MP-RN1208D

Digital Sound Processor

Milestone PRO

User Manual

www.milestonepro.co.in

Contents

1. Overview

This document introduces a digital audio processing system, including both hardware and software components. The software adopts a drag-and-drop architecture for configuring audio processors and provides tools for device programming, control, and management. Its interface is simple and intuitive, allowing users to easily adjust module parameters, manage presets, and recall settings.

Key features of the system include drag-and-drop programming, a graphical control interface, dynamic signal flow visualization, and intelligent auto-snap feature. The signal flow is fully visualized, offering a “what you see is what you get” experience. End users can control the device via a web-based interface for more convenient remote operation.

On the hardware side, devices of the same model can be configured for dual-machine hot backup using a heartbeat detection mechanism. This ensures system reliability and stability in critical audio processing or sound reinforcement applications.

When using this software, users can configure and recall all parameters and presets within the entire audio system through simple mouse operations. The interface is intuitive and easy to navigate, enabling users to get started quickly.

This manual also provides detailed operating instructions to help users better understand the software's functions and applications, thereby enhancing overall usability and user satisfaction.

2. Hardware

2.1 Safety Instructions

1. Read these instructions.
2. Keep these instructions.
3. Heed all warnings.
4. Follow all instructions.
5. Do not use the device near water.
6. Clean only with a dry cloth.
7. Do not block any ventilation openings. Install in accordance with the manufacturer's instructions.
8. Do not install near any heat sources such as radiators, heat registers, stoves, or other equipment (including amplifiers) that produce heat.
9. Use proper grounding connections when connecting this device to a power outlet. Do not defeat the safety purpose of a polarized or grounding-type plug.

A polarized plug has two blades, one wider than the other. A grounding-type plug has two blades and a third grounding prong. The wide blade or the third prong is provided for your safety. If the provided plug does not fit into your outlet, consult an electrician to replace the obsolete outlet.

10. Protect the power cord from being walked on or pinched, particularly at plugs, convenience receptacles, and the point where they exit from the apparatus.
11. Only use attachments/accessories specified by the manufacturer.
12. Use only with a cart, stand, tripod, bracket, or table specified by the manufacturer, or sold with the apparatus. When using a cart, use caution when moving the cart/apparatus combination to avoid injury from tip-over.
13. Unplug during lightning storms or long periods unused.
14. Refer all servicing to qualified service personnel. Servicing is required when the apparatus has been damaged in any way, such as when the power-supply cord or plug is damaged, liquid has been spilled or objects have fallen into the apparatus, the apparatus has been exposed to rain or moisture, does not operate normally, or has been dropped.

The lightning flash with an arrowhead symbol within an equilateral triangle is intended to alert the user to the presence of uninsulated “dangerous voltage” inside the product enclosure that may be sufficient to constitute a risk of electric shock.

The exclamation point within an equilateral triangle is intended to alert the user to important operating and maintenance instructions included in the accompanying documentation.

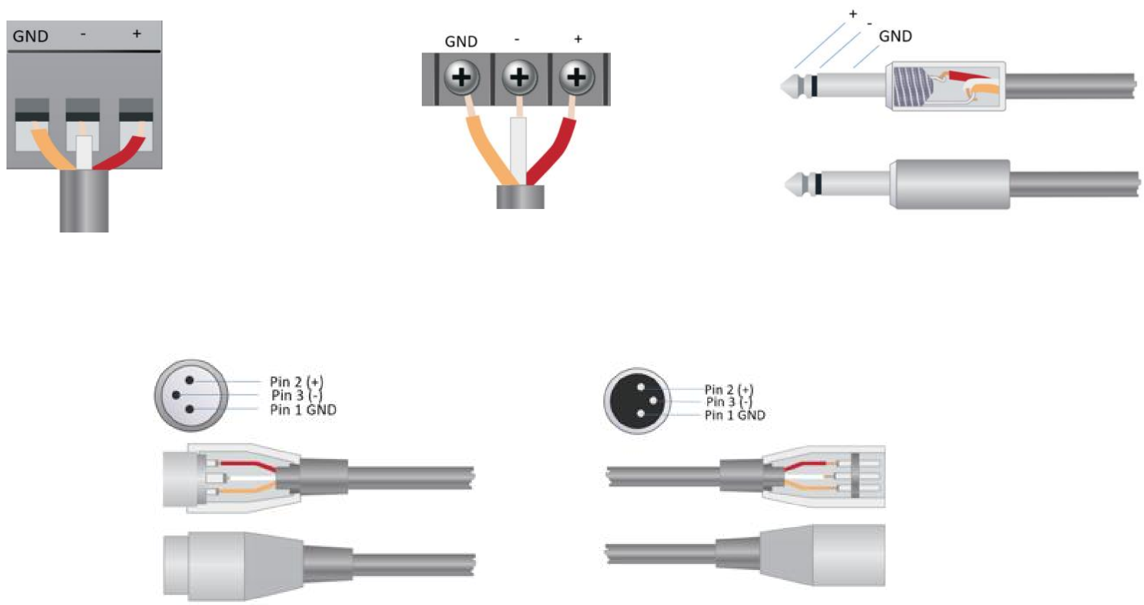
**Warning:** To reduce the risk of electric shock, do not defeat the safety purpose of the polarized plug provided with this equipment. Do not modify the plug or connect it to an extension cord, receptacle, or outlet unless the blades can be fully inserted.

2.2 Audio Analog and Network Wiring Reference

Balanced Analog Audio Connections

Any of these connectors may appear on either side of a balanced audio connection.

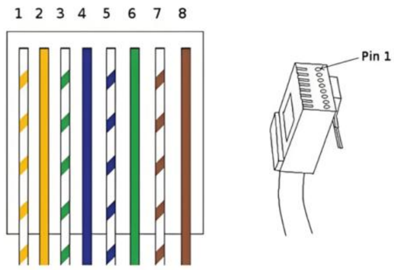
Note: For XLR connectors, the female connector is typically used for outputs, while the male connector is used for inputs.



Unbalanced Analog Audio Connections

RCA connectors and 1/4-inch TS connectors are unbalanced interfaces. They typically use a single-conductor shielded cable and may be used on either side of an unbalanced connection.

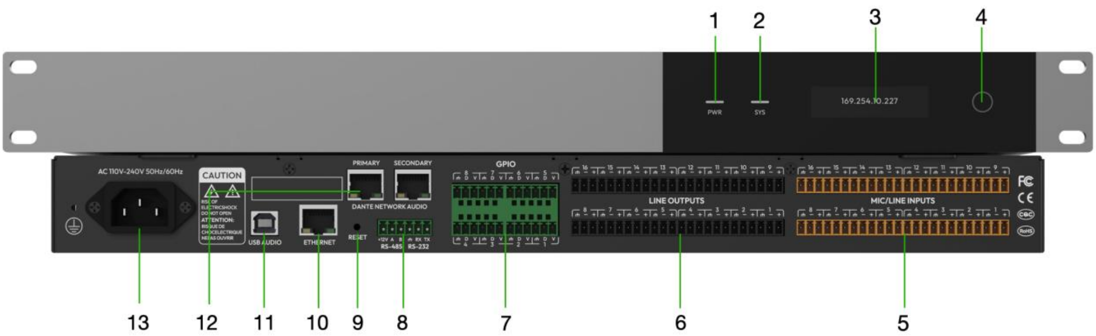
RJ45 Network Cable Pinout



T-568B

2.3 Hardware Introduction

For 1616 with Dante



(1) Power Indicator: steady white light indicates normal operation.

(2) Operation Status Indicator: blinking white light indicates normal operation.

(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) Analog Mic / line Input: 16-channel balanced analog audio input, independent preamp, phantom power.

(6) Analog Line Output: 16-channel balanced analog output, you can use the software to independently control the level and the mute.

(7) GPIO: Used to connect the control terminal or central control equipment.

(8) RS-232/RS-485: Used for the third-party device to control this equipment or for the equipment to control the third party device.

(9) RESET switch: Restore factory setting.

(10) Ethernet Port: used to connect the Open Designer software for programming, management and control of equipment

(11) USB-B sound card: to provide 2x2 audio, used to connect to the PC for remote meetings or for recording and playback.

(12) Dante Ports: Redundant 1000 Base-T Ethernet ports provides 64 (32x32) channels of Dante network audio.

(13) Power Supply: accept detachable IEC cable power supply (AC 110-240V, 50/60Hz, 60W maximum)

3. Operating Environment

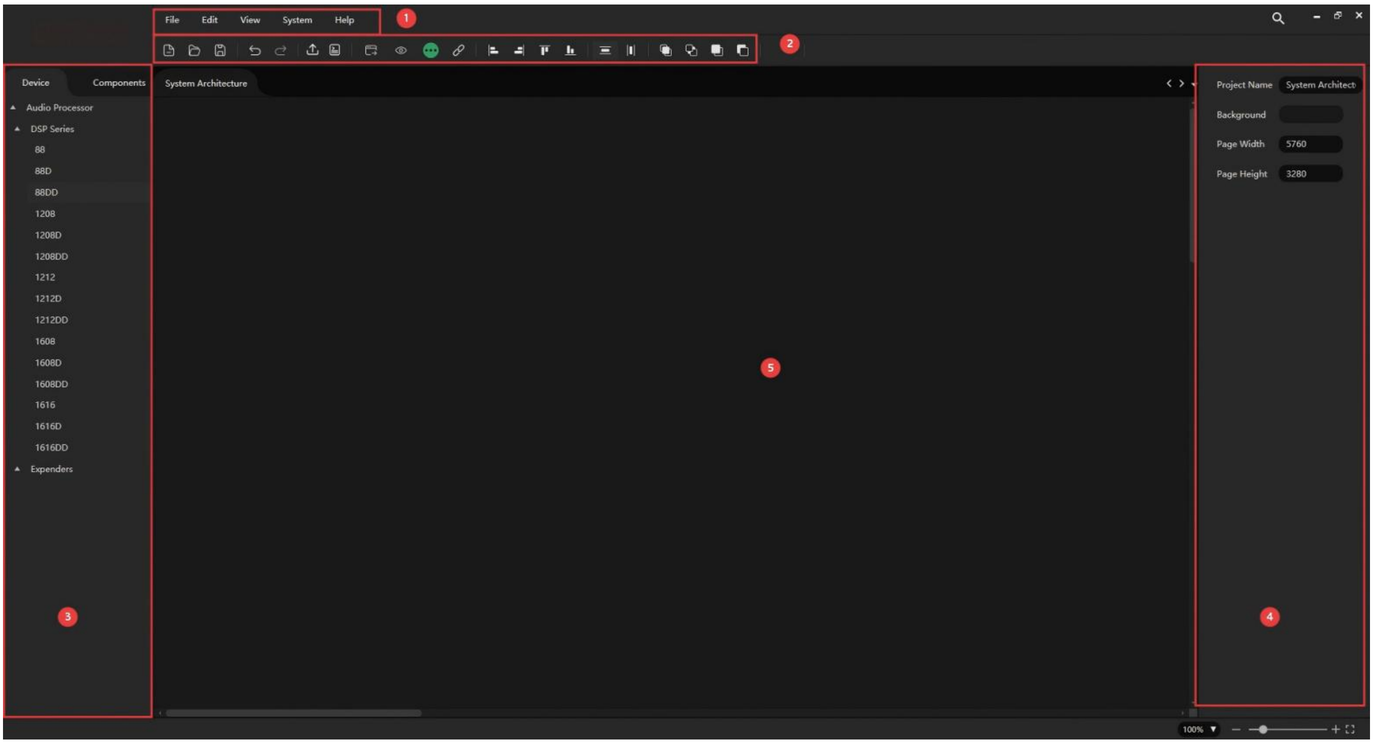
3.1 Hardware

- Processor: Intel Pentium/Celeron dual-core or above
- Memory: 4 GB or above
- Storage: 64 GB or above
- Resolution: 1920 * 1080 or above

3.2 Software

- Operating System: Windows 7 or above
- Environment Support: .NET Framework 4.5.1 or above

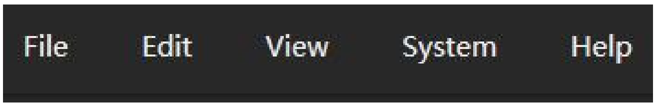
4. Software Interface Introduction



The software is divided into 5 main areas:

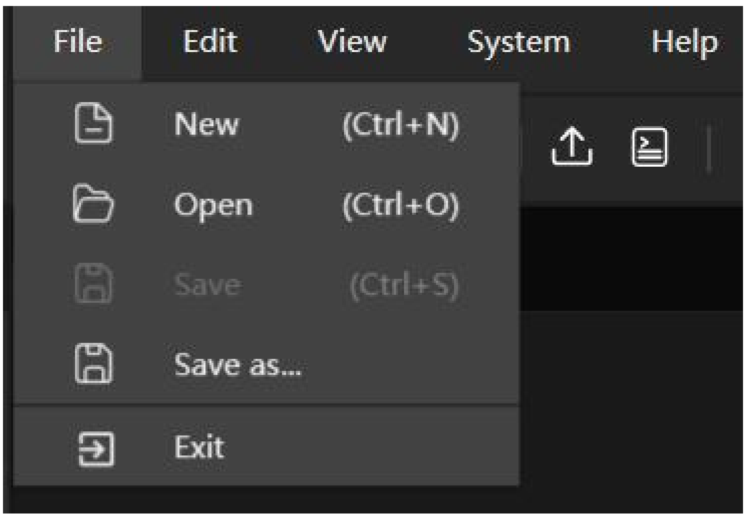
1.Menu bar 2. Toolbar 3. Device bar 4. Properties bar 5. Editing area

4.1 Menu Bar



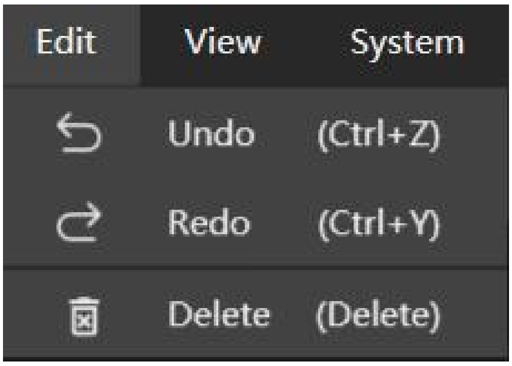
The menu bar contains five options: File, Edit, View, System, and Help.

(1) File



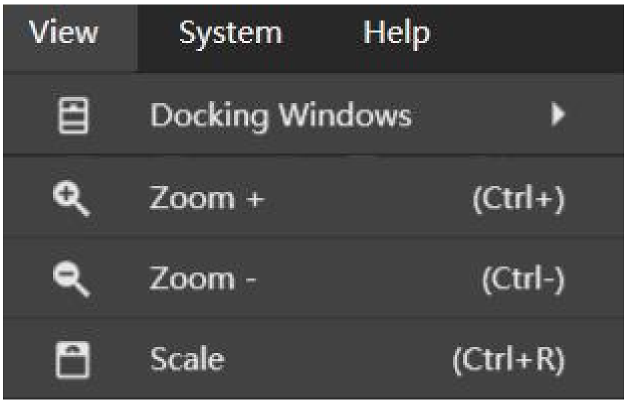
New: Creates a new project. If there are unsaved changes, the system will prompt you to save before proceeding.
Open: Opens an existing project from a selected path.
Save: Saves the current project to the default path.
Save as: Saves the current project to a specified path.
Exit: Exit control of the program and shut down the software.

(2) Edit



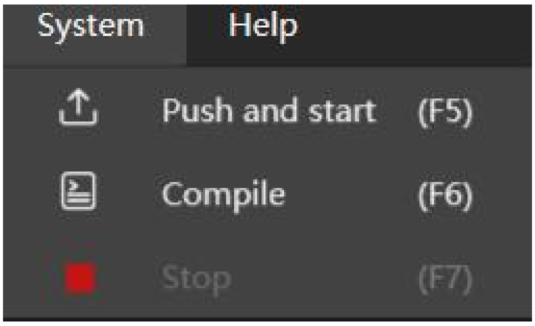
Undo: Reverts the last action. Shortcut: Ctrl + Z
Redo: Restores the last undone action. Shortcut: Ctrl + Y
Delete: Deletes the selected item. Shortcut: Delete

(3) View



Docking Windows: Dock windows and select which functions to display in the software interface.
Zoom In: Enlarges the editing viewport (up to 400%). Shortcut: Ctrl + or hold Ctrl and mouse wheel up.
Zoom Out: Reduces the editing viewport (down to 50%). Shortcut: Ctrl - or hold Ctrl and mouse wheel down.
Scale: Adjusts zoom proportionally. Shortcut: Ctrl + R

(4) System

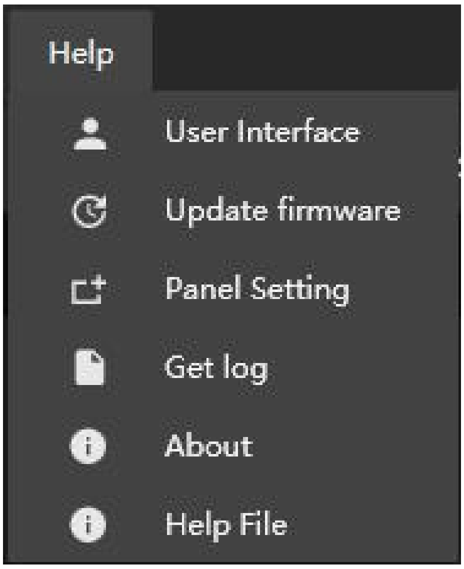


Push and start: Uploads the current edited program to the device and runs it. Shortcut: F5

Compile: Compiles the current program and enters simulation mode. The software is offline (non real-time) in this state. Shortcut: F6

Stop: Stops control of the online running program or simulation and back to the edit mode. Shortcut: F7

(5) Help



User Interface: Enter the user interface editor, where you can design and configure control functions and UI layout.

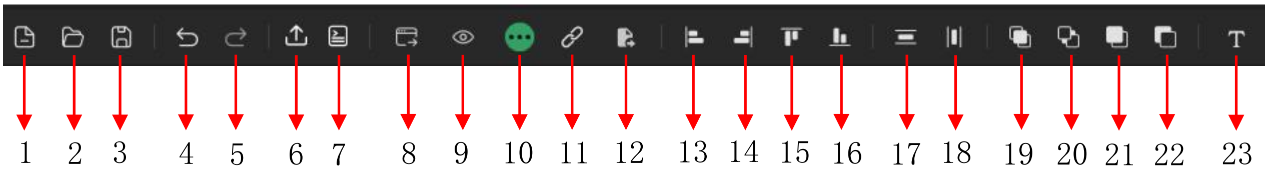
Update Firmware: Perform firmware upgrades for hardware devices directly within the software.

Panel Setting: Setup compatible control panels.

Get Log: Select the corresponding device and retrieve log records. The system saves up to six logs, each with a maximum size of 5 MB. A new log is generated after every power cycle. The seventh log will overwrite the oldest one in sequence(FIFO). To obtain them, the device must be connected; otherwise, logs remain stored only in the hardware.

About: Displays software information and version.

4.2 Tool Bar



(1) New Project: Creates a new project. If the current project contains unsaved changes, the system will prompt you to save it.

(2) Open Project: Opens an existing project from a selected file path.

(3) Save Project: Saves the current project to the default directory.

(4) Undo: Reverts the last action. Shortcut: Ctrl + Z

(5)Redo: Restores the previously undone action. Shortcut: Ctrl + Y

(6)Upload and Run: Uploads the program to the device and switches to real-time online mode. Shortcut: F5

(7)Compile: Required when adjusting parameters. Must be closed when adding modules or editing module connections. Shortcut: F6

(8)Output Window: Displays real-time compilation logs and system information (not audio output levels).

(9)Aerial View: Displays a overview of the entire programming layout.

(10)Signal Path: When enabled, selecting any connection highlights all associated signal paths from start to end.

(11)Auto Connect: When multiple modules are selected, this function automatically connects corresponding channels.

(12)Preset Edit Mode: When enabled (in non-compiled state), you may add or select a preset, then drag the desired modules from the program into the preset list to store their parameters. Multiple modules can be selected and dragged simultaneously.

Notice that when preset edit mode is enabled, all DSP modules are locked to preset editing functions only. Exit this mode to resume normal editing.

(13)Align Left: Aligns selected modules to the left.

(14)Align Right: Aligns selected modules to the right.

(15)Align Top: Aligns selected modules to the top edge.

(16)Align Bottom: Aligns selected modules to the bottom edge.

(17)Vertical Arrange: Moves selected modules into a vertically aligned, closely spaced arrangement.

(18)Horizontal Arrange: Moves selected modules into a horizontally aligned, closely spaced arrangement.

(19)Bring Forward: Moves the selected module one layer up.

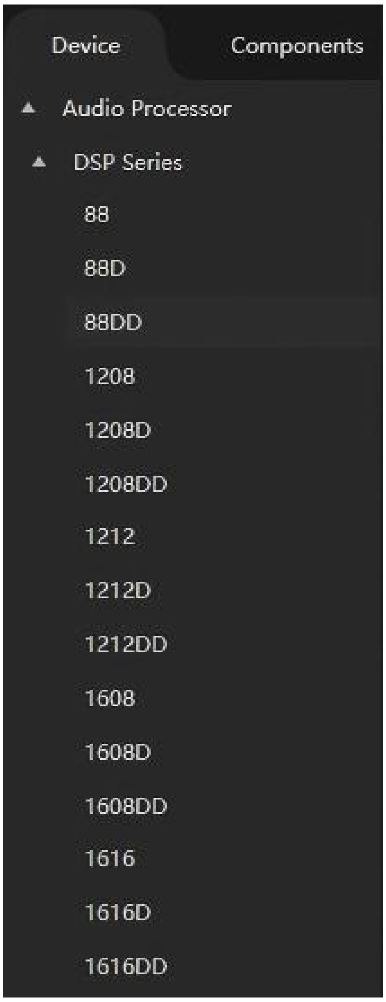
(20)Send Backward: Moves the selected module one layer down.

(21)Bring Top: Moves the selected module to the top layer.

(22)Send to Bottom: Moves the selected module to the bottom layer.

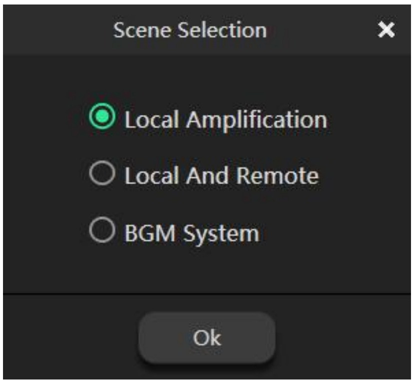
(23)Text: Adds a text note for annotation or comments.

4.3 Device Bar



The Device Bar lists all device models supported by the software. In offline editing mode, simply drag the desired device model into the editing area, then double-click it to enter its program.

4.4 Three Default Application Scenes

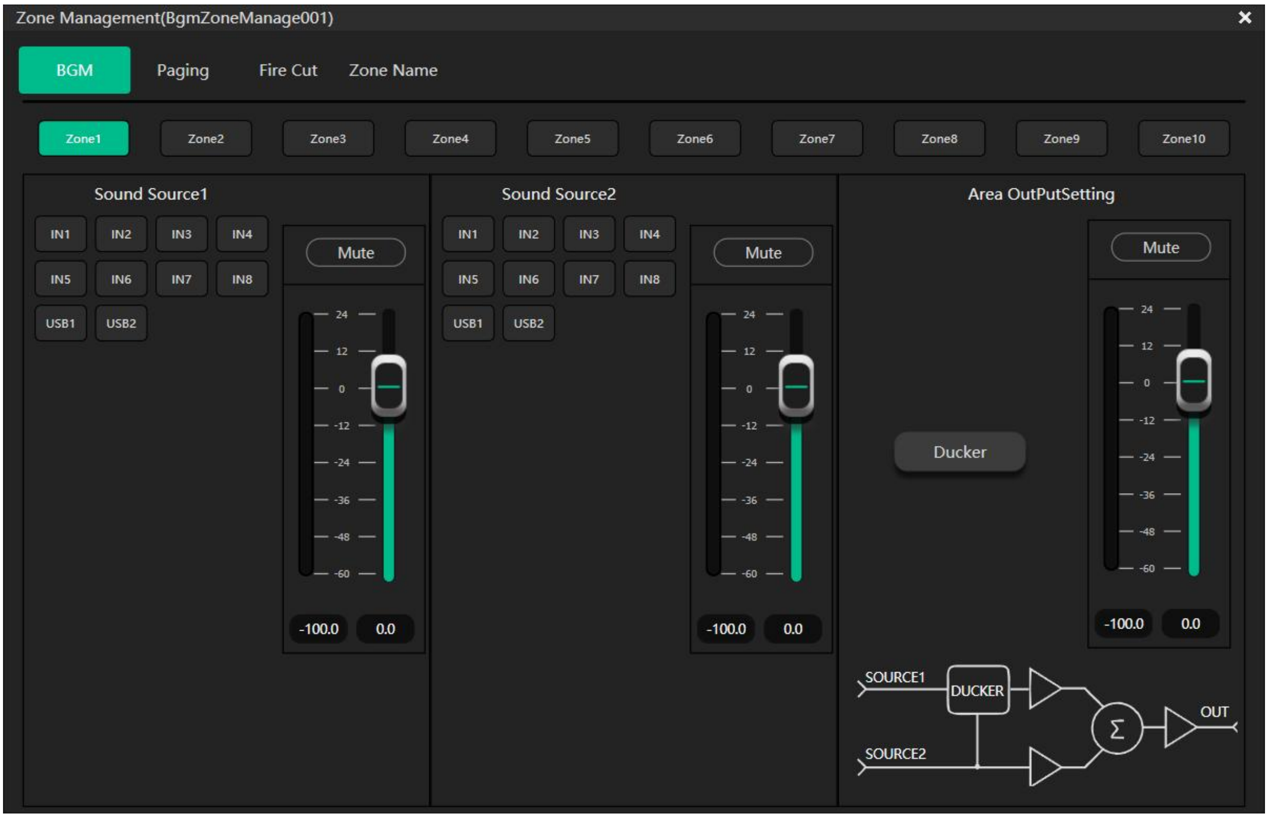


When a corresponding processor model is selected and you double-click to enter the program interface, the software will prompt you with a scene-selection window. Three default application modes are available:

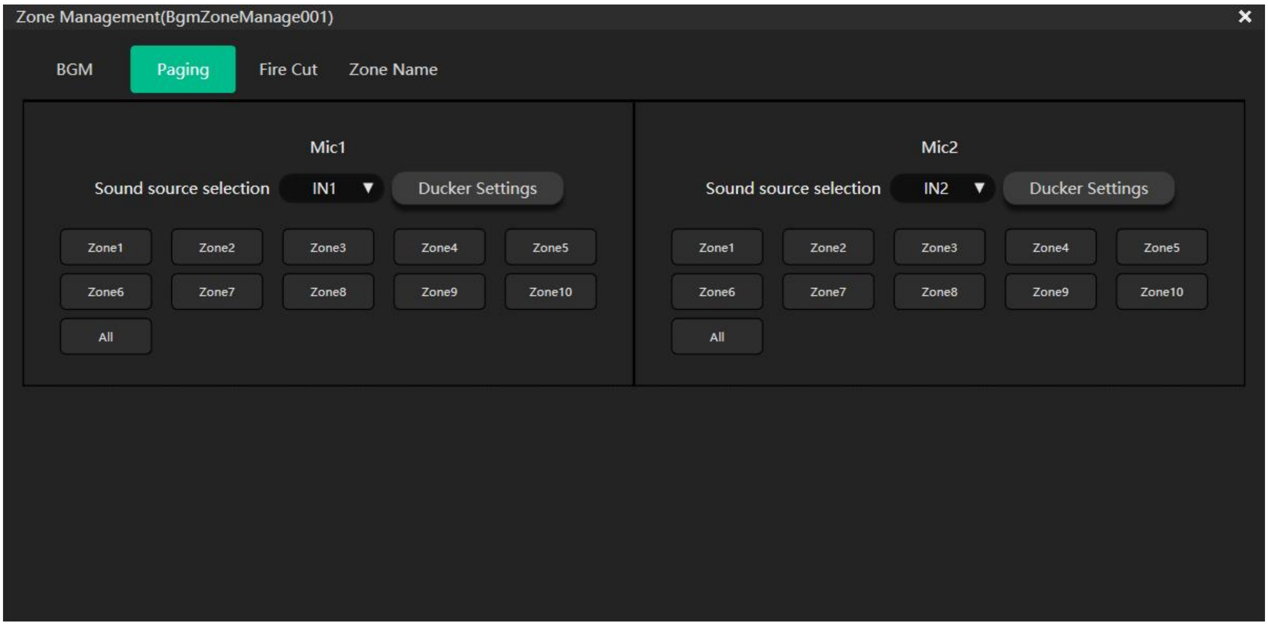
(1) Local Reinforcement: In the Local Reinforcement mode, the software automatically displays ATS auto-mixing and the corresponding matrix layout according to the selected device model. All wiring paths are preconfigured. All you need to do is dragging the required processing modules into the desired channels.

(2) Local + Remote: In the Local + Remote (conference) mode, ATS auto-mixing, AEC echo cancellation, matrix routing, and other key modules appear.
In this mode, the system splits all input signals into two paths: One path is preserved for local reinforcement; the other is processed through AEC in parallel to avoid affecting the local system. The AEC engine supports up to eight independent AEC channels. You only need to assign the input sources to the AEC channels through the matrix.

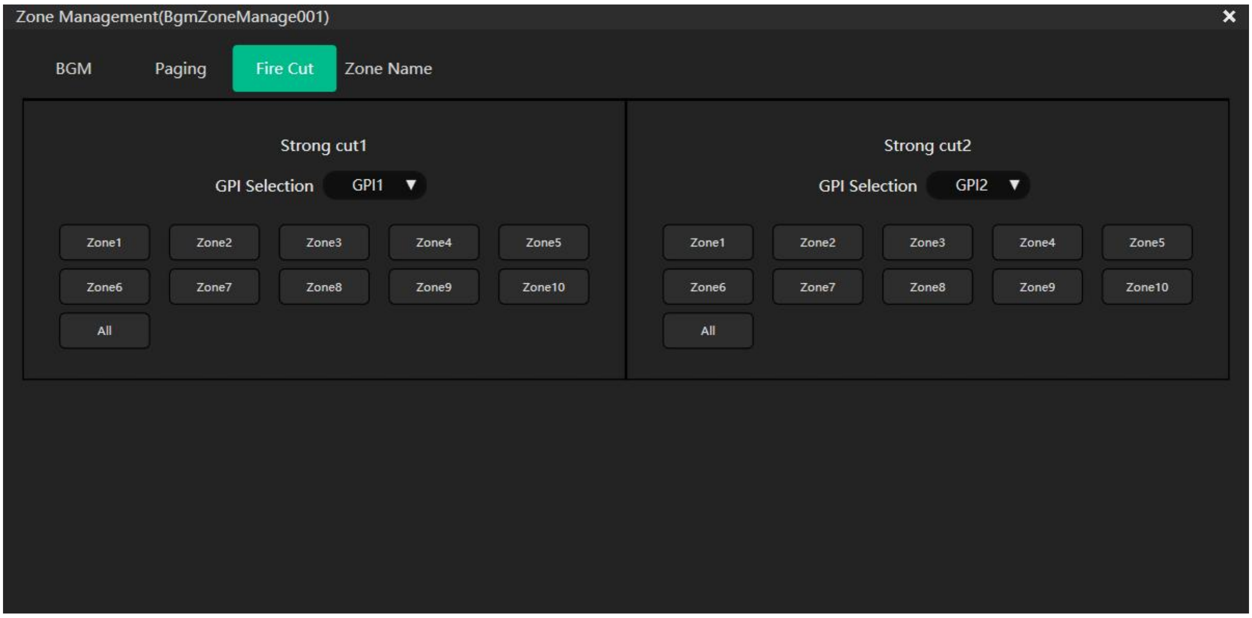
(3) BGM System: The BGM mode highlights the zone-management module, which controls background-music source selection and zone-level audio routing. You can configure the system according to usage requirements.
We provide two categories of audio sources, designed for central control room sources and zone-specific local sources.
Priority between central sources and zone sources can be configured (zone source priority higher than central source, or vice versa), depending on the application.



This mode also integrates a paging feature. When a paging microphone is connected and configured (including channel selection, parameters, and zone assignment), paging announcements will override any active BGM playback within the selected zones.



To address emergency signaling, the system supports fire-alarm linkage through GPI triggering. When an external fire system provides a trigger signal, the audio system will automatically mute the designated zones according to the configuration.



Note: Once an application is selected, it cannot be changed during the current programming session.

If a different scene is required from the original design, follow these steps:

(1) Load a new processor module.

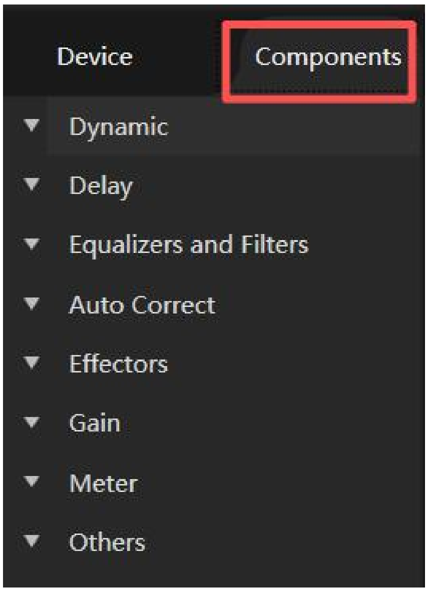
(2) Delete the previous processor module from the project.

(3) Assign the new module the same IP address as the physical device.

(4) Select the desired application mode.

(5) Upload the program again.

4.5 Component Library

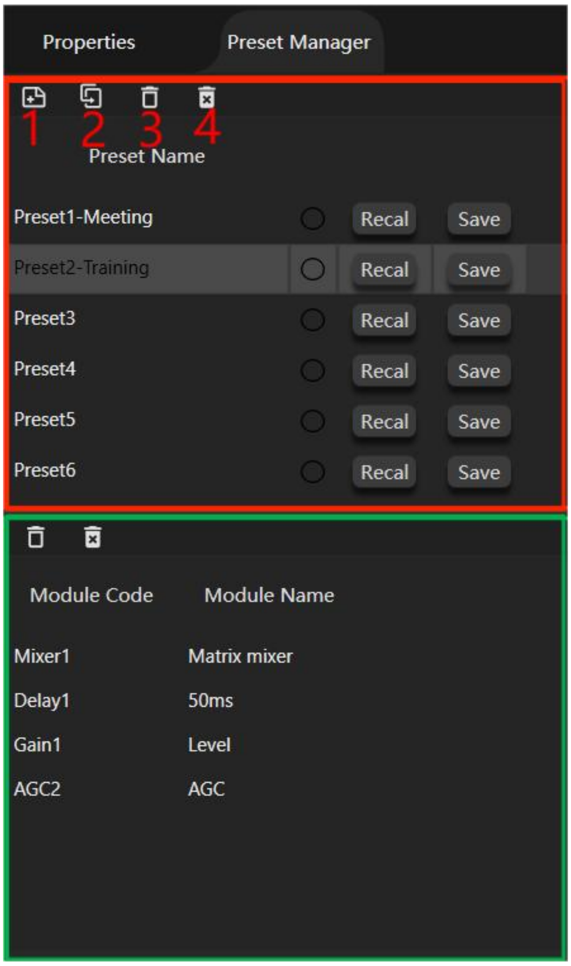


The Component Library displays all DSP processing modules of the model. Based on actual scenarios and application needs, you can select modules and drag them into the editing area for signal flow processing, editing, and debugging.

4.6 Properties Bar

In non-compile mode, when different modules are selected, a red border will appear. The module's parameters can then be edited in the "Properties" bar on the right side of the software. These include modifying module parameters, viewing or changing the ID, adjusting font color, position, and other attributes.

4.7 Preset Manager



(1) Add Preset Scene

(2) Duplicate Preset Scene

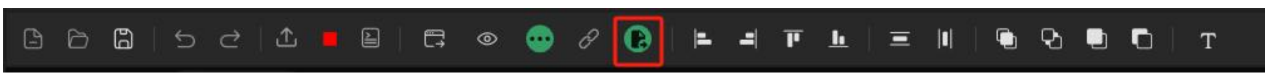
(3) Delete Selected Preset Scene

(4) Clear All Preset Scenes

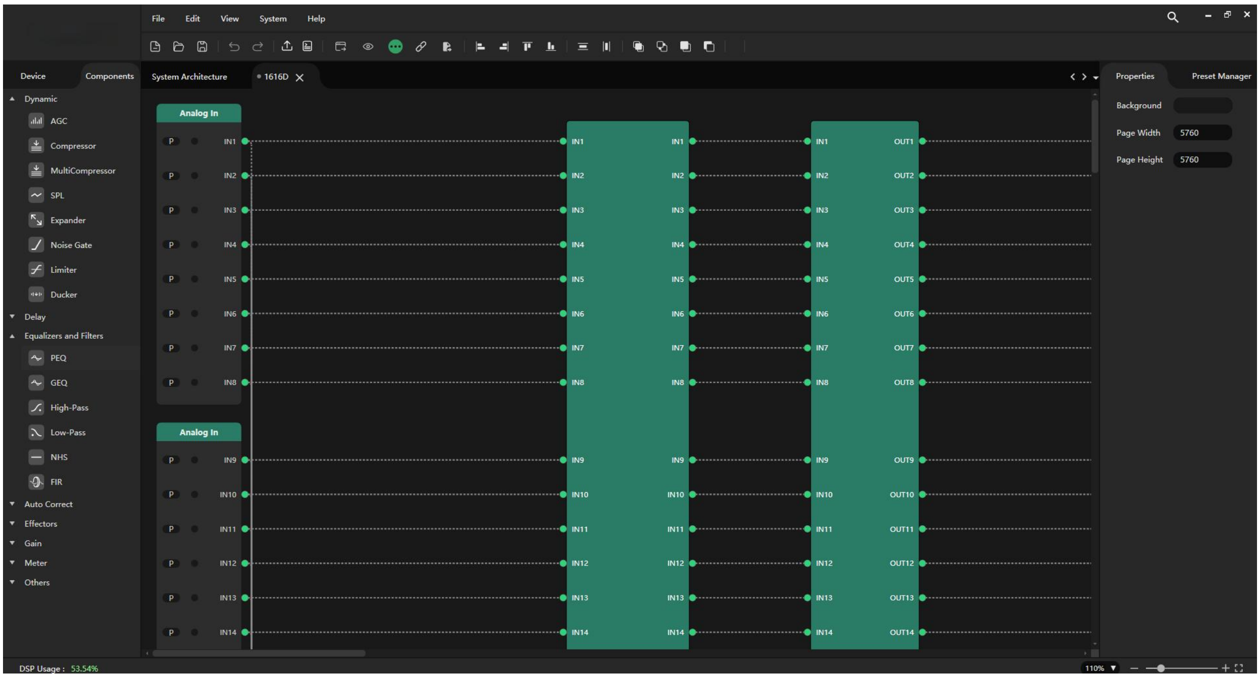
The red area represents the preset section, where you can add or select a preset and save or recall its contents. Double-click the preset name to rename it.

The green area shows the saved content of the selected preset. Modules can be directly dragged into this area to save their parameters for the preset.

Note: Before configuring any preset-related modules, you must enable **Preset Edit Mode** from the toolbar. After completing the preset configuration, turn off this mode to return to normal editing.



4.8 Editing Area

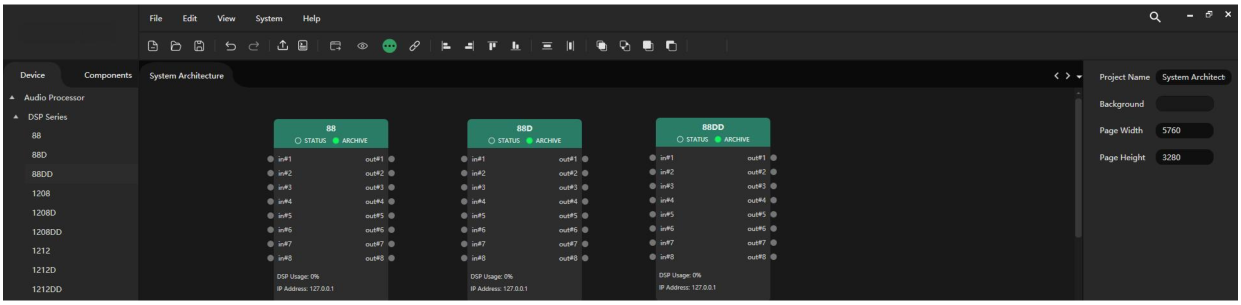


The central and largest area is the project editing area, where all DSP module editing, debugging, and connections are carried out.

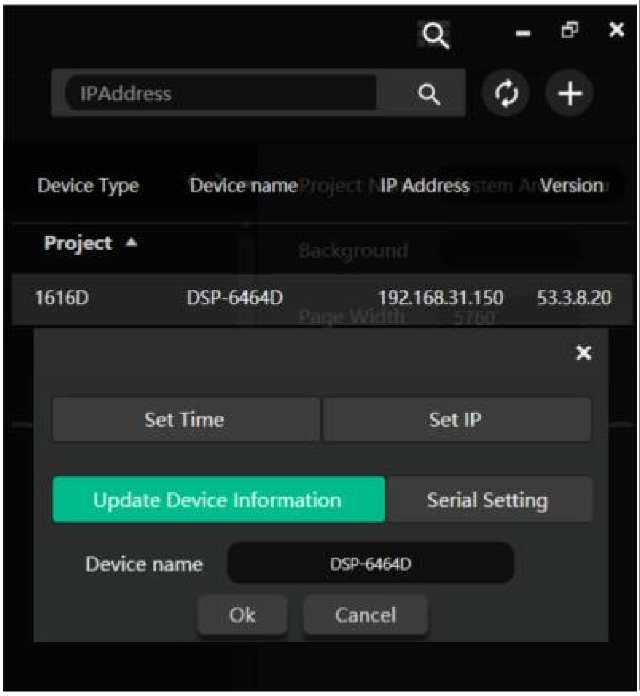
5 Software Usage

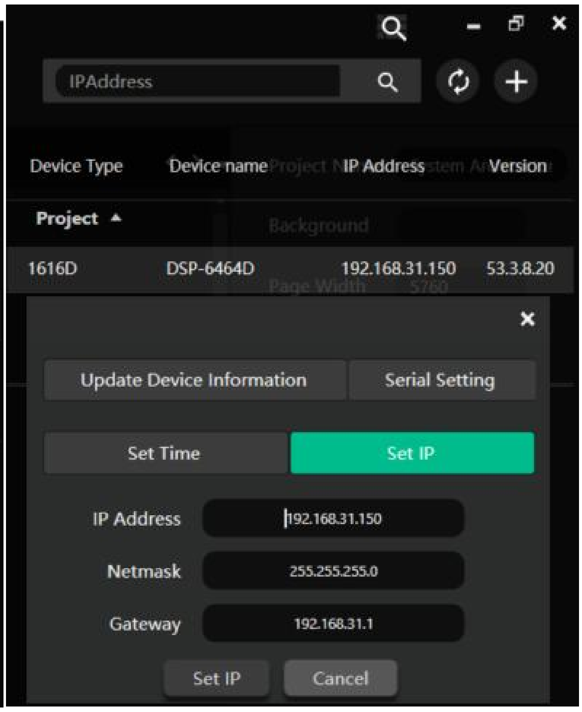
5.1 Device Discovery

In the upper right corner of the main software interface, click the Device Search button (the magnifying glass icon). A list will appear on the right side of the interface showing all devices detected on the network. The device discovery list displays the device type, device name, IP address, and firmware version of all recognized devices.



In the online device list on the right, double-click a device. In the pop-up dialog box, you can modify the device's basic information, serial port settings, and system time, as shown in the figure below:

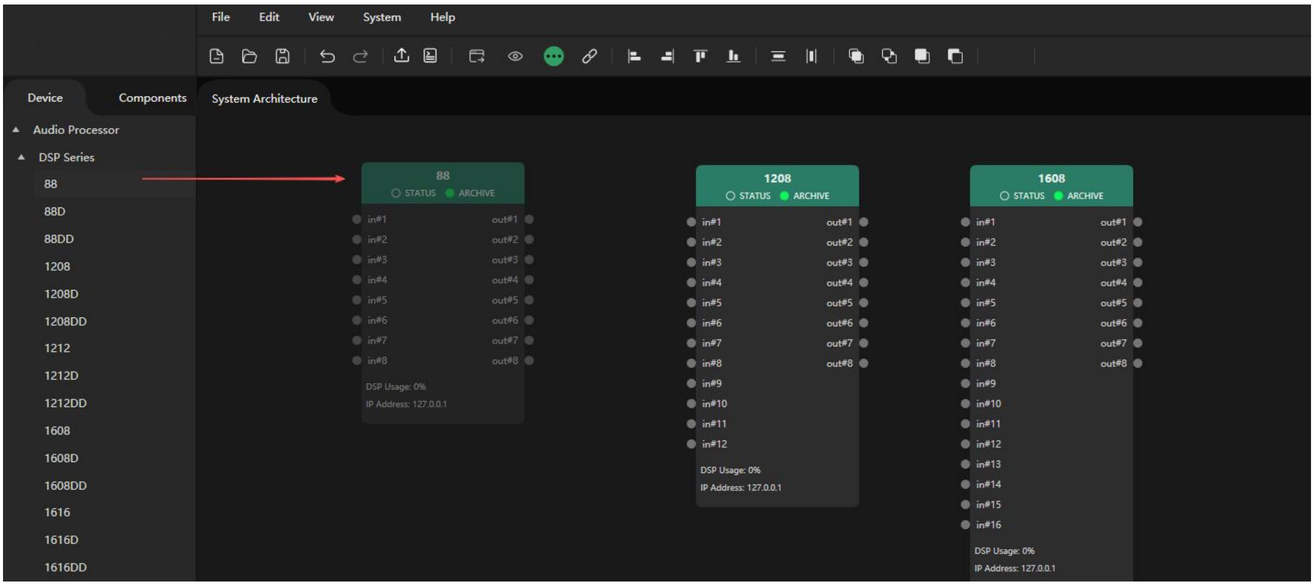




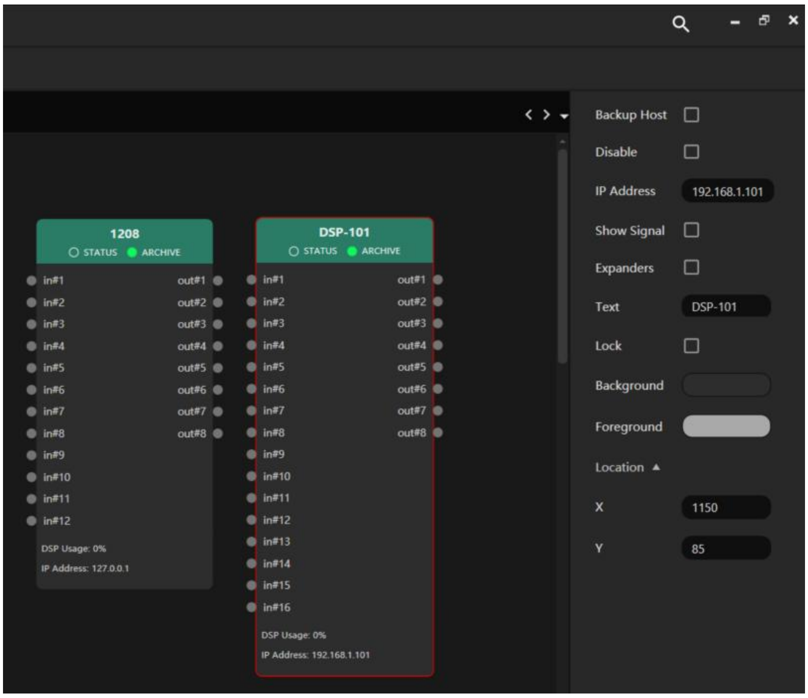
Note: As long as the devices are within the same LAN, they can be discovered in the Device Discovery list even if they are not on the same subnet. However, to perform program download or upload operations, the PC must be on the same subnet as the device. In addition, under the time settings, the device time can be synchronized with the computer time.

5.2 Add New Devices

In the system architecture interface of a program, you can select the target device from the device panel on the left side of the software's main interface, then holds down the left mouse button and drags it into the main window in the center of the software, as shown in the figure below:



Click to select the created device module, and the property configuration of the created device module will be displayed in the properties bar on the right side of the window, allowing you to configure the following:

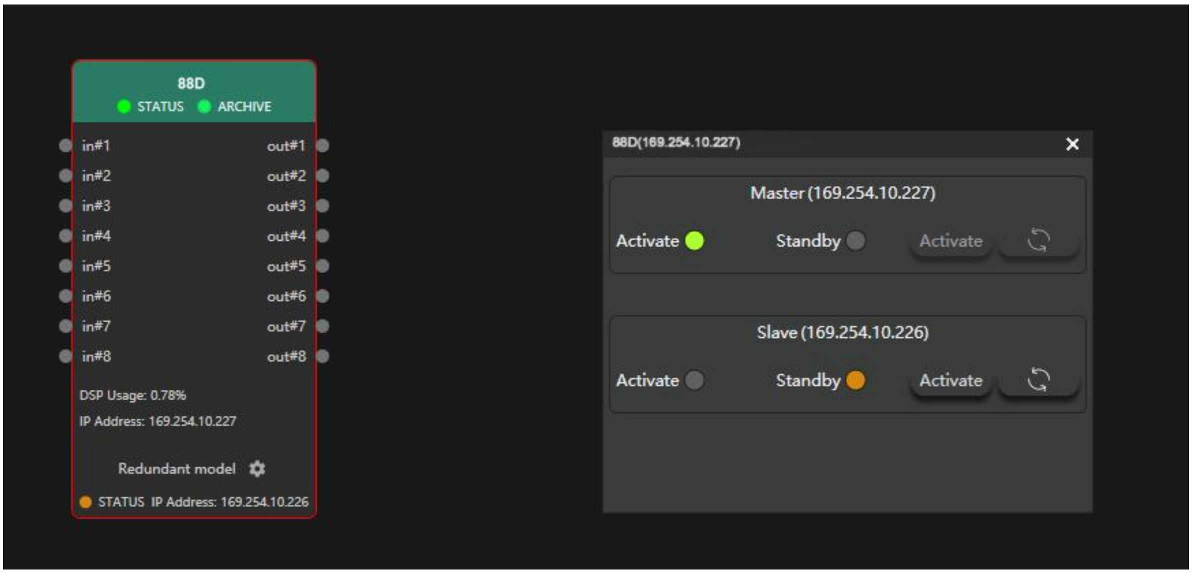


5.3 Hot-standby (Dual Redundancy)

When configuring dual-device hot-standby, select the “Backup Host” option and an input field for the backup device's IP address will appear. Simply enter the IP address of the backup one. Please note that the master and slave devices must be of the same model; devices of different models cannot synchronize channels.

The hot-standby system uses a heartbeat packet mechanism for mutual handshake detection. Both network and RS-232 detection methods are supported. By default, network detection is enabled, with the query interval set to a minimum of 1 second (this value can be adjusted). Under the condition that both devices are the same model and set as mutual backups, they will share the same project file. In other words, the backup device does not require separate programming. While the primary device is active and outputting signals, the backup device remains silent. Once the primary device loses the heartbeat handshake, the system will automatically switch to the backup device, and the backup device will immediately release the audio signal.

Additionally, after enabling the hot-standby system setting, a “Redundant model” option will appear on the device module. By opening the settings on it, you can view the status of both master and slave devices, showing whether they are activated or ready. Manual switching between the devices is also supported.



5.4 IP Address

The IP address of the designed module should be set corresponding to the target device hardware (which IP can be viewed and set in the device discovery list);
Meanwhile, when dragging a device from the discovery list to a designed module (instead of blank space) quickly assigns it an IP address.
Note: Incorrect IP configuration here will prevent the program from being uploaded to the processor normally.

5.5 Show Signal

When enabled, in the program, when there is an audio signal, the line will display a real-time dynamic audio signal flow, allowing us to quickly monitor the signal.

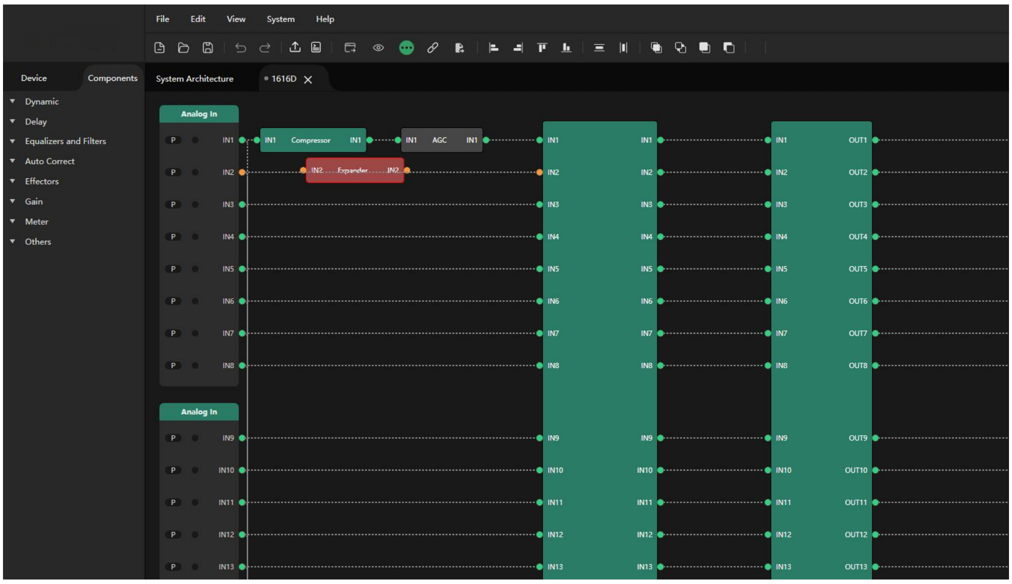
5.6 Text and Colors

The display name of the device module, as well as changes to the module and font display colors.
The software supports designing and managing programs for multiple devices under a same program. To edit or control each device, simply double-click the device module or the name of an already opened device's standalone tab to enter the program design page of that device. Additionally, it can be saved as an integrated program file or stored in each individual device.

5.7 Detailed Programming Workflow

5.7.1 Module Connections

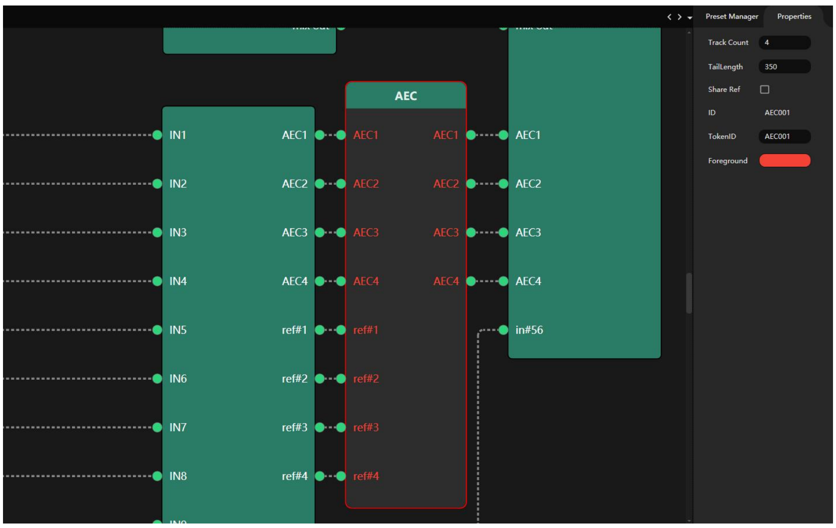
When entering the programming interface, all inputs and outputs and the related application structure are connected. Based on this default layout, you may expand the processing module library on the left panel, select any processing block, and drag it onto the target node of the signal path. The software will display an orange dual-ended marker indicating where the module will be inserted, and the system will automatically align and snap it into place for a more user-friendly programming experience.



If a module is inserted by mistake or is no longer required, simply select and delete it using keyboard. To reposition any processing module, drag it to the desired node, and no additional rewiring is required. The system intelligently manages signal snapping and layout organization. When the number of modules increases, the interface automatically applies overflow management to maintain module spacing and alignment, ensuring a clean and intuitive signal-flow overview.

5.7.2 Editing Module Properties

Click a module to display its property configuration panel on the right side of the window. You may modify attributes including label text, processing band count, text color, and other related settings.

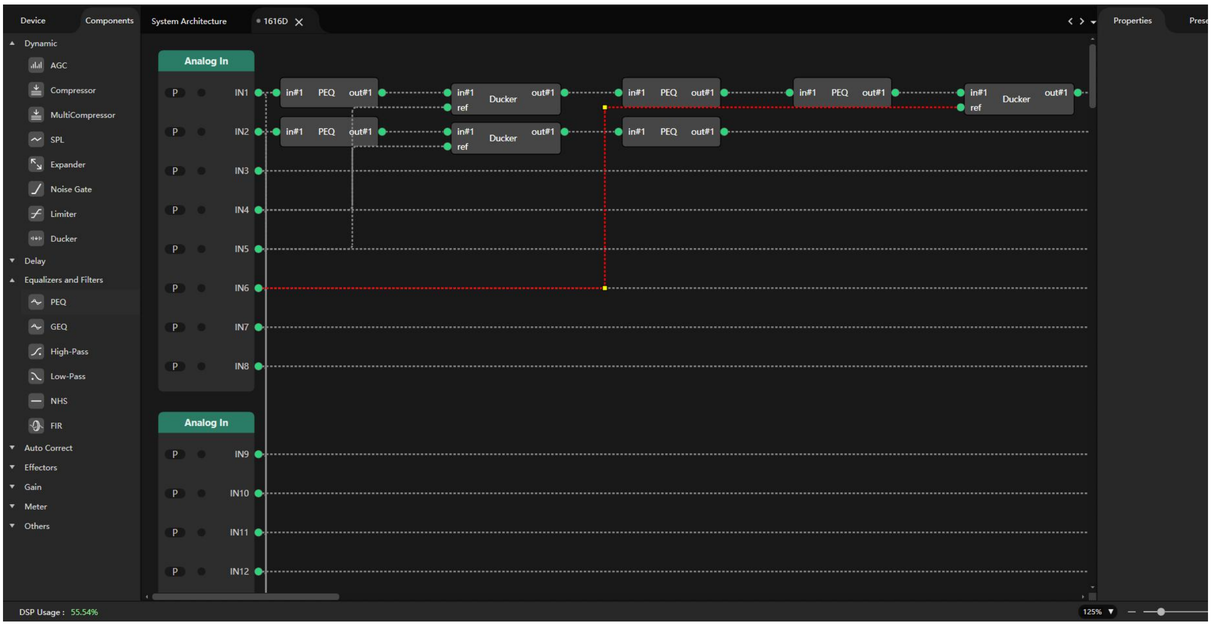


5.7.3 Component Library

The software provides a wide range of DSP audio processing modules. Once inside the device module interface, you can drag required processing blocks from the component library on the left to the appropriate channel to complete audio routing, control, and signal processing.

Available modules include:
Level control, level meters, HP/LP filters, parametric EQ, feedback suppressors, delay processors, compressors, FIR filters, reverb and echo effects, and more.

Two modules: Continuity SPL and Priority Ducker, require a reference signal input. Their reference input pins are exposed for manual connection by you. These are the only modules that require manual wiring.

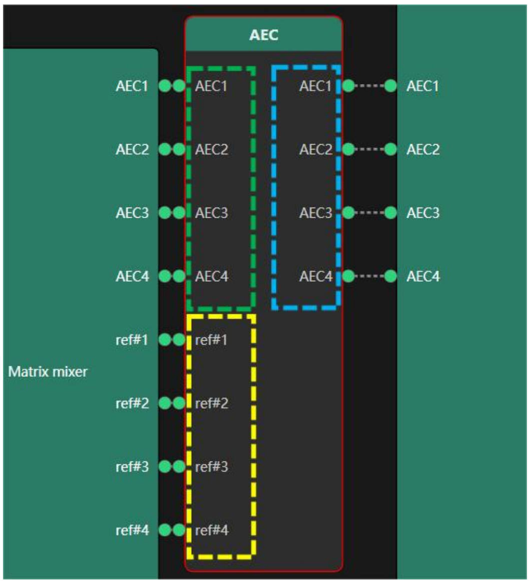


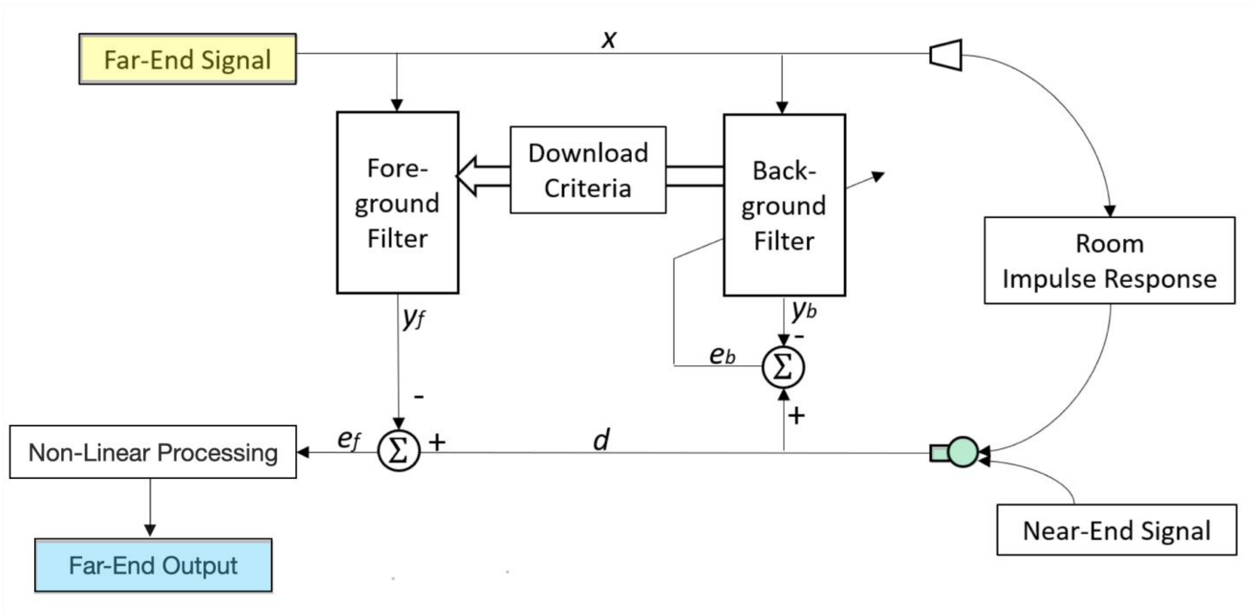
5.7.4 AEC (Acoustic Echo Cancellation)

AEC is used to eliminate acoustic echo in two-way audio communication. In conferencing scenarios, remote speech may be picked up again by local microphones and sent back to the far end, forming audible echo. AEC applies adaptive algorithms to detect and remove this echo in real time, improving clarity and stability.

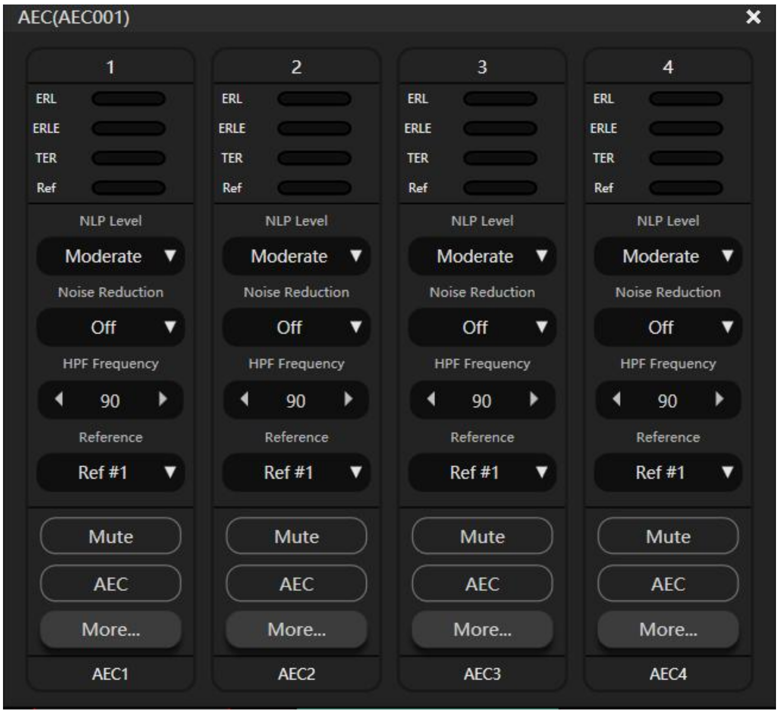
AEC is widely applied in video conferencing, voice communication, and remote education, anywhere that full-duplex audio is required. The processor uses a dedicated high-performance chipset to run AEC algorithms, removing unwanted echo by referencing the far-end signal and subtracting its residual presence from the microphone channel.

Since the purpose of AEC is to prevent the far-end from hearing their own voice, the algorithm operates primarily for remote-side clarity.





Each AEC channel includes a corresponding AEC Ref module. Assign the far-end signal as the reference and the local microphone signal as the AEC input. Once enabled, the system computes automatically. Adjustments to microphones, loudspeakers, or gain structure may trigger brief recalculation.



The AEC module also includes AGC that can be used to maintain consistent far-end signal level.

ERL (Echo Return Loss): Natural attenuation between the reference (far-end) signal and its echo picked up by the microphone before AEC is applied. ERL < 10 dB may indicate poor acoustic design.

ERLE (Echo Return Loss Enhancement): Additional attenuation achieved by the AEC algorithm. Higher values indicate better echo suppression.

TER (ERL + ERLE): Total echo reduction from source to AEC output.

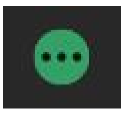
REF: Required far-end reference signal. Excessive nonlinear distortion will degrade AEC performance.

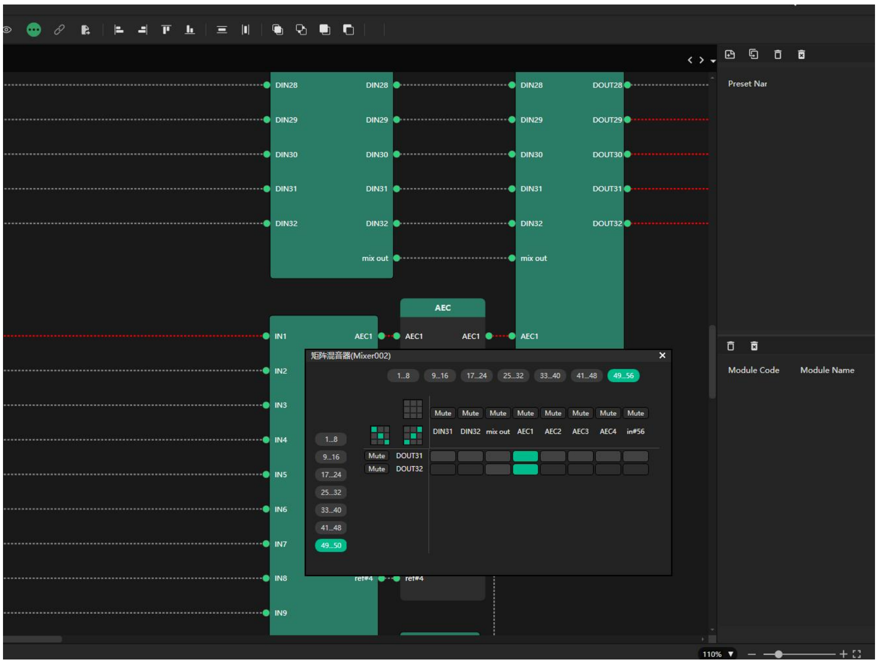
NLP Level: Non-linear processing extends echo suppression when linear AEC cannot fully eliminate nonlinear residual echo. Three modes: Conservative, Moderate (default), Aggressive.

5.7.5 Signal Path Trace

In complex audio systems, users can quickly locate the signal path for troubleshooting.

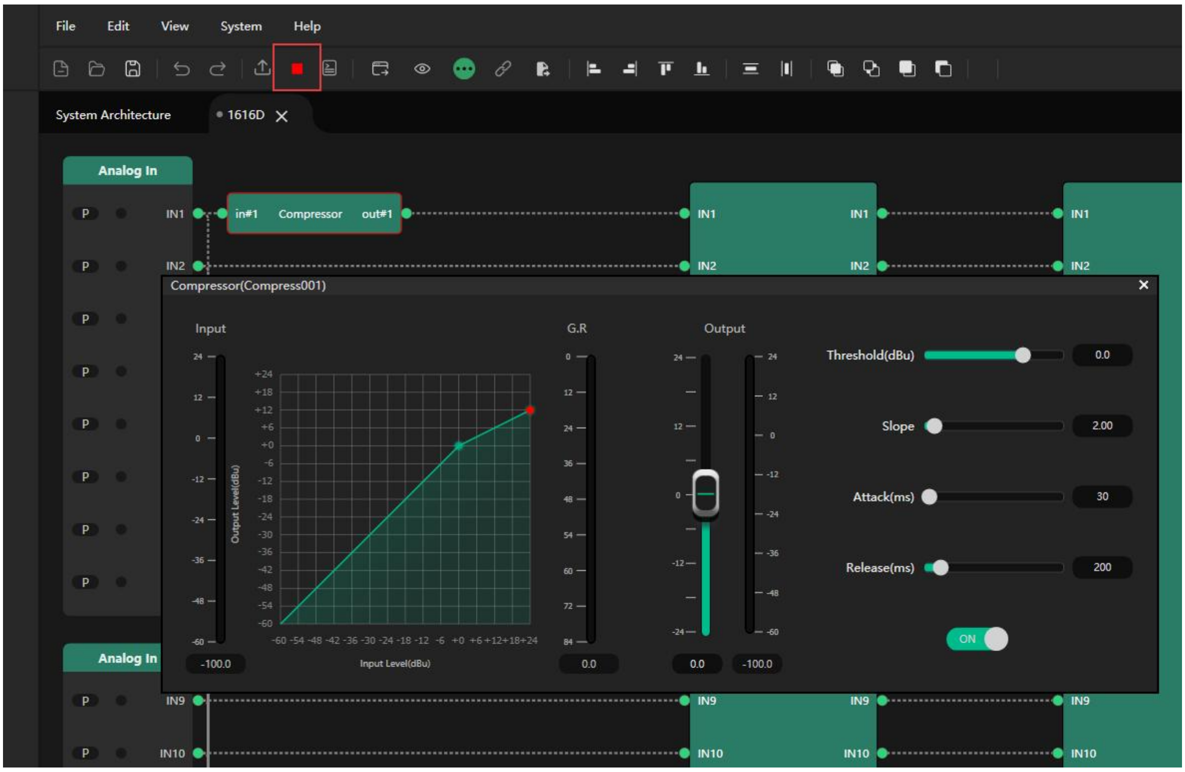
Enable "Signal Path" in the toolbar, then click any desired node. The full signal flow will be highlighted in red.





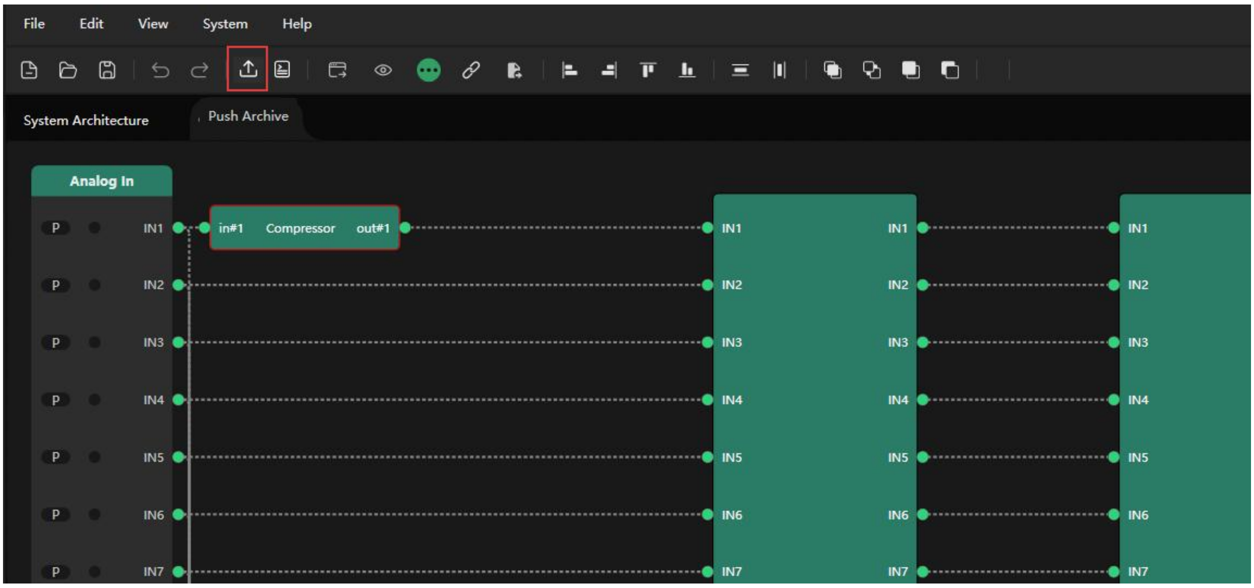
5.7.6 Simulation Run

After completing the program design, users may compile and run a offline simulation by clicking the compile icon or pressing F6. Parameters become adjustable only after successful compilation.



5.7.7 Uploading / Stopping Program

Once design is complete, upload it to the processor by clicking the upload icon or pressing F5.



After successful upload, the toolbar changes from Compile Mode to Run Mode, also device status indicator turns green, during which modules cannot be moved or added. Press F7 or stop execution to resume editing.

Notes:

(1) DSP usage shown in the lower-left corner must not exceed 100%.

(2) Device module IP must match the actual hardware IP.

(3) The color of the module represent:

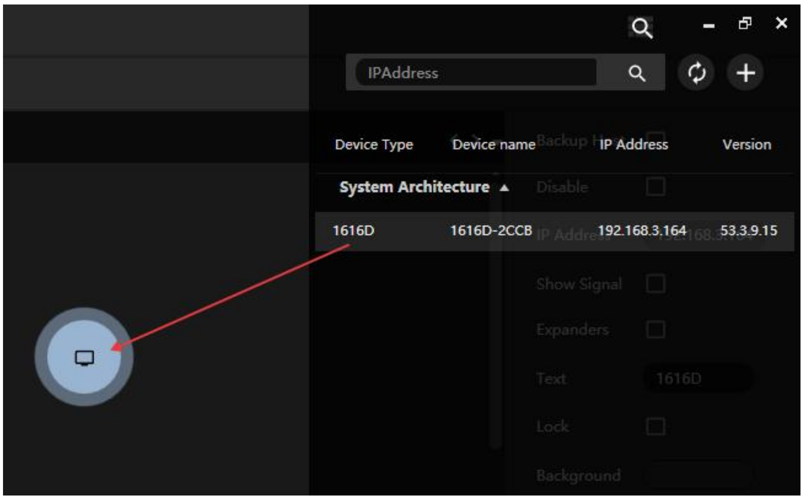
Grey = connected but disabled;

Green = connected and active;

Red = unconnected.

5.7.8 Downloading Program From Devices

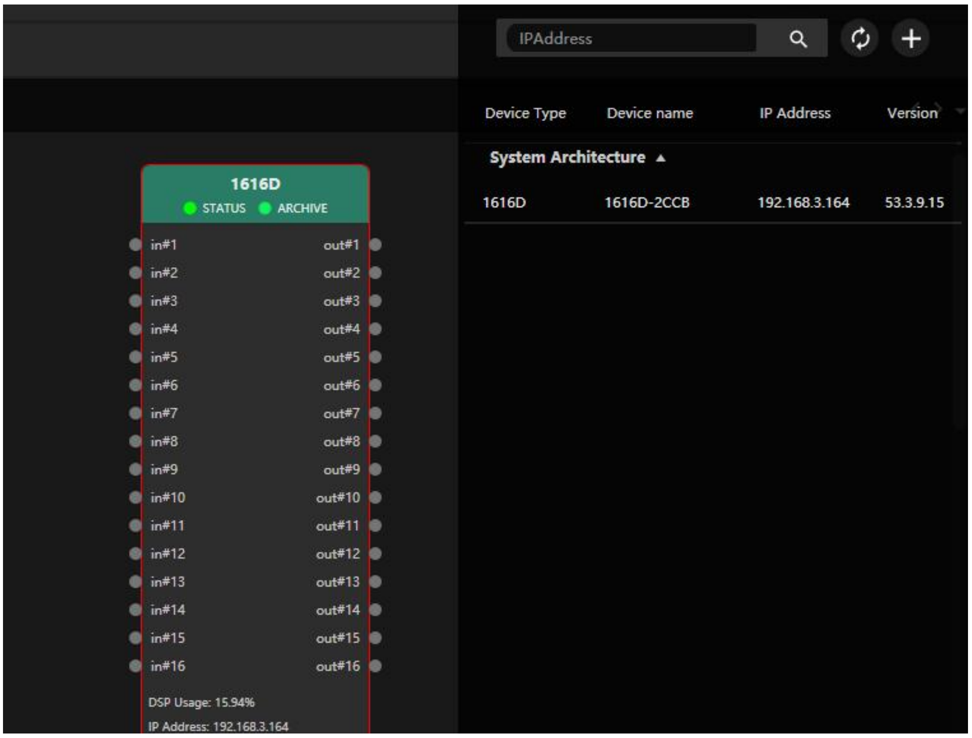
You may view, control, and modify programs stored in devices on the local network by dragging the target device from the discovery panel into the blank space of the system architecture interface.



Notes:

PC must share the same subnet as the target device.

After download, the software enters Run Mode and device status indicators turn green.



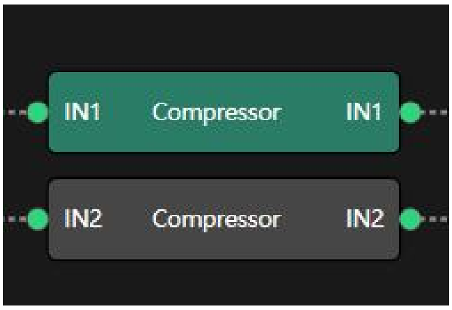
If device IP has changed, exit Run Mode, modify the module IP property, and re-upload.

5.7.9 Save / Save As

Once editing is complete, you may "Save" or "Save as.." file in your computer. Newly created programs must be saved to a local directory.

5.7.10 DSP Processing Modules

All DSP modules use a color-coded state system for intuitive operation:
Green = module enabled
Grey = connected but disabled



DSP level meters are displayed in dBu, with a maximum of +24 dBu (meter limit).
Device maximum input level is +18 dBu.
Standard applied:

0 dBu = –18 dBFS (EBU-R68)

Note: Values shown should not be interpreted as digital full-scale meters.

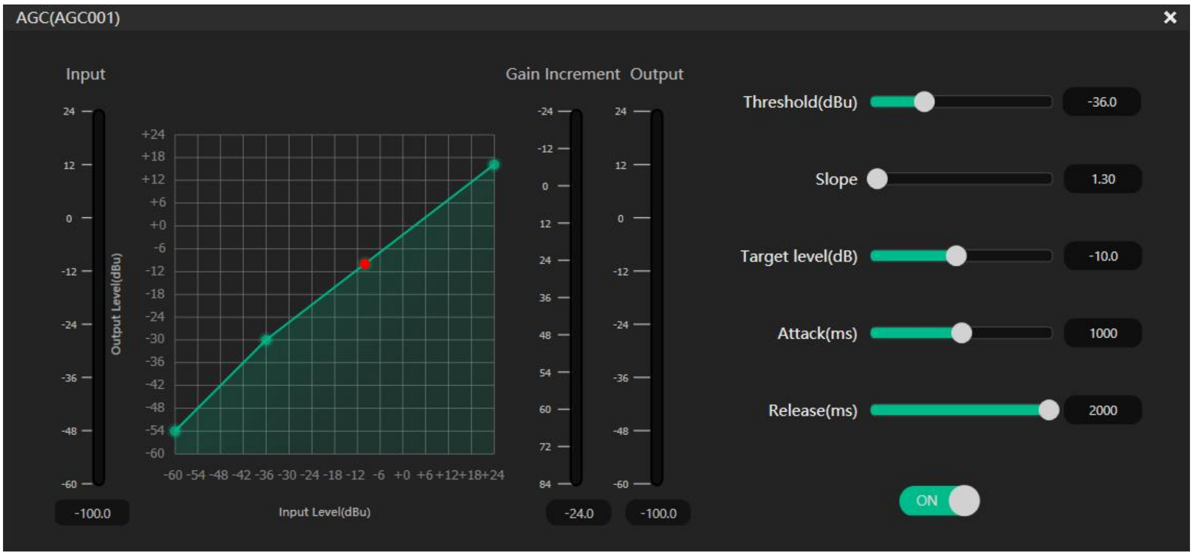
Automatic Gain Control (AGC)

Automatic Gain Control (AGC) is a special type of compressor. It boosts signals that fall below the threshold and attenuates signals that exceed it, effectively combining upward and downward compression in one module. When the threshold is set to a relative level and paired with a high ratio, the processed signal will converge toward the target threshold regardless of the input level. The primary goal is to reshape audio with a wide dynamic range into a signal that stays close to the target level, ensuring a stable and narrow dynamic range.

In simple terms, AGC uses the configured threshold and ratio to reduce loud signals (downward compression) and raise quiet signals (upward compression).

With AGC, playback signals such as background music or voice reproduction can maintain stable output levels under different acoustic conditions. It can also be used to keep speech signals consistent when recording or sending to remote endpoints, ensuring the remote audio can be reinforced locally with stable levels.

However, when AGC is applied to microphone channels for local reinforcement, improper parameter settings may trigger acoustic feedback and potentially damage equipment. To avoid such risks, you should cautiously enabling AGC on microphone reinforcement channels unless parameters need to be carefully and conservatively configured.



AGC provides the following parameters:

Threshold: When the input level falls below this threshold, AGC does not engage and the signal passes through unchanged. This value is typically set above the noise floor and below the level of the desired signal.

When the input level exceeds the threshold, AGC behaves in two ways:

If target level > input level > threshold, the signal is boosted according to the configured ratio.

If input level > target level, the signal is attenuated according to the configured ratio.

Slope: The proportional relationship between the change in input level (above or below the target level, and above the threshold) and the resulting change in output level.

Target Level: The level that AGC drives the processed signal toward. When the input level is higher than the target, the controller applies downward compression to reduce the signal. When the input level is lower than the target, the controller applies upward compression to raise the signal.

Attack Time: The time required for the output to transition from its initial level to the final processed level once the input moves above or below the target level (and is above the threshold). This is also referred to as the “build-up time”.

Release Time: The reaction time needed for AGC to return from its active state to a neutral, 1:1 output once the input signal drops below the threshold.

Compressor

A compressor is primarily used to reduce the dynamic range of signals that exceed a defined threshold, while signals below the threshold remain unchanged.

The compressor provides the following control parameters:

Threshold: When the input level exceeds this value, the compressor begins attenuating the signal according to the configured ratio. Any portion of the signal that surpasses the threshold is considered “overshoot”. In normal operation, the compressor attenuates this overshoot, and the greater the overshoot, the stronger the attenuation.

Slope: The compression ratio that determines how much overshoot is reduced. A lower ratio slope results in less attenuation, making the output closer to the original input.

Once the input passes the threshold, the slope defines the relationship between the change in input level and the change in output level.

For example:

With a 2:1 ratio, if the input exceeds the threshold by 2 dB, the output exceeds it by only 1 dB.

With a 1:1 ratio, no compression is applied, and the other parameters become irrelevant.

The ratio range is 1~20; higher ratios make the compressor behave more like a limiter.

Attack Time & Release Time: Attack time is the duration required for the compressor to fully apply gain reduction once the input crosses the threshold.

Release time is the duration required for the compressor to return to a non-compressed state after the signal falls back below the threshold.

If the attack or release time is set too short, you may experience “pumping” or “breathing” artifacts. To avoid these issues while preserving the natural transient response, it is usually desirable for the initial portion of the signal to pass through with minimal influence.

Sudden or dramatic changes in gain can also create pumping. Attack time determines how quickly gain is reduced, and release time determines how quickly gain is restored, both must be set properly to avoid such artifacts.

Output Gain: Also known as compensation gain. If compression significantly reduces the signal level, you can raise the output gain to restore loudness, making the signal fuller and more present. This gain adjustment applies uniformly across the entire processed signal and is independent of other compressor parameters.

Multi Compressor

By defining crossover points, the signal can be separated into low-mid and mid-high bands for individual control. As the speaker moves closer to the microphone and low-frequency energy increases, the compressor reduces the excess low-frequency gain, keeping the voice clear, natural, and consistent without large dynamic shifts.

Parameter Description:

Threshold: Defines the input level at which compression begins. When the input exceeds the threshold, gain reduction is applied. The higher the input rises above the threshold, the greater the attenuation relative to uncompressed signal level.

Ratio: Determines the amount of gain reduction applied once the signal exceeds the threshold. A lower ratio results in lighter compression, allowing higher peaks; a higher ratio increases control.

For example, with a 2:1 ratio, if input rises 2 dB above threshold, output increases only 1 dB.

A 1:1 ratio applies no compression. Adjustable range: 1:1 to 20:1.

Attack & Release: Attack time controls how quickly gain reduction engages once the threshold is exceeded, maintaining natural transient articulation.

Release time determines how quickly gain returns after the signal falls back below threshold.

These settings prevent pumping and breathing artifacts caused by rapid gain fluctuations.

Compensation Gain: Compensates for level reduction caused by compression. Gain is applied only to the affected frequency band, independent of other compressor parameters, ensuring tonal balance while restoring output loudness.

G.R. Meter & Output Meter: G.R. (Gain Reduction) displays the amount of compression applied.

Output shows the final level after compression.

Gain reduction is shown as an inverted meter.

Example:

Input: -6 dB

Threshold: -30 dB

Ratio: 2:1

The input exceeds the threshold by 24 dB, resulting in 12 dB reduction.

Thus:

GR meter = -12 dB

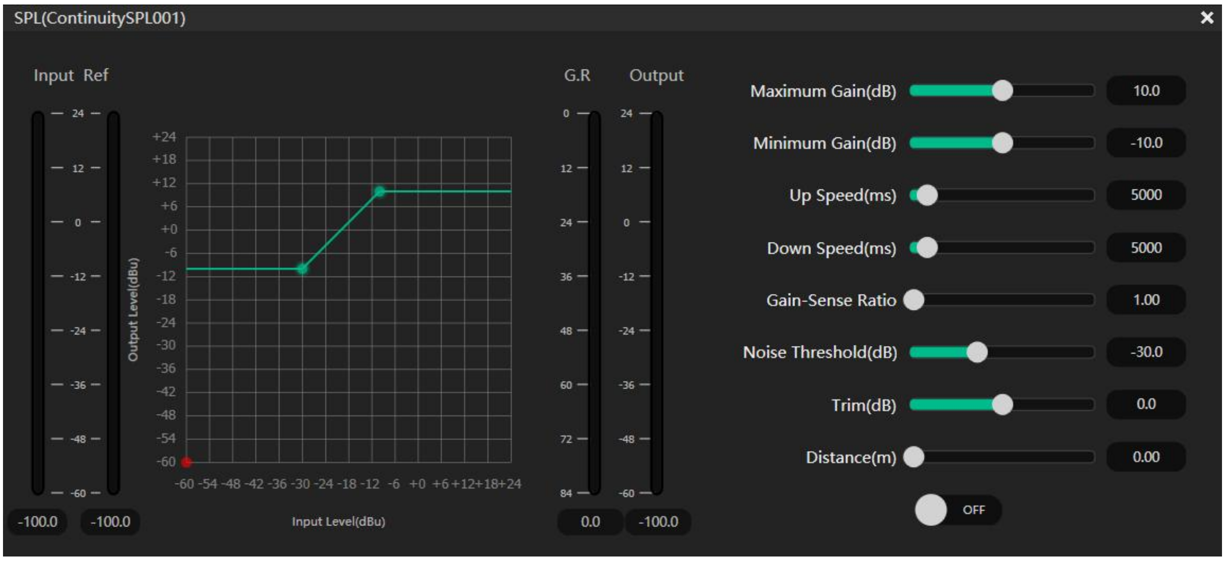
Output level = -6 dB - 12 dB = -18 dB.

Continuity SPL (Environmental Loudness Compensation)

The Continuity SPL dynamically adjusts output level based on the level of a reference signal, typically the ambient noise captured by a microphone.

For example, in a shopping mall installation, a microphone can monitor the environmental noise level and automatically adjust background music volume.

When the environment becomes noisy, the background music level increases proportionally; when the environment becomes quiet, the music level decreases to maintain comfortable listening conditions.



Parameter Description:

Max Gain: Defines the maximum achievable output gain.

Min Gain: Defines the minimum permissible output gain.

Up Speed: Determines how long it takes for the output level to increase to the target value once the ambient level exceeds the threshold.

Down Speed: Determines how long it takes for the output level to decrease to the target value once the ambient level falls below the threshold.

Gain-Sense Ratio: Controls how aggressively the output gain follows changes in the reference signal level (gain scaling factor).

Noise Threshold: Ambient level threshold above which gain increases and below which gain is reduced.

Digital Trim Gain: Allows fine level adjustment of the processed signal, boosting or attenuating the target level as needed.

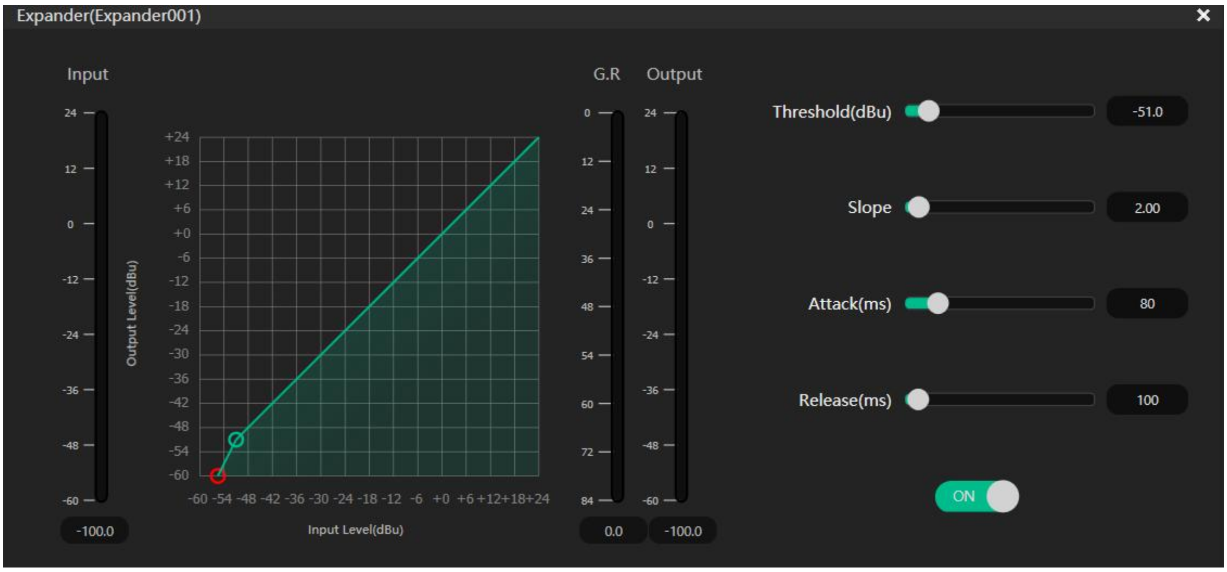
Distance: Defines the distance between the reference microphone and the SPL channel output loudspeaker, for acoustic energy decay compensation and control accuracy improvement.

Expander

An expander works in principle opposite to a compressor: it increases the dynamic range of audio signals.

While a compressor reduces the level of signals above the threshold, an expander attenuates signals that fall below the threshold. In other words, an expander makes quiet sounds even quieter.

When the ratio reaches around 20:1, an expander behavior approaches that of a noise gate. In fact, a noise gate can be considered a type of expander with a very high expansion ratio.



Parameter Description:

Threshold: The input level below which the expander becomes active.

Typically set slightly above the ambient noise floor or unwanted background signal level.

Slope: The gain reduction slope below the threshold.

Higher ratios approach gate-like behavior.

Attack Time: Time required for the expander to apply gain reduction after the input drops below the threshold.

Shorter attack times allow the expander to react quickly to transients.

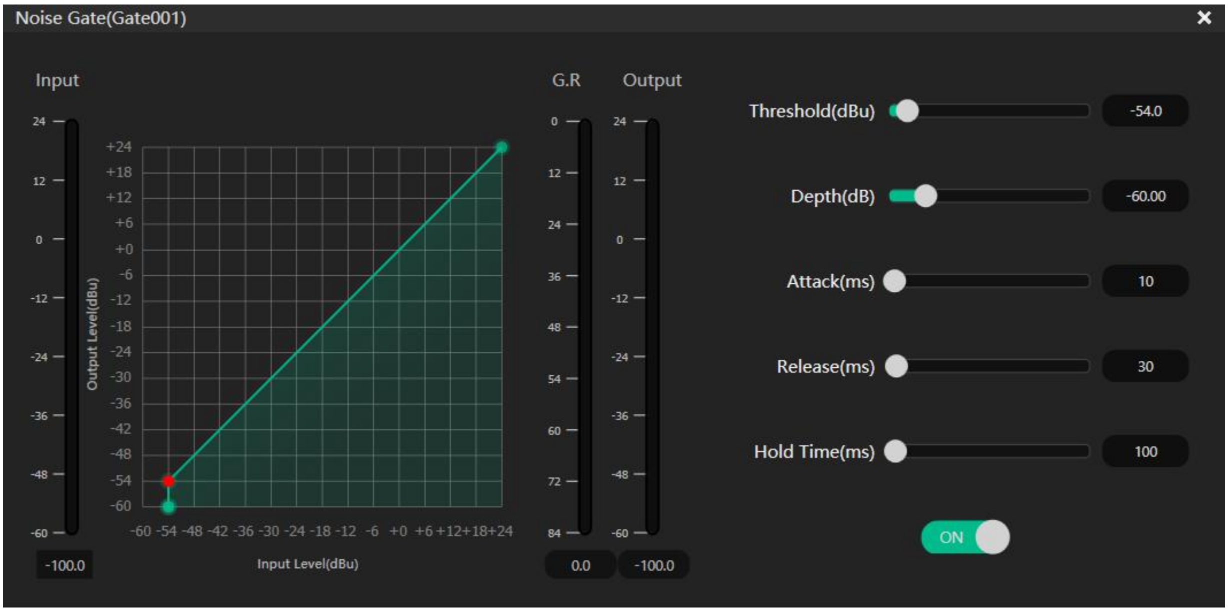
Release Time: Time required for gain to return to normal after the input rises above the threshold.

Both the attack time and the release time simply define how quickly the gain reduction changes. For example, the attack time controls how fast the gain moves from -40 dB to -50 dB, while the release time determines how fast it returns from -50 dB to -40 dB.

These time constants are independent of the threshold setting: when the input level rises above the threshold, the expander stops operating and the release time takes effect. Once the signal drops below the threshold again, the expander re-engages and the attack time becomes active.

Noise Gate

A noise gate is used to attenuate or mute signals that fall below a specified threshold. Signals below the threshold are typically considered unwanted noise or residual floor signals. When the input level is low (below the threshold), the gate blocks the signal; when the level is high, the gate opens.



Parameter Description:

Threshold: Signals above the threshold pass through.

Signals below the threshold are attenuated or muted.

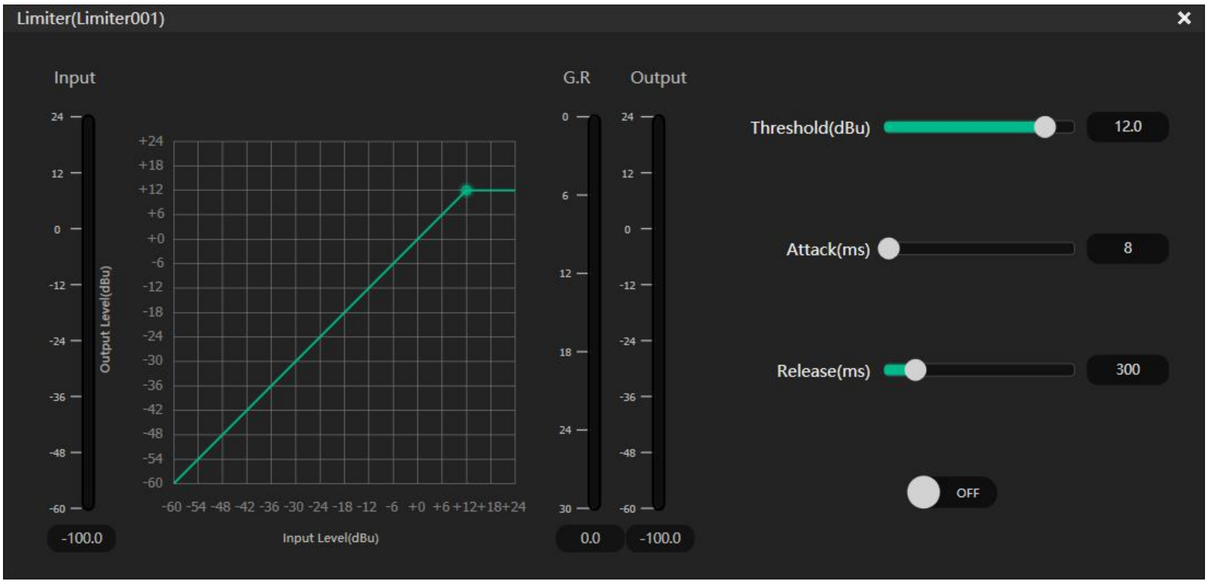
Depth: Determines how much the signal is attenuated when the gate works, expressed in dB.
Attack Time: Time required for the gate to fully work and apply attenuation once the signal falls below the threshold.
Release Time: Time required for the gate to open after the input rises above the threshold. (Reverse behavior of attack time.)
Hold Time: Prevents the gate from closing immediately once the input falls below the threshold. For example, with a 100 ms hold time, the gate will maintain its open state for 100 ms before beginning attenuation, even if the signal keeps below the threshold during that time.
Noise gates typically use peak detection, and because peak levels fluctuate much more aggressively than RMS levels, the signal may cross the threshold repeatedly within a short period. This can cause the gate to open and close rapidly, resulting in chattering distortion.
To prevent this, the noise gate in this module incorporates a 5 dB hysteresis range: the opening threshold is 5 dB higher than the closing threshold. This hysteresis prevents signals around the threshold from repeatedly triggering the gate, effectively suppressing chattering distortion.

Peak Limiter

A limiter has one primary function: to ensure that the signal never exceeds the specified threshold level under any circumstances. Its core operating principle focuses on preserving the signal below the threshold while aggressively preventing peaks from surpassing it.

A limiter typically employs two stages of gain control:
(1) Mild limiting as the signal approaches the threshold, preparing for gain reduction without fully suppressing overshoot.
(2) Hard limiting once the signal exceeds the threshold, applying an extremely fast and aggressive attenuation to prevent any overshoot.

This process responds extremely quickly and relies solely on peak detection, meaning gain reduction is triggered by the instantaneous peak level rather than the average level.



Parameter Description:

Threshold: Any signal exceeding the threshold will be attenuated.

The amount of attenuation depends on how far the signal overshoots, but the final output is limited to the threshold level.

Attack Time: Time required for the limiter to reduce the signal to the threshold level once it crosses the limit.

Release Time: Time required for the signal to return to its natural level after falling back below the threshold.

Since peak limiting is used, the limiter reacts as soon as the peak reaches the threshold.

This module is typically placed at the output stage for system protection.

Configure the threshold based on the rated or peak voltage tolerance of the power amplifier or loudspeaker.

Settings are usually expressed in dBu (0 dBu defined as 0.775 V).

Normally when used for protection, the attack time is typically set fast, while the release time is set relatively slow.

Priority Ducker

A ducker attenuates the input signal when the reference signal exceeds a defined threshold. Once triggered, the input channel level is reduced while the reference signal remains audible, creating a “ducking” effect. This feature is commonly used in applications such as announcements over background music.

Parameter Description:

Threshold: When the reference signal exceeds this level, attenuation is applied to the input channel.

When the reference signal drops below the threshold, the input gradually returns to its original level.

Depth: The maximum attenuation applied to the input signal when ducking is active.

Attack Time: The time it takes for the input signal to begin attenuating after the reference signal rises above the threshold.

Release Time: The time required for the input channel to return to its original level once the reference signal falls below the threshold.

Hold Time: Maintains the ducking effect for a specified duration after the reference signal falls below the threshold.

Prevents rapid toggling during short pauses.

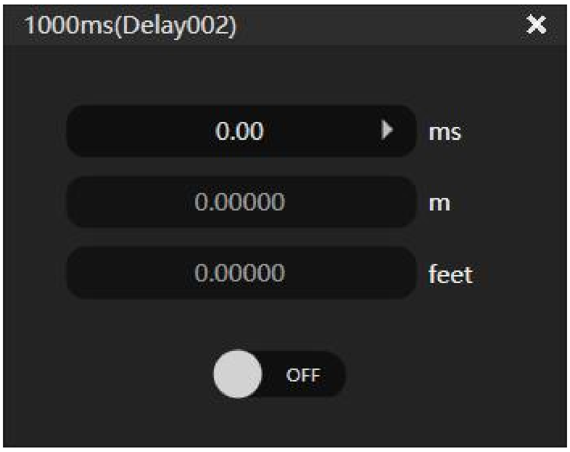
Mixing Output:

Enabled: The reference signal is mixed with the attenuated input signal for output (simultaneous output).

Disabled: The reference signal only acts as a trigger and is not passed to the output.

Delay

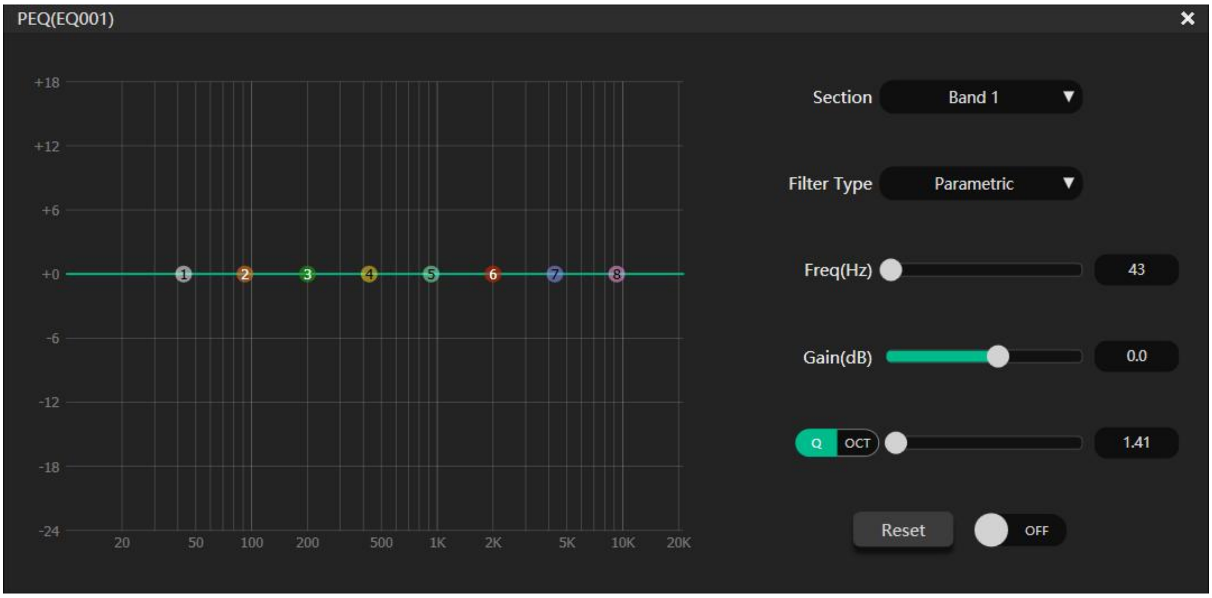
The delay module is available in three types: 50 ms, 100 ms, and 1000 ms, representing the maximum delay time of each module (i.e., 0~50 ms, 0~100 ms, 0~1000 ms). You may select the appropriate delay module type based on actual usage requirements. The primary function of this module is to apply delay to the audio signal at the current node or channel by the configured amount, which can be used for signal phase alignment or sound image localization.



The delay time can be set in milliseconds, meters, or feet, and these three units can be converted directly between one another.

Parametric Equalizer (PEQ)

The parametric equalizer is mainly used to correct frequencies in the audio signal that are boosted or missing, regardless of whether these frequency bands are wide or narrow. The equalizer can also narrow or widen the bandwidth or modify the amplitude of specific components in the spectrum. In simple terms, an equalizer allows you to shape the tonal balance of the signal, though it may introduce some degree of phase alteration.



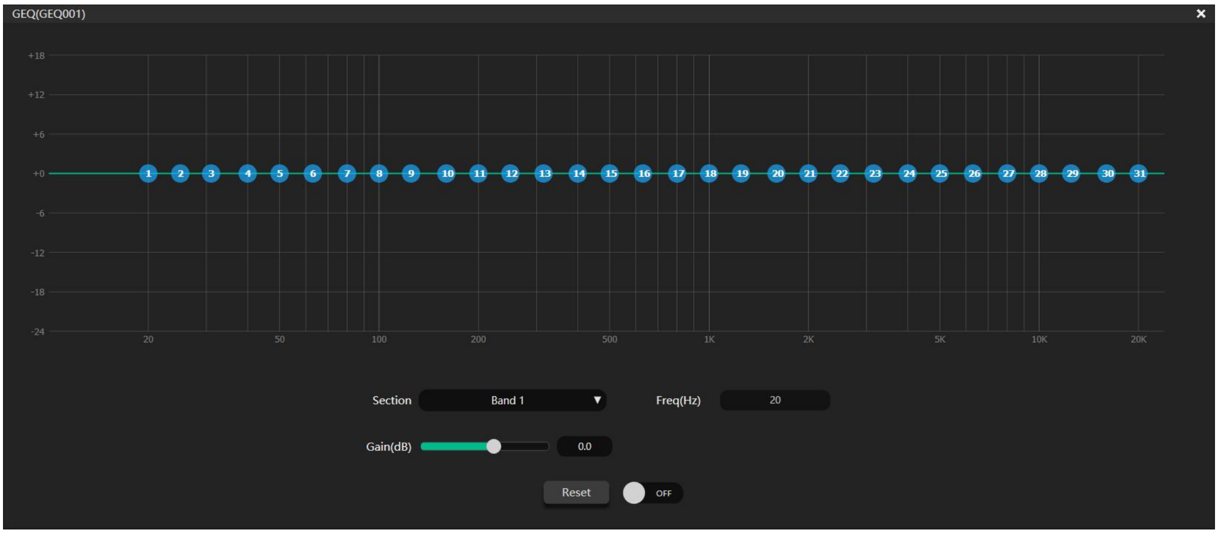
Parameter Description:

Filter Types: The default filter type is parametric EQ. First and last frequency bands support replaceable high-pass and low-pass filters; the second and second-to-last bands support replaceable shelving filters. Each filter type has different characteristics and performs different functions.

Pass Filters: The reference frequency used by these filters is called the cutoff frequency. A high-pass or low-pass filter allows frequencies on one side of the cutoff to pass freely while continuously attenuating frequencies on the other side.
High-Pass Filter (HPF):
Allows frequencies above the cutoff to pass, while attenuating frequencies below the cutoff according to the defined slope.
Low-Pass Filter(LPF):
Allows frequencies below the cutoff to pass, while attenuating frequencies above the cutoff according to the defined slope.
Shelving Filters:
High-Shelf Filter:
Boosts or attenuates all frequencies above the selected frequency.
Low-Shelf Filter:
Boosts or attenuates all frequencies below the selected frequency.
Frequency (Hz): Center frequency of the filter.
Gain (dB): Boost or attenuation applied at the center frequency.
Q / Bandwidth: Controls the overall bandwidth of the filter around the center frequency. Q-value and octave bandwidth are interchangeable, and you may select either method depending on your workflow.
The gain determines the maximum boost or cut at the center frequency, while the bandwidth (i.e., the affected frequency range) is defined using either the Q-value or the octave span. Bandwidth is measured between the two -3 dB cutoff points located above and below the center frequency.
This module provides 8 bands of PEQ by default, configurable up to 32 bands as needed.
Reset: Restores all EQ parameters back to the initial flat state, including filter types, center frequencies, and gains. A confirmation dialog will appear before the reset is applied.

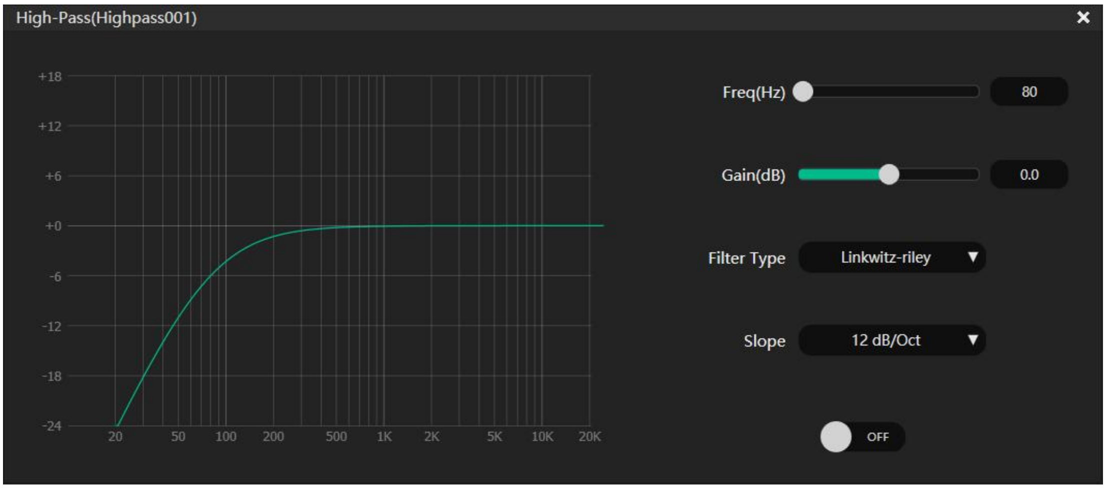
31-Band Graphic Equalizer

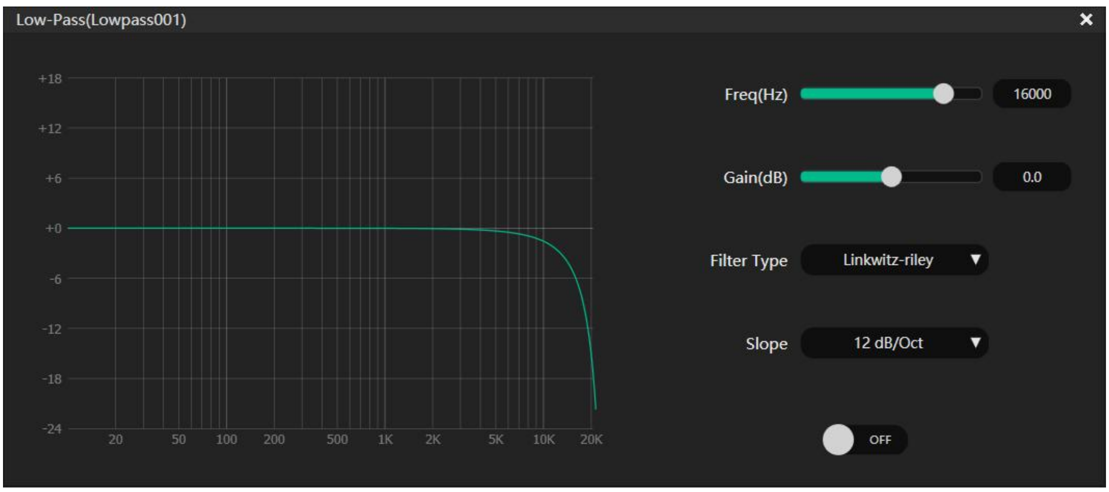
The 31-band graphic equalizer uses a constant-Q filter design with fixed frequencies. There are 31 frequency bands, each with a dedicated slider for boosting or cutting. Regardless of the amount of gain change, the bandwidth remains constant. It divides the 20 Hz~20 kHz range into 31 segments, corresponding to standard 1/3-octave bands.



High-Pass & Low-Pass Filters

High-pass and low-pass filters are primarily used to filter out high or low frequency components of an audio signal. Setting the appropriate cutoff frequency and slope helps achieve the desired filtering effect.





Available Parameters:

Filter Type: Choose from Bessel, Butterworth, and Linkwitz-Riley.

Butterworth has the flattest passband.

Frequency: The cutoff frequency.

Bessel & Butterworth define cutoff at -3 dB

Linkwitz-Riley defines cutoff at -6 dB

Gain: Gain settings affect the boost or attenuation of the entire signal frequency band.

Slope: Available options are 6, 12, 18, 24, 30, 36, 42, and 48 dB/Oct.

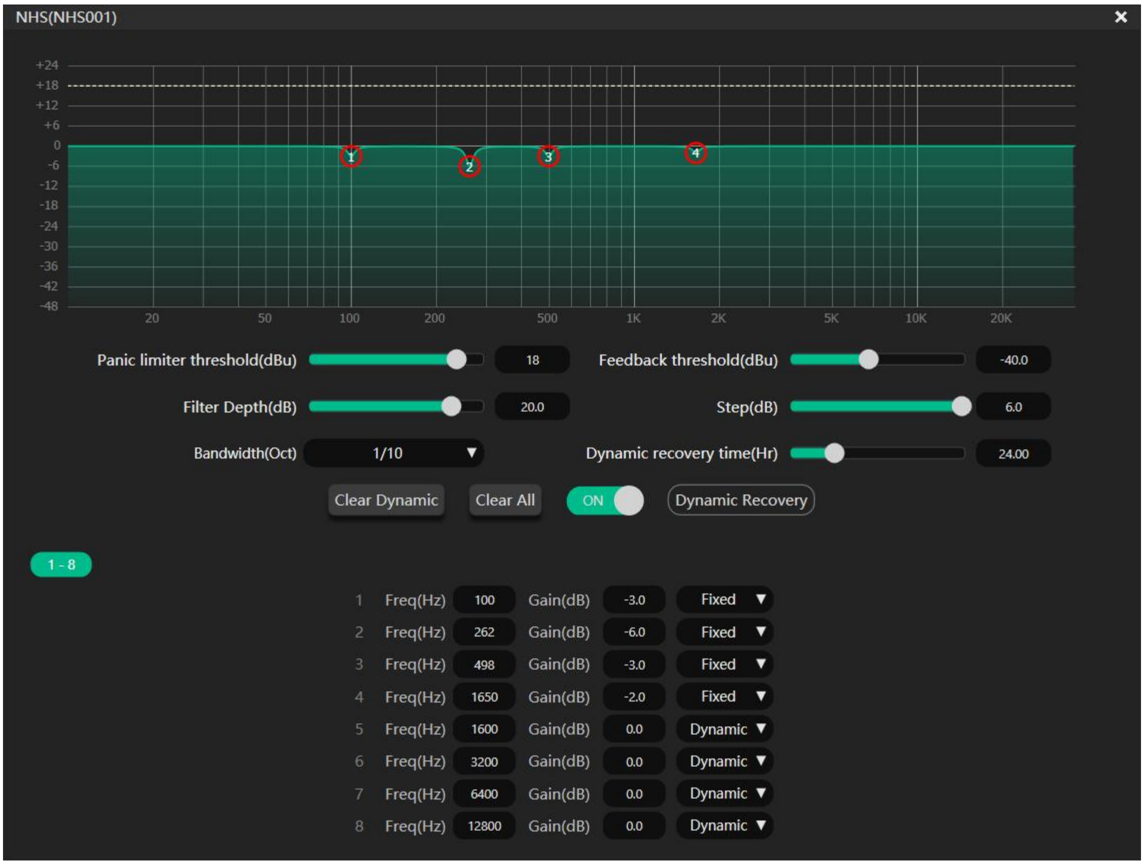
Example: 24 dB/Oct means the signal attenuates 24 dB per octave.

Notch Howling Feedback Suppression (NHS)

When using the feedback suppression module, it should always be combined with proper system design and engineering acoustic environment, rather than replacing them. Traditional methods such as limiting the number of open microphones, shortening the distance from the sound source to the microphone, selecting highly directional microphones and speakers to reduce feedback, and flattening the room frequency response through equalization adjustment are still required. Only after completing the above basic optimizations can the feedback suppressor be used to obtain additional acoustical gain. Note: The feedback suppressor cannot magically fix system design flaws, nor can it break through the physical limits of the system; forced use may instead damage the sound signal.

The feedback suppressor automatically detects and suppresses acoustic feedback in the system. It differentiates between actual feedback and intended audio content. When feedback is detected at a particular frequency, a notch filter is automatically inserted at that frequency with initial shallow attenuation. If feedback persists, the notch filter's depth increases according to parameters until the feedback disappears or reaches the limit.

After setting the notch filter automatically or manually, the filter can be locked to prevent parameter changes during use; at the same time, the filter settings can be copied to a dedicated filter module (such as an equalizer). The 8 filters equipped with the device will cycle through the filters set to "Auto" mode to clear filter settings that are only needed temporarily.



Available Parameters:

Panic Limiter Threshold: Defines the level above which any signal is considered "definite feedback."

When input exceeds the panic threshold:

a) Output gain is reduced instantly

b) Output level is limited to prevent runaway feedback

c) Filter sensitivity increases to maximum for faster detection

Once the level falls below the threshold, gain and sensitivity return to normal.

Setting this value to +24 dBu effectively disables this feature.

Feedback Threshold: This parameter defines the level below which a signal is never considered feedback. It prevents the module from misidentifying background noise or low-level hum as feedback.

Filter Depth: Specifies the maximum attenuation applied by a single filter. Proper configuration prevents excessive attenuation that may distort the signal and avoids degraded feedback-suppression performance in systems with strong narrowband resonances.

Step Size: The speed at which gain reduction occurs.

Bandwidth:

1/5, 1/10, 1/20 Octave.

Uses constant-Q filtering; bandwidth does not widen with deeper cuts.

Dynamic Recovery Time:

Defines, in hours, how long the system holds the applied feedback suppression before automatically restoring the gain at that frequency. Only applies when dynamic mode is enabled.

Notch Mode (Dynamic / Fixed):

Dynamic: Filter participates in the automatic cycle; used when new feedback frequencies appear.

Fixed: Support manually adjustable frequency and depth; does not auto-adjust.

Clear Dynamic: Instantly clears all dynamic filters.

Clear All: Clears all notch filters, both fixed and dynamic.

Recommended Tuning Workflow

(1) Calibrate loudspeaker frequency response

(2) Calibrate room frequency response

(3) Ensure proper gain structure (e.g., -18 dBFS nominal input, amplifier gain adjusted for 75~85 dBA SPL)

(4) Apply HPF/LPF, EQ, automixing, routing

(5) Remove unnecessary frequency content

(6) Configure feedback suppression parameters and reserve several dynamic filters for unexpected conditions

If speakers are involved, it is recommended to add a compressor/limiter for protection in case all notch filters are used up or gain becomes excessive.

FIR Filter

Finite Impulse Response (FIR) filters are a type of linear time-invariant digital filter whose output is the weighted sum of current and past input samples. After an impulse input, the output decays to zero within a finite number of samples. A major advantage is the ability to achieve linear phase, meaning all frequencies experience equal delay, which is critical for audio systems that require phase-coherent reproduction.

This FIR module supports both automatic and manual design using imported measurement data. It allows exporting FIR data as CSV, and importing standard CSV or SECFIR files.

Bands: 8-band EQ for manual adjustment

Filter Type: HPF, LPF, or EQ

Frequency: Center frequency

Gain: Boost/cut

Q: Bandwidth adjustment

Phase Auto:

Automatically generates a calibrated phase curve between the selected start and end frequencies. Smaller octave smoothing produces a smoother curve.

Magnitude Auto: Automatically generates a calibrated magnitude response in the specified frequency range.

Import: To start FIR design, measurement data (TXT format) containing magnitude and phase can be imported from the computer.

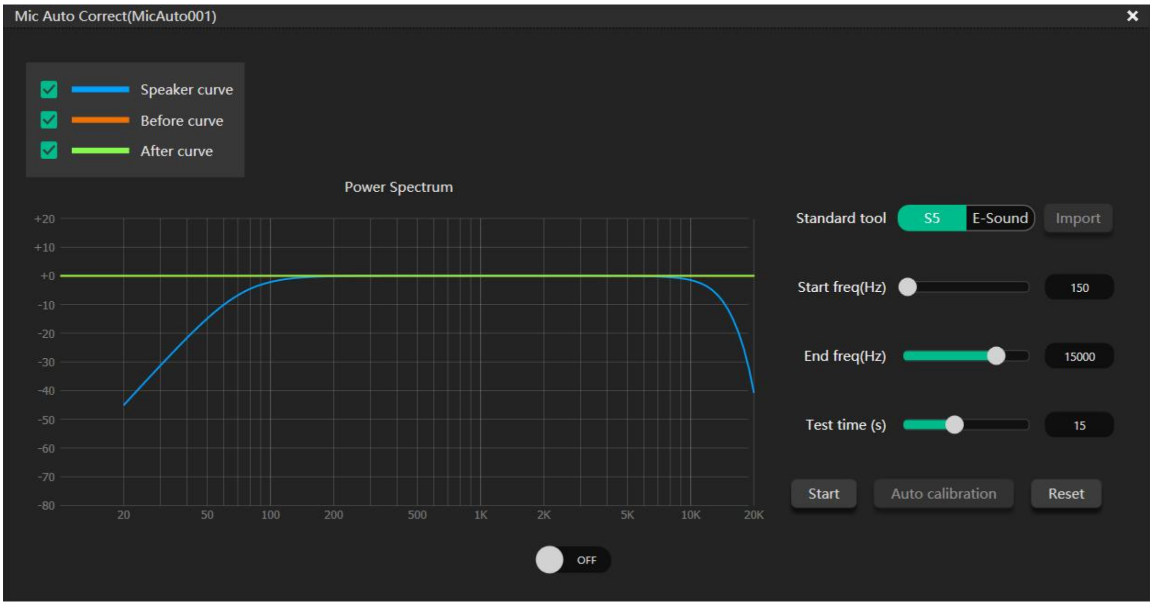
Export: Allows exporting FIR files with optional window functions:

Rectangle, Triangle, Hanning, Hamming, Blackman, Kaiser.

Up to 1024 taps are supported. The higher the tap count, the greater the processing latency.

Microphone Auto Adjustment (Beta)

For optimal sound reproduction during system tuning, it is essential to establish a proper gain structure for the sound reinforcement system while also calibrating microphones and loudspeakers. As audio professionals, we understand that every microphone and loudspeaker exhibits different performance characteristics and quality levels. To restore sound as close to its original form as possible, device calibration becomes critical. To realize this process, the DSP includes a Microphone Auto-Calibration Module.



Measurement Tools:

S5:

The auto-calibration process requires an auxiliary tool: a reference loudspeaker.

We provide a dedicated reference model named S5. The S5 is our proprietary measurement loudspeaker. Its frequency response curve is built into the module for accurate reference.

External Tools:

If you are using a different loudspeaker, you may import its frequency response file in CSV format.

The purpose is to ensure that the auxiliary loudspeaker's characteristics do not negatively affect the calibration results.

The reference response curve is displayed in the graph for intuitive comparison.

Start Frequency: Defines the lowest frequency included in the auto-calibration range.

Frequencies below this point will not be processed.

End Frequency: Defines the highest frequency included in the auto-calibration range.

Frequencies above this point will not be processed.

Test Duration: The module generates an internally built pink noise signal (20 Hz~20 kHz), which is routed through the matrix to the loudspeaker.

The Test Duration parameter (in seconds) determines how long the signal is emitted.

Measurement Procedure

Before initiating the test:

Align the loudspeaker's acoustic axis directly toward the microphone.

Maintain a recommended distance of approximately three times the loudspeaker's cabinet height, along the high-frequency axis.

Adjust both loudspeaker output and microphone input so that the captured signal reaches approximately 0 dBu, ensuring an ideal measurement level.

Start: Press Start to begin pink-noise playback.

Because the first test is often used as preparation, initial readings may not be suitable as usable calibration data. Once levels and positions are confirmed, run the measurement again for accurate results.

After testing, an orange curve (Pre-Calibration Curve) is generated.

This curve represents the microphone's effective response (influenced by device performance and acoustic environment), computed using an algorithm that take reference loudspeaker characteristics into account.

Auto Calibration: The Auto-Calibration button becomes available only after test data has been generated.

Before proceeding, ensure that Start Frequency and Stop Frequency are correctly set for the intended correction range.

For conference microphones, full-band calibration is unnecessary, since voice-focused ranges are typically sufficient.

When auto calibration is executed, the module optimizes the response curve to achieve a flatter frequency response within the designated range.

After the process, a green curve (Post-Calibration Curve) is generated.

This curve represents the microphone's optimized response after calibration. Notice that it’s not the EQ correction curve itself.

Reset: The Reset function clears all adjustments and parameters, returning the module to its default state as if newly added.

ON/OFF: This switch applies or bypasses the final optimized curve produced by the calibration algorithm.

Important Notes:

Before enabling the calibrated microphone response:

(1) Disable the measurement loudspeaker routing in the matrix.

(2)Turn off the auxiliary loudspeaker.

(3)Ensure the optimized microphone channel is muted until ready.

If these precautions are not followed, enabling the calibration output will route the microphone signal directly to the loudspeaker under measurement conditions, which may cause acoustic feedback.

Loudspeaker Auto Adjustment (Beta)

In addition to the microphone auto-tuning module, the system also integrates a Loudspeaker Auto-Calibration engine. In sound reinforcement systems, the perceived performance of loudspeakers is highly subjective. While the factory performance of a loudspeaker plays a decisive role in overall sound quality, system-level calibration is often overlooked, leading to inaccurate judgments and incorrect tuning practices.

Professional engineers typically calibrate each loudspeaker or loudspeaker group. It is important to distinguish between loudspeaker calibration (near-field measurement) and system calibration (far-field measurement). Generally speaking:

(1) Near-field calibration: Minimizes room interference to accurately reflect loudspeaker driver performance.

(2) Far-field calibration: No longer pursues a "flat" curve due to acoustic realities (including room reflections, sound summation and cancellation, and psychoacoustic response curves). A perfectly flat far-field curve is neither achievable nor desirable.

This module aims to provide a streamlined loudspeaker calibration method without requiring complex measurement setups or advanced software.

Measurement Tools:

S-MIC:

Built-in calibration file included (Recommended).

External Measurement Mic:

Import frequency response file (CSV format).

This compensates for microphone frequency response coloration and ensures accurate analysis.

A reference response curve is displayed for verification.

Start Frequency: Lowest frequency to be included in auto-calibration. Signals below this range remain unaffected.
End Frequency: Highest frequency for calibration. Signals above this range remain unaffected.
Test Time: The module generates an internally built pink noise signal (20 Hz~20 kHz), which is routed through the matrix to the loudspeaker.

Measurement Procedure
Measurement requirements:
(1) Place the measurement microphone on-axis, at a distance = three times loudspeaker cabinet height
(2) Adjust gain so mic level close to 0 dBu
Start: Press start to trigger sweep playback.
Sweep runs for the configured duration. The pre-calibration curve (orange) represents the loudspeaker's approximate response under near-field conditions, compensating for mic response but still influenced slightly by equipment and room.
Auto Calibration: Auto-calibration becomes available after measurement data is acquired.
Before triggering, verify start/end frequency ranges accord with the loudspeaker’s usable bandwidth. For example:
Column speakers typically roll off above ~100 Hz (-6 dB points)
Point-source LF extension varies by driver size & enclosure design
The module adjusts the response to a smooth and natural compensated target, not an artificially flat response.
A post-calibration curve (green) is generated, representing the optimized result.

Note: The curve displayed is the final response, not the EQ correction curve.

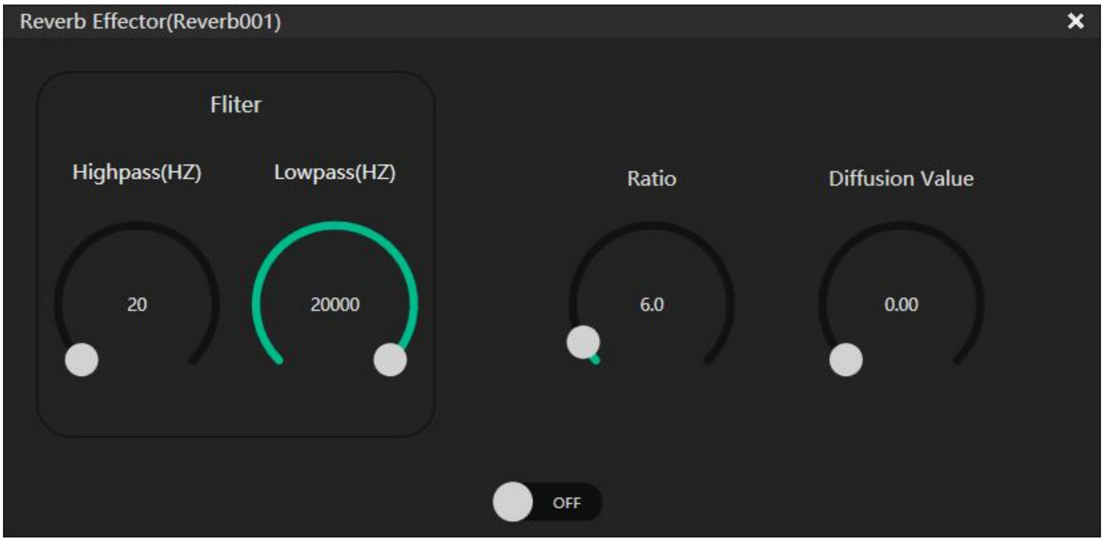
Reset: Resets all parameters and measurement data as if it's a new added module.
ON/OFF: Applies or bypasses the optimized response

Important:
Before enabling the result, disconnect the measurement mic and disable its routing.
Otherwise feedback will occur due to open-loop gain with loudspeakers active.

Reverb Effect

This module provides a classic plate-style digital reverb, extending use cases beyond typical DSP processors which often lack built-in effects.

Reverb simulates reflections in enclosed spaces. Direct sound reaches the listener first, followed by early reflections and dense late reverberation decaying over time. The algorithm models this psycho-acoustic phenomenon.



Available Parameters:

Filters (HPF/LPF):

Defines bandwidth of the signal entering the reverb engine to avoid undesirable frequency buildup.

HPF: Removes excessive low frequencies

LPF: Removes excessive high frequencies

Dry/Wet Ratio:

Blend between original and processed signal at the ratio.

0% = dry signal only

100% = fully wet reverb signal

Reverberation Time: Defined as the adjustment of RT60 (Time required for reverb to decay by 60 dB), associated with perceived room size.

Pre-Delay: Time between direct sound and first reflection, affecting sense of room size and source distance.

Internal advanced parameters (e.g., decay, diffusion, room size) are model-locked for usability.

Delay Effect

Compared to the acoustically complex nature of reverb, which is composed of thousands of dense reflections, reverberation creates a highly clustered and continuous sound field. This dense reflection pattern effectively simulates spatial environments, but excessive reverb may blur the mix and introduce masking effects, reducing clarity.

In contrast, delay produces a much sparser structure, typically consisting of only several discrete echoes. When the goal is not to reproduce a highly realistic acoustic environment, but instead to add perceived space and depth, or to create a slightly diffused spatial effect, delay processing is often more suitable.

In simple terms, a delay effect generates multiple echoes from the input signal, each progressively reduced in level, until the repeated signal eventually fades out.

Available Parameters:

Filter (HPF/LPF): The delay processor includes both high-pass and low-pass filters. These filters are used to shape the input signal before it enters the delay engine, ensuring only the desired frequency content is processed. This prevents undesirable frequency components from producing unwanted resonances or buildup in the delayed signal.

HPF: Removes low-frequency energy to avoid low-frequency accumulation.

LPF: Removes high-frequency content to prevent overly bright or harsh repeated transients.

Delay Time: Delay time is the primary control parameter of a delay processor. It defines the interval between echoes and can be modulated to cycle between shorter and longer values.

For example, if the delay time is set to 100 ms, echoes will appear at approximately:100 ms -> 200 ms -> 300 ms -> 400 ms -> ...

Feedback: Without feedback, only a single echo will be produced after the input signal passes through the delay line.

Feedback determines how many echoes are generated and how quickly they decay.

For example, with a fixed 100 ms delay time and a feedback value of 5, echoes will typically be heard at:100 ms -> 200 ms -> 300 ms -> 400 ms -> 500 ms ->...

The feedback amount controls the attenuation (or in some creative cases, amplification) applied to each successive echo.

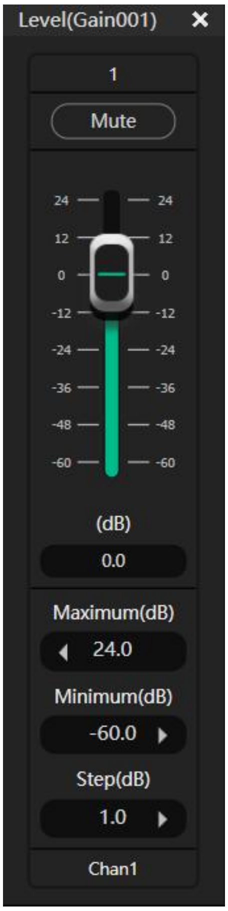
In attenuation mode, the decay between echoes can be approximated by:

$$\text{Decay} = 20\lg(N*0.05)\ \text{dB}$$

where N is the feedback value.

A lower feedback value results in faster decay and fewer audible repeats.

Level Control

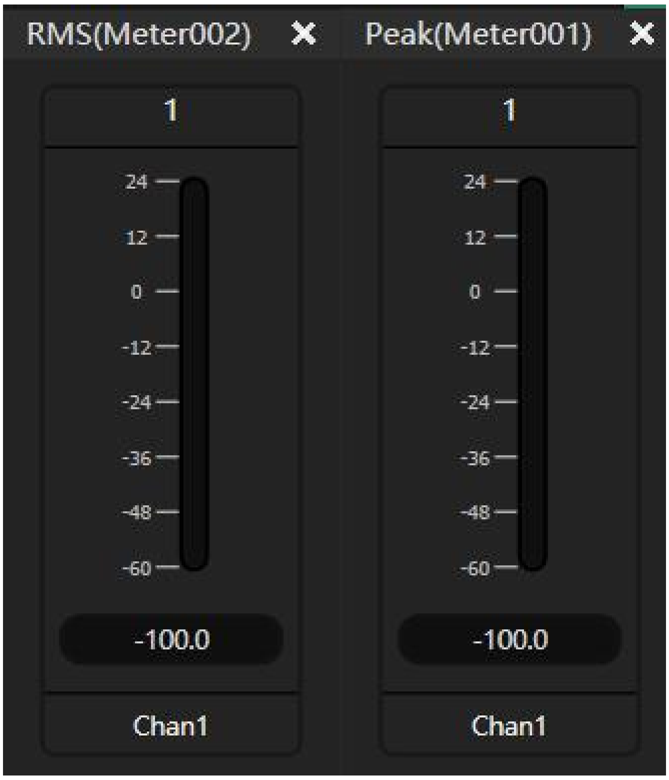


To manage signal level, a gain control module can be inserted at specific nodes within the DSP signal chain. This allows independent volume adjustment at different processing stages.

The user may define both maximum and minimum gain limits to constrain the control range. Additionally, gain can be adjusted in configurable step increments, allowing precise increase or attenuation of the signal level as needed.

Meter

Two types of level meters are provided in the component list: a Peak Meter and an RMS Meter.



Peak Meter

The Peak Meter measures the maximum instantaneous signal level: the True Peak value. It displays the highest voltage excursion of the waveform, either positive or negative, relative to the neutral position.

For example, with a sinusoidal signal, the peak level corresponds to the amplitude of the waveform crest or trough relative to zero.

RMS Meter

The RMS (Root Mean Square) Meter measures the effective signal level. For periodic signals (e.g., a sinusoidal wave), the RMS value is calculated by squaring the signal, averaging it over one period, and then taking the square root.

RMS level reflects the power content of the signal and correlates with perceived loudness and actual delivered energy.

Peak and RMS meters may display different responses for the same input signal, as they represent fundamentally different measurements of signal level.

Gating Automixer

In conference environments, when multiple microphones are open at the same gain level and only one participant is speaking, the overall clarity often decreases. The unused microphones will pick up room noise, reflections, and reverberation. When these signals are mixed together, the combined audio quality is significantly degraded, and the sound reinforcement system becomes highly susceptible to feedback, limiting usable gain before feedback.

To address this issue, unused microphones must be attenuated or muted. An Automatic Mixer performs this process automatically, reacting much faster than manual adjustment. The DSP includes a gate-based automatic mixer algorithm for local reinforcement need. All input channels are supported regardless of processor model.

Channel mute and fader controls are applied after the automatic gain stage. The fader controls the direct output level, adjusting these parameters does not affect automixer behavior.

For channels like media playback should be excluded from auto-mixing by not engaging the automixing function for that channel, its gain will remain unaffected and its level will not influence other channels.

Notice that channel mute prevents direct output, yet the signal still participates in the mix bus. . Even if a active channel is muted post-automix, a strong input signal may still reduce gain on other channels. To prevent this, make sure to mute the channel before the automixer node or disable automix for that channel.

Each channel in the auto-mixer provides a “Default” button to skip automixer, making it always on and only affected by channel mute and fader controls. This is suitable for sources requiring fixed level handling, such as priority microphones that must remain always-on.

Priority Control:

Priority allows higher-priority channels to override lower-priority channels and influences the automix algorithm. Values range from 0 (lowest) to 10 (highest), with a default of 5. The priority can be adjusted via slider or numeric entry. A higher value increases the channel's weighting. When two channels have the same level, the channel with higher priority opens first.

Note: Extreme priority settings (e.g., 0 vs. 10) should be used with caution. If a high-priority microphone picks up background noise or loudspeaker spill, it may suppress lower-priority microphones even when not actively in use.

If this occurs, consider inserting a noise gate/expander before the automix block on the high-priority channel and set the threshold so background noise does not trigger activation.

The automixer is gate-based, meaning each channel is either open or closed.

Available Parameters of the automixer:

Gain: Controls the overall output gain of the automatic mixer.

Hold Time: Determines how long a channel remains open after the speaker stops talking.

This prevents the channel from closing during short pauses between words or phrases.

Off Attenuation: Specifies the level reduction applied to a channel when its input signal does not exceed the gate threshold and the channel remains closed.

Sensitivity: Controls how much the signal must exceed the ambient noise reference level to open the channel.

Lower sensitivity values make it easier for quiet input signals to activate the channel, even in low-noise environments.

NOM Attenuation (Number of Open Microphones Attenuation): Adjusts the overall attenuation applied as more microphones become active.

This parameter reflects the attenuation when two mics are open, and the additional attenuation applied each time the number of open mics doubles.

For example, when set to 3 dB:

2 open mics = -3 dB

4 open mics = -6 dB

8 open mics = -9 dB

In general, doubling the number of open microphones adds the defined attenuation value.

Maximum NOM: Defines the maximum number of microphones allowed to open simultaneously.

Noise Threshold (ATS): Uses Automatic Threshold Sensing (ATS) technology. No adjustment is required.

Channels open when the signal exceeds the automatic threshold, and close when below it.

Gain-sharing Automixer

The processor is equipped with a gain-sharing auto mixer. Both gating auto mixer and gain-sharing auto mixer can be configured by modifying the "Mixer Type" parameter in the properties bar. The core logic of gain-sharing is: "the total output gain remains constant, while the gain of each channel is distributed according to its signal level." The principle is as follows:

Using eight microphones with identical parameters as an example:
If only one person is speaking, the active microphone receives the maximum gain, while the gains of the other microphones are attenuated. If two persons speak simultaneously (with similar levels), both microphones share equal gain, and each is attenuated by 3 dB. By the same rule, every time the number of active microphones doubles, the gain of each active microphone is reduced by 3 dB. In this way, the gain of each speaking microphone depends on the number of active channels and parameter settings, but the total system output gain remains constant (determined by the speaking microphones).

In the auto-mix matrix, each channel has both a Individual Out and a Total Auto Mix Out. Both outputs are affected by the mute status and channel fader. In addition, the Total Auto Mix Out is also controlled by the master mute. Refer to the signal flow diagram in the appendix for details.

The Auto Mixer contains two groups of parameters: Global Parameters and Channel Parameters.

Global Parameters:

Gain: Use it to control the main output level of the automatic mixer.

Slope: Affects how much low-level channels are attenuated. A higher slope results in greater attenuation of low-level channels. Its behavior is similar to the ratio control of an expander and is recommended to be set to 2.0 or close to 2.0.

(1) When set to 1.0, automatic mixing on all channels is effectively disabled.

(2) When set to 2.0, ideal gain sharing is achieved. This is the recommended default value.

(3) When set to 3.0, the gain adjustment range increases and may cause unnatural sound.
Note that the larger the slope value and the more channels that are active, the greater the overall system attenuation.

Response Time: A faster response time ensures that the initial transient of speech is not cut off. A slower response time provides smoother operation.

The automatic mixer is designed so that microphones turn on much faster than they turn off, so even with a response time set to 100 ms, the initial syllables are usually not cut off. If a slow response time of several seconds is set, the automatic mixer maintains a longer hold time and the previously active channel will remain on for several seconds.

ON/OFF: The button is used to enable or disable the automatic mixing function.

Channel Parameters:

Active Button: Each channel has an Auto Mix on/off button. Channels that need to participate in auto mixing must have this enabled. When disabled, the channel is excluded from the auto-mix process.

Priority: Priority allows higher-priority channels to "override" lower-priority ones, affecting the auto-mix algorithm. The range is 0-10; the higher the value, the higher the priority.

Mute: Channel mute and fader control are located after the auto mixer. Therefore, even if a channel is muted, if its signal level is high, it will still cause the gain of other channels to decrease.

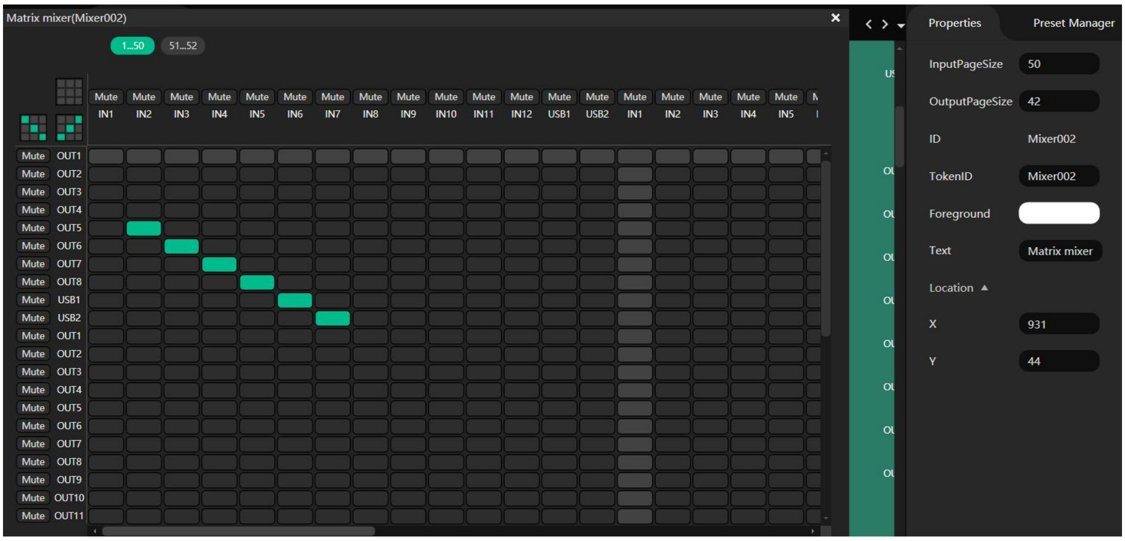
Fader: Adjusting the fader increases or decreases the relative level (weight) of the corresponding channel within the auto mix.

Matrix Mixer

For devices equipped with different processor models, the corresponding matrix modules have already been configured within the program interface, with the necessary link connections established.

To complete the routing setup, simply route the input channels to their respective output channels as required.

When selecting a matrix module in edit mode, you can modify the matrix display page size in the Properties Bar. By default, the matrix pagination size is set to 8*8.



During signal routing, you can use the gray cross-hair guide to visually confirm the routing relationship between input and output channels, ensuring the accuracy of your operations.

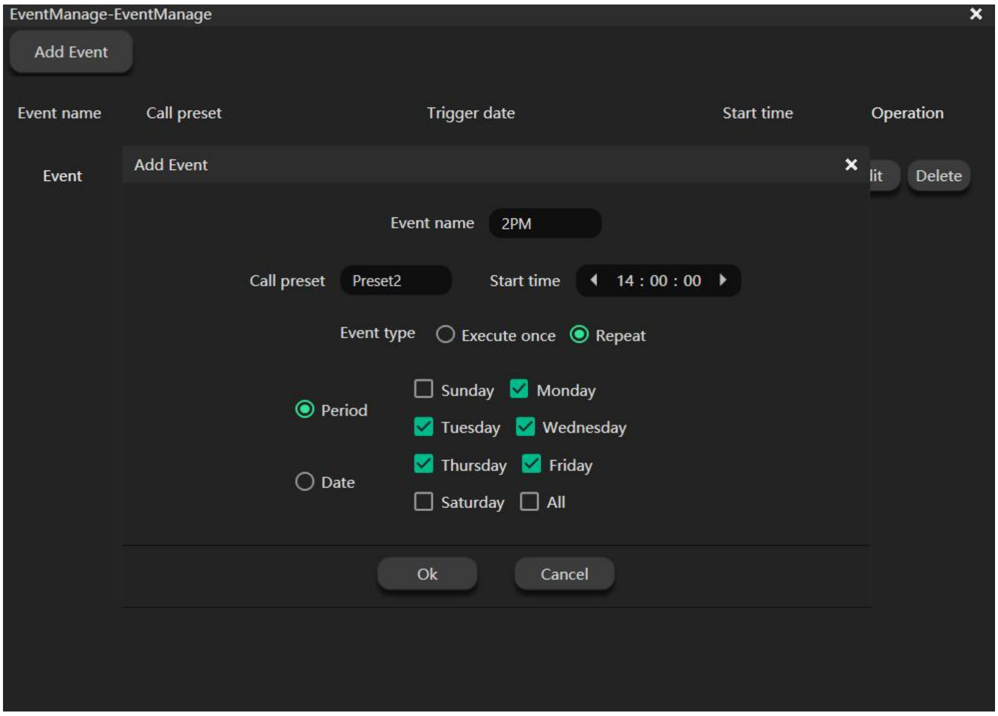
When an input-to-output connection requires level adjustment, right-click on the matrix routing node. A menu will appear with options such as Gain. Selecting Gain opens the Volume Control Window, where you can set the desired level.

The system also supports a range of quick operation functions, including:
(1) Copy Gain / Paste Gain (send level)
(2) Turn Row On / Off
(3) Turn Column On / Off
(4) Turn All Off
(5) Reset All Gain Values

Event Manage

At the bottom of the DSP diagram interface, an Event Management Control is built in. By double-clicking the module, you can open the event configuration window to set up corresponding events.

The Event Management function is designed to automatically execute predefined actions according to the configured or scheduled time and presets.



You can define an Event Name, and based on your requirements, specify whether the event should be executed once or on a recurring schedule.

For scheduled execution, you can set the date and time of the event.

In recurring mode, two types of repetition are available:

Period Mode: Select any combination of weekdays (Monday to Sunday) for cycle execution.

Date Mode: Specify exact dates (year, month, and day) for execution. Multiple dates can be added, and the execution will repeat according to your settings.

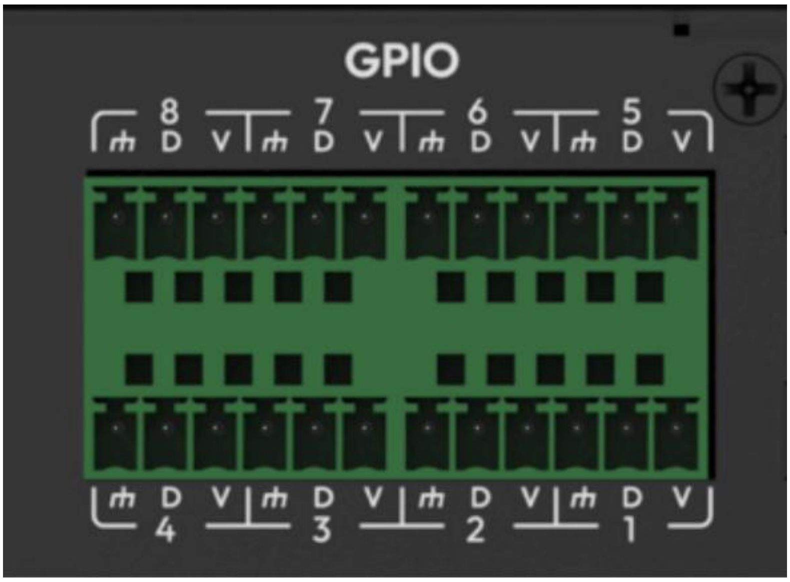
The Event Management system supports multiple event entries. You can add new events from the upper-left corner of the window. Once events are created, they can be edited or deleted in the Event Table, allowing you to modify or remove previously configured settings as needed.

Note: All event management settings must be configured in Online mode. If the program is compiled offline and then uploaded to the device, all previously configured event items will be cleared.

GPIO

At the bottom of the DSP diagram interface, there is a GPIO Control Module. By double-clicking the module, you can configure the GPI (Input) or GPO (Output) settings. This function is used in conjunction with the GPIO interface on the processor's rear panel.

All processor models in this series are equipped with eight (8) GPIO channels. In the software, you can define each channel's function: either GPI (Input) or GPO (Output), and set corresponding trigger and control types according to your requirements.

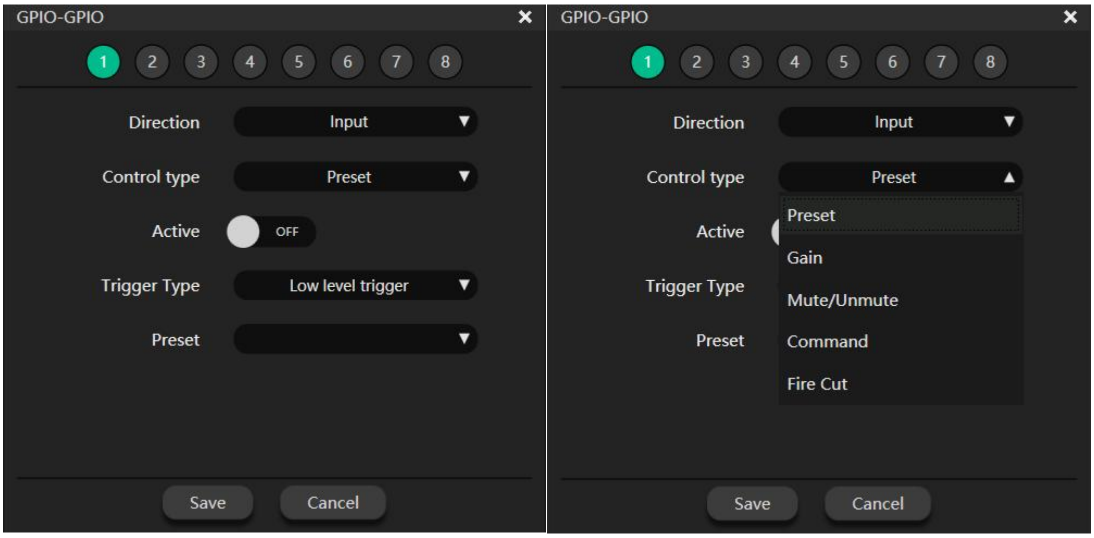


GPIO Hardware Interface Definition:
Each GPIO group on the processor's rear panel consists of three terminals:
V, D, and GND (Ground). The GND terminal is shared and can be paired with either V or D.
Each group can only be configured as one type (either input or output) within the software.

V (Voltage): Represents the voltage supply line.
D (Data): Represents the data signal line.
GND (Ground): Common reference terminal.

When the GND terminal is connected with the V terminal to an external device, the processor provides a constant supply voltage of approximately 3.3 V, with an output resistor of 10 ohms.

When the GND terminal is connected with the D terminal, the external device can trigger the processor through closed circuit, open circuit, or voltage signal.
Voltage input is supported in the range of 0~5 V. When the voltage exceeds 5 V, the processor's circuit protection mechanism will activate to prevent damage.
Similarly, the D-GND circuit supports both power receiving and power supplying modes, with a supply voltage of approximately 5 V.
This output can be triggered by the processor through preset configurations, level detection control, or mute switching, enabling a 5 V voltage output signal.



Software Configuration:

Using the GPIO Function Control in the software, you can define GPIO direction (Input or Output).

When set as Input (GPI), external devices trigger the processor to perform specific actions.

When set as Output (GPO), the processor triggers external devices when certain internal actions occur.

6. Central Control Protocol and Command Rules

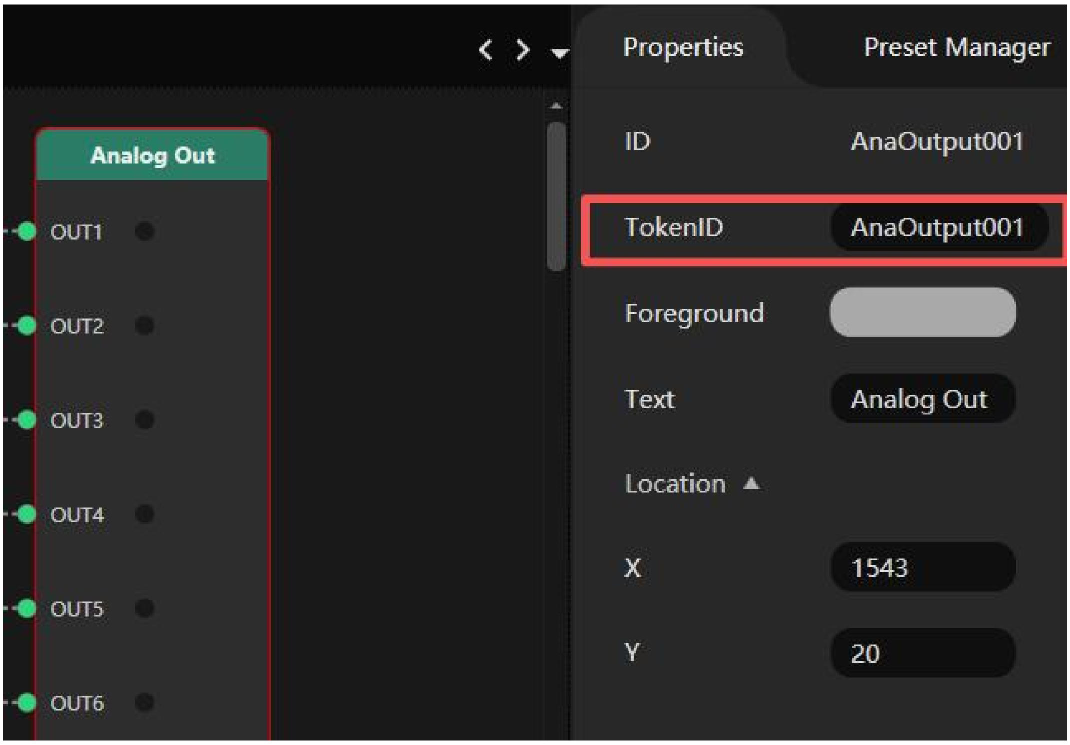
6.1 Control Protocol

This device series supports third-party central control integration.
Control data packets are transmitted via WebSocket communication, supporting both ASCII and JSON data formats.

The device supports the following communication protocols:
TCP (Client Mode): Port number **8001** (fixed, not configurable)
UDP: Port number **50000** (fixed, not configurable)
Serial (RS-232 / RS-485): Default baud rate **115200** (modifiable via the Device Discovery interface)

6.2 Command Rules

In control command coding, whether using ASCII or JSON format, control operations are executed based on the module ID or a custom Token ID.
These IDs can be queried within the program (note: if multiple modules share the same function, each will have a unique ID) and can also be modified (default IDs are pre-assigned but can be customized).



Example 1: Mute (ASCII Format)

For example, to mute one of the analog input channels, first locate the corresponding module ID.

If the module ID is AnaInput001, all control commands related to this module must reference this ID.

To mute Channel 3, the ASCII command is:

Set AnaInput001 Mute 2 1#

Explanation:

Set: Represents a configuration or control command.

AnaInput001: Module ID (you may also use the corresponding Token ID, which can be customized).

Mute: Mute function.

2: Channel index, counted from 0. Thus, 2 represents Channel 3.

When controlling multiple channels simultaneously, you may use 0, 1, 2, 3, which correspond to Channels 1~4.

1#: Apply mute; while 0# means unmute.

Example 2: Volume Adjustmente (ASCII Format)

If you want to adjust volume, the control command would be:

Set AnaInput001 Step 2 1#

This follows the same structure as above:

Step: Indicates step-based gain adjustment.

2: Channel index (Channel 3).

1#: Increase gain by +1 dB; changing to 2# increases gain by +2 dB. Using a negative value applies attenuation, e.g., -1# decreases gain by -1 dB.

Conclusion: Command structure rule (ASCII Format)

The general syntax is:

Set <ID> <Function> <Channel> <Action>

If you need detailed command definitions, please refer to the control-code specification table for the corresponding system.

7. User Interface

7.1 Overview

The User Interface (UI) is a key feature of the audio processor. It allows users to create and customize control interfaces according to specific needs or on-site layouts, enabling better system control and monitoring.

Through the user interface, users can easily access various functions of the audio processor and obtain intuitive visual feedback, helping them better understand the system's status and operation. Typical operations include adjusting volume, switching input sources, selecting audio effects, and configuring mixing and routing.

This series supports Web-based access UI, allowing cross-platform operation without additional software. Simply open a browser and enter the device's IP address.
Note: The controlling device and the processor must be on the same local network (LAN).

Key Features

(1) Customization: You can design and personalize your own control interfaces to meet specific application needs.

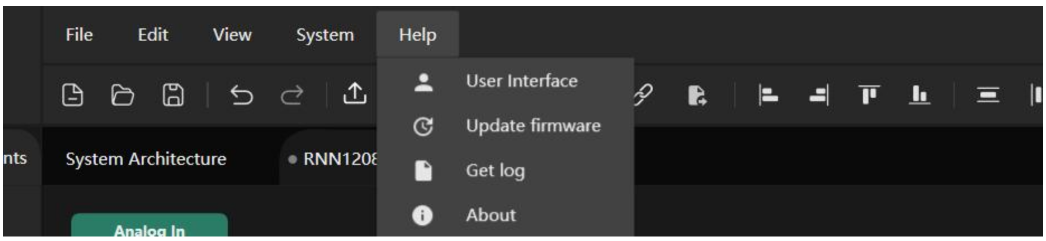
(2) Ease of Use: The interface employs intuitive graphics and controls (see illustration below).

(3) Efficiency: The user interface improves workflow efficiency by allowing quick access to and control of all system components.

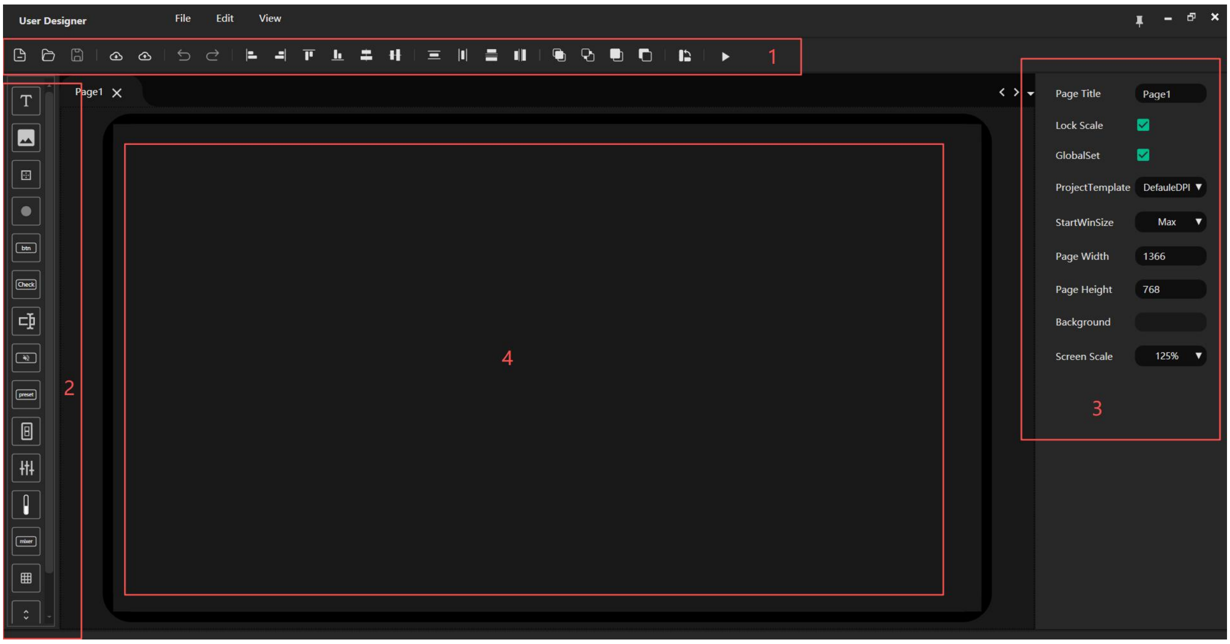
(4) Real-Time Feedback: Real-time monitoring provides visual status updates, ensuring accurate system supervision and control.

In summary, the user interface provides a personalized, intuitive, and efficient way to control and monitor the audio system.

To open the User Interface Editor, go to the software menu bar and click:
“Help” -> “User Interface.”



The editing interface is divided into five main areas:



Menu Bar / Toolbar: New page, Open, Save, Undo, Redo, Ruler, Alignment tools, Arrange tools, and Layer controls.

Component Panel: Text, Image, Border, Status Indicator, Buttons, Preset, Gain, Channel, Level, and Mixing Matrix controls.

Properties: Used to modify properties and parameters for selected components.

Editing Area: The main workspace where all modules and UI elements are edited and configured.

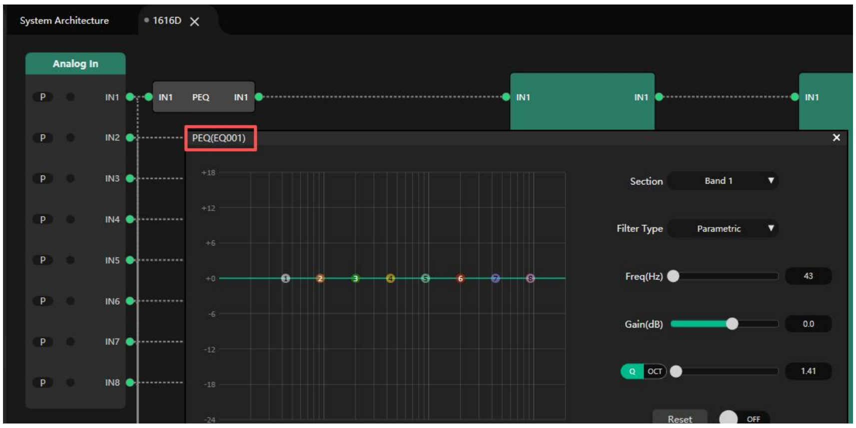
It is recommended to copy directly from the software modules that are already functional in your program.

Note: Controls from the left component library may differ slightly in properties from those copied from the main software.

7.2 Creating a User Interface

Within the user interface editor, drag and drop controls from the component panel into the Editing Area, then configure each component's settings. Then you can customize the appearance and layout, including color, size, position, and font.

Each control that corresponds to a processor module must have the correct Module ID and IP address configured to function properly. Module IDs can be found in the programming interface, displayed in properties bar or parentheses at the top-left of each module window.



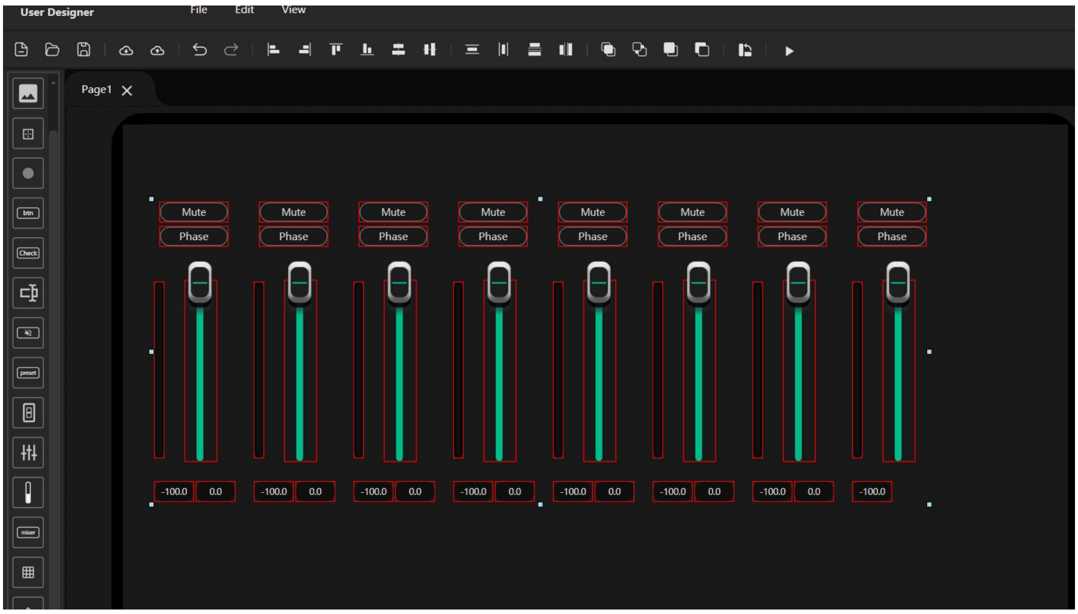
Copying Controls from the Programming Interface:

You can quickly copy existing functional controls from the programming interface to the user interface:

(1) Switch to the edit mode (non-compile mode).

(2) Double-click the module and select the desired control and feedback elements (use mouse box for group selection or Ctrl+A to select all).

(3) Press Ctrl+C to copy, then Ctrl+V to paste into the User Interface editing area.

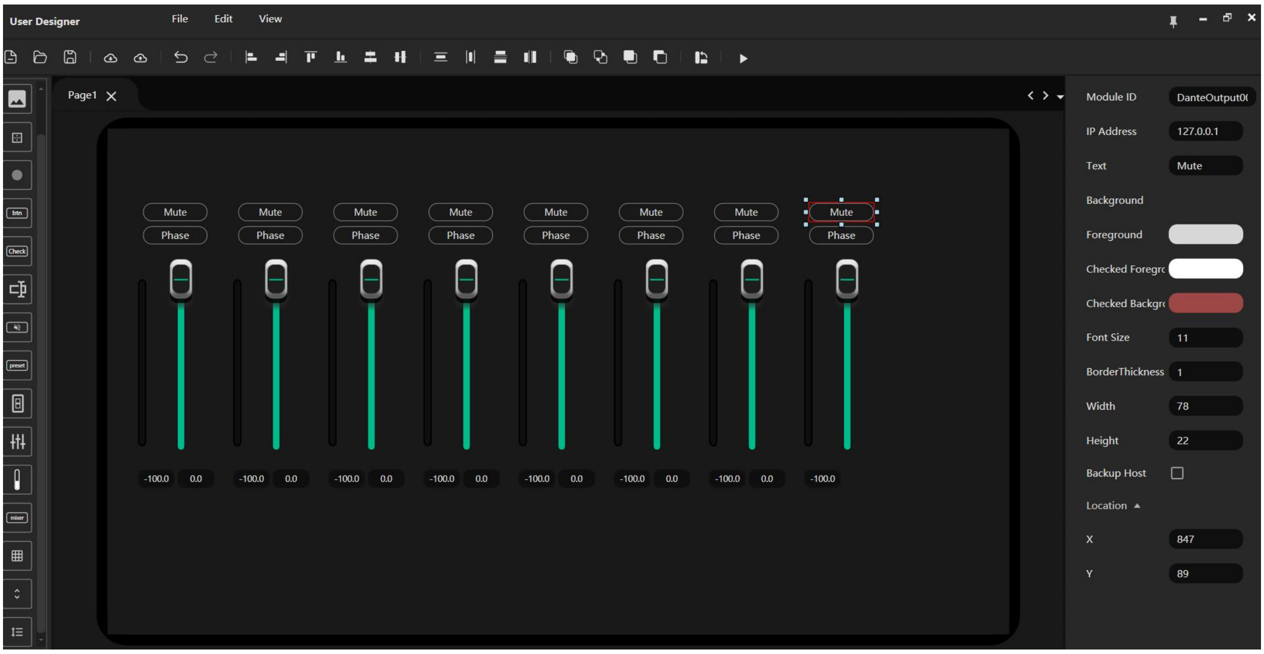


Note: Before creating or editing a user interface, always press F7 or click “Stop” to exit compile or online run mode.

If multiple software instances are open, controls can only be copied from the instance that opened the User Interface Editor.

7.3 Properties Adjustment

During UI designing, you can edit component properties in the right-hand Properties Bar, including: Text, Font, Color, Size, and Orientation, associated Device ID and IP Address and so on.



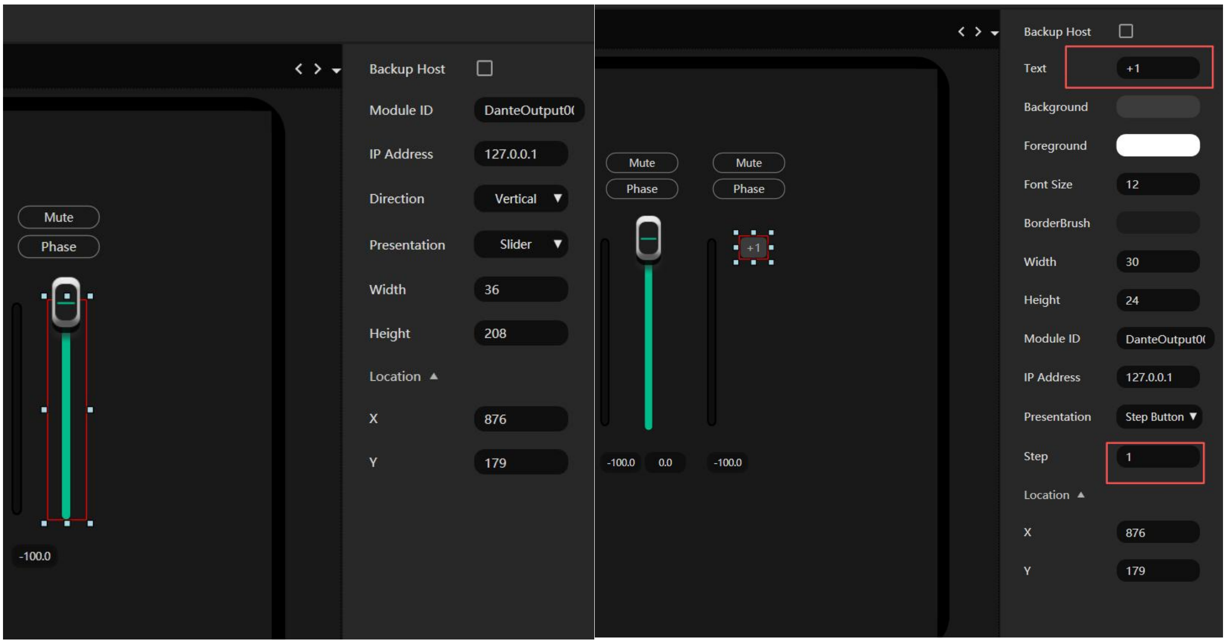
7.4 Control Presentation Variation

Certain controls allow changes to adapt to user preferences or interface styles.

For example, the Gain Sliders that copied from the main software can be converted into Step Button Style to perform incremental volume adjustments:

(1) Select the gain slider control and change presentation from "Slider" to "Step Button".

(2) Set the Step Value to 1, -1, or any value up to ±5.



This converts the fader into button-based controls for fixed step adjustments.

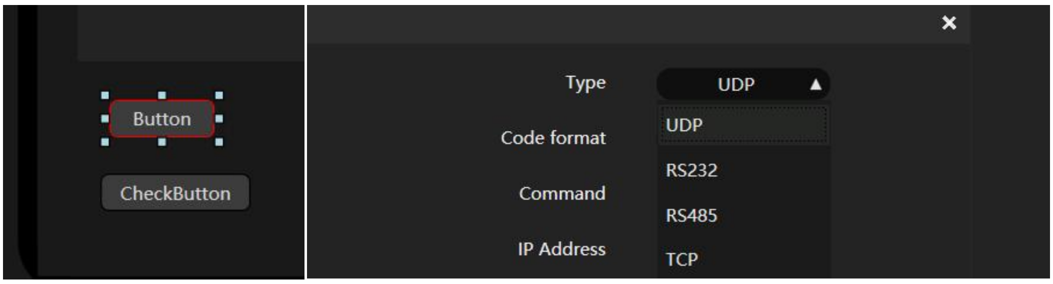
Among all interface controls, two button types: Button and CheckButton, support custom command editing:

(1) Right-click the control and select “Edit Command”.

(2) A pop-up window will appear, allowing you to configure custom commands.

Supported Protocols: RS-232 / RS-485(Serial), TCP / UDP (Network)

Data Formats: HEX (Hexadecimal) or ASCII Encoding

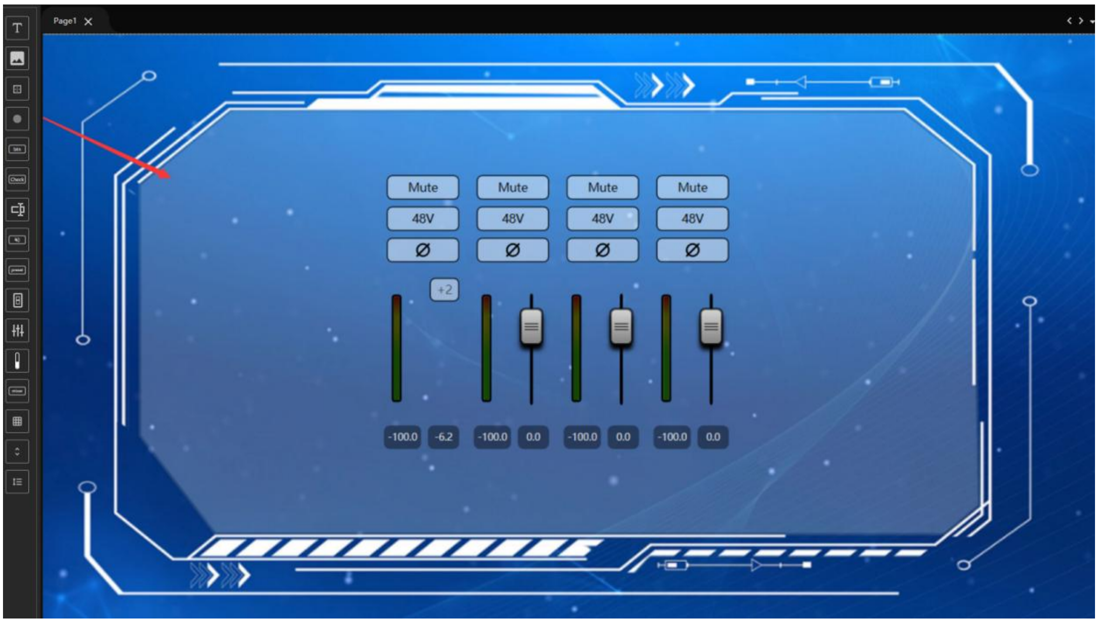


7.5 Pictures Incorporation

The user interface supports adding pictures or backgrounds to enhance visual appeal and usability.

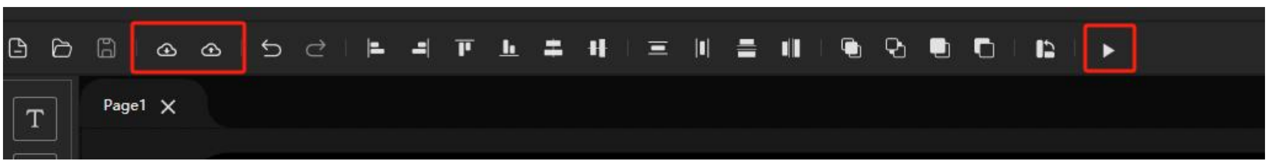
By integrating images, you can highlight key control areas, provide operation hints, or include descriptive elements to make the UI interface more intuitive and user-friendly.

To add an image:
(1) Drag a "Picture" Control from the Component Library into the editing area.
(2) In the Properties, select an image file from your local computer.



7.6 Uploading and Accessing the User Interface

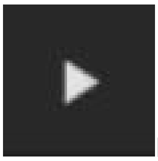
After completing your interface design, you can upload it directly to the device via the Toolbar Upload function.



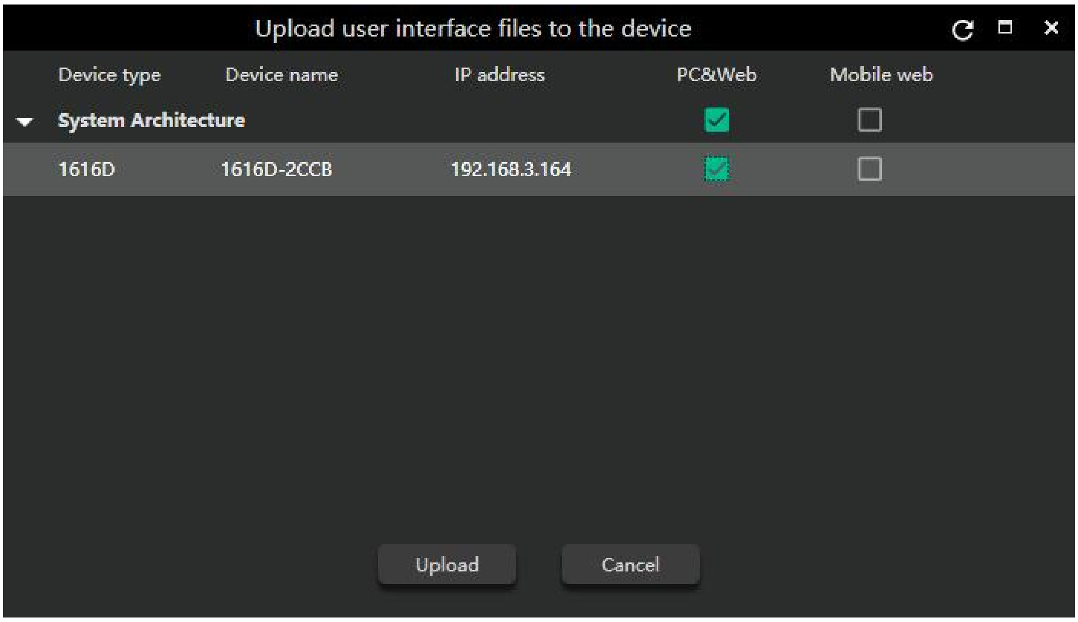
Toolbar icons perform the following functions:

Download the UI file from the device for further editing

Upload the current UI design to the device for control use

Run Interface for real-time control and testing with connected hardware

Recommended Workflow:
Test-run the interface locally to ensure all controls and displays function properly.
Once confirmed, upload the interface to the desired device.



Before uploading, ensure that you select the correct target device, and the appropriate resolution (PC Web or Mobile Web), which determine the canvas size and resolution used during interface design.

Once uploaded successfully, users within the same local network (wired or wireless) can access the interface by entering the device IP address in a web browser.

Thank you for reading this manual section.

We hope it helps you quickly understand and operate the device.

If you encounter any errors, unclear explanations, or technical issues, please contact us for prompt clarification and correction.