CNS6692 Professional Audio Processor

User Manual

Thank you for choosing our products. To ensure optimal use of this device, please read the manual carefully before operation.

About this Manual

This manual includes device description, operating instructions and other related information of the device. Please read this manual carefully before use.

This manual is provided for reference only. Any updates or revisions will be made without prior notice.

This manual is intended solely as guidance for user operation and is not to be used for maintenance or repair purposes.

V2.2

2025-10-20

Caution

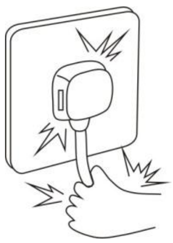
- When the power switch of this device is in the "OFF" state, the device is not completely disconnected from the mains. For safety, please unplug the power cord from the outlet when the device is not in use.
- This device can not be exposed to water droplets or splashes, and objects filled with liquids, such as vases, can not be placed on the device.
- Do not open the cover of the device to avoid the risk of electric shock. If necessary, have it repaired by qualified personnel.
- Terminals marked with the ϟ symbol indicate hazardous live parts. Connections to these terminals must be performed by instructed personnel only.
- The device is connected to the mains through the power cord plug. In case of malfunction or danger, disconnecting the plug from the outlet will cut off the device from the mains. Therefore, ensure that the power outlet is positioned so the plug can be easily accessed and removed.

Warning

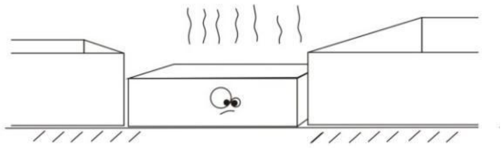
To prevent personal injury, equipment damage, and property loss, please follow the basic precautions below.

This symbol indicates what is "prohibited". This symbol indicates what is "required".

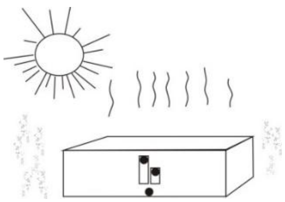
■ Check whether the power cord is damaged. Do not pull the power cord but hold the plug directly to remove the plug, otherwise it may cause electric shock, short circuit or fire.



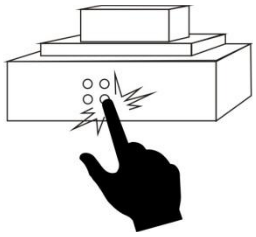
■ Do not block the vents during operation. All vents should remain unobstructed to prevent overheating.



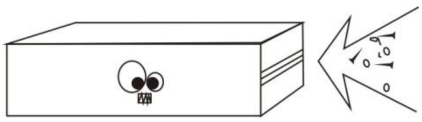
■ Do not operate or store the device in environments with excessive dust, vibration, extreme cold or heat.



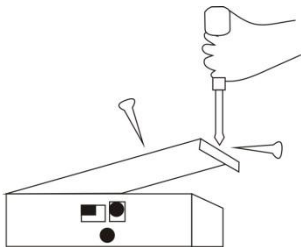
■ Do not place something heavy on the device. Do not use excessive force when operating switches or buttons or connecting external audio sources.



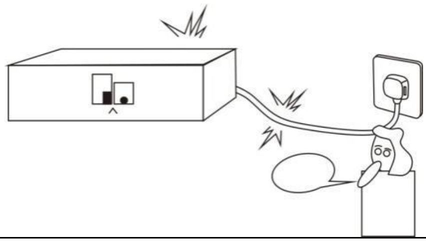
■ Do not allow any external objects (such as paper and metal) to enter any gaps or openings of the device. If this happens, disconnect the device from the mains immediately.



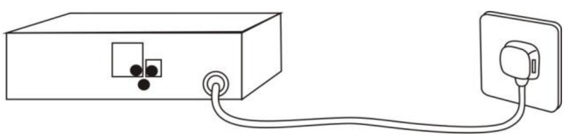
■ Do not attempt to disassemble or modify the internal parts of the device in any way.



■ If the device suddenly stops working, or emits an unusual odor, or produces smoke during operation, unplug the power cord immediately to prevent electric shock, fire or other hazards, and have the device serviced by qualified personnel.



■ If the device is not in use for a long time, unplug the AC power cord or turn off the wall outlet for zero power consumption.



Contents

1. Brief Introduction

This series of devices supports 12×12 analog channels with a high-quality 21-level preamplifier circuit and a DSP processing bus structure to deliver high-fidelity sound reproduction. It is widely applicable to large venues such as theaters, concert halls, remote video conferencing rooms, stadiums, churches, conference centers, and theme parks, meeting diverse sound reinforcement needs. The device is designed for simplicity in operation and features intelligent control functions.

1.1 Applications

- Lecture halls
- Small, medium and large conference rooms
- Courtrooms
- Auditoriums
- Multifunctional halls, and more

1.2 Features

- 24-bit/48kHz sampling frequency, high-performance AD/DA converters, and a 40-bit floating-point DSP processor.
- High-precision input sensitivity adjustment with 21 levels, 3dB steps, and a maximum input gain of 60dB.
- Full-featured matrix mixing function, offering exceptional and clear sound, including EQ, GATE, COMP, LIMIT, AGC, and more.
- GPIO supports 8-channel scene switching for quick operation, with RS232 and RS485 central control commands.
- 60 scene presets with a user-friendly interface, supporting fast switching between Chinese and English.
- TCP/IP control protocol for PC connection, enabling web-based quick control and parameter adjustment.

2. Technical Specifications

Input/Output	12×12
Input/Output Type	Analog
Signal Processor	40-bit floating-point DSP 400MHz
Bit Depth	24-bit
Sampling Frequency	48kHz
Input Impedance	＜10KΩ
Output Impedance	200Ω
A/D Dynamic Range	118dB
D/A Dynamic Range	118dB
Max Input Level	+18dBu
Max Output Level	+18dBu
Transmission Delay	4.1ms
Output Channel Delay	0ms～1500ms (Step:0.02ms)
THD+N	≤0.005% (+4dBu 1kHz)
Frequency Response	20Hz～20kHz ±0.3dB
CMRR	≥55dB
Crosstalk	≤-100dB
SNR	≥111dB (20–20KHz)
Dynamic Range	≥111dB (20–20KHz)
Noise Floor	≤-93dBu (20–20KHz)
Connection Type	TCP/IP, RS232, RS485
Preset	60
Size	482×253×44mm
Net/Gross Weight	3.3kg / 3.8kg

3. Functional Structure

3.1 Front Panel

3.2 Rear Panel

4. Quick Guide

The web controller page for the professional audio processor is a tool designed for users to quickly interact with and adjust various parameters of one or multiple units. It allows configuration settings to be saved in the memory of the processor. This product offers high efficiency with a clear and well-structured interface. The UI of this product adopts a self-developed web page control library, featuring a rational design that greatly enhances the user experience.

4.1 Operating Environment

The professional audio processor web controller is compatible with PC computers.

4.2 Software Installation

This operation page requires Google Chrome or Edge browser for further use. If not installed, please download and install from the official Google website or Microsoft website.

4.3 Audio Input and Output Connections

Note: The device supports both balanced and unbalanced inputs and outputs. Below are the connection methods for balanced and unbalanced signals.

Input

When the input source is balanced, the positive (+), negative (-), and ground (⏚) outputs of the source correspond to the positive (+), negative (-), and ground (⏚) inputs of the device, as shown in Figure 1.

When the input source is unbalanced, the positive (+) and ground (⏚) outputs of the source correspond to the positive (+) and ground (⏚) inputs of the device, as shown in Figure 2.

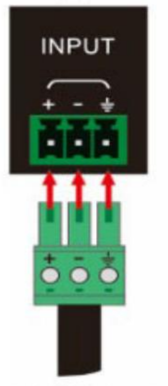


Figure 1 Balanced Input Device

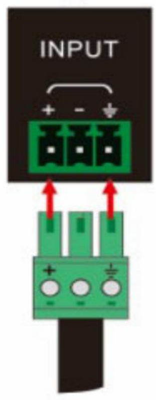


Figure 2 Unbalanced Input Device

Output

When the downstream device has a balanced input, the positive (+), negative (-), and ground (⏚) inputs of the downstream device correspond to the positive (+), negative (-), and ground (⏚) outputs of the device, as shown in Figure 3.

When the downstream device has an unbalanced input, the positive (+) and ground (⏚) inputs of the downstream device correspond to the positive (+) and ground (⏚) outputs of the device, as shown in Figure 4.

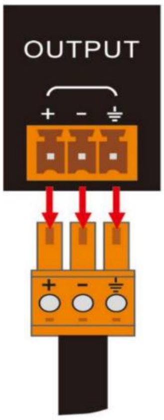


Figure 3 Balanced Receiving Device

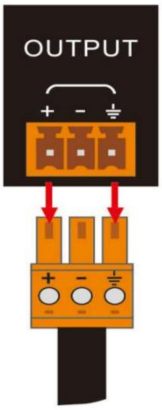
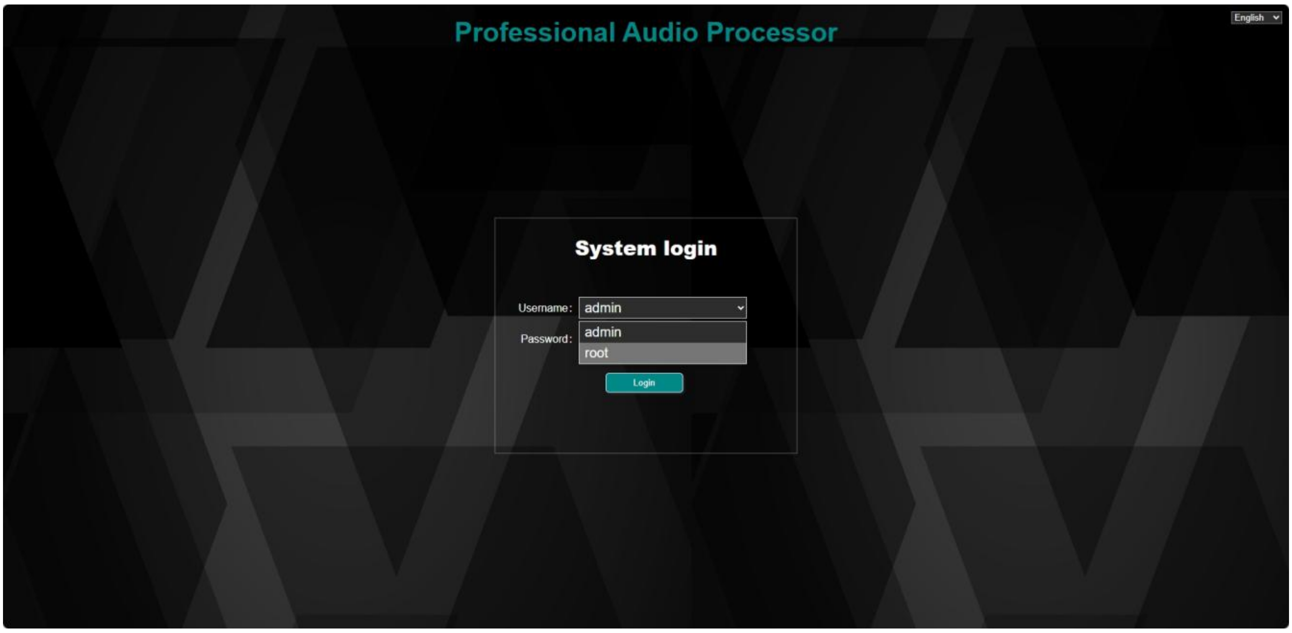


Figure 4 Unbalanced Receiving Device

4.4 Operation Interfaces

Operating procedures: Connect the computer to the professional audio processor with a network cable, and power on the professional audio processor. Open Chrome or Edge browser and enter the URL: 192.168.1.30 (factory default).



Username: admin, Password: 123123 (factory default)

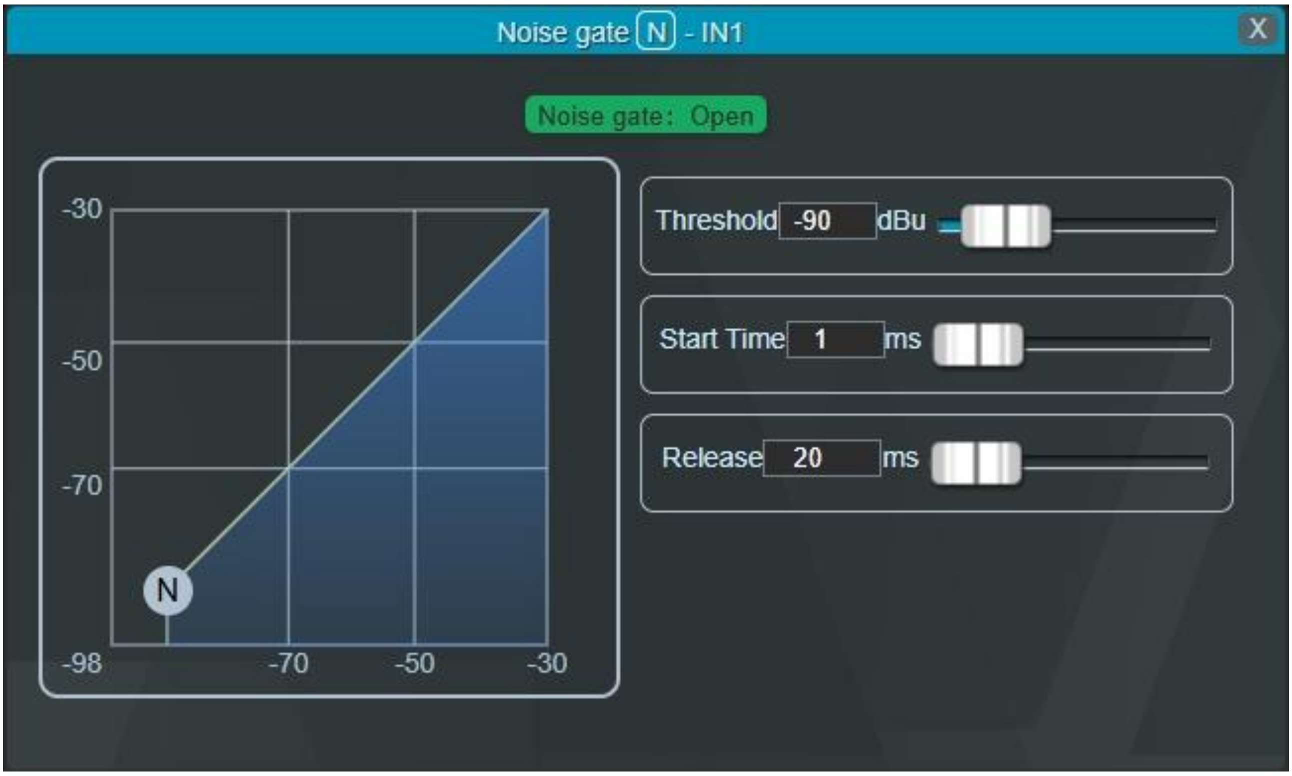
Username: root, Password: 888888 (factory default)

Processor Function Overview

As shown in the figure above, the input/output controls are positioned on the left and right sides of the interface. The menu bar at the top includes the following functions, in order: Input Module, Noise Gate, Input Equalizer, Input Compressor, Full-Featured Matrix Mixing, Output Crossover, Output Equalizer, Output Delayer, Output Limiter, and Output Module.

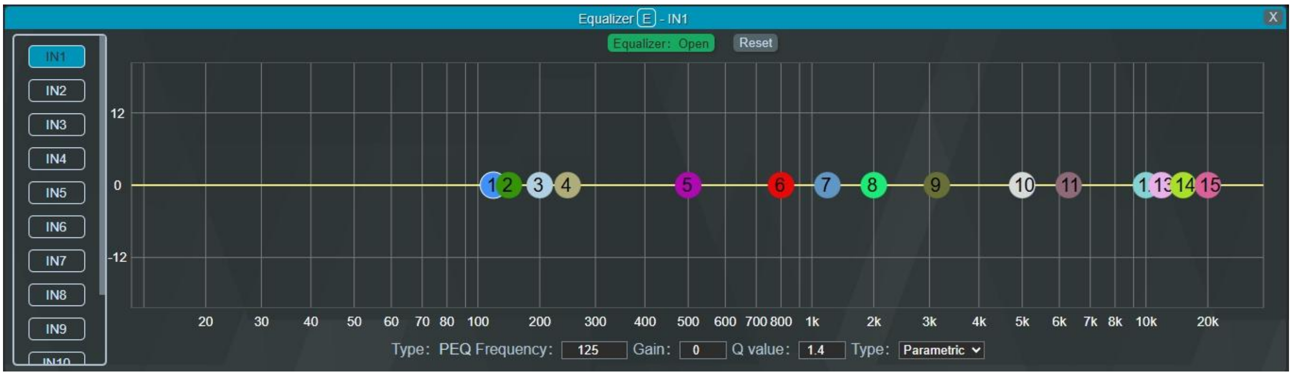
There are two types of input signals: one is the external analog signal, and the other is the test signal generated internally by the device. Only one of these signals can be selected at a time.

- Sensitivity: For analog input, the sensitivity can be set from -60~0, with a 3dB step.
- Mute: When selected, the channel is muted.
- Test Signal: Enabling the test signal will automatically block the analog input signal.
- Phantom Power: Used for powering condenser microphones. Do not enable this for line inputs or non-condenser microphones to avoid damage.
- Automatic Gain: The threshold is the minimum effective level, and the target threshold is the maximum effective level. The gain value is the gain applied by the automatic gain control.



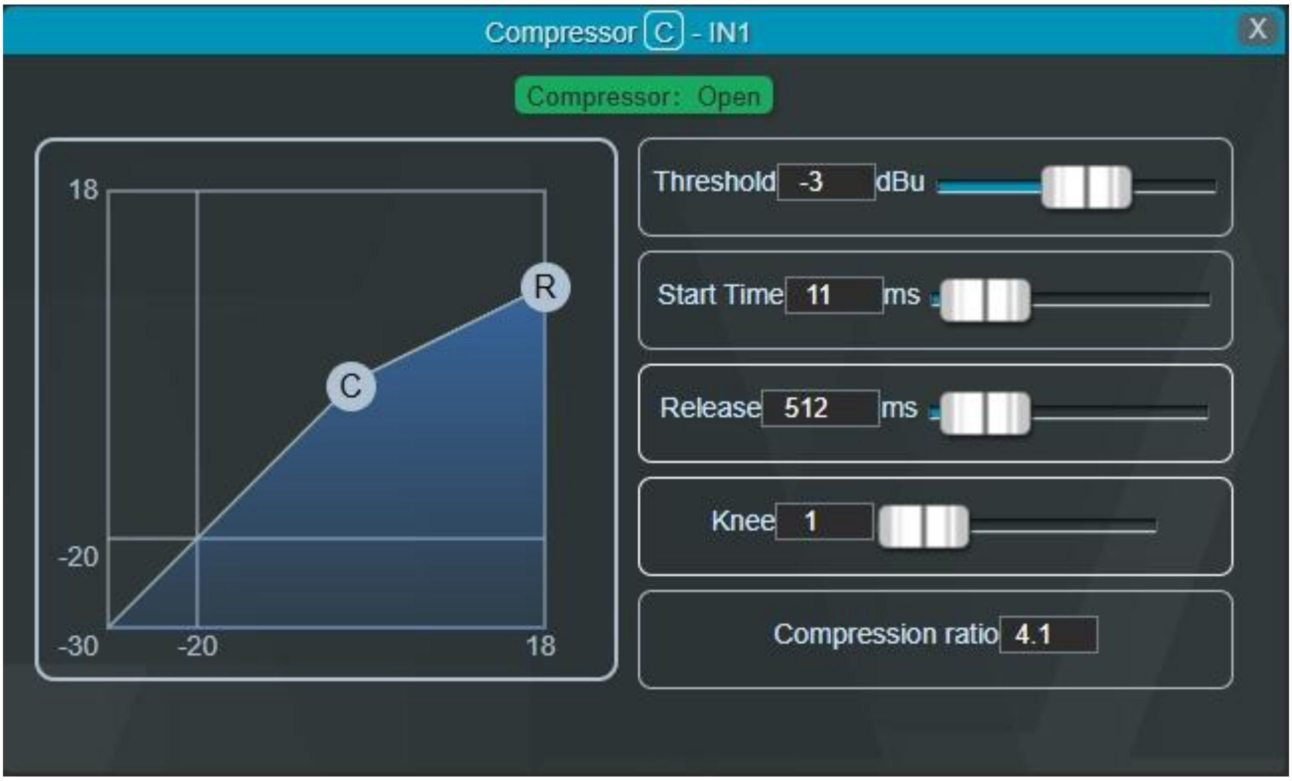
The noise gate is used to filter out the background noise coming from the previous stage based on user settings. When the input signal is lower than the “threshold”, the noise gate will mute the input signal. When the input signal is greater than the “threshold”, it will pass through with a 1:1 ratio, meaning the output level equals the input level.

- On/Off: Determine whether the noise gate is active.
- Attack Time: The time required for an input signal below the noise gate “threshold” to transition from mute to output.
- Release Time: The time required for the input signal to return from mute to its original non-muted state.



- On/Off: Determine whether the equalizer is active.
- Reset: Reset all equalizer points to their initial state.
- PEQ Frequency: The central frequency to be processed by the equalizer.

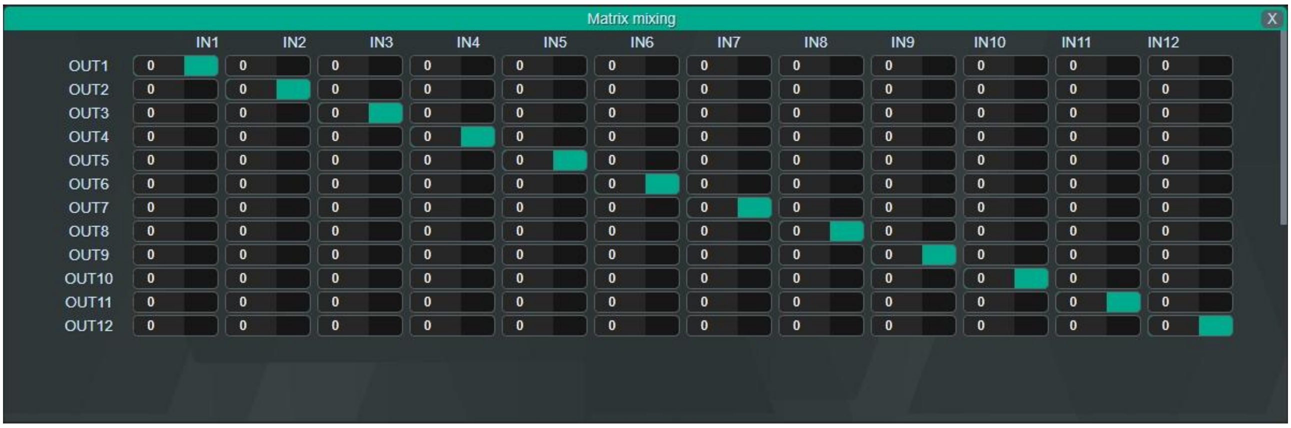
- Gain: The gain/attenuation value at the central frequency point.
- Q Value: The range of influence around the central frequency; a higher value results in a smaller bandwidth and a more focused effect.
- Type: The equalizer type can be selected from the following options: High-pass filter, Low-pass filter, High-shelf filter, Low-shelf filter, Parametric equalizer, and All-pass filter.



The compressor is used to reduce the dynamic range of the signal above a user-defined threshold. Signals below the threshold remain unchanged.

- Threshold: The signal level above which the gain reduction begins. This point is the turning point in the input/output curve. For peak limiting, the threshold should be set just below the peak level.
- Compression Ratio: The ratio of input to output compression.
- Attack Time: The speed at which the compressor begins to reduce gain. A shorter attack time results in a more significant instantaneous change in the signal, and rapid gain reduction may cause discomfort to the ear.
- Release Time: The time it takes for the compressor to return to normal gain after the signal falls below the threshold. A faster release time increases the subjective level, while a slower release time is more effective for maintaining a controlled level.
- Knee: The setting of the turning point. A larger knee value results in smoother compression (gentler but slower), while a smaller knee value leads to more abrupt compression (harsher but

faster).

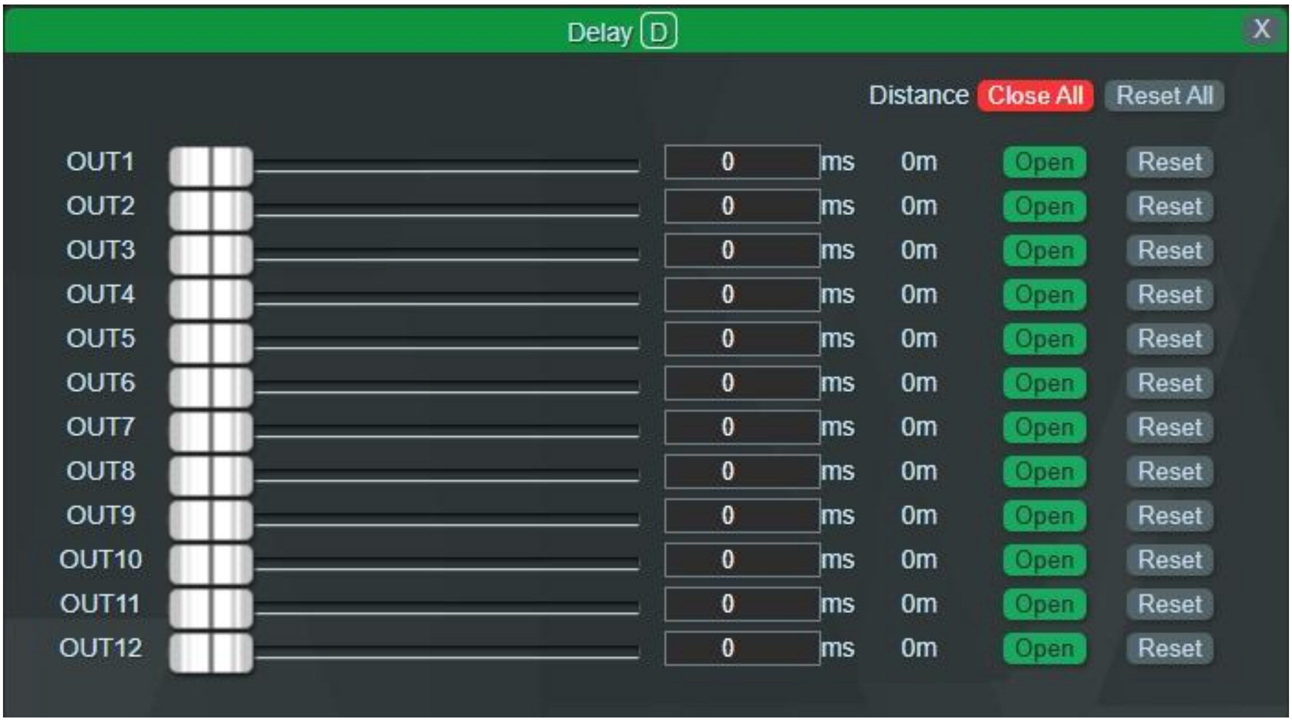


The full-featured mixing matrix is used to mix input signals and send them to the output with adjustable send levels.

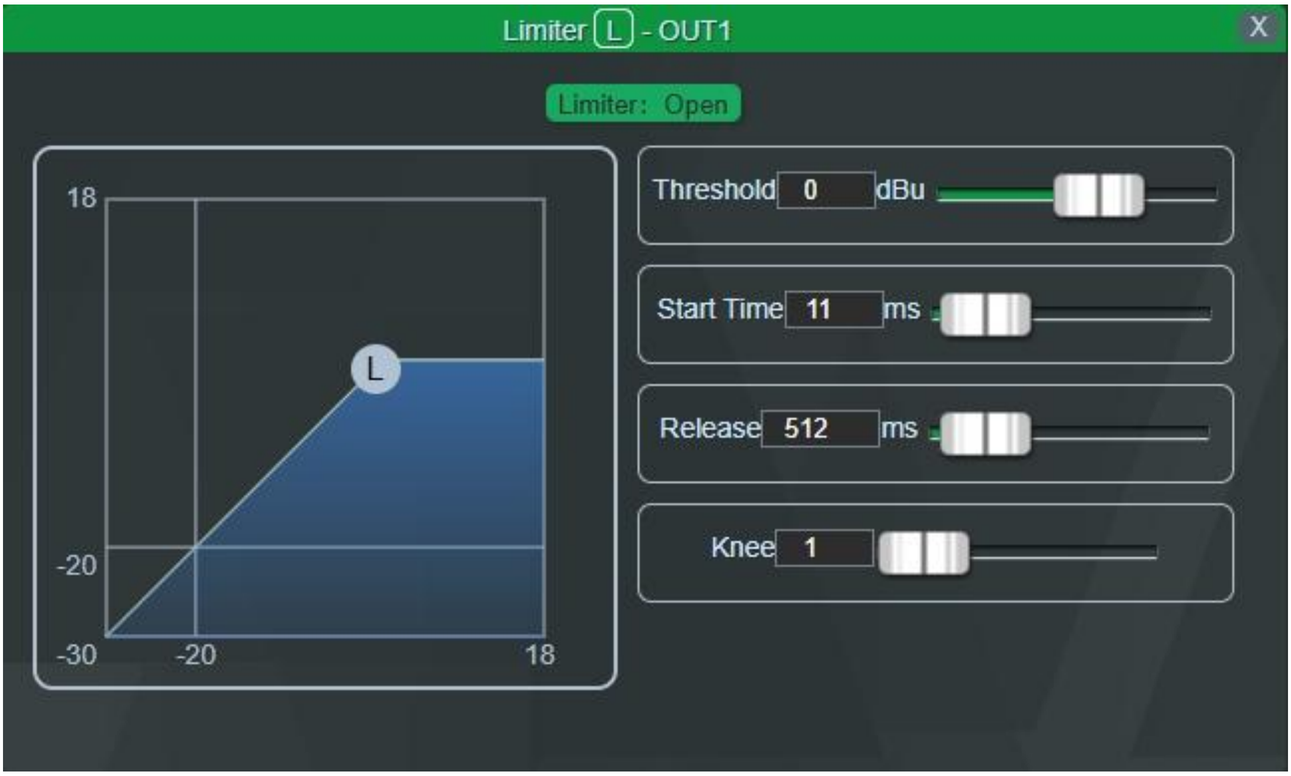
- Reset: Reset the mixing matrix to its initial state.
- Select All: Mix all input and output channels in the entire matrix.
- Selected: Click to highlight the selected channels.
- Send Level: Range from -18dB to +12dB.

- High-pass On/Off: Enable or disable the high-pass filter.
- Low-pass On/Off: Enable or disable the low-pass filter.
- High-pass Frequency: The cutoff frequency for the high-pass filter.

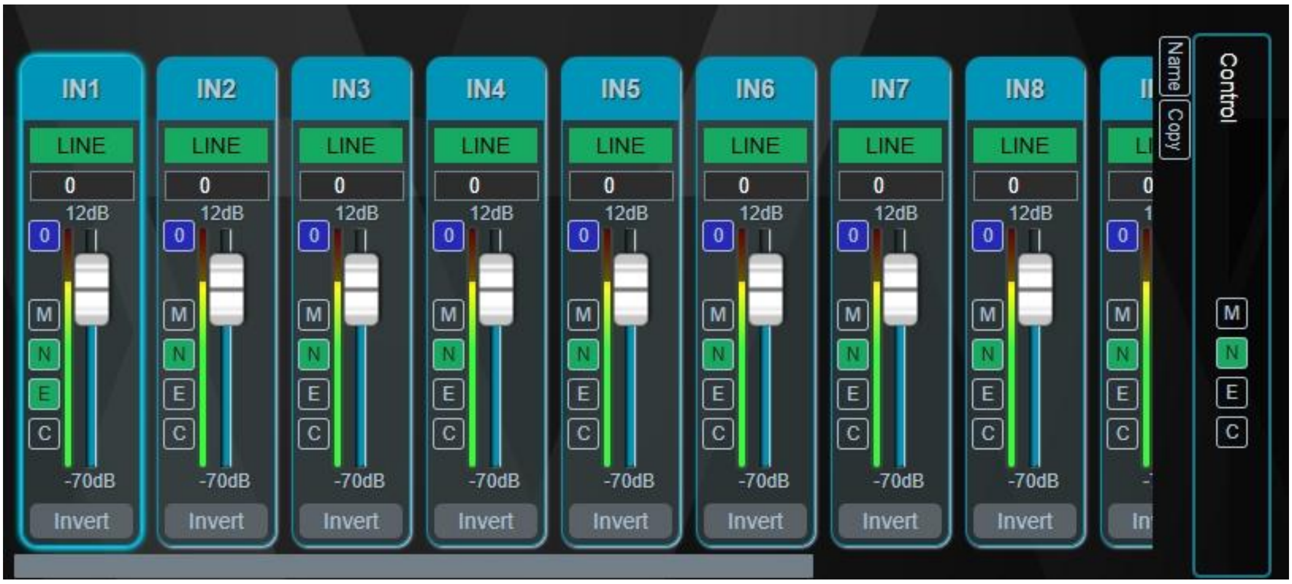
- Low-pass Frequency: The cutoff frequency for the low-pass filter.
- Filter Type: Options include Butterworth, Bessel, Linkwitz-Riley.
- Slope: Butterworth, Bessel: 6, 12, 18, 24, 30, 36, 42, 48 dB/Oct; Linkwitz-Riley: 12, 24, 36, 48 dB/Oct.



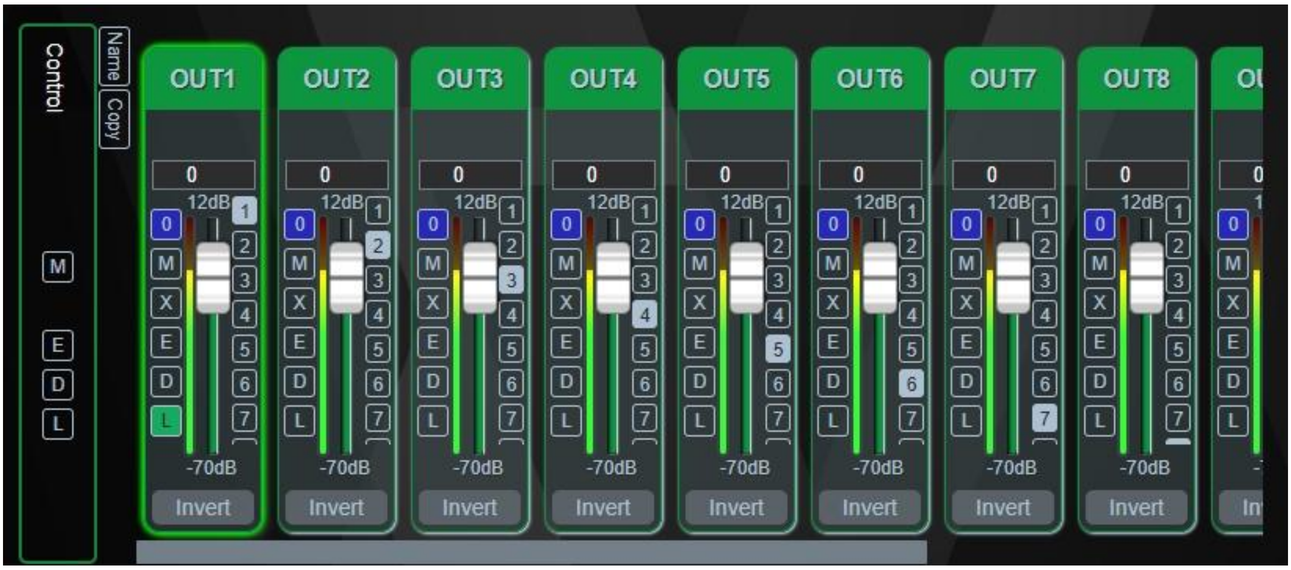
The time interval for the signal to travel from the input to the output of the processor, with an adjustable range of 0~1500ms.



- On/Off: Enable or disable the limiter.
- Threshold: The starting level for limiting. When the signal exceeds this threshold, the limiting process is activated.
- Attack Time: The speed at which the limiter begins to reduce gain. A shorter attack time results in a more significant instantaneous change in the signal, and rapid gain reduction may cause discomfort to the ear.
- Release Time: When the input signal falls below the set threshold, the channel will not immediately close. Instead, it will delay the closure based on this set release time. If the signal exceeds the “threshold” within this period, the channel will remain open.
- Knee: The setting of the turning point. A larger knee value results in smoother limit (gentler but slower), while a smaller knee value leads to more abrupt limit (harsher but faster).



Quick input operations include one-click mute for all inputs, one-click toggle for enabling/disabling the noise gate on all inputs, one-click toggle for enabling/disabling the input equalizer on all inputs, one-click toggle for enabling/disabling the input compressor on all inputs, and corresponding inverse operations for the inputs. Additionally, features include channel naming, copying/pasting channel parameters, and adjusting channel faders. The input channel level is displayed in front of the fader.



Quick output operations include one-click mute for all outputs, one-click toggle for enabling/disabling the equalizer on all outputs, one-click toggle for enabling/disabling the delayer on all outputs, one-click toggle for enabling/disabling the limiter on all outputs, enabling/disabling the crossover, and corresponding inverse operations for the inputs. Additionally, features include channel naming, copying/pasting channel parameters, and adjusting channel faders. The output channel level is displayed behind the fader.

5. Packing List

No.	Item	Quantity
1	Professional Audio Processor	1
2	Power Cord	1
3	5-Pin Pluggable Terminal Block	1
4	10-Pin Pluggable Terminal Block	1
5	12-Pin Pluggable Terminal Block	6
6	Rack Mounting Screws and Washers	4
7	User Manual	1