BLANC DSP48

Digital Speaker Manager

User Manual

Note:

Dante functionality is available only on applicable DSP48 variants and may not be included in all configurations.

Preface

The purpose of this section is to ensure that the user is able to use the product correctly through this manual in order to avoid danger in operation or property damage. Before using this product, please read the product manual carefully and keep it for future reference.

Outlined

This manual applies to Digital Speaker Manager.

This manual describes the functions and use of the various functional modules of the Digital Speaker Manager, and guides you through the installation and commissioning of the Digital Speaker Manager.

Symbol Conventions

The symbols that may be found in this document are defined as follows.

Symbol	Description
Note	Provides additional information to emphasize or supplement important points of the main text.
Caution	Indicates a potentially hazardous situation, which if not avoided, could result in equipment damage, data loss, performance degradation, or unexpected results.
Danger	Indicates a hazard with a high level of risk, which if not avoided, will result in death or serious injury.

Safety Instructions

Danger

To ensure reliable use of the equipment and the safety of personnel, please observe the following during installation, use and maintenance:

- During the installation and use of the equipment, all electrical safety regulations of the country and the region of use must be strictly observed.
- When installing the equipment, make sure that the power supply of the equipment is AC 100V-240V, 50/60Hz.
- Keep the working environment well ventilated so that the heat generated by the equipment during operation can be discharged in time to avoid damage to the equipment due to excessive temperature.

- Always unplug the unit's power adapter from the AC power outlet before: A. Removing or reinstalling any part of the equipment; B. Disconnecting or reconnecting any electrical plug or connection of the equipment. Do not operate with electricity.
- There are AC high-voltage parts in the equipment, non-professionals should not disassemble them without permission to avoid the risk of electric shock. Do not repair the equipment privately to avoid aggravating the damage.
- Do not spill any corrosive chemicals or liquids on or near the equipment.
- If the unit emits smoke, odour or noises, turn off the power immediately and unplug the power cord, and contact your dealer or service centre.
- If the appliance is not working properly, contact the shop where you purchased the appliance or the nearest service centre and do not disassemble or modify the appliance in any way. (We cannot be held responsible for problems caused by unapproved modifications or repairs).

Caution

- Do not drop objects on the equipment or vibrate the equipment vigorously, and keep the equipment away from locations with magnetic field interference. Avoid installing the equipment in a place where the surface vibrates or is susceptible to shock (neglecting this may damage the equipment).
- Do not use the equipment in high temperature, low temperature or high humidity environments. Refer to the equipment's data sheet for specific temperature and humidity requirements.
- Use the unit indoors, not in an exposed installation where it may be exposed to rain or extreme humidity.
- When the equipment is not used for a long period of time or in a humid and dewy environment, the main power supply of the equipment should be switched off.
- When cleaning the equipment, please use a sufficiently soft dry cloth or other alternatives to wipe the internal and external surfaces, do not use alkaline detergent to wash, and avoid hard objects to scratch the equipment.
- Please keep all the original packaging materials of the equipment properly, so that in case of problems, use the packaging materials to pack the equipment and send it to the agent or return it to the manufacturer for processing. We will not be responsible for any accidental damage in transit not caused by the original packaging materials.

Note

- Requirements for the quality of installation and commissioning personnel Qualifications or experience in the installation and commissioning of audio and video systems and qualifications to perform related work, in addition to the knowledge and operational skills listed below.

  − Basic knowledge and installation skills of audio and video systems and components.

- Basic knowledge and skills in low voltage cabling and wiring of low voltage electronics.
- Basic audio and networking knowledge and skills and the ability to read and understand the contents of this manual.

TABLE OF CONTENTS

Chapter 1 Product Introduction

1.1 Introduction

The Digital Speaker Manager Features an innovative structural design, it incorporates premium AD/DA conversion chips with support for 96kHz sampling rates and over 123dB dynamic range, achieving ultra-low noise down to -92dBu. Audio inputs and outputs support bothanalogandDante protocols. Its DSP utilizes a 40-bit floating-point ultra-high-performance processor, effortlessly handling complex audio algorithms.

The software provides a complete end-to-end speaker crossover solution. Each output channel features a stable, linear-phase 896-order custom FIR filter with 1500ms input/output channel delay. Common algorithms include Compressor, Expander, Graphic Equalizer, Parametric Equalizer, Dynamic Equalizer, Matrix Mixer, Crossover, RMS Limiter and Peak Limiter. Additional functions encompass input/output channel duplication, channel cross-binding, multi-device grouping for simultaneous tuning, and 50 scene presets. The device supports 100MHz Ethernet and high-speed USB interface, enabling automatic device discovery and rapid data communication for instant connectivity. Featuring an 800×268 high-resolution RGB LCD screen with flexible operation and quick-access function zones, it delivers full control capabilities independent of PC software.

This product is primarily designed for speaker control and processing, serving applications such as live sound reinforcement, theaters, nightclubs, concert halls, and other environments requiring speaker management.

1.2 Functions

- ☐ Input 4 analog signals and output 8 analog signals; inputs 4 digital signals via Dante and outputs 8 digital signals, enabling signal distribution management for multi-speaker scenarios;

- ☐ Built-in high-performance 40-bit floating-point DSP processing chip with 32-bit/96kHz processing capability, featuring high-performance A/D and D/A converters;

- ☐ 800×268 resolution high-definition RGB LCD screen with a user-friendly, innovative GUI interface;

- ☐ Stable, linear-phase 896-tap custom FIR filters; supports importing third-party software-generated FIR parameters with graphical interface display;

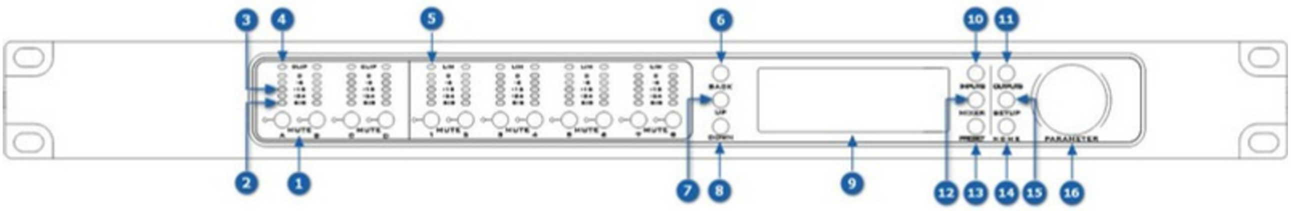
- ☐ Input channel components: Mute, Expander, Gain, Delay (0-1500ms), 31-band Graphic Equalizer, 12-band Parametric Equalizer, 3-band Dynamic Equalizer, Compressor;

- ☐ Output Channel Components: Delay (0-1500ms), Crossover (Butterworth, Bessel, Linkwitz-Riley filter types, 896 taps FIR filter), 12-band Parametric Equalizer, Gain, RMS Limiter,PeakLimiter;

- Professional control software for Windows, macOS, Linux, iPad and Android, connecting via USB or RJ45 cable for device software configuration;
- Software supports adding and routing management of multiple online devices;
- Input and output channel duplication functionality supported;
- Input and output channel cross-binding functionality supported;
- Supports multi-device grouping with 50 scene presets for saving, recalling, importing, and exporting;
- Features RS485 interface and standard Ethernet control interface with open protocol documentation.

Chapter 2 Interface Description

2.1 Front Panel



① Channel Mute Button and Mute Indicator;

② Channel Signal Indicator;

③ Channel Level Indicator;

④ Input Channel Clipping Indicator;

⑤ Output Channel Limiting Indicator;

⑥ BACK Button: Exit current menu or return to previous level;

⑦ UP Button: Previous item, for fine parameter adjustment;

⑧ DOWN button: Next item, for fine parameter adjustment;

⑨ LCD Display;

⑩ INPUTS button: Enter input components configuration interface;

⑪ OUTPUTS button: Enter output components configuration interface;

⑫ MIXER button: Enter matrix mixer interface;

⑬ SETUP button: Enter the setup interface;

⑭ PRESET button: Enter the scene preset configuration interface;

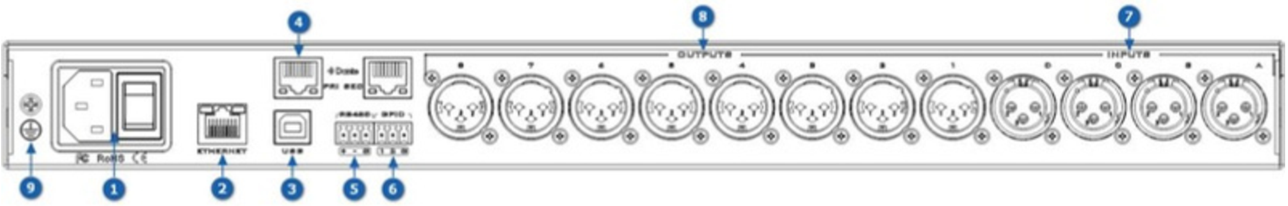
⑮ HOME button: Return to the main interface;

⑯ PARAMETER encoder knob:

1) Rotate: Adjust the selected parameter or scroll through the list;

2) Press: Confirm the current selection or execute the corresponding operation.

2.2 Rear Panel



① Power Connector: Connects to AC 100V-240V 50/60Hz power supply; toggle switch controls device power;

② ETHERNET: Network control interface, through the connection of this network port, the client computer can debug and monitor the device;

③ USB Type-B Interface: USB control interface, through the connection of this port, the client computer can debug and monitor the device;

④ Dante: Dante network audio transmission interface, equipped with primary and secondary dual network interface, can be used for redundant backup of Dante network signals;

⑤ RS485 Interface: Connects to control terminals or central control device;

⑥ GPIO Interface: GPIO input;

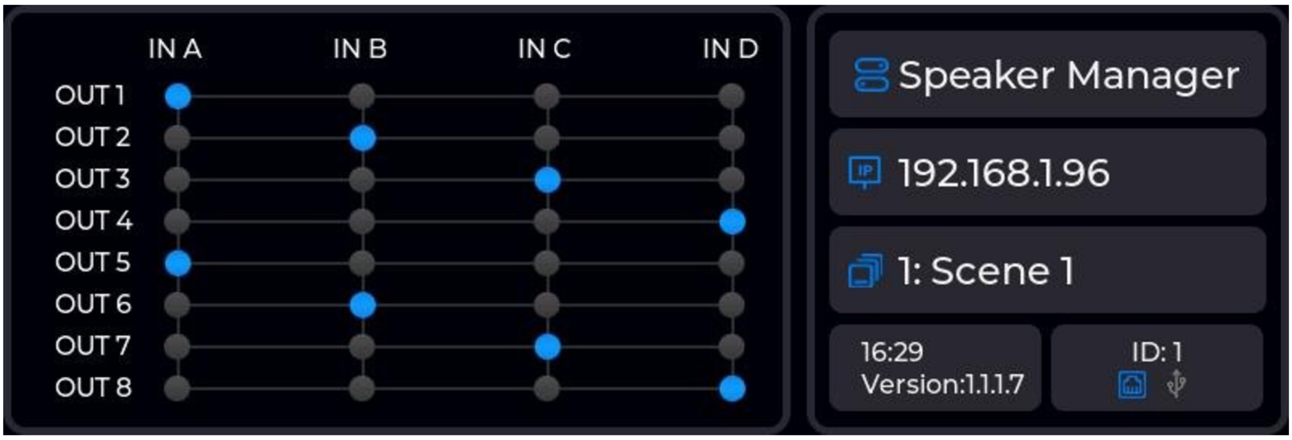
⑦ INPUT: Analog input interface, connects to instruments, mixing consoles, PC and other devices;

⑧ OUTPUT: Analog output interface, connects amplifiers, powered speakers and similar devices;

⑨ Ground screw: Used to ground the chassis, play accidental leakage safety protection, electrostatic balance and other protective measures.

Chapter3 FrontPanelOperationUserGuide

3.1 Home



① Displays connection status between Inputs (IN A～D) and Outputs (OUT 1～8) in a matrix form;

② Displays device name;

③ Displays device's current IP address;

④ Displays device's current scene preset;

⑤ Displays device runtime and current software version;

⑥ ID: Displays the unique ID when the device is configured as an RS485 slave;

⑦ Network Icon: Displays the current network connection status. The icon turns blue when the control software uses a network connection;

⑧ USB Icon: Displays the USB connection status. The icon turns blue when the control software uses a USB connection.

3.2 Input Component Configuration

Input channel components are used to adjust various parameters within the signal chain of each input channel.

① **Source:** Input sources can be selected as Analog or Dante;

② **Expander:** Expander component;

③ **Gain:** Controls the Gain of the channel;

④ **Delay:** Delay component;

⑤ **GEQ:** Graphic Equalizer component;

⑥ **PEQ:** Parametric Equalizer component;

⑦ **DEQ:** Dynamic Equalizer component;

⑧ **Compressor:** Compressor component.

3.2.1 Input Source

Input sources can be selected as Analog or Dante. The Analog Input provides line-level input for devices with line-level outputs, and inputs for instruments, mixing consoles, etc. The Analog Input converts the analog input signal to a processed digital. Connections are made using one XLR female connector.

3.2.2 Input Expander

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

① **Threshold (-80～0):** Sets the point from which the attenuation is calculated based on the Ratio setting. This is where the Expander starts working. When the input signal falls below the Threshold Level, the Expander compresses the signal at a preset Ratio to increase its dynamic range. When the input signal exceeds the Threshold Level, the signal is output at a 1:1 Ratio, preserving its original dynamic range. The Threshold setting requires adjustment based on the specific application scenario;

Example:

- If the: Threshold Level is -30dB; Ratio is 2.5; Input level is -40dB
- Then the Adjusted Output is:
- [(Input Level - Threshold Level) * Ratio] + Threshold Level = Output Level
- {[-40dB - (-30dB)] * 2.5} + (-30dB) = -55dB.

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

③ **Attack Time (1 ～ 500):** The time required for an input signal less than the Expander Threshold Level to enter the expansion state and to output at the set expansion ratio. Shorter attack time enable the Expander to respond quickly to signal changes, making it suitable for processing rapidly changing audio signals; longer attack time provide smoother transitions, preventing overly abrupt processing effects, and are ideal for gentle signals like vocals or music;

④ **Release Time (1～500):** The time required for the input signal level to return from the extended state to the original non-extended state. Shorter release time enables rapid restoration of signal dynamics, making it suitable for fast-changing audio, but it may induce pumping effects (rapid fluctuations in signal level); longer release time provides smoother transitions and reduces pumping effects, but it may cause the signal restoration process to appear sluggish. Therefore, release time settings should be flexibly adjusted based on audio characteristics and processing requirements;

⑤ **Input Level:** Graphically displays the level of the input signal;

⑥ **Gain Reduction:** Graphically displays the amount of attenuation between the signal processed by the Expander and the input signal. The Gain Reduction reflects the degree of signal attenuation applied by the Expander;

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

⑧ **Enable/Disable:** Enable or Disable the Expander for the current channel. When the Expander is disabled, audio is passed through without any change.

3.2.3 Input Gain

① **Gain:** Select the Gain Fader, the channel Gain Fader adjustment method is based on the step fine-tuning of the up and down arrow keys in steps of 1 dB, the range is -72～12dB;

② **Input Level:** Graphically displays the level of the input signal;

③ **Polarity :** Inverts the polarity of the output signal. In audio processing, phase is a critical parameter that determines the starting point and direction of a signal waveform. Through phase inversion, the phase of an audio signal is reversed by 180 degrees. In multi-speaker systems, inconsistent signal phases between different speakers can cause sound cancellation or interference. The phase inversion function allows adjustment of the phase to ensure sound clarity and consistency;

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

3.2.4 Input Delay

The Delay component is primarily used to apply time delays to audio signals, enabling various audio effects and optimizing audio system performance. By activating the delay during signal processing and imposing specific time delays on the audio signal, it alters the signal propagation time to achieve specialized sound processing. Users can configure the delay component with fixed delay durations according to their requirements. The delay component supports a delay time range of 0～1500 milliseconds, accommodating diverse application scenarios.

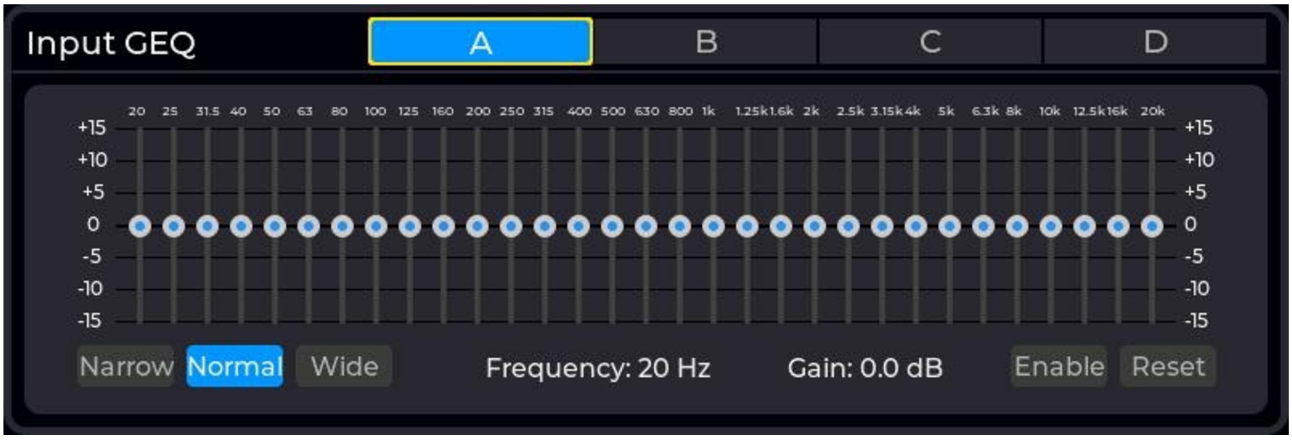
- Reverb Effect: By setting an appropriate delay time, the reverb effect simulates the reflection and diffusion of sound in a space, enhancing the sound's spatial and three-dimensional quality, as if you were immersed in a specific acoustic environment.
- Echo Effect: By using a delay unit to generate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

① **Delay time:** Delay time range (0～1500ms);

② **Enable/Disable:** Enable or Disable the Delay for the current channel. When the Delay is disabled, audio is passed through without any change.

3.2.5 Input Graphic Equalizer

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



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

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

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

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

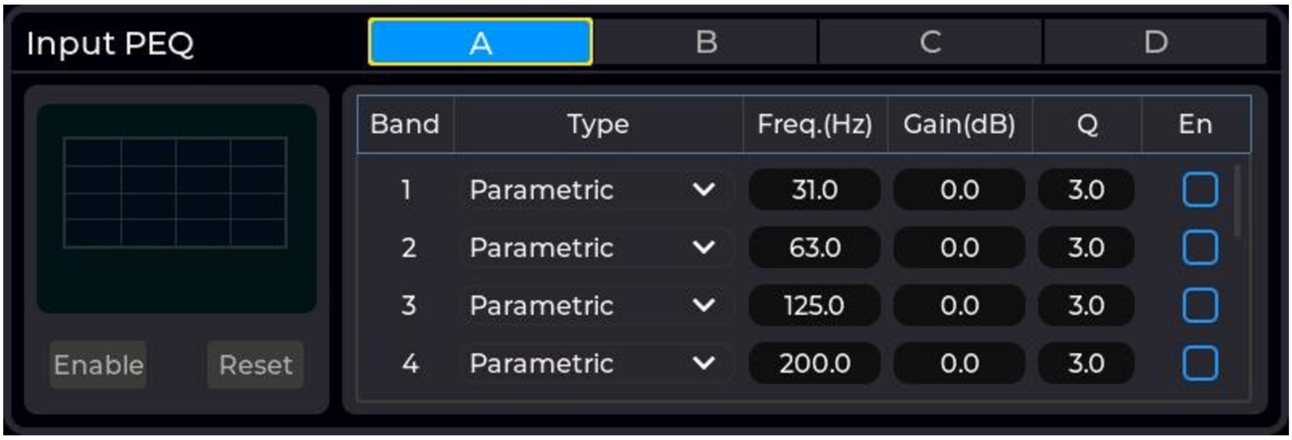
⑤ **Gain:** Controls the output gain of an individual frequency band, positive values indicate an increase in gain for that frequency band, while negative values indicate a decrease in gain for that frequency band;

⑥ **Enable/Disable:** Enable or Disable the Graphic Equalizer for the current channel. When the Graphic Equalizer is disabled, audio is passed through without any change;

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

3.2.6 Input Parametric Equalizer

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



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

- **Parametric Equalization Filter:** Parametric Equalization Filter is an adjustable filter used to precisely boost or attenuate specific frequencies in an audio signal. It optimizes the frequency response of audio by adjusting the center frequency, gain, and bandwidth;
- **Low-Pass Filter:** Allows low-frequency signals to pass through and cut off high-frequency signals based on the set frequency, typically used to remove high-frequency noise or increase low-frequency components;
- **High-Pass Filter:** Allows high-frequency signals to pass through and cut off low-frequency signals based on the set frequency, typically used to remove low-frequency interference or extract high-frequency features;
- **All-Pass Filter:** Passes signals of all frequencies with identical gain but alters the signal's phase response. An all-pass filter does not modify the amplitude spectrum of the signal. Instead, it improves system transient response, corrects frequency response defects caused by phase issues, or creates intriguing sonic effects by precisely adjusting phase and delay relationships.
- **Band-Pass Filter:** Allows only signals within a specific frequency range (pass band) to pass through, while attenuating or blocking all frequency signals outside this range (low and high frequencies);
- **Low-Shelf:** The Low shelf is a gain increased or attenuated for the frequency portion below the set center frequency, typically used to increase the richness of low frequencies or reduce low frequency rumbling;
- **High-Shelf:** The High shelf is a gain increased or attenuated for the frequency portion above the set center frequency, typically used to increase the clarity of high frequencies or reduce the harshness of high frequencies.

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

③ **Gain:** Controls the Gain for an individual frequency band. Users can adjust the gain to increase or decrease the signal strength of specific frequencies. For example, if the sound in a certain frequency range of an audio signal is too loud or too soft, the volume can be balanced by increasing or decreasing the gain. The adjustment range of the gain is usually from negative values (attenuation) to positive values (increase). This setting is disabled when selecting All-Pass, Band-Pass, Low-Pass, High-Pass filter;

④ **Q-Factor:** The Q-factor refers to the range of frequencies affected around the center frequency. Set the Q-factor for a single band in the equalizer, ranging from 0.4 octaves to 128 octaves (default value is 3.00). This setting is disabled when selecting Low-Shelf, High-Shelf, Low-Pass, or High-Pass, unless marked as Vari-Q (Variable Q-factor). The Q-factor determines the precision and scope of equalization adjustments. Higher Q-factor yield narrower bandwidth and more precise frequency range control; lower Q-factor provide wider bandwidth and broader frequency influence;

⑤ **Enable/Disable:** Enable or Disable the Parametric Equalizer for an individual frequency band. When an individual frequency band is disabled, audio is passed through without any change;

⑥ **Enable All/Disable All:** Enable or Disable All the Parametric Equalizer for the current channel. When the Parametric Equalizer is disabled, audio is passed through without any change;

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

3.2.7 Input Dynamic Equalizer

The Dynamic Equalizer component is an intelligent audio signal processing tool that combines the precise frequency control capabilities of a multi-band equalizer with the dynamic processing characteristics of a compressor. Unlike traditional static equalizers that apply fixed gain changes at all times, dynamic equalizers automatically adjust the gain of specific frequency bands based on the dynamic changes in the audio signal. This means it can boost or attenuate the input signal level when the signal level exceeds a set threshold and the frequency matches the preset frequency, while remaining static during normal signal conditions. This prevents excessive processing of the audio.

① **Threshold (-54～0):** Sets the level where Dynamic Equalizer begins to work. This is the point from which the amount of dynamic gain is calculated based on the Ratio setting;

② **Mode:** Controls the direction of dynamic response, with four modes available for selection:

- **Cut Above:** Attenuates when the signal exceeds the Threshold Level, similar to dynamic clipping or compression, often used to control rumble or sibilance;
- **Boost Above:** Boosts when the signal exceeds the Threshold Level, used to increase dynamic peaks in the signal, adding impact or brightness to a specific frequency band;
- **Cut Below:** Begins attenuating when the signal falls below the Threshold Level, used to remove background noise, tail sounds, and other low-level content;
- **Boost Below:** Begins boosting when the signal falls below the Threshold Level, used to increase the presence of weak signal segments.

③ **Ratio (1 ～ 4):** The ratio between the Input and the Output as measured from the Threshold Level. Controls the response rate of dynamic gain changes or the perceived compression intensity, influencing the "steepness" of dynamic processing;

④ **Filter Type:** Including Parametric Equalization, Low-Shelf, High-Shelf Filter;

- **Parametric Equalization Filter:** Parametric Equalization Filter is an adjustable filter used to precisely boost or attenuate specific frequencies in an audio signal. It optimizes the frequency response of audio by adjusting the center frequency, gain, and bandwidth;
- Low-Shelf: The Low shelf is a gain increased or attenuated for the frequency portion below the set center frequency, typically used to increase the richness of low frequencies or reduce low frequency rumbling;
- High-Shelf: The High shelf is a gain increased or attenuated for the frequency portion above the set center frequency, typically used to increase the clarity of high frequencies or reduce the harshness of high frequencies.

⑤ **Frequency:** Sets the center Frequency of an individual band. Frequency is one of the core parameters in audio processing. In audio signals, different frequencies correspond to different sound characteristics. For example, low frequencies are typically associated with heavy drum sounds or bass, mid-frequencies involve vocals and the timbre of most instruments, and high frequencies are related to bright timbres or details. By selecting the appropriate center frequency, users can precisely enhance or attenuate the timbre of specific instruments, optimize the clarity of vocals, or resolve frequency issues in audio;

⑧ **Max Effect:** Controls the Gain or Attenuation for the center frequency, constituting the maximum dynamic gain variation achievable ("+" or "-", depending on the mode). **Note:** This is the "limit" for dynamic variation; the actual effect depends on the signal level;

⑨ **Q-Factor:** The Q-factor refers to the range of frequencies affected around the center frequency. Set the Q-factor for a single band in the equalizer, ranging from 0.4 octaves to 128 octaves (default value is 3.00). The Q-factor determines the precision and scope of equalization adjustments. Higher Q-factor yield narrower bandwidth and more precise frequency range control; lower Q-factor provide wider bandwidth and broader frequency influence;

⑩ **Attack Time (1 ～ 1000):** Attack time is how fast the dynamic gain reacts to a signal outputting at the set Ratio when the input signal meets the Threshold Level condition;

⑪ **Release Time (1～1000):** Release time is how fast the dynamic gain reacts when the signal no longer meets the Threshold Level condition and gain is restored to its non-limited level;

⑫ **Enable/Disable:** Enable or Disable the Dynamic Equalizer for an individual frequency band; When an individual frequency band is disabled, audio is passed through without any change.

3.2.8 Input Compressor

The purpose of the Compressor component is to control the dynamic range of the Output above a set Threshold Level, thereby optimizing audio balance and consistency. Compressor are widely used in music production, live sound reinforcement, broadcasting, and voice processing, helping users control audio signal peaks, prevent distortion, and enhance overall audio clarity and audibility by raising the average signal level

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

⑥ **Threshold (-80～0):** Sets the level where compression begins. This is the point from which the amount of attenuation is calculated based on the Ratio setting. A level below the Threshold Level is not compressed, anything above the Threshold Level attenuation is applied;

Example:

- If the: Threshold Level is -30dB; Ratio is 2.5; Input level is -10dB
- Then the Adjusted Output is:
- [(Input Level - Threshold Level) / Ratio] + Threshold Level = Output Level
- {[-10dB - (-30dB)] / 2.5} + (-30dB) = -22dB.

⑦ **Ratio (1 ～ 20):** The ratio between the Input and the Output as measured from the Threshold Level. The closer the Ratio is to 20, the smaller dynamic changes in the Output level. As the Ratio is adjusted closer to 1, the dynamic range of the Output increases;

⑧ **Attack Time (1～1000):** Attack time is how fast the compressor reacts to a signal crossing the set threshold going up. Short attack time compressors can quickly capture signal peaks, making them suitable for percussion instruments, but if the attack time is too short, it can produce "breathing sounds" and lose naturalness; Long attack times provide smooth transitions, making them suitable for vocals and other gentle signals, preserving more dynamics and details;

⑨ **Release Time (1～1000):** Release time is how fast the compressor reacts when the signal drops below the threshold and gain is restored to its non-limited level. Fast release time can increase signal loudness, but is prone to suction effects; Slow release time provides a smooth transition and reduces suction effects, but may sound sluggish. Settings should be balanced according to audio characteristics;

⑩ **Output Gain (-24～30):** Controls the Gain of the output, used to compensate for the reduction in signal level caused by compression processing. When an audio signal is compressed, its overall volume decreases. The function of Output Gain is to restore the compressed signal to a volume level close to that before compression by raising the output signal level;

⑪ **Input Level:** Graphically displays the level of the input signal;

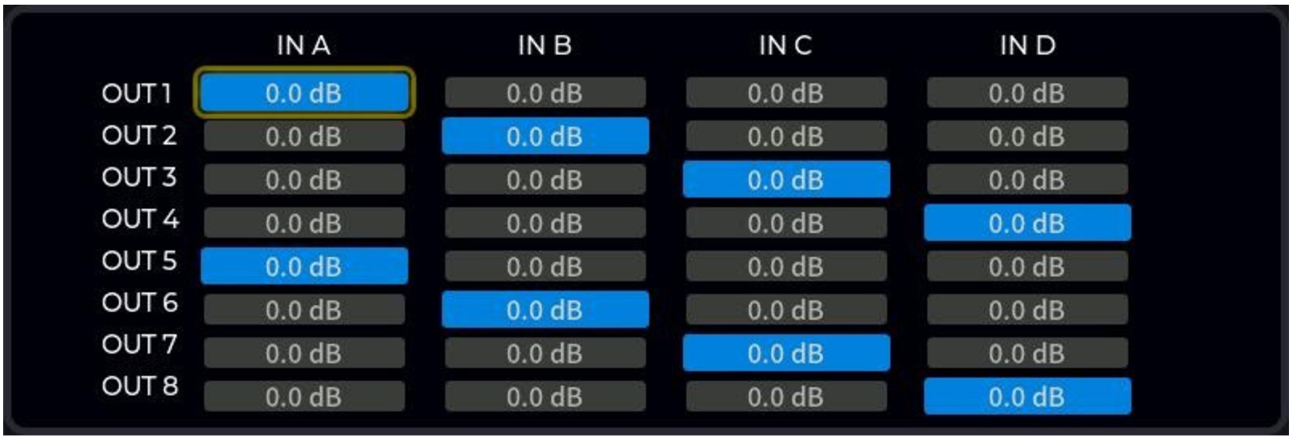
⑫ **Gain Reduction:** Graphically displays the amount of attenuation between the signal processed by the Compressor and the input signal. The Gain Reduction reflects the degree of signal attenuation applied by the Compressor;

⑬ **Output Level:** Graphically displays the level of the output signal;

⑭ **Enable/Disable:** Enable or Disable the Compressor for the current channel. When the Compressor is disabled, audio is passed through without any change.

3.3 Matrix Mixer

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



Control whether each channel matrix mix is enabled or disabled.

3.4 Output Component Configuration

Output channel components are used to adjust various parameters within the signal chain of each output channel.

① **Delay:** Delay component;

② **X-Over:** X-Over component;

③ **PEQ:** Parametric Equalizer component;

④ **Gain:** Controls the Gain of the channel;

⑤ **RMS Limiter:** RMS Limiter component;

⑥ **Peak Limiter:** Peak Limiter component;

3.4.1 OutputDelay

Reference Input Component Configuration - Input Delay.

3.4.2 Output X-Over

The Crossover component divides the audio input signals into 3 frequency bands: Low-Pass, Band-Pass and High-Pass, you can set the Slope rates and filter types (Butterworth, Linkwitz-Riley, Bessel) for each filter in each band. Crossover play an indispensable role in fields such as audio processing, sound system design, and professional audio production.

① **Filter Type:** The Type setting includes: Bessel, Butterworth, Linkwitz-Riley filters. You can select any combination of two of these filters for a band's high-pass and low-pass frequencies;

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

② **Slope:** Determines the rate of change of attenuation at the high-pass and low-pass frequencies of the band. The slope setting establishes the crossover region between two adjacent bands. The Slope includes 6dB/Oct, 12dB/Oct, 18dB/Oct, 24dB/Oct, 32dB/Oct, 36dB/Oct, 42dB/Oct, 48dB/Oct;

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

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

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

⑤ **FIR Filter:** Finite impulse response filter, extract desired frequency components from complex signals. Their core capability lies in their precisely designed frequency response and phase linearity.

⑥ **FIR File:** Displays the FIR file imported using the control software;

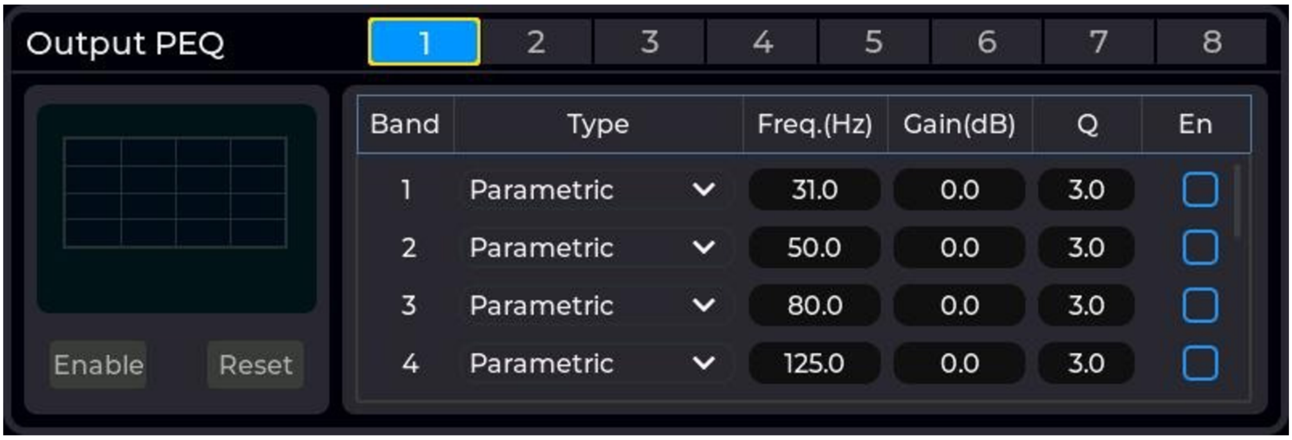
⑦ **Taps:** FIR files maximum 896 taps;

⑧ **Latency:** Displays the latency calculated from the FIR filter file;

⑨ **Latency Link:** Aligns the output channel's latency.

⑩ **FIR Enable/Disable:** Enable or Disable the FIR filter for the current channel. When the FIR filter is disabled, audio is passed through without any change.

3.4.3 Output Parametric Equalizer



Reference Input Component Configuration - Input Parametric Equalizer.

3.4.4 OutputGain

Reference Input Component Configuration - Input Gain.

3.4.5 OutputLimiter

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

Each output channel is equipped with two-stage Limiters. One stage is an RMS Limiter that responds to the RMS level of the signal, while the other stage is a Peak Limiter specifically designed to handle signal peaks.

- **RMS Limiter** continuously monitors the input signal's RMS, which more closely approximates the loudness perceived by the human ear. It activates only when the signal's average energy exceeds a preset threshold, reducing gain accordingly;
- **Peak Limiter** continuously monitors the instantaneous peaks of the input signal. Should the amplitude surpass the set threshold, it operates according to a defined attack time to "clip" the signal's highest level.

① **Threshold (-80 ～ 0):** Sets the level at which the Limiter has an effect, and the level at which the output is held;

② **Attack Time (1 ～ 1000):** Attack time is how fast the Limiter reacts to a signal crossing the set threshold going up. Short attack time compressors can quickly capture signal peaks, making them suitable for percussion instruments, but if the attack time is too short, it can produce "breathing sounds" and lose naturalness; Long attack times provide smooth transitions, making them suitable for vocals and other gentle signals, preserving more dynamics and details;

③ **Release Time (1～500):** The release time determines how long it takes for the input signal to return to its original dynamics from maximum attenuation. It dictates how long the limiter continues to attenuate the signal after it falls below the Threshold Level. Shorter release time enables rapid signal dynamics release, suitable for fast-changing audio but prone to Pumping effect (rapid fluctuations in signal level). Longer release time provides smoother transitions and reduces Pumping effect, though it may make the signal release process appear sluggish;

④ **Input Level:** Graphically displays the level of the input signal;

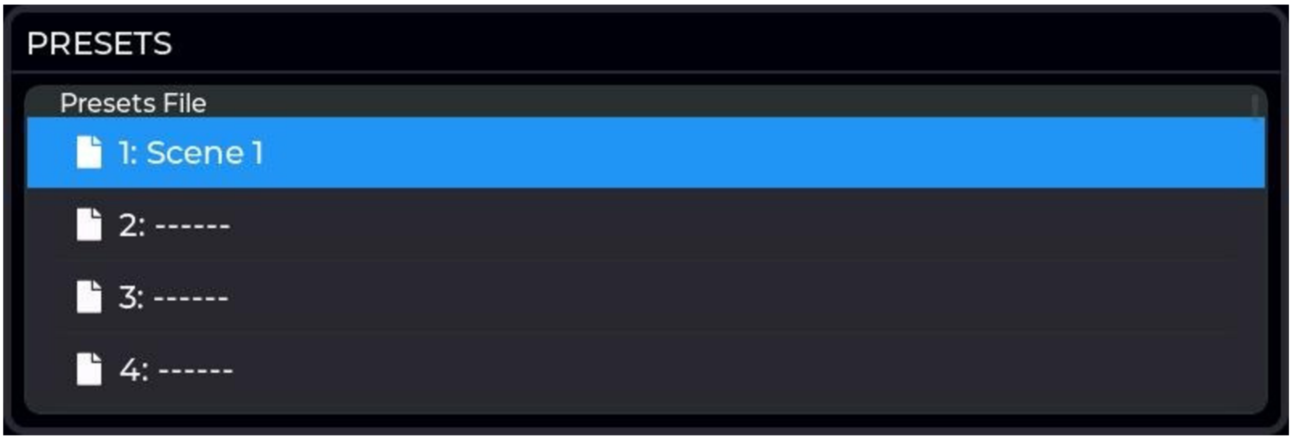
⑤ **Gain Reduction:** Graphically displays the amount of attenuation applied to the Channel, Gain Reduction reflects the degree to which the limiter attenuates the signal. For example, if the input signal exceeds the threshold by 3dB, the limiter may attenuate the signal by 3dB, resulting in a compression of 3dB;

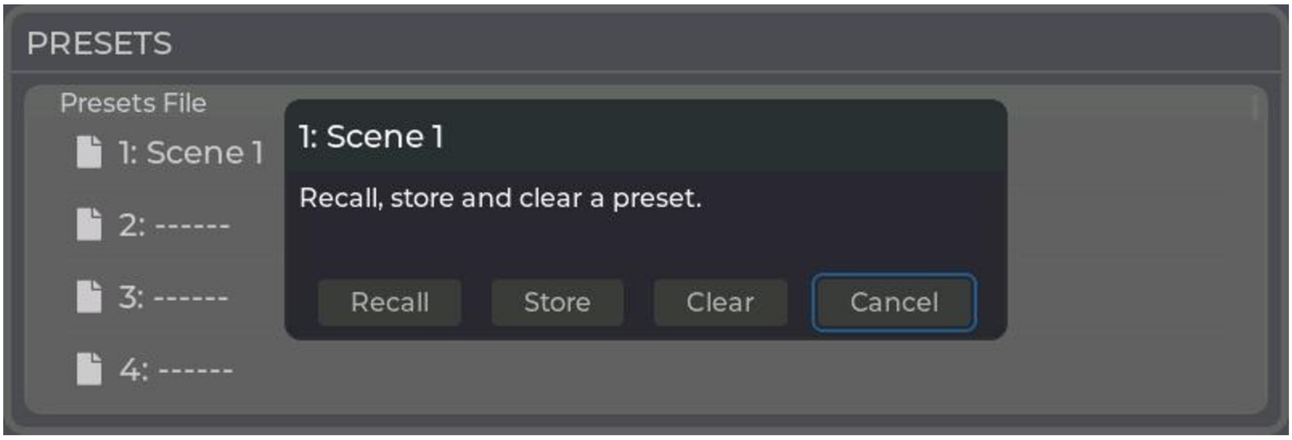
⑥ **Output Level:** Graphically displays the level of the output signal.

3.5 Presets

The scene presets provides users with a convenient scene management platform that supports scene Store, Recall, Delete. Through this component, users can flexibly manage various scene files to meet the needs of different scenarios while ensuring the accuracy and traceability of scene parameters.

The Speaker Manager provides 50 scene presets archive locations.





① **Store:** Stores the current scene configuration to the specified archive location;

② **Recall:** Recall a scene preset from the specified archive location;

③ **Clear:** Clear a scene preset configuration from the specified archive location.

3.6 Setup

① **Device Network:** Configurable device network;

② **Dante Network:** Configurable Dante network for the device;

③ **LCD:** Adjustable LCD backlight brightness and screen off time for the device display;

④ **GPIO Configuration:** Provides 2 GPIO input interfaces with 4 selectable modes:

1) Disabled: Disables GPIO functionality;

2) Fire Alarm Mute Contact Closed: Mutes all channels upon trigger;

3) Fire Alarm Mute Contact Open: Unmutes all channels upon trigger;

4) Preset Recall Contact Closed: Recalls specified preset.

3.6.1 RS485

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

① **Baud Rate:** Setting the baud rate determines the data transmission rate for serial communication. Users can select an appropriate baud rate based on communication requirements and device compatibility. Multiple baud rates are available, including 9600, 19200, 38400, 57600, and 115200. The default baud rate is 9600;

② **Mode:** Selectable between Slave and Master. Typically, an RS485 network has only one master, while one or more slaves can exist on the same network. The Master does not respond to control commands on the RS485 bus; instead, it forwards control commands received through the network port that are not addressed to itself via the RS485 bus. Slaves respond to control commands on the RS485 bus that are addressed to them;

③ **Unique ID:** If the mode is set to slave, configure the slave's unique ID. Multiple slaves require distinct unique IDs;

④ **Communication Status:** Displays the RS485 communication status.

3.6.2 System

The system interface displays the device's current sample rate, clock source, DSP resource, device version, DSP version, and health status information.

① **Sample Rate:** Displays the device's current sample rate, synchronized with Dante's sample rate. **Note:** If Dante's sample rate is set to 48k, the device sample rate will display 48k; if Dante's sample rate is set to 96k, the device sample rate will display 96k;

② **Language:** Supports English and Chinese;

③ **Reset:** Restore default settings.

Chapter 4 Control Software User Guide

4.1 Software Installation

The client control software for the Speaker Manager is stored on a portable USB drive, which is placed inside the packaging box alongside the main unit. Insert the portable USB drive into the USB port of the PC where installation is desired. Navigate to the portable USB drive, double-click the Manager software package, and follow the on-screen prompts to complete the installation.

Note: Before installing the client software, ensure your Windows operating system has .NET 8.0 Desktop Runtime installed.

4.2 Main Interface

① **Menu;**

② **Device List:** Automatically searches for online devices and displays them in the list;

③ **Input Channel Selection:** Selects or indicates the currently configured channel;

④ **Matrix Mixing:** Click to enter the matrix mixing interface;

⑤ **Output Channel Selection:** Select or indicate the currently configured channel;

⑥ **Scene Preset:** Select the scene preset to use;

⑦ **Input Components:** Includes input Source selection, Mute, Expander, Gain, Polarity, Delay, Graphic Equalizer, Parametric Equalizer, Dynamic Equalizer, Compressor;

⑧ **Matrix Mixer:** Signal connections display input-to-output mixing status;

⑨ **Output Components:** Delay, Crossover, FIR Filter, Parametric Equalizer, Gain, Polarity, Limiter,Mute;

⑩ **Input Controls:** Channel name customization, Invert, Gain Control, Mute;

⑪ **Output Controls:** Channel name customization, Invert, Gain Control, Mute;

⑫ **Input Link:** Performs linked operations on selected channels to configure multiple channels simultaneously;

⑬ **Output Link:** Performs linked operations on selected channels to configure multiple channels simultaneously.

4.3 Menu

4.3.1 File

① Add Device: Add different device models;

② Connect: Connect the selected device in the device list;

③ Disconnect: Disconnect the selected device in the device list;

④ Exit: Exit the software.

4.3.2 Edit

① Copy: Copy the selected channel configuration;

② Paste: Paste the copied channel configuration onto the selected channel.

4.3.3 Tools

① Settings: Access the settings interface;

② Presets: Access the preset interface;

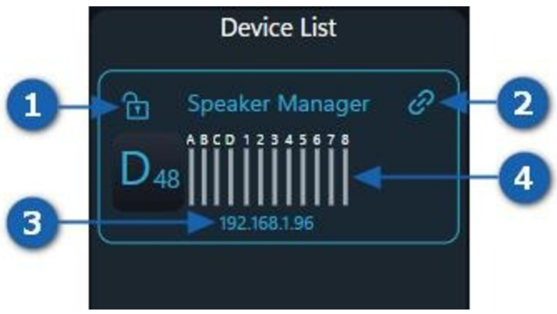
③ Security: Access the security interface;

④ Upgrade: Update device firmware.

4.3.4 Help

About: Software version information.

4.4 Device List



① Security Lock: :Login does not require a password; :Login requires a password;

② Connection Status: :Disconnected; :Disconnected. **Note:** When using a USB connection, :Disconnected; :Disconnected;

③ Device IP Address;

④ Channel Level: Displays channel level after connection.

4.5 Control Software Connection

4.5.1 Control Software Network Configuration

The default IP address of the Speaker Manager is set to obtain an IP address automatically. Connecting to the Speaker Manager via the network requires an IP address assigned by a DHCP server. Users can modify the Speaker Manager's IP address in the device settings interface on the front panel display. After modification, ensure the client host IP address is within the same subnet as the Speaker Manager to enable the client software to connect properly.

Note: The IP address of the Speaker Manager can also be modified within the control software after the client software has successfully logged in.

4.5.2 Connection Steps

After installing the client software, open the software. The device list will automatically display available devices. Select the corresponding device connection icon in the device list to connect.

After a successful connection, the disconnected icon for the corresponding device in the device list changes to a connected icon.

4.5.3 Disconnecting

Select the corresponding device in the device list. The "Disconnect" option is available in the "File" menu, or you can directly click the disconnect icon to disconnect the device.

4.6 Input Component Configuration

① **Source:** Input sources can be selected as Analog or Dante. The Analog Input provides line-level input for devices with line-level outputs, and inputs for instruments, mixing consoles, etc. The Analog Input converts the analog input signal to a processed digital. Connections are made using one XLR female connector;

② **Mute:** Mutes the input signal;

③ **Expander:** Expander component;

④ **Gain:** Controls the Gain of the channel;

⑤ **Polarity:** Inverts the polarity of the output signal. In audio processing, phase is a critical parameter that determines the starting point and direction of a signal waveform. Through phase inversion, the phase of an audio signal is reversed by 180 degrees. In multi-speaker systems, inconsistent signal phases between different speakers can cause sound cancellation or interference. The phase inversion function allows adjustment of the phase to ensure sound clarity and consistency;

⑥ **Delay:** Delay component;

⑦ **GEQ:** Graphic Equalizer component;

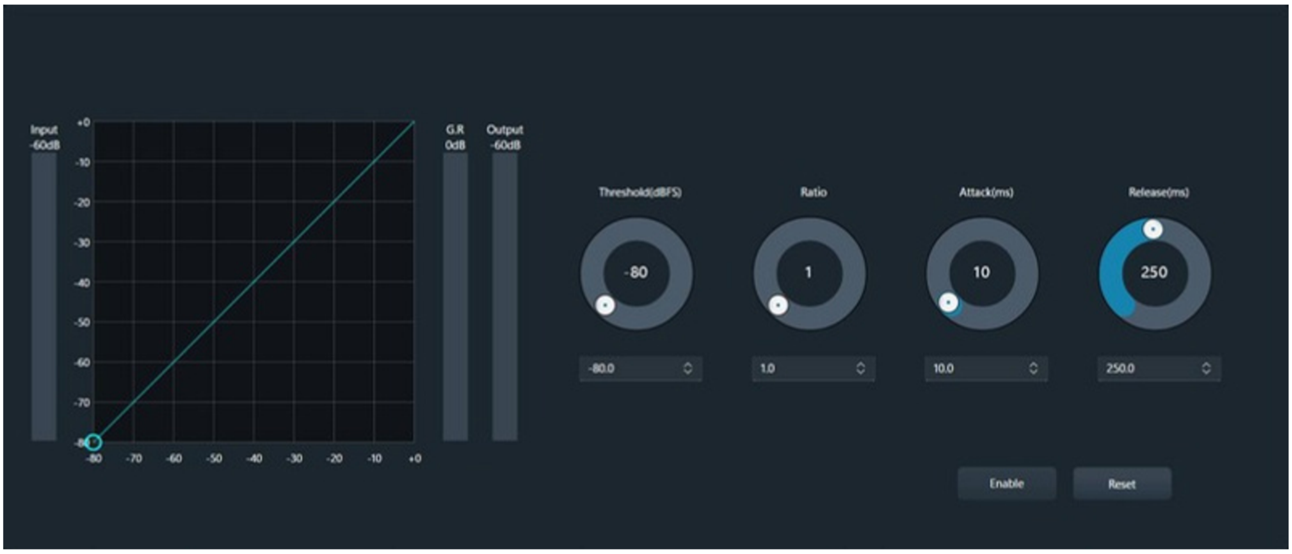
⑧ **PEQ:** Parametric Equalizer component;

⑨ **DEQ:** Dynamic Equalizer component;

⑩ **COMP:** Compressor component.

4.6.1 Input Expander

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



① **Threshold (-80～0):** Sets the point from which the attenuation is calculated based on the Ratio setting. This is where the Expander starts working. When the input signal falls below the Threshold Level, the Expander compresses the signal at a preset Ratio to increase its dynamic range. When the input signal exceeds the Threshold Level, the signal is output at a 1:1 Ratio, preserving its original dynamic range. The Threshold setting requires adjustment based on the specific application scenario;

Example:

- If the: Threshold Level is -30dB; Ratio is 2.5; Input level is -40dB
- Then the Adjusted Output is:
- [(Input Level - Threshold Level) * Ratio] + Threshold Level = Output Level
- {[-40dB - (-30dB)] * 2.5} + (-30dB) = -55dB.

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

③ **Attack Time (1～500):** The time required for an input signal less than the Expander Threshold Level to enter the expansion state and to output at the set expansion ratio. Shorter attack time enable the Expander to respond quickly to signal changes, making it suitable for processing rapidly changing audio signals; longer attack time provide smoother transitions, preventing overly abrupt processing effects, and are ideal for gentle signals like vocals or music;

④ **Release Time (1～500):** The time required for the input signal level to return from the extended state to the original non-extended state. Shorter release time enables rapid restoration of signal dynamics, making it suitable for fast-changing audio, but it may induce pumping effects (rapid fluctuations in signal level); longer release time provides smoother transitions and reduces pumping effects, but it may cause the signal restoration process to appear sluggish. Therefore, release time settings should be flexibly adjusted based on audio characteristics and processing requirements;

⑤ **Input Level:** Graphically displays the level of the input signal;

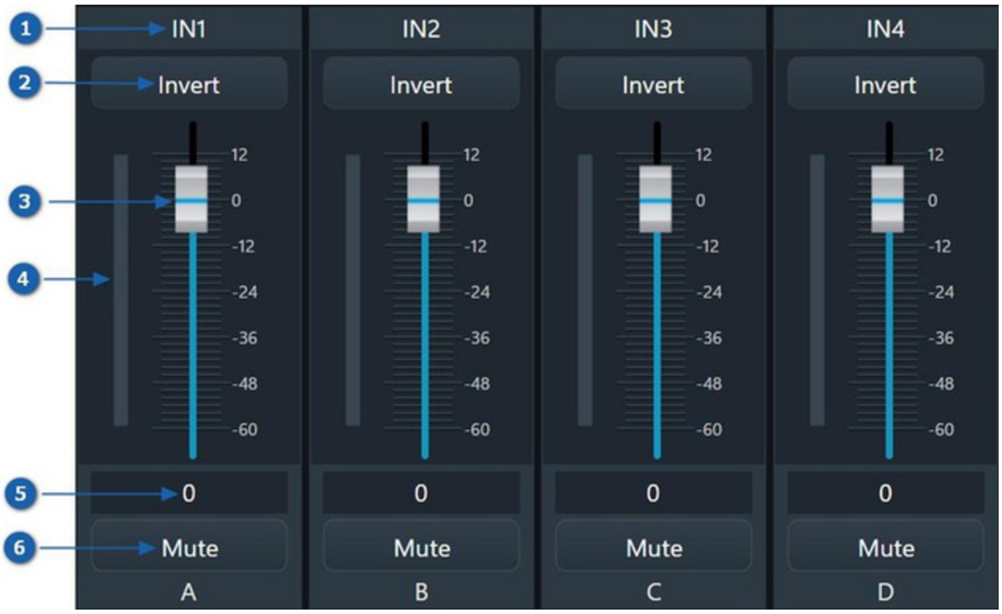
⑥ **Gain Reduction:** Graphically displays the amount of attenuation between the signal processed by the Expander and the input signal. The Gain Reduction reflects the degree of signal attenuation applied by the Expander;

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

⑧ **Enable/Disable:** Enable or Disable the Expander for the current channel. When the Expander is disabled, audio is passed through without any change;

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

4.6.2 Input Gain



① **Input Channel Name:** Customizable channel name;

② **Invert:** Inverts the polarity of the output signal. In audio processing, phase is a critical parameter that determines the starting point and direction of a signal waveform. Through phase inversion, the phase of an audio signal is reversed by 180 degrees. In multi-speaker systems, inconsistent signal phases between different speakers can cause sound cancellation or interference. The phase inversion function allows adjustment of the phase to ensure sound clarity and consistency;

③ **Fader:** Click to select the Gain Fader, the channel Gain Fader adjustment method is based on the step fine-tuning of the up and down arrow keys in steps of 1 dB;

④ **Input Level:** Graphically displays the level of the input signal;

⑤ **Gain:** Can manually enter the volume within the range of -72～12dB;

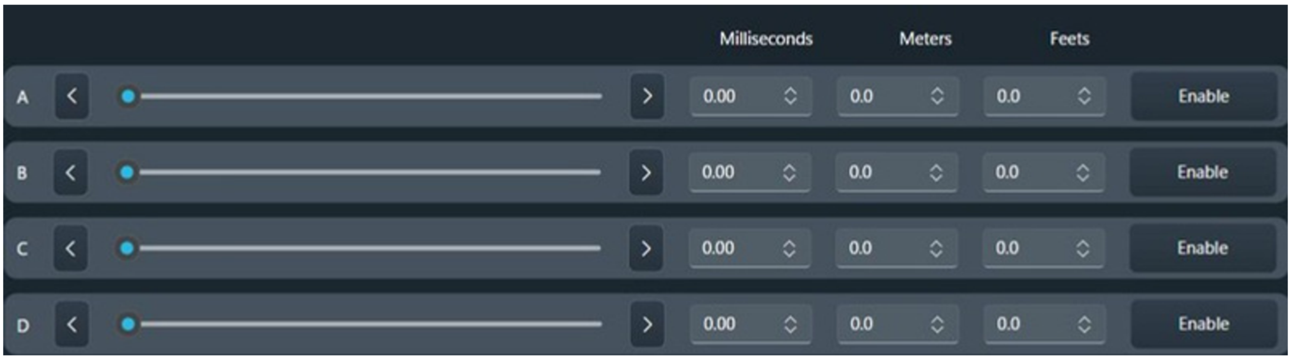
⑥ **Mute:** Mutes the current channel.

4.6.3 Input Delay

The Delay component is primarily used to apply time delays to audio signals, enabling various audio effects and optimizing audio system performance. By activating the delay during signal processing and imposing specific time delays on the audio signal, it alters the signal propagation time to achieve specialized sound processing. Users can configure the delay component with fixed delay durations according to their requirements. The delay component supports a delay time range of 0～1500 milliseconds, accommodating diverse application scenarios.

- **Reverb Effect:** By setting an appropriate delay time, the reverb effect simulates the reflection and diffusion of sound in a space, enhancing the sound's spatial and three-dimensional quality, as if you were immersed in a specific acoustic environment.

- **Echo Effect:** By using a delay unit to generate repeated sound signals, natural echoes are simulated, enhancing the layering and depth of the sound.
- **Sound Optimization:** In larger performance venues, delay units can be used to assist with speaker processing. By applying different delays to different speakers, sound is distributed evenly throughout the space, avoiding sound overlap and interference, and optimizing the overall sound field effect.



① **Delay time:** Delay time range (0～1500ms);

② **Delay distance:** Delay distance range (0～510m). This provides an alternative method for setting delay times in units of distance, ranging from 0 meters to 510 meters. Entering delay times as distances is often more convenient in practical situations. If needed, distances can also be entered in feet;

③ **Enable/Disable:** Enable or Disable the Delay for the current channel. When the Delay is disabled, audio is passed through without any change.

4.6.4 Input Graphic Equalizer

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

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

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

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

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

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

⑥ **Enable/Disable:** Enable or Disable the Graphic Equalizer for the current channel. When the Graphic Equalizer is disabled, audio is passed through without any change;

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

4.6.5 Input Parametric Equalizer

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

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

- **Parametric Equalization Filter:** Parametric Equalization Filter is an adjustable filter used to precisely boost or attenuate specific frequencies in an audio signal. It optimizes the frequency response of audio by adjusting the center frequency, gain, and bandwidth;
- **Low-Pass Filter:** Allows low-frequency signals to pass through and cut off high-frequency signals based on the set frequency, typically used to remove high-frequency noise or increase low-frequency components;
- **High-Pass Filter:** Allows high-frequency signals to pass through and cut off low-frequency signals based on the set frequency, typically used to remove low-frequency interference or extract high-frequency features;
- **All-Pass Filter:** Passes signals of all frequencies with identical gain but alters the signal's phase response. An all-pass filter does not modify the amplitude spectrum of the signal. Instead, it improves system transient response, corrects frequency response defects caused by phase issues, or creates intriguing sonic effects by precisely adjusting phase and delay relationships.
- **Band-Pass Filter:** Allows only signals with in a specific frequency range (passband) to pass through, while attenuating or blocking all frequency signals outside this range (low and high frequencies);
- **Low-Shelf:** The Low shelf is a gain increased or attenuated for the frequency portion below the set center frequency, typically used to increase the richness of low frequencies or reduce low frequency rumbling;
- **High-Shelf:** The High shelf is a gain increased or attenuated for the frequency portion above the set center frequency, typically used to increase the clarity of high frequencies or reduce the harshness of high frequencies.

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

③ **Gain:** Controls the Gain for an individual frequency band. Users can adjust the gain to increase or decrease the signal strength of specific frequencies. For example, if the sound in a certain frequency range of an audio signal is too loud or too soft, the volume can be balanced by increasing or decreasing the gain. The adjustment range of the gain is usually from negative values (attenuation) to positive values (increase). This setting is disabled when selecting All-Pass, Band-Pass, Low-Pass, High-Pass filter;

④ **Q-Factor:** The Q-factor refers to the range of frequencies affected around the center frequency. Set the Q-factor for a single band in the equalizer, ranging from 0.4 octaves to 128 octaves (default value is 3.00). This setting is disabled when selecting Low-Shelf, High-Shelf, Low-Pass, or High-Pass, unless marked as Vari-Q (Variable Q-factor). The Q-factor determines the precision and scope of equalization adjustments. Higher Q-factor yield narrower bandwidth and more precise frequency range control; lower Q-factor provide wider bandwidth and broader frequency influence;

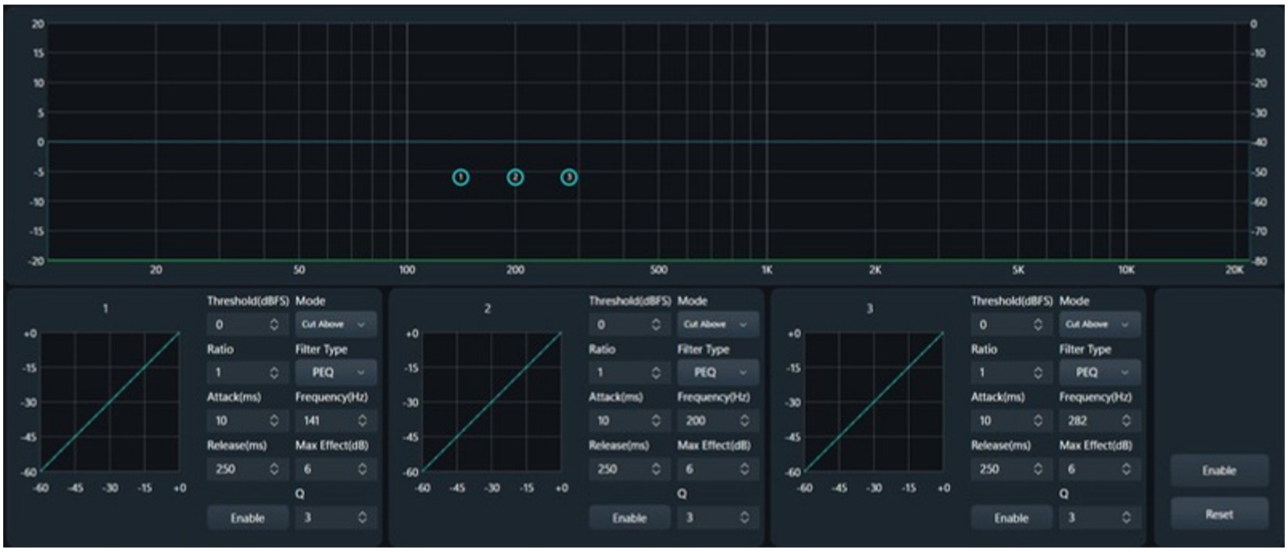
⑤ **Enable/Disable:** Enable or Disable the Parametric Equalizer for an individual frequency band. When an individual frequency band is disabled, audio is passed through without any change;

⑥ **Enable All/Disable All:** Enable or Disable All the Parametric Equalizer for the current channel. When the Parametric Equalizer is disabled, audio is passed through without any change;

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

4.6.6 Input Dynamic Equalizer

The Dynamic Equalizer component is an intelligent audio signal processing tool that combines the precise frequency control capabilities of a multi-band equalizer with the dynamic processing characteristics of a compressor. Unlike traditional static equalizers that apply fixed gain changes at all times, dynamic equalizers automatically adjust the gain of specific frequency bands based on the dynamic changes in the audio signal. This means it can boost or attenuate the input signal level when the signal level exceeds a set threshold and the frequency matches the preset frequency, while remaining static during normal signal conditions. This prevents excessive processing of the audio.



① **Threshold (-54～0):** Sets the level where Dynamic Equalizer begins to work. This is the point from which the amount of dynamic gain is calculated based on the Ratio setting;

② **Mode:** Controls the direction of dynamic response, with four modes available for selection:

- **Cut Above:** Attenuates when the signal exceeds the Threshold Level, similar to dynamic clipping or compression, often used to control rumble or sibilance;
- **Boost Above:** Boosts when the signal exceeds the Threshold Level, used to increase dynamic peaks in the signal, adding impact or brightness to a specific frequency band;
- **Cut Below:** Begins attenuating when the signal falls below the Threshold Level, used to remove background noise, tail sounds, and other low-level content;
- **Boost Below:** Begins boosting when the signal falls below the Threshold Level, used to increase the presence of weak signal segments.

③ **Ratio (1 ～ 4):** The ratio between the Input and the Output as measured from the Threshold Level. Controls the response rate of dynamic gain changes or the perceived compression intensity, influencing the "steepness" of dynamic processing;

④ **Filter Type:** Including Parametric Equalization, Low-Shelf, High-Shelf Filter;

- **Parametric Equalization Filter:** Parametric Equalization Filter is an adjustable filter used to precisely boost or attenuate specific frequencies in an audio signal. It optimizes the frequency response of audio by adjusting the center frequency, gain, and bandwidth;
- **Low-Shelf:** The Low shelf is again increased or attenuated for the frequency portion below the set center frequency, typically used to increase the richness of low frequencies or reduce low frequency rumbling;
- **High-Shelf:** The High shelf is again increased or attenuated for the frequency portion above the set center frequency, typically used to increase the clarity of high frequencies or reduce the harshness of high frequencies.

⑤ **Frequency:** Sets the center Frequency of an individual band. Frequency is one of the core parameters in audio processing. In audio signals, different frequencies correspond to different sound characteristics. For example, low frequencies are typically associated with heavy drum sounds or bass, mid-frequencies involve vocals and the timbre of most instruments, and high frequencies are related to bright timbres or details. By selecting the appropriate center frequency, users can precisely enhance or attenuate the timbre of specific instruments, optimize the clarity of vocals, or resolve frequency issues in audio;

⑧ **Max Effect:** Controls the Gain or Attenuation for the center frequency, constituting the maximum dynamic gain variation achievable ("+" or "-", depending on the mode). **Note:** This is the "limit" for dynamic variation; the actual effect depends on the signal level;

⑨ **Q-Factor:** The Q-factor refers to the range of frequencies affected around the center frequency. Set the Q-factor for a single band in the equalizer, ranging from 0.4 octaves to 128 octaves (default value is 3.00). The Q-factor determines the precision and scope of equalization adjustments. Higher Q-factor yield narrower bandwidth and more precise frequency range control; lower Q-factor provide wider bandwidth and broader frequency influence;

⑩ **Attack Time (1 ～ 1000):** Attack time is how fast the dynamic gain reacts to a signal outputting at the set Ratio when the input signal meets the Threshold Level condition;

⑪ **Release Time (1～1000):** Release time is how fast the dynamic gain reacts when the signal no longer meets the Threshold Level condition and gain is restored to its non-limited level;

⑫ **Enable/Disable:** Enable or Disable the Dynamic Equalizer for an individual frequency band; When an individual frequency band is disabled, audio is passed through without any change;

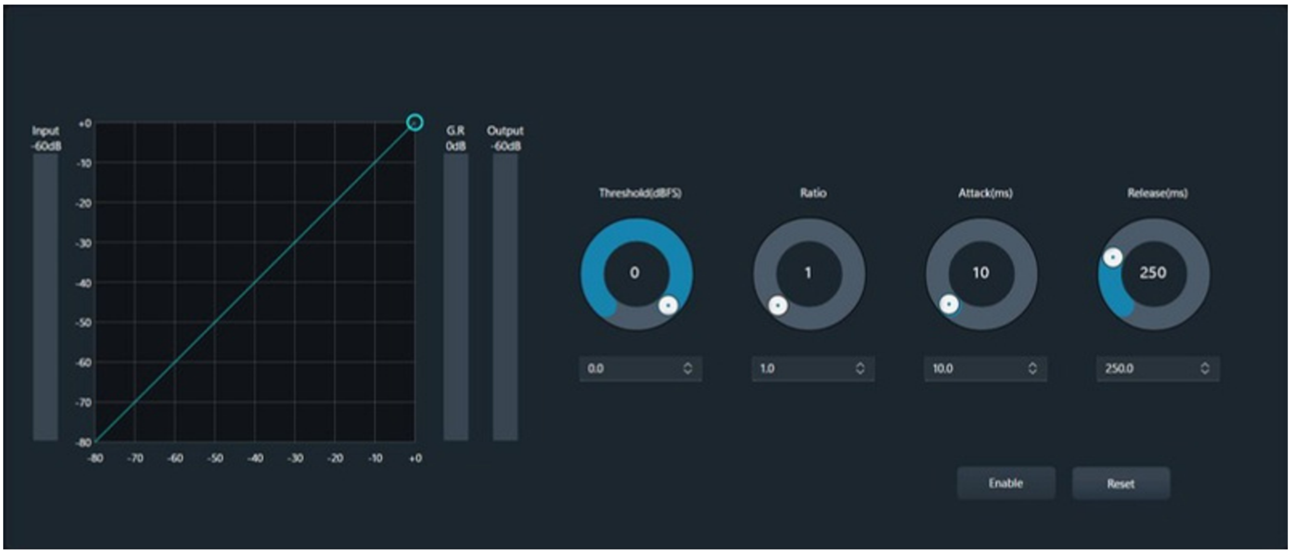
⑬ **Enable All/Disable All:** Enable or Disable All the Dynamic Equalizer for the current channel; When the Dynamic Equalizer is disabled, audio is passed through without any change;

⑭ **Reset:** All band filter parameters are restored to the default.

4.6.7 Input Compressor

The purpose of the Compressor component is to control the dynamic range of the Output above a set Threshold Level, thereby optimizing audio balance and consistency. Compressor are widely used in music production, live sound reinforcement, broadcasting, and voice processing, helping users control audio signal peaks, prevent distortion, and enhance overall audio clarity and audibility by raising the average signal level

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



① **Threshold (-80～0):** Sets the level where compression begins. This is the point from which the amount of attenuation is calculated based on the Ratio setting. A level below the Threshold Level is not compressed, anything above the Threshold Level attenuation is applied;

Example:

- If the: Threshold Level is -30dB; Ratio is 2.5; Input level is -10dB
- Then the Adjusted Output is:
- [(Input Level - Threshold Level) / Ratio] + Threshold Level = Output Level
- {[-10dB - (-30dB)] / 2.5} + (-30dB) = -22dB.

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

③ **Attack Time (1～1000):** Attack time is how fast the compressor reacts to a signal crossing the set threshold going up. Short attack time compressors can quickly capture signal peaks, making them suitable for percussion instruments, but if the attack time is too short, it can produce "breathing sounds" and lose naturalness; Long attack times provide smooth transitions, making them suitable for vocals and other gentle signals, preserving more dynamics and details;

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

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

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

⑦ **Gain Reduction:** Graphically displays the amount of attenuation between the signal processed by the Compressor and the input signal. The Gain Reduction reflects the degree of signal attenuation applied by the Compressor;

⑧ **Output Level:** Graphically displays the level of the output signal;

⑨ **Enable/Disable:** Enable or Disable the Compressor for the current channel. When the Compressor is disabled, audio is passed through without any change;

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

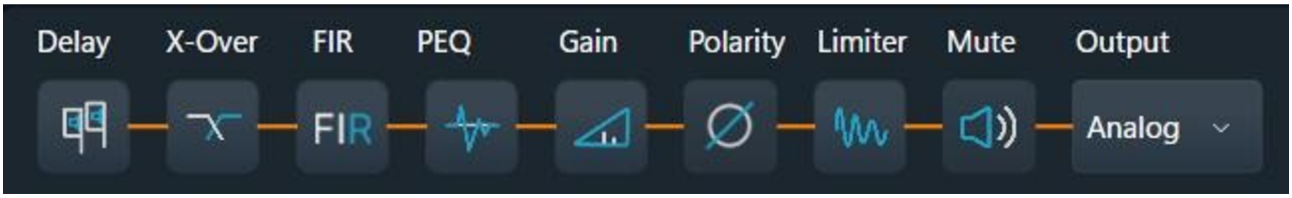
4.7 Matrix Mixer

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

① **Mix Output Gain (-24～30):** Adjust the mix output gain using the fader;

② **Enable:** Control whether each channel matrix mix is enabled or disabled.

4.8 Output Component Configuration



⑦ **Delay:** Delay component;

⑧ **X-Over:** X-Over component;

⑨ **FIR:** FIR Filter;

⑩ **PEQ:** Parametric Equalizer component;

⑪ **Gain:** Controls the Gain of the channel;

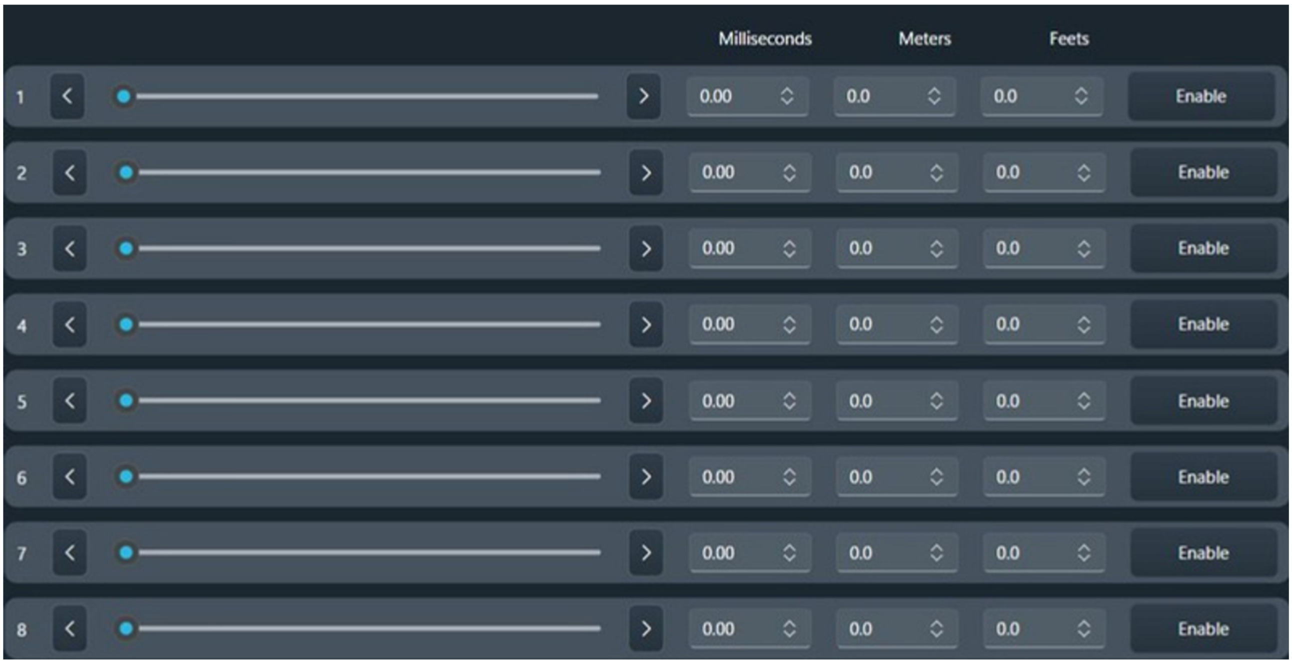
⑫ **Polarity:** Inverts the polarity of the output signal. In audio processing, phase is a critical parameter that determines the starting point and direction of a signal waveform. Through phase inversion, the phase of an audio signal is reversed by 180 degrees. In multi-speaker systems, inconsistent signal phases between different speakers can cause sound cancellation or interference. The phase inversion function allows adjustment of the phase to ensure sound clarity and consistency;

⑬ **Limiter:** Limiter component;

⑭ **Mute:** Mutes the output signal;

⑮ **Output:** Analog Output provides one channel of line-level output for device. The Analog Output converts the processed digital signal to analog. Connections are made using one XLR male connector.

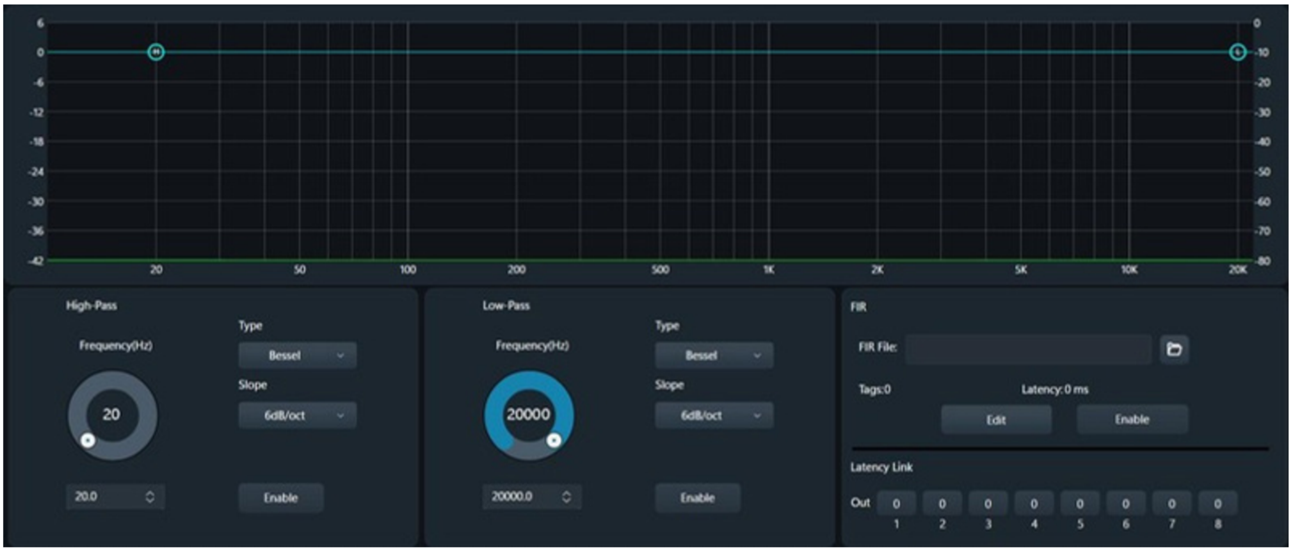
4.8.1 Output Delay



Reference Input Component Configuration - Input Delay.

4.8.2 Output X-Over

The Crossover component divides the audio input signals into 3 frequency bands: Low-Pass, Band-Pass and High-Pass, you can set the Slope rates and filter types (Butterworth, Linkwitz-Riley, Bessel) for each filter in each band. Crossover play an indispensable role in fields such as audio processing, sound system design, and professional audio production.



① **Filter Type:** The Type setting includes: Bessel, Butterworth, Linkwitz-Riley filters. You can select any combination of two of these filters for a band's high-pass and low-pass frequencies;

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

② **Slope:** Determines the rate of change of attenuation at the high-pass and low-pass frequencies of the band. The slope setting establishes the crossover region between two adjacent bands. The Slope includes 6dB/Oct, 12dB/Oct, 18dB/Oct, 24dB/Oct, 32dB/Oct, 36dB/Oct, 42dB/Oct, 48dB/Oct;

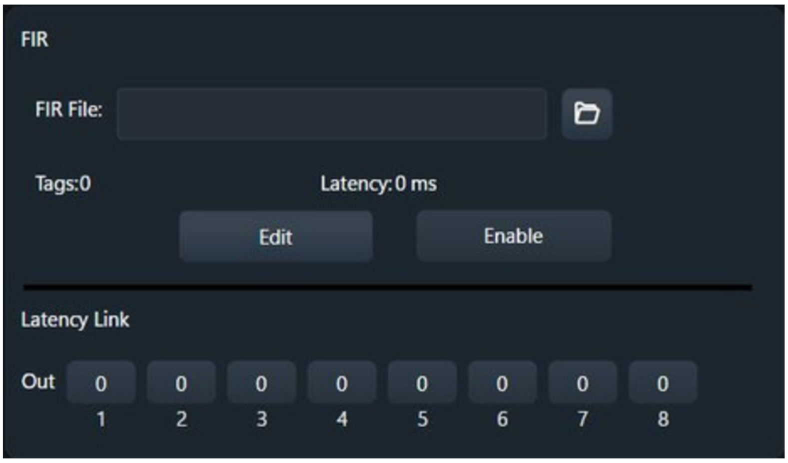
- **Low slope (Such as 6dB/Oct, 12dB/Oct):** The transition is relatively smooth, suitable for scenarios requiring a gentle transition, but the crossover effect is not clean enough and may result in frequency band overlap;
- **High slope (Such as 24dB/Oct, 48dB/Oct):** The transition is steep, resulting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

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

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

4.8.3 FIR filter

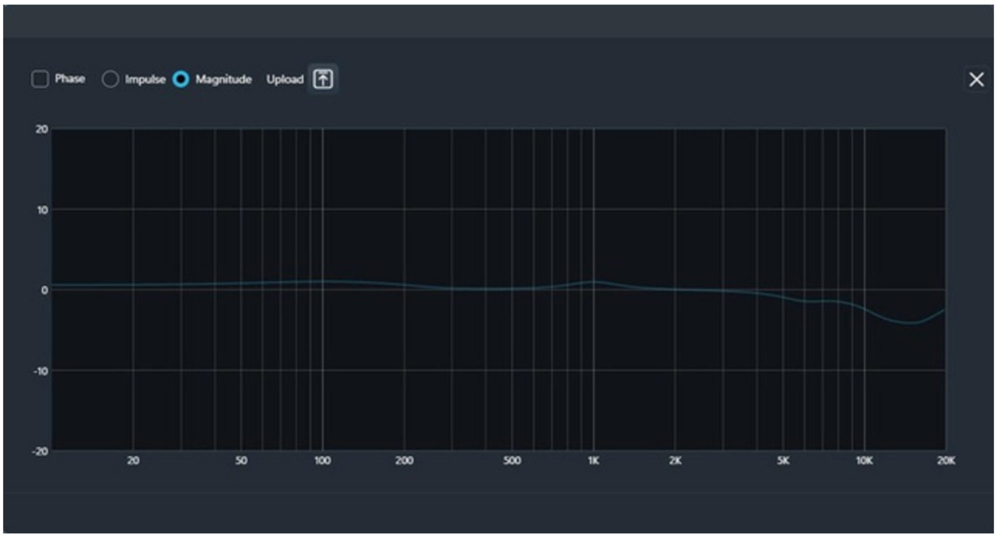
FIR filter, namely finite impulse response filter, extract desired frequency components from complex signals. Their core capability lies in their precisely designed frequency response and phase linearity.



① **Import:** Import FIR filter files with a maximum 896 tags;

② **Enable:** the FIR filter for the current channel. When the FIR filter is bypassed, audio is passed through without any change;

③ **Edit:** Access the edit interface to upload imported FIR filter files to the device. View the Phase, Impulse, and Magnitude of the FIR file's frequency response curve;



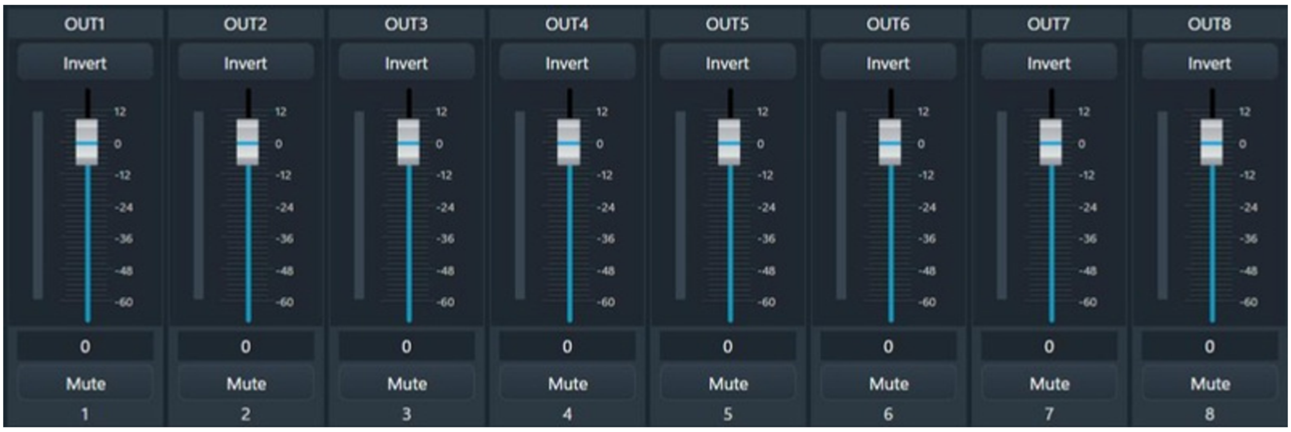
④ **Latency:** Displays the latency calculated from the FIR filter file;

⑤ **Latency Link:** Aligns the output channel's latency.

4.8.4 Output Parametric Equalizer

Reference Input Component Configuration - Input Parametric Equalizer.

4.8.5 Output Gain



Reference Input Component Configuration - Input Gain.

4.8.6 Output Limiter

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

Each output channel is equipped with two-stage Limiters. One stage is an RMS Limiter that responds to the RMS level of the signal, while the other stage is a Peak Limiter specifically designed to handle signal peaks.

- **RMS Limiter** continuously monitors the input signal's RMS, which more closely approximates the loudness perceived by the human ear. It activates only when the signal's average energy exceeds a preset threshold, reducing gain accordingly;
- **Peak Limiter** continuously monitors the instantaneous peaks of the input signal. Should the amplitude surpass the set threshold, it operates according to a defined attack time to "clip" the signal's highest level.

① **Threshold(-80～0):** Sets the level at which the Limiter has an effect, and the level at which the output is held;

② **Attack Time(1~1000):** Attack time is how fast the Limiter reacts to a signal crossing the set threshold going up. Short attack time compressors can quickly capture signal peaks, making them suitable for percussion instruments, but if the attack time is too short, it can produce "breathing sounds" and lose naturalness; Long attack times provide smooth transitions, making them suitable for vocals and other gentle signals, preserving more dynamics and details;

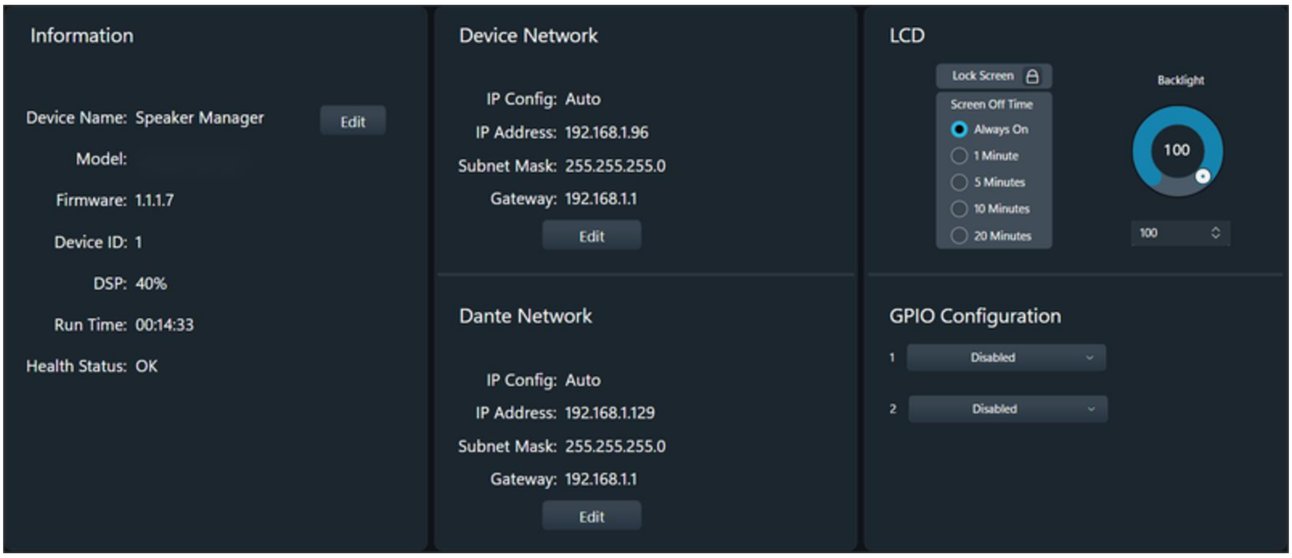
③ **Release Time (1～500):** The release time determines how long it takes for the input signal to return to its original dynamics from maximum attenuation. It dictates how long the limiter continues to attenuate the signal after it falls below the Threshold Level. Shorter release time enables rapid signal dynamics release, suitable for fast-changing audio but prone to Pumping effect (rapid fluctuations in signal level). Longer release time provides smoother transitions and reduces Pumping effect, though it may make the signal release process appear sluggish;

④ **Input Level:** Graphically displays the level of the input signal;

⑤ **Gain Reduction:** Graphically displays the amount of attenuation applied to the Channel, Gain Reduction reflects the degree to which the limiter attenuates the signal. For example, if the input signal exceeds the threshold by 3dB, the limiter may attenuate the signal by 3dB, resulting in a compression of 3dB;

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

4.9 Settings



④ **Device Name:** Customizable name;

⑤ **Device Network:** Configurable device network;

⑥ **Dante Network:** Configurable Dante network for the device;

⑦ **LCD:** Adjustable LCD backlight brightness and screen off time for the device display;

⑧ **GPIO Configuration:** Provides two GPIO input interfaces with 4 selectable modes:

5) Disabled: Disables GPIO functionality;

6) Fire Alarm Mute Contact Closed: Mutes all channels upon trigger;

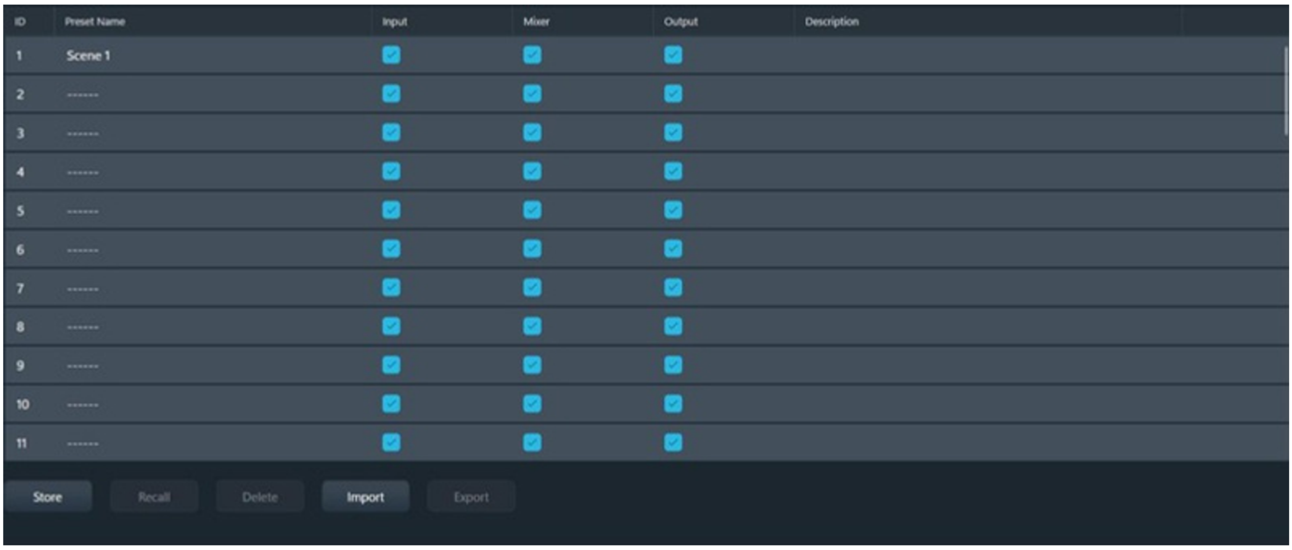
7) Fire Alarm Mute Contact Open: Unmutes all channels upon trigger;

8) Preset Recall Contact Closed: Recalls specified preset.

4.10 Presets

The scene presets provides users with a convenient scene management platform that supports scene Store, Recall, Delete, Import and Export. Through this component, users can flexibly manage various scene files to meet the needs of different scenarios while ensuring the accuracy and traceability of scene parameters.

The Speaker Manager provides 50 scene presets archive locations.



④ **Store:** Stores the current scene configuration to the specified archive location. You can customize the name, add a description, and choose to save "Input", "Mixer" and "Output";

⑤ **Recall:** Recall a scene preset from the specified archive location;

⑥ **Delete:** Delete a scene preset configuration from the specified archive location;

⑦ **Import:** Import a scene preset file into the specified archive location;

⑧ **Export:** Export a scene preset from the specified archive location to a local file.

4.11 Security

The security lock enable, lock, and permission management for various processing components along input and output signal paths. This interface prevents accidental operation by non-professionals, ensuring long-term stable operation of the equipment within a fixed audio processing solution.

① **Password Required:** Enable to require password entry during connection;

② **Change Password:** Modify the login password. The default password is blank. Passwords must consist solely of numeric characters and be ≤4 characters in length;

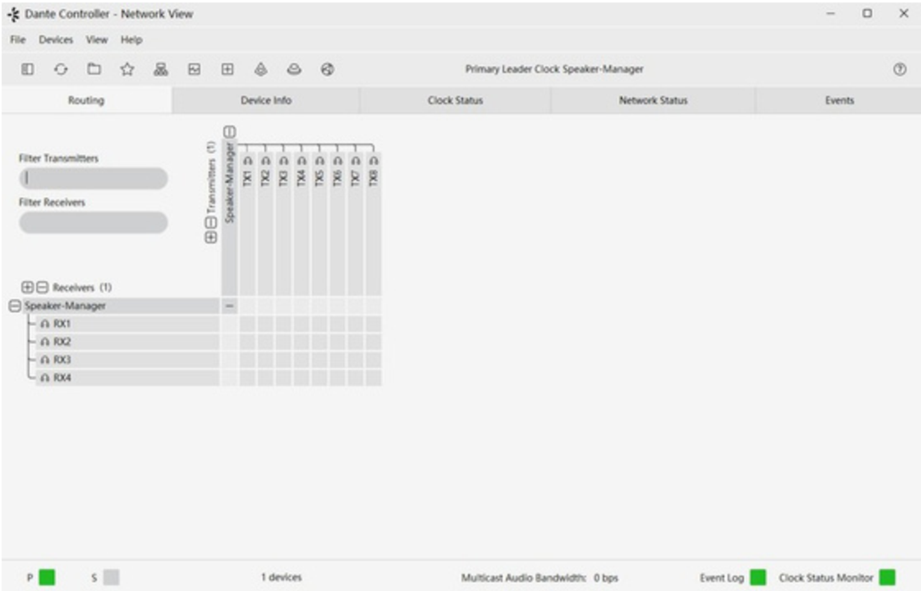
③ **Component Lock:** Enable the corresponding lock to lock the component;

④ **Save:** Click Save after modifying lock status to synchronize changes to the device.

Chapter 5 Dante Network Audio Routing

Note: Only the Dante edition supports Dante network transmission protocol.

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



The software "Dante Controller" is free to download from the company of Audinate (the owner of Dantetechnology).Toinstallthesoftwareonthecomputer,pleasevisitthelink: https://www.getdante.com/products/software-essentials/dante-controller.

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

Note:

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

Note: Only the Dante edition supports Dante network transmission protocol.

Chapter 6 FAQ

1. Channel no sound

First, check whether the audio source, audio input and output wiring is normal; Second, check whether the mute function of the corresponding audio channel is enabled, if the mute switch has been turned on, please turn off the mute switch; Third, check whether the settings of the corresponding channel's input processing, matrix mixing and output processing are normal; Fourth, check that the audio signals are routed correctly in the Dante Controller. If the problem is not solved, please contact the distributor or manufacturer.

2. The control software cannot search the device

First, check whether the device display enters the system normally; second, check whether the network connection is normal; third, ensure the network accessibility between the configuration host and the device; fourth, reset the device to factory settings and restart it. If the problem is not solved, please contact the distributor or manufacturer.

3. Network connection failure

Network connection failure is usually caused by different network segments of the device. If the LAN and the processor network segments are different, you can connect the device directly via PC, login to the device configuration interface, change the processor network segment to be the same as the LAN and then access the LAN. (**Note:** If the LAN is automatically obtaining IP, please check DHCP).

4. Current noise in output channel

Please check whether the device is well grounded, which usually requires the grounding screw on the left side of the rear panel of the chassis to be connected to the metal enclosure such as the cabinet through a metal wire. If the problem still exists, please check the wiring of the input devices. If the input devices are unbalanced (two wires), please connect the "+" and "G" of the input connector of the device.

5. How to recognize system noise

After the system is set up, there is noise troubleshooting: First, unplug the device output audio cable, there is noise, please check the causes of the back stage equipment; Second, restore the output wiring, mute the corresponding output channel, there is noise, if unbalanced connection, try to shorten the connecting line, to avoid the introduction of interference, if balanced connection, try to disconnect the ground wire; Third, cancel the corresponding channel mute, unplug the device input audio cable, there is noise, reset the device to factory settings and restart the device;Fourth, restore the input wiring, turn off the audio source, there is noise, check the input connection, refer to the second point of the processing; Fifth, check the audio source is there is noise. If the problem is not solved, please contact the distributor or manufacturer.

6. RS485 center control command does not work

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

Chapter 7 Packing List

Device	Control software USB flash drive	Power Cable	Quick Guide	Certificate of Conformity
1PCS	1PCS	1PCS	1PCS	1PCS

Chapter 8 Specification

Category	**Parameter Item**	**Parameter Description**
Peripherals	Input Interfaces	4 Analog + 4 Dante
	Output Interfaces	4 Analog + 8 Dante
	Display	Embedded GUI interface, displays full function operation, and quick

		adjustment of parameters
	Control Interfaces	1 RJ45 interface, 1 USB-B interface, 1 RS485 interface, 2 GPIO control interface
Audio processing	Processor	ADI SHARC ADSP-21489 450 MHz high performance 40-bit floating-point DSP processor; 32-bit A/D and D/A converter, 96kHz sampling rate
	Input Channel	Functional components: Mute, Expander, Gain, Delay (0-1500ms), 31-band Graphic Equalizer, 12-band Parametric Equalizer, 3-band Dynamic Equalizer, Compressor.Physical interface: Balanced XLR female connectors.
	Output Channel	Functional components: Delay (0-1500ms), Crossover (Butterworth, Bessel, Linkwitz-Riley filter types, 896 taps FIR filter), 12-band Parametric Equalizer, Gain, RMS Limiter, Peak Limiter.Physical interface: Balanced XLR male connectors.
	Input Impedance	Balanced: 20KΩ
	Output Impedance	Balanced: 100Ω
	Common Mode Rejection Ratio	>60dB@50Hz
	Input to Output Dynamic Range	≥110dB
	Frequency Response	20Hz～20KHz, ±0.2dB
	Noise Floor	-92dBu
	Signal to Noise Ratio	110dB
	THD+N	≤0.0015% @1kHz, +4dBu

	Channel Isolation	105dB@1kHz
	Input Level Range	≤+22dBu (A-Weighting)
	Crossover	Three types of high and low pass filters: Butterworth, Bessel and Linkwitz-Riley
	Equalizer	Parametric Equalizer: Frequency: 20～20kHz, Gain: -20～+15dB, Bandwidth: 0.4～128 Graphic Equalizer: Frequency: 20～20kHz, Gain: -15～+15dB
	Maximum Output Level	22dBu
	Maximum Input Level	22dBu
	Analog/Digital Dynamic Range	123dB
	Digital/Analog Dynamic Range	123dB
General specification	Operating Voltage	AC 100V～240V, 50Hz/60Hz
	Maximum Power	30W
	Operating Temperature and Humidity	0℃～40℃, 10%～90%RH, No condensation
	Chassis	1U
	Product Dimensions (L×W×H)	482.4mm×210.5mm×44mm
	Net weight	2.8kg
	Package Dimensions (L×W×H)	590mm×430mm×110mm
	Package Weight	3.3kg

Warranty Regulations

The warranty period of this product is 1 year.

In the warranty period of non-man-made damage caused by the product performance failure can enjoy three packages of service.

Warranty card by the sales unit stamped after the effective. The alteration is invalid!

The following conditions (including, but not limited to, this) are not covered by the three-package service:

1. No warranty card or missing valid invoice or the date has exceeded the validity period of the three packages of services;
2. Not in accordance with the requirements of the product instructions for use, maintenance, management and damage caused;
3. The product model or code on the warranty voucher does not match the physical goods;
4. Damage caused by the dismantling and repair of non-authorised service providers;
5. Normal discolouration, wear and tear and consumption during the use of the product are not covered by the warranty;
6. The product cannot be used due to the user's own network reasons, please consult customer service staff.