module will be “gray” instead of “green”. Now if we need to control the DSP; we need to exit the Run State first. Then click the DSP module; change its IP address in the Properties Window to match the its actual IP address. Then you are ready to upload the program file through above steps.

5.3.9 Save

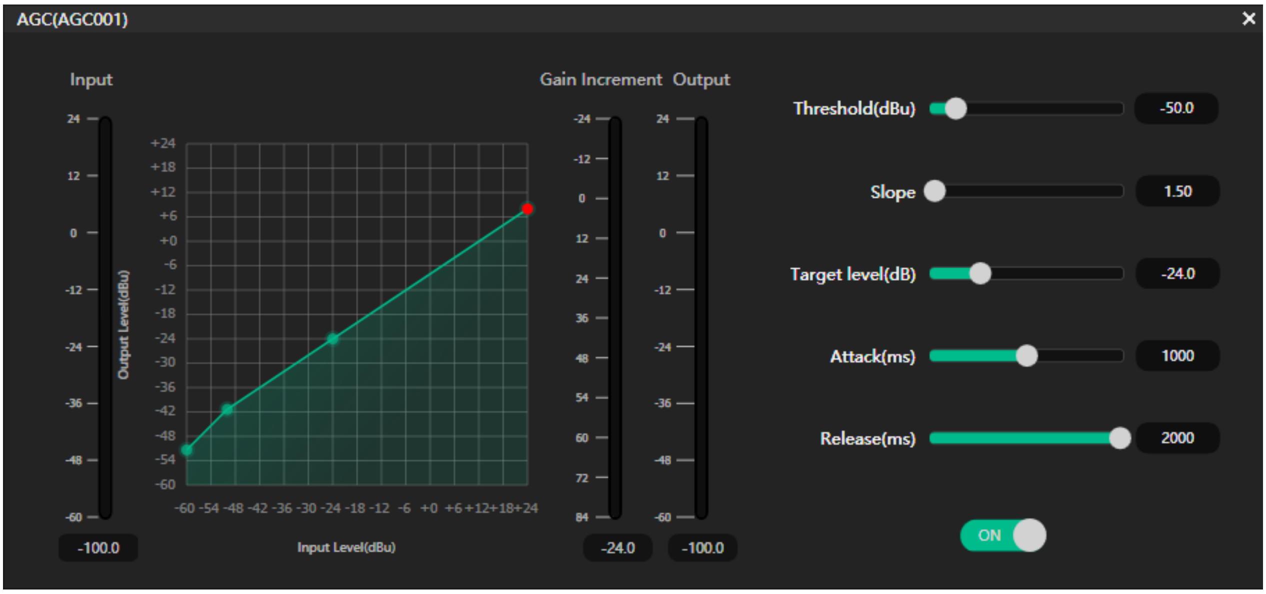
You can save current program file in your PC when there are no further modifications needed. Newly created programs file usually require you to choose a local path first.

5.3.10 DSP Components

Automatic Gain Control

Automatic gain control (AGC) is a special case of a compressor; with a threshold set at a very low level; slow to moderate attack time; long release time; and low ratio. Its purpose is to raise signals from 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 AGC from regular compressors/limiters.

Using AGC can normalize; for example; the voltage level of CD players playing BGM (background music); FGM (foreground music); or waiting music; in order to eliminate some variations in the level of paging microphone.



AGC includes the following controls and switches:

Threshold: When the signal level is below this threshold; the input/output ratio remains 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 is recommended.

Slope (Ratio): The ratio between portion that exceeds the threshold value of the input signal and of the output signal.

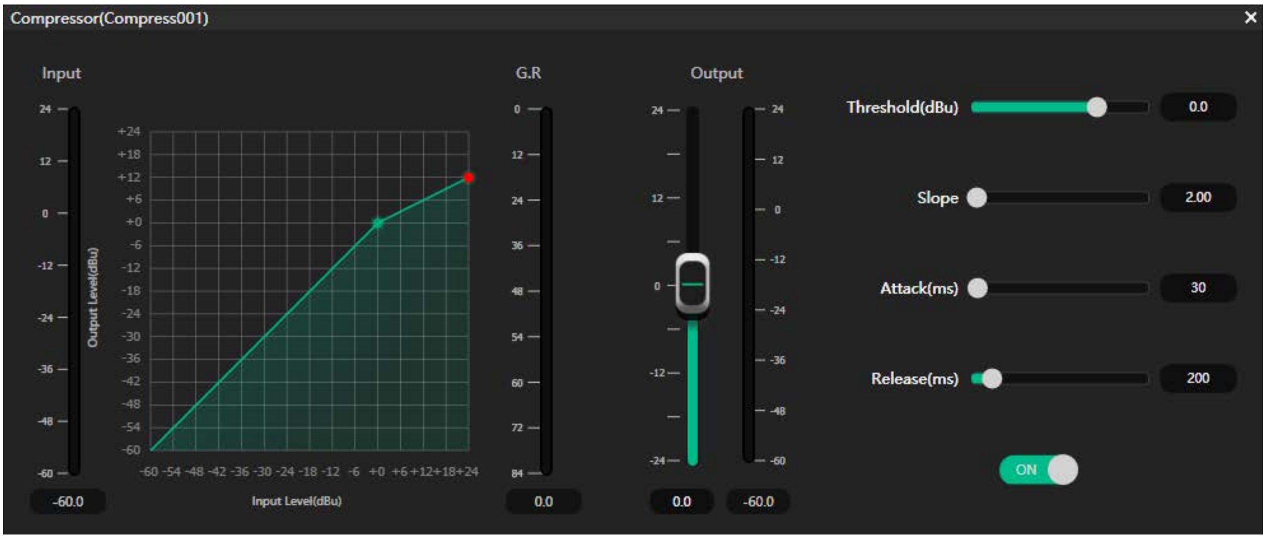
Target threshold: The required output signal level. If the signal is above this threshold; the controller will compress the signal according to the ratio.

Attack time: Control the response speed when signal go above the threshold level.

Release time: Control the recover speed time when signal go below the threshold signal.

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:



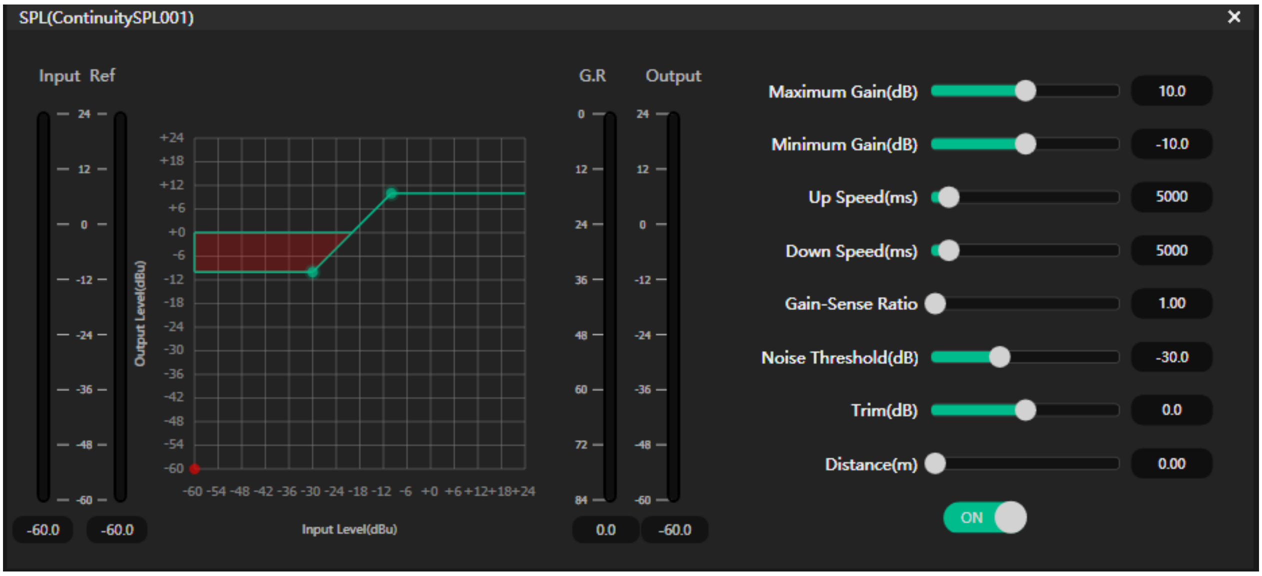
Threshold: When the signal level exceeds the threshold; the compressor/limiter begins to decrease the gain. Any signal that exceeds the threshold will be considered an overshoot signal and its level will be reduced under normal circumstances. The larger the range of signals exceeding the threshold; the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

Slope (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 between portion that exceeds the threshold value of the input signal and of the output signal. 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 will be 1dB. A compression ratio of 1:1 indicates that the compressor has not proportionally attenuated the signal. The adjustable range of compression ratio is 1-100.

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 speed of the compressor. Otherwise; 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 effect from happening. The attack time determines the speed of gain attenuation; while the release time determines the speed of gain recovery.

Output gain: Also known as compensation gain. 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.

SPL(ContinuitySPL)



Automatically adjusts output volume based on ambient noise sensing and processing.

Maximum Gain: Maximum level that can be adjusted to.

Minimum Gain: Minimum level that can be adjusted to.

Up Speed: The time it takes to raise the volume to the target value after the threshold is reached upward.

Down Speed: The time it takes to reduce the volume to the target value after the threshold is reached downward.

Gain-Sense Ratio: The ratio is higher; the change in raising or reducing gain is larger.

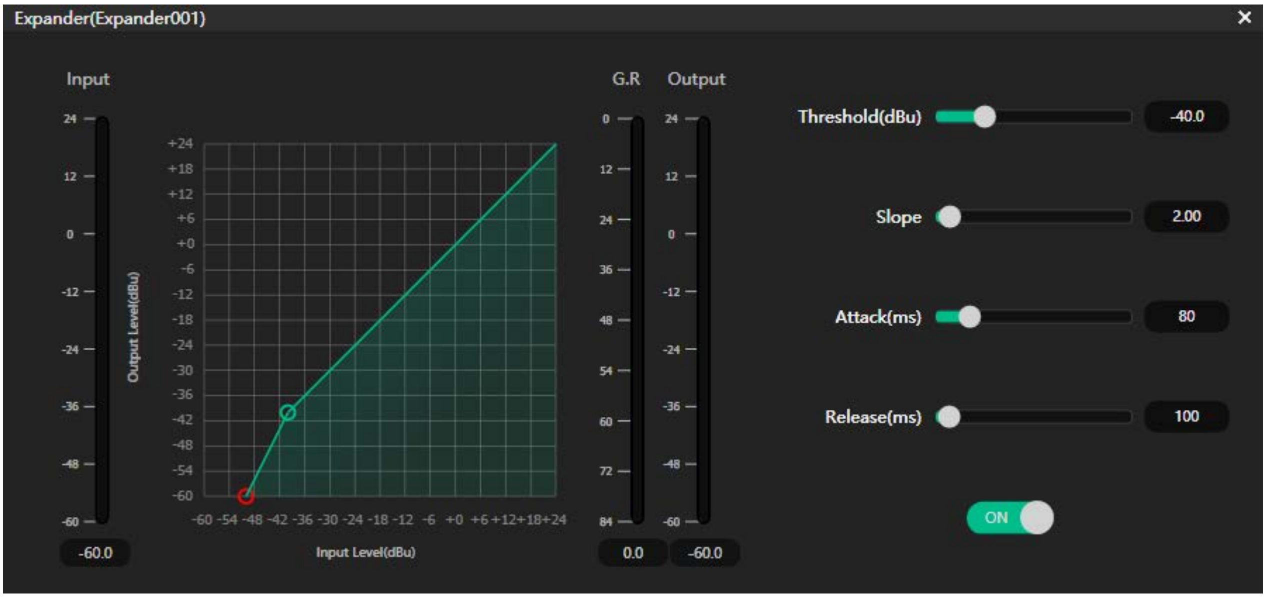
Noise Threshold: When noise is higher than this value; volume raise; when noise is lower than this value; volume reduce.

Trim: This can be used to adjust gain.

Distance: The distance between reference signal to local signal.

Expander

Expander can expand the dynamic range of the signal; which is theoretically opposite to compressor. 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. 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.



Threshold: The signal must exceed this level to pass through the expander. Normally; it is usually set to the environmental noise level.

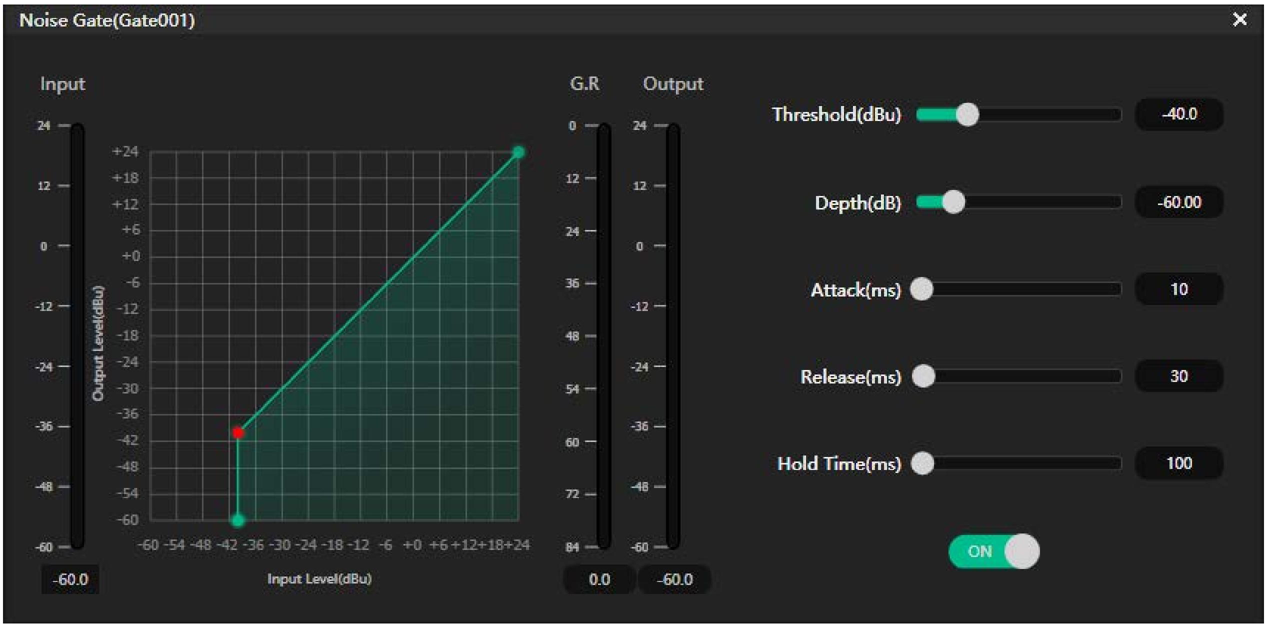
Slope(Ratio): The slope below the threshold point on the gain curve. When the ratio is set high; expander appear to be noise gate.

Attack time: When input signal reach above threshold; the duration time expander needed to open and let signal passing through. Faster attack times let shorter transient signal open the expander.

Release time: The time required for the expander to work after the input signal drops below the threshold. Whether it is attack time or release 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; while the speed at which the gain decays from 0dB to -40dB is controlled by the release time. 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.

Noise Gate

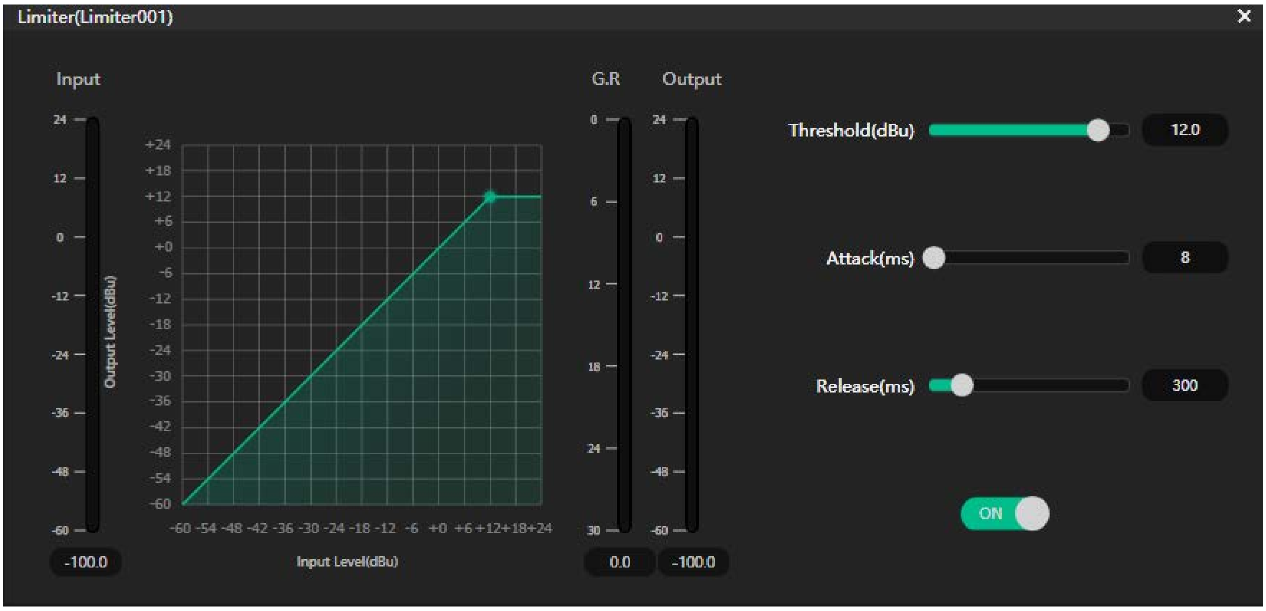
The main purpose of a noise gate is to attenuate signals below a threshold; and this attenuated signal is usually noise.



Threshold: signals exceeding the threshold pass through; signals below the threshold attenuate.
Depth: The amount of attenuation; which determines how much signal below the threshold needs to be attenuated.
Attack time： Refers to the speed at which the noise door opens and let signal passing through.
Release time: The release time is opposite to the attack time and refers to the speed at which the noise gate begins to work.
Hold time: When there is no Hold time; the signal will immediately begin to decay after the signal goes below threshold. If you set a Hold time of 10ms; for example; the noise gate will maintain its original output for 10ms and only begin to decay after 10ms.

Peak Limiter

The job of Limiter is 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. Limiter relates to how the gain attenuation starts to occur before the signal experiences overshoot. There are two processing stages. In the first stage; only slight among of limitation are applied; but overshoot signals are not processed. In the second stage; if signals experience overshoot; they will attenuate drastically.



Threshold: When the signal level exceeds the threshold; the limiter begins to decrease the gain.

Attack time: When input signal reach above threshold; the duration time limiter needed to work and attenuate signal to the threshold.

Release time: When input signal falls below threshold; the duration time signal needed to recover its original level.

Limiter operate on peak levels. It is useful for protecting loudspeakers. To set the threshold parameter; please refer to the rated voltage of the loudspeaker to do the calculation (Note: 0dBu≈0.775V). Typically; the attack time is usually set faster and the release time slower when doing loudspeaker protection.

Priority Ducker

When the level value of one channel exceeds the specified threshold; the level of the other channel will be attenuated; which is what ducker do.

Threshold: Ducker begins to attenuate signal when the reference signal is above the threshold. And Ducker recovers when reference signal goes below the threshold.

Depth: The amount of signal attenuation.

Attack time: When reference signal reach above threshold; the duration time ducker needed to work and attenuate the other channel.

Release time: When the reference signal is lower than the threshold value; the ducked channel will recover to the original gain.

Hold time: Hold time refers to how long the ducked channel needs to maintained attenuation after the reference signal goes below the threshold.

Crossover

Depending on the type of mode; there are different slope parameter settings. Again; the choice of slope configuration is based on actual needs.

The 2-Way crossover can divide signal into the H for high-frequency section and the L for low-frequency section.

The 3-Way crossover can divide signal into the H for high-frequency section; M for mid-frequency section and the L for low-frequency section.

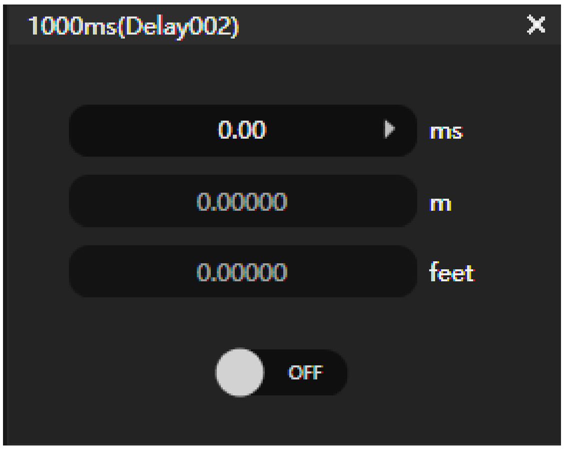
Freq: The frequency of the corresponding crossover point can be set according to the characteristics of the loudspeakers.

Type: There are 3 different shape to choose from; Bessel; Butterworth and Linkwitz-Riley. You can select the appropriate type for crossover setting according to the actual situation of signal coupling at the crossover point.

Slope: Depending on the type of filter; there are different slope parameter settings. Again; the choice of slope configuration is based on actual needs.

Delay

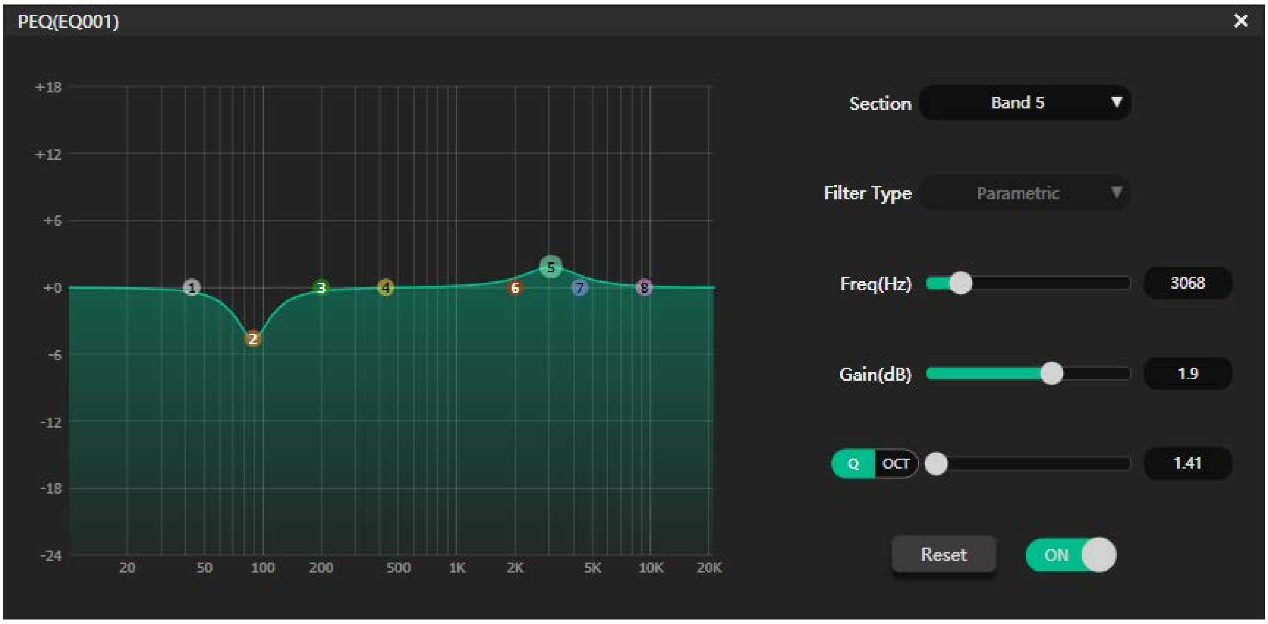
There are three types of delay; 50ms; 100ms and 1000ms; which correspond to the maximum delay time. You can choose the needed delay time according to the actual situation.



Filling in milliseconds will show corresponding meters or feet. So that you can have a reference for setting delay time.

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 gain of certain components in their spectrum. In simple terms; an equalizer can alter the timbre and phase of an audio signal.



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

High & Low pass: The reference frequency of the pass filter is called the cutoff frequency. The pass 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.

Freq (Hz): The center frequency of the parameter filter and cutoff frequency of the pass filter.

Gain (dB): Gain increase or attenuation at the center frequency position of parameter filter; overall gain increase or attenuation of the pass filter.

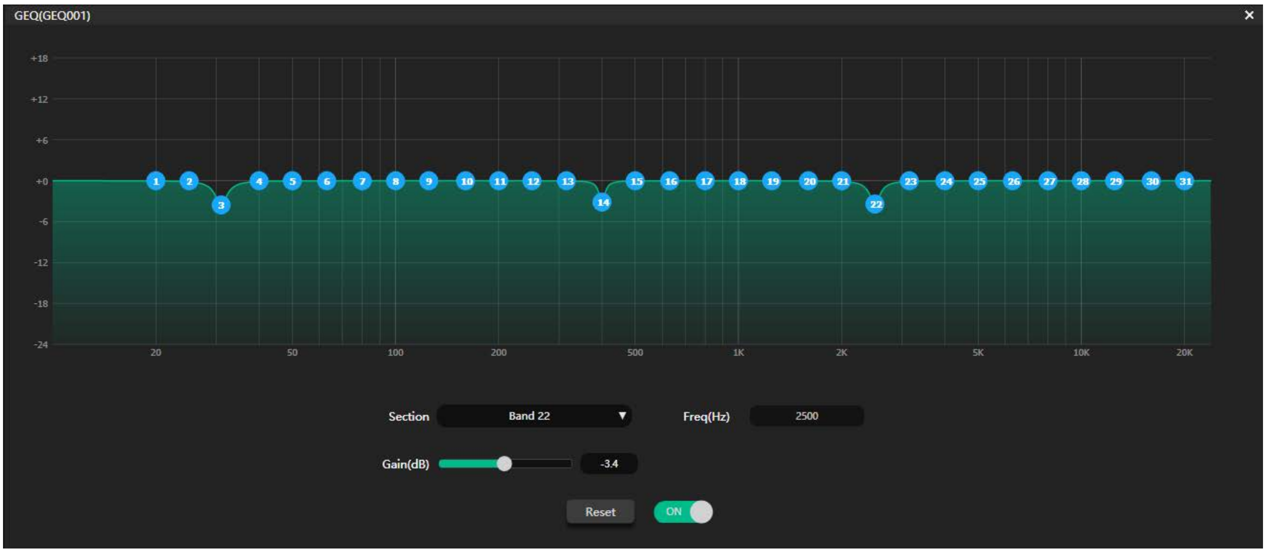
Q Value/Oct: Bandwidth setting of the filter at the center frequency; which determine the range of influence on the frequency band of the audio signal. Q value and bandwidth can be converted to each other and can be set according to specific needs.

The module is an 8-band parametric equalizer by default; you can set the required number of bands in the Properties Window according to your needs; and it can support up to 32 bands in one PEQ.

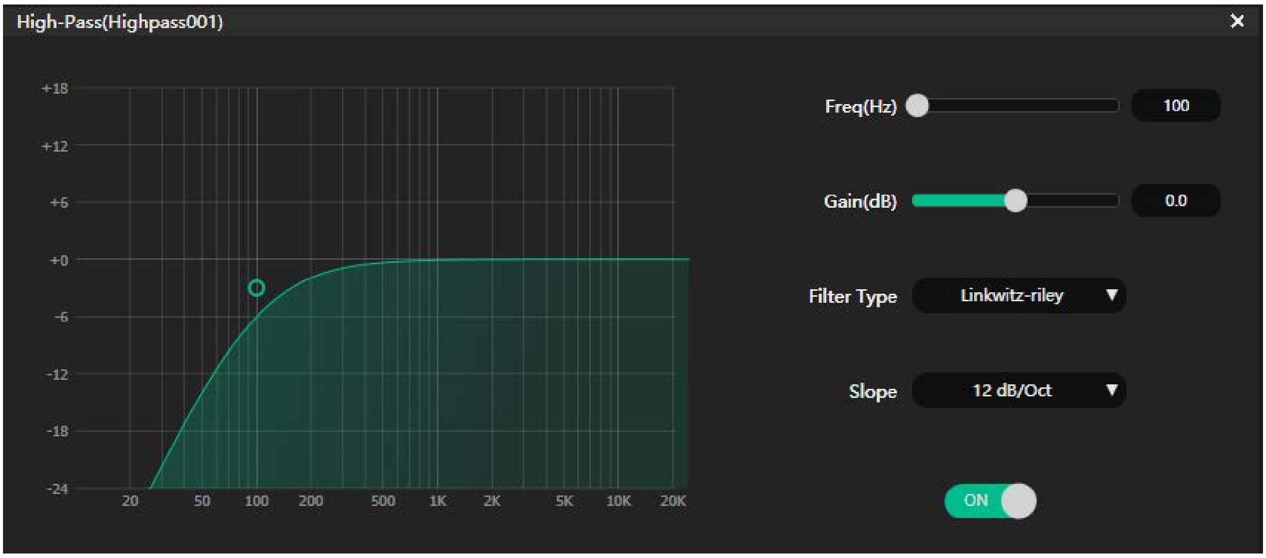
GEQ(31-band Graphic EQ)

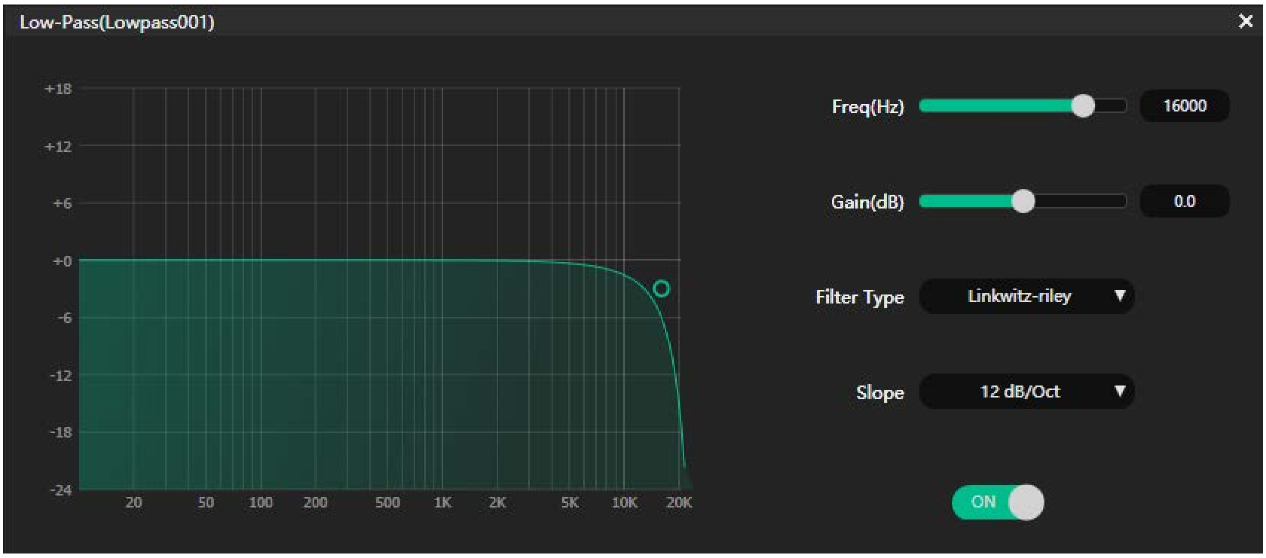
Using constant Q-value technology; each frequency point is equipped with a push-pull potentiometer; which ensures that the bandwidth of the filter remains unchanged regardless of whether a certain frequency is increased or decreased. The commonly used professional graphic

EQ divides the 20Hz~20kHz signal into 31 segments for adjustment; which is 1/3 octave between each adjacent band.



High Pass & 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: Three filter types can be chosen from; 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.

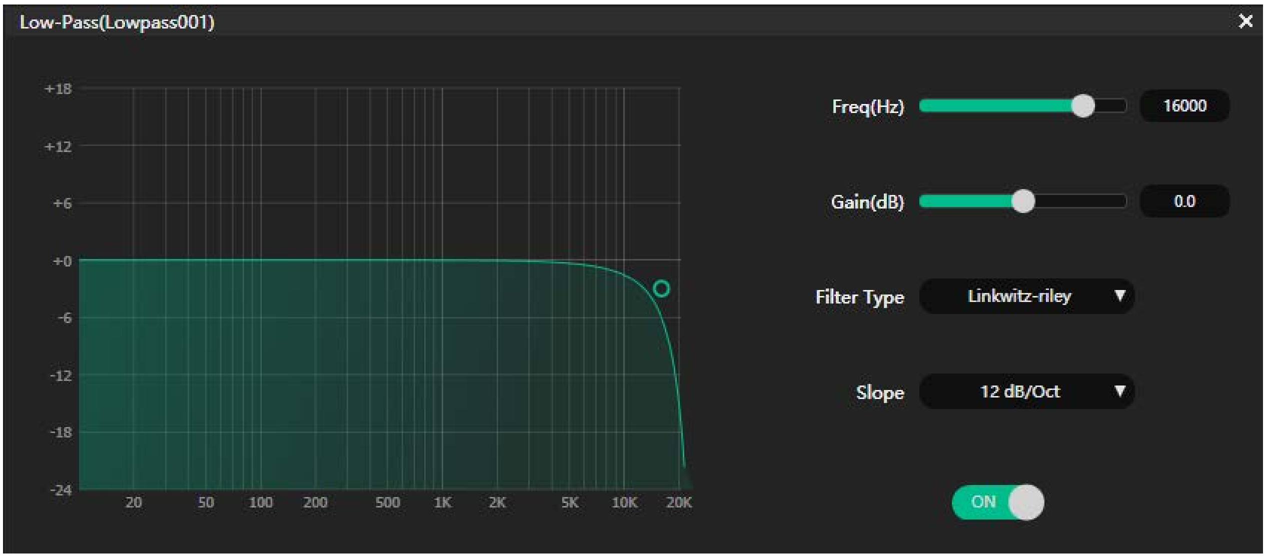
NHS (Notch-filter-based Howling Suppression)

NHS also known as feedback suppressor. When using any kinds of feedback suppression; they should always be combined with good system design and engineering practice. Traditional methods should still be omitted; 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. Afterwards; feedback suppressor 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 NHS 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.

The filters can be locked to prevent them from changing during the performance. The filters can be fixed as a dedicated notch filter module. When they set as dynamic filters; they are automatically cycled. In this situation; filters that are only temporarily needed can be refresh.



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 detection sensitivity increases to detect feedback faster.

Once the output level drops below the threshold; the gain recovers and the sensitivity return to normal. This value refers to the peak level of the full-scale digital signal. When this value is set to maximum (stands for 0 dBFS); it is equivalent to turning off this function.

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.

Step size: The speed at which the gain attenuates.

Bandwidth: 1/5; 1/10 and 1/20 Oct is available. 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.

Dynamic Recovery Time: This parameter is the holding time required for the system to return to normal gain (i.e.; stop attenuating the feedback frequency) after the feedback suppressor has finished detecting and suppressing the feedback signal.

Dynamic/Fixed: Each notch filter has two modes; dynamic or fixed. When set to dynamic; the filter will participate in the filter cycle and detect new feedback. When 8 filters are used up; the module will search for the dynamic filter setting and use it to suppress the new feedback. When set to fixed; 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 save them in the preset.

Clear Dynamic: Click this button to instantly clear all dynamic filters.

Clear All: Click this button to instantly clear all filters; no matter they are dynamic of fixed. This operation is typically performed when re-debugging the NHS module.

Feedback suppressor can be used as tool for system debugging to find feedback frequencies or as preventive measures during normal operation. If you want to achieve high system sound transmission gain and feedback suppression effect; the following steps for debugging is recommended:

(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. Reduce the panic threshold to lower the level of feedback occurrence.

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

(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 or the dynamic state to handle feedback that may occur during the performance. Another possible approach is to copy the notch filter parameters to an equalizer. This allows for the addition of more filtering capabilities.

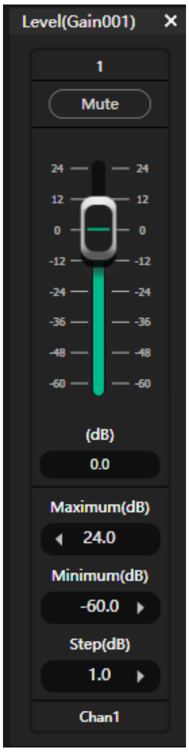
If the devices include speakers; it is recommended to use a compressor/limiter module for additional protection before above steps. Setting appropriate limiters will ensure that the speaker is not damaged; even if all notch filters are used up or feedback suppressors cannot control feedback when system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

FIR Filter

FIR (Finite Impulse Response) filter is one of the LTI (Linear and Time-invariant) filter in digital signal processing. Its output is a weighted sum of the current and past input samples. When a pulsed signal is input; its output decays to zero after a finite number of samples. This is why it is called a finite impulse response filter. An important advantage of FIR filters in terms of application is that they can be designed to have a linear phase; which means that signals with different frequency components pass through the filter with phase delays that are linearly related to frequency. In communication systems and audio processing; it is important to maintain the phase relationship of the signal. For example; in audio; if the filter does not have a linear phase; then the sound signal may have a timbre distortion after it has been filtered. This is because the time it takes for sound components of different frequencies to reach the human ear is altered by a nonlinear phase delay.

FIR files can be imported for frequency response phase correction; supporting .csv and .secfir formats.

Level Control

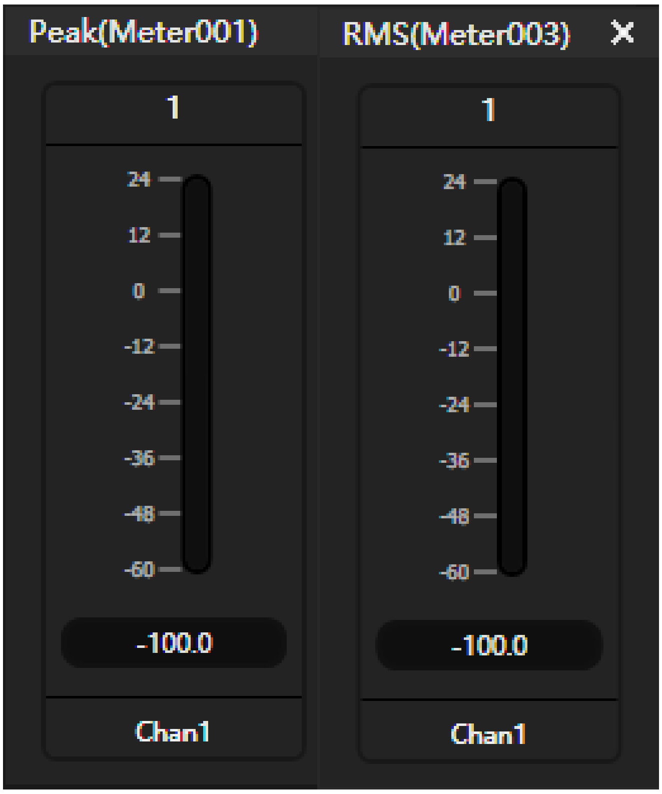


The maximum or minimum control level can be set.

The step size can be set to increase or attenuate in certain steps.

Meter

There are two types of level meters; Peak Meter and Root Mean Square (RMS) Meter.



The Peak Level Meter is used to measure the maximum instantaneous level reached by a signal. It shows the peak value of the signal voltage; which is the maximum value of the signal waveform in the positive or negative direction away from the zero level over a period of time. The peak level of a sine wave; for example; is the amplitude from the peak or trough.

RMS level meter measures the RMS value of a signal. It reflects the power characteristics of the signal and is related to the actual energy delivered by the signal.

Automixer

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

The DSP is equipped with automixers that each one supports up to 128 channels of audio signal input. Each channel in the automixer can be set to inactive. In this way; it can bypass and without being regulated by the automixer. For example; if you want to keep the chairman's microphone in a normally unmuted state; or there are channels suitable for fixed volume like BGM.

Channel mute and fader are both adjusted after automixer; and any adjustments to them will not affect the automixing algorithm. In this case; the level gain of other channels can be reduced whether this channel is muted or not when its level goes high. The mute button on each channel will mute that channel in the mix out and also direct out. The channel fader 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. Channel mute and fader are both adjusted after automixer; and any adjustments to them will not affect the automatic mixing algorithm. In this case; the level gain of other channels can be reduced whether this channel is muted or not when its level goes high. The mute button on each channel will mute that channel in the mix out and also direct out. The channel fader 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 automixing 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 value between 0-10.

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 gain. For example; if the priority of channel 1 is set to 6 and the priority of channel 2 is set to 3. The input levels of the two channels are the same; channel 1 will receive an additional 6dB of automatic gain compared to channel 2. During use; it should be noted that the slope setting of the automixer can also affect the automatic 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 should be the same; please keep all settings at the default level 5.

Note: You should be cautious 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 before automixer on the highest priority channel; while setting the threshold to a level where the threshold or expander will not be opened by background noise.

Gain-sharing Automixer

Gain: Control the mix out volume.

Slope: Affects the attenuation of lower level channels. When the slope is higher; channels with lower levels will experience more attenuation. The slope control working mode is similar to the ratio control on the expender. Value around 2.0 is recommended. If set to 1.0; the effect is equivalent to turning off automixing 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: Faster speed 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. Desire speed of turning on the microphone is much faster than off; which means even a 100ms response usually does not affect the initial words spoken. If set to a slower speed of a few seconds; the previous active channel will hold for a longer time and keep its state within a few seconds.

Active: Each channel has an automixing on/off button; and each channel that need to participate in automixing needs to turn this on. It can also be turned off; as this channel does not participate in automixing.

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.

Gating Automixer

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

Gain: Control the mix out volume.

Hold time: Adjust the length of time the channel remains open after the talker ends speaking. This function ensures that brief pauses between words and sentences during the speech will not cause the channel to close.

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

Sensitivity: 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 sets high; the channel will be easily triggered even in very quiet environment.

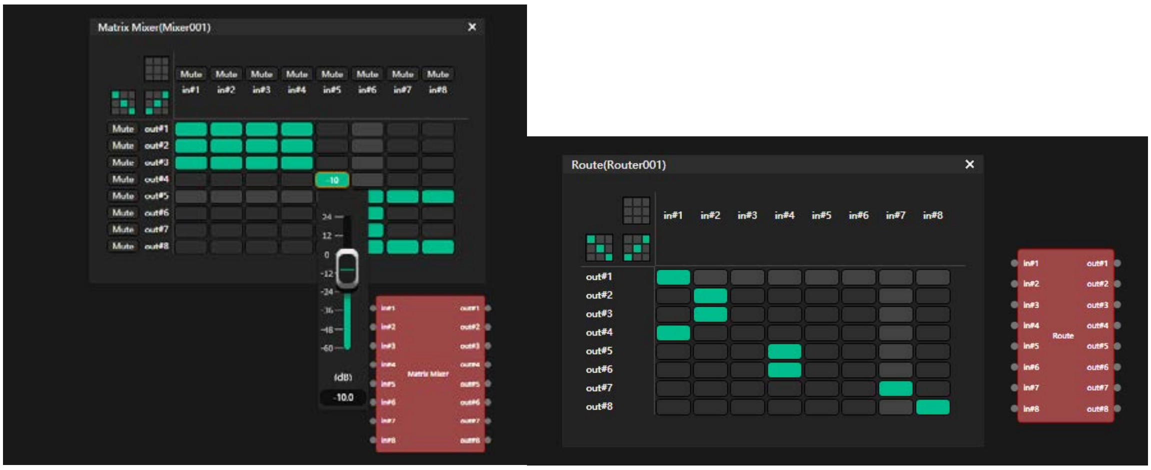
NOM Atten: The NOM attenuation is the attenuation value for all channels when more microphones are turned on. This parameter reflects the attenuation when two microphones; as well as doubling of the number of microphones are turned on. For example; when set to 3dB; opening two microphones will attenuate 3dB; opening four microphones will attenuate 6dB; and opening eight microphones will attenuate 9dB. In other words; 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.

Matrix mixer and Route

There are two types of matrix; which is Matrix mixer and Route. The differences between them are two. Matrix mixer allow each output receive more than one input; but Router can’t. Meanwhile; matrix mixer can adjust send level of the crosspoint.



6 User Interface

6.1 Overview

User Interface is a key feature which allows you to create and design your UI surface according to the actual need for the application.

Through User Interface; it is much easier to visit each function or feedbacks from the DSP. There are many operations you can access such as adjusting the channel volume; switch to another source or recall presets.

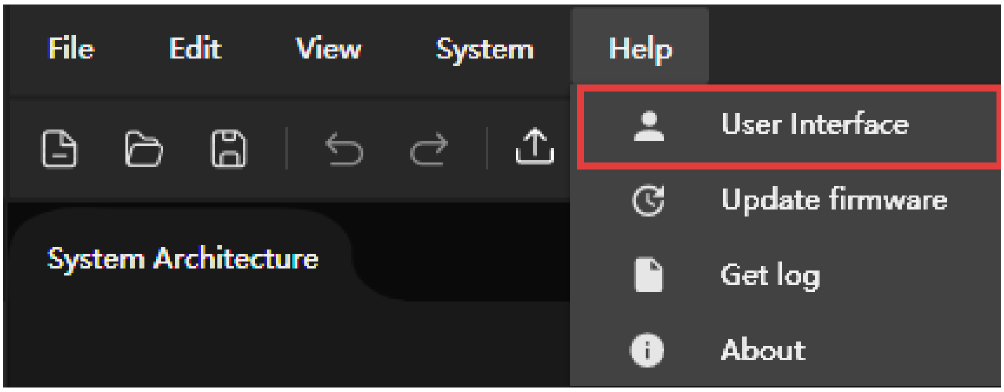
(1) Customizable: You are able to build your own interface so that you can meet different requirement.

(2) Simple: Intuitive pictures and components make it easy to operate as shown below. There is no need for complicate programming and technical knowledge.

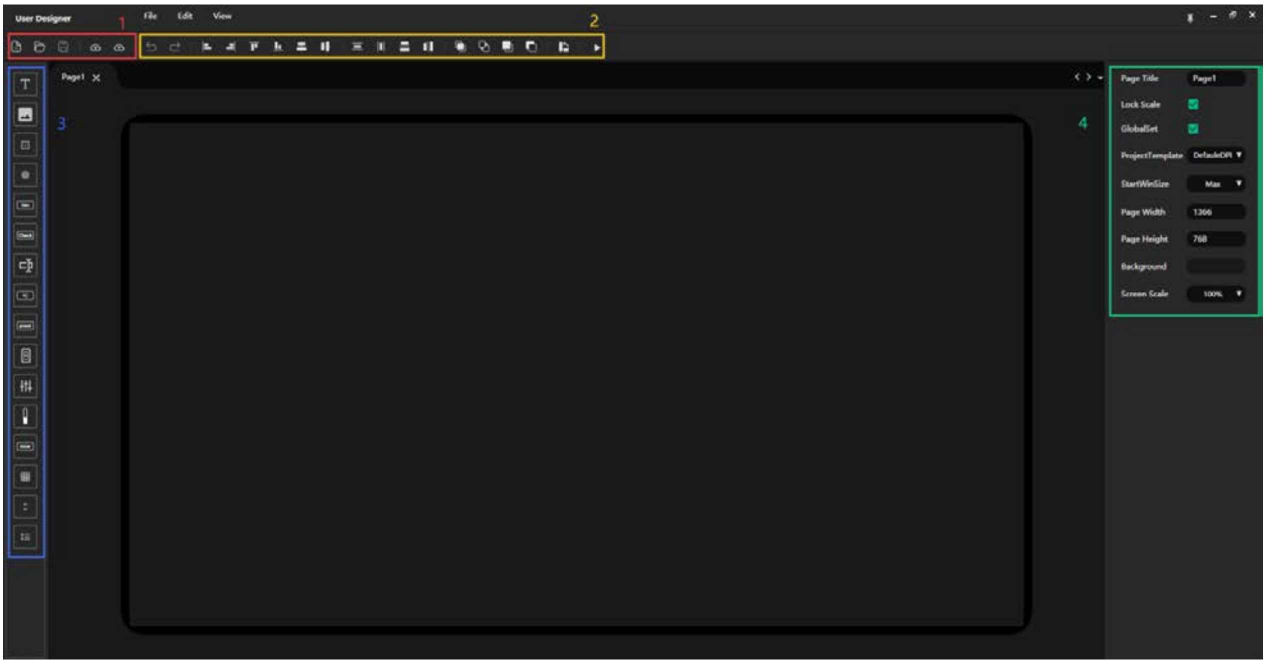
(3) Productive: Quick access for the users to the system improves productivity.

(4) Real time: Real-time feedback helps users better understand the status and activity of the system for better monitoring and control.

You can find the access of “User Interface”editing window under menu “Help”.



The editing window is shown below:



There are five areas in the User Interface editing window; 1 for Menu bar; 2 for Tool Bar; 3 for Components Windows; 4 for Properties Window and 5 for Editing Area.

Menu bar: File; Edit; View.

Tool Bar: New; Open; Save; Download the UCI to your local computer; Web control interface uploaded to the machine; Undo; Redo; Align Left; Align Right; Align Top; Align Bottom; Vertical Align Center; Horizontal Align Center; Equ-V-non distance; Equ-H-non distance; Equ-V distance; Equ-H distance; Bring Forward; Send to Back; Bring Top; Sent to Button; Rotate; Run.

Components: Text; Pic; Border; Button; Check Button; Volume Value; Mute Button; Preset Button; Channel; Gain; Level; Mixer Button; Matrix Mixer; Matrix Gain; Level Scale.

Properties: Modify properties of each component.

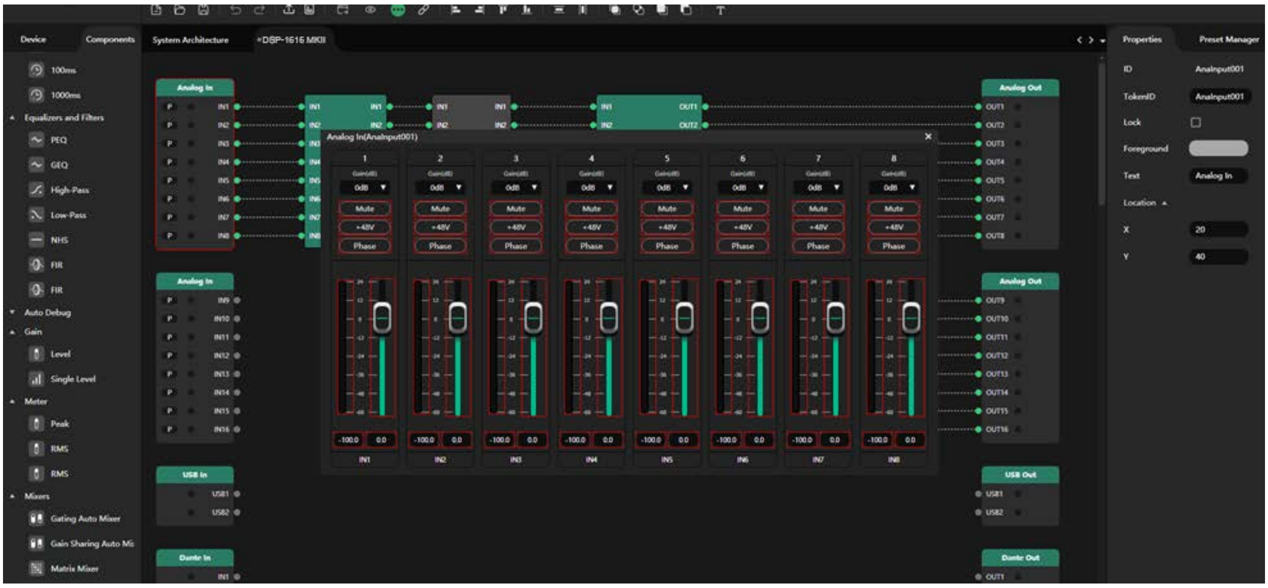
Editing Area: User Interface editing area; where all configuration of components takes place.

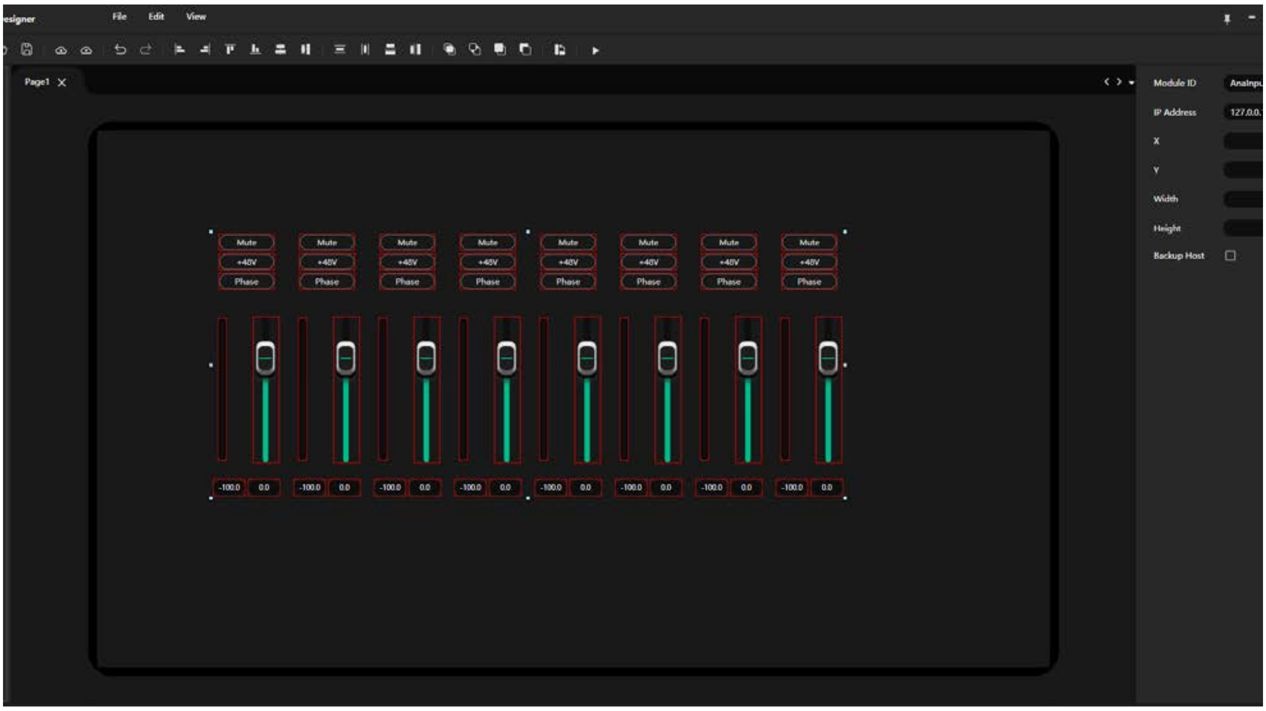
Note: The properties of the controls in the left component library are slightly different from those of the controls copied from the software. It's recommended that you copy the functional modules you need to from software directly.

6.2 Configuration

Design various controls by drag components into the editing area first. You can adjust color; size; position; and font of the components. Then you can edit the functionality of controls; by determining target device of the command and how they react to user input.

The other way is to copy module control components from the software. For example; click select or box select the components of the Analog Input; then use Copy (Ctrl+C); Paste (Ctrl+V) into the editing area of the user interface. As shown in the picture:



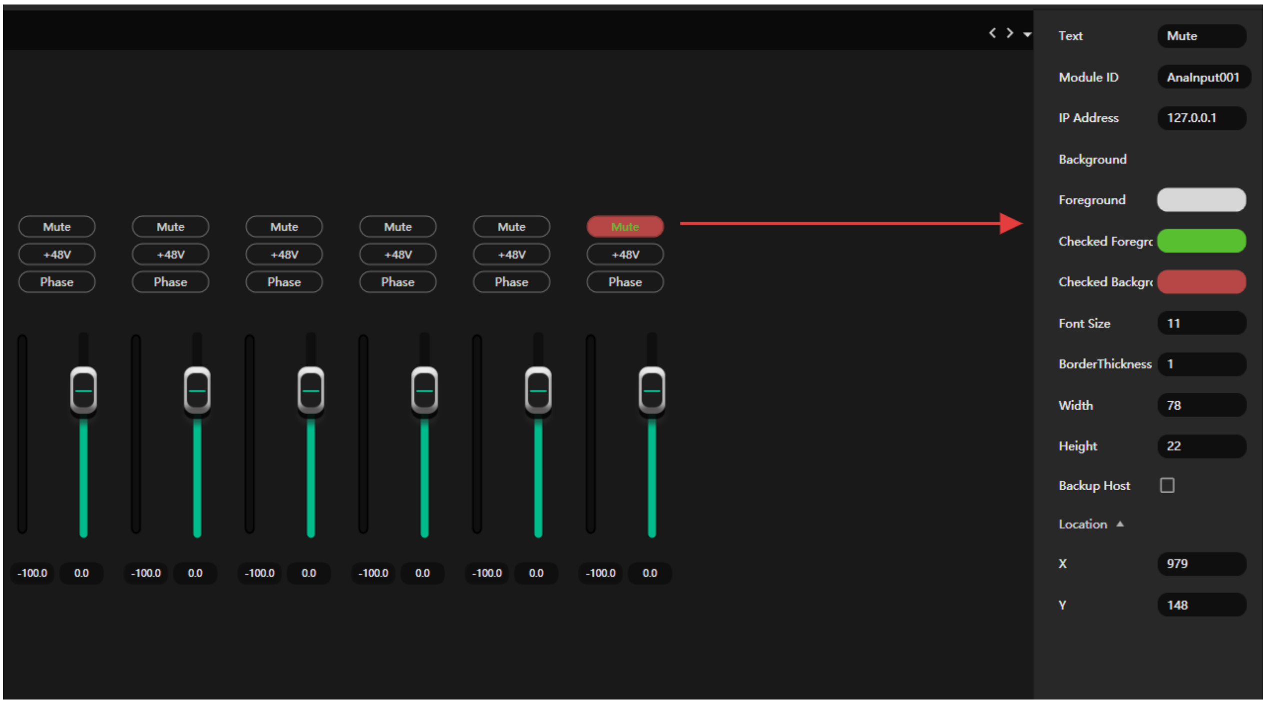


Note: User interface could not be edited until you press F7 or “Stop” first to take the project out of compiling or online mode.

If you open multiple software; you can only copy components from the software that opens the User Interface editing window.

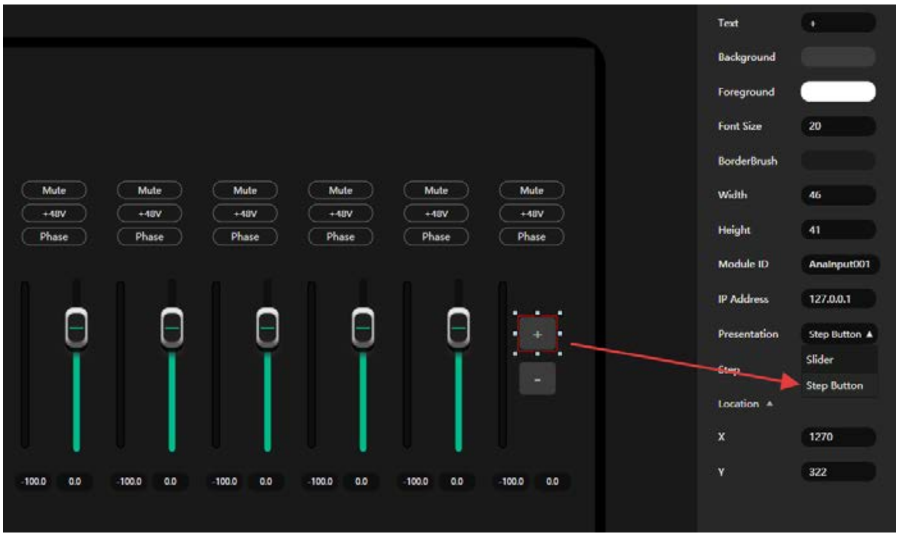
6.3 Basic Adjustment

You can specify properties such as font; color; size; orientation; and IP address of the corresponding device to be edited in the Properties Window on the right side.



6.4 Advanced Adjustment

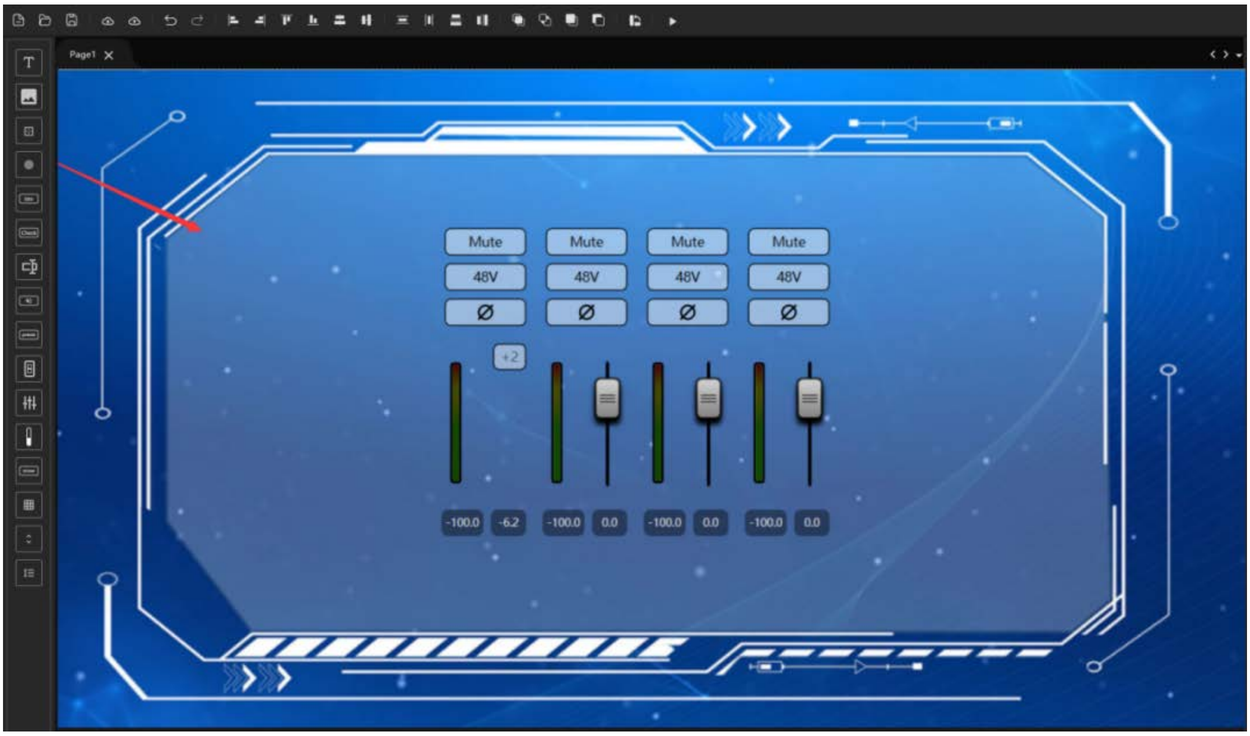
The control and display types of some component can be changed to suit the user prefer style. For example; the level fader can be changed to the mode of step button. After change the Presentation type from “Slider” to “Step Button”; and then modify the step size to “1”and “-1”;you can as shown in the following figure:

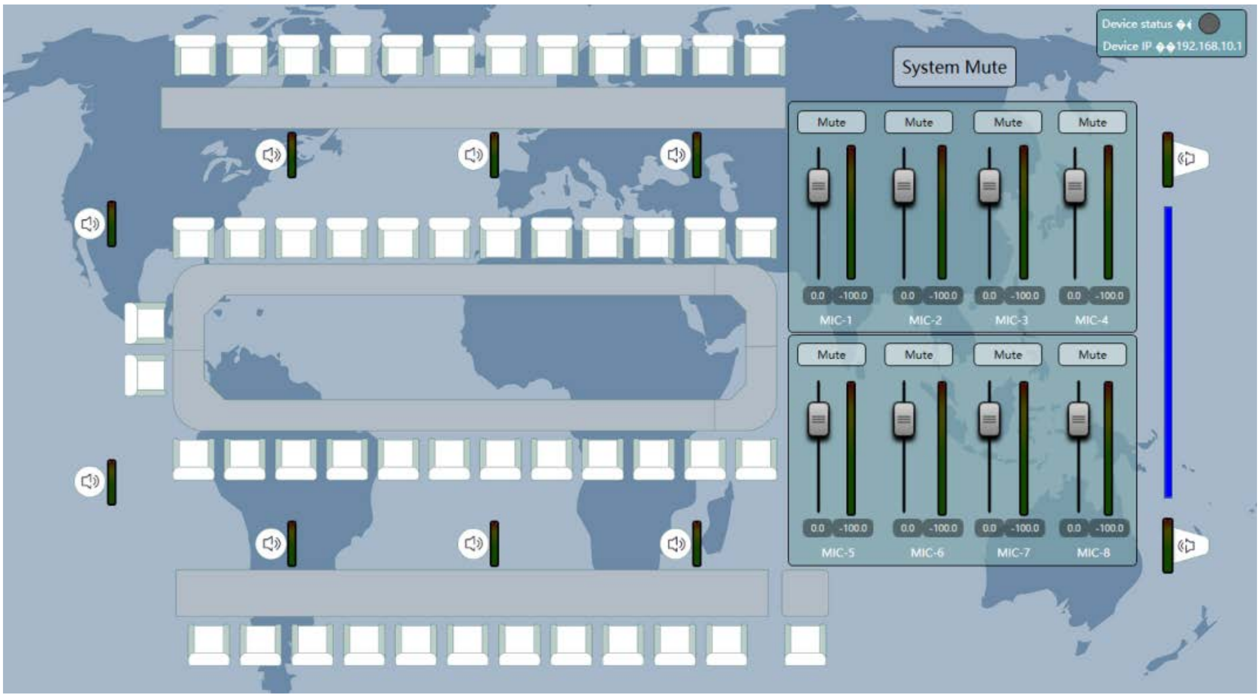


6.5 Picture

The user interface supports the layout of pictures; which can add more visual elements to the GUI interface; making the GUI interface more beautiful; intuitive and easy to understand. You can provide guidelines and hints in the GUI interface; mark important controls and operating areas; as well as provide instructions to facilitate quick understanding and use of the system.

Select a Pic component then you can select an image from the local computer path in the Properties Window; as shown below:


Page1
Mute
Mute
Mute
Mute
48V
48V
48V
48V
+2
-100.0
-6.2
-100.0
0.0
-100.0
0.0
-100.0
0.0


Device status
Device IP 192.168.10.1
System Mute
Mute
Mute
Mute
Mute
MIC-1
MIC-2
MIC-3
MIC-4
Mute
Mute
Mute
Mute
MIC-5
MIC-6
MIC-7
MIC-8