

# Digital Audio Processor

## Instruction Manual



This manual is an example of how to use the full-featured audio processor matrix, and can be used as a reference for other processor models.

This manual is the copyright of our manufacturer, without permission, any unit or individual may not be part of this manual or all of its contents for commercial purposes.

This manual is intended only as a user operating instructions, not for maintenance purposes.

## Content

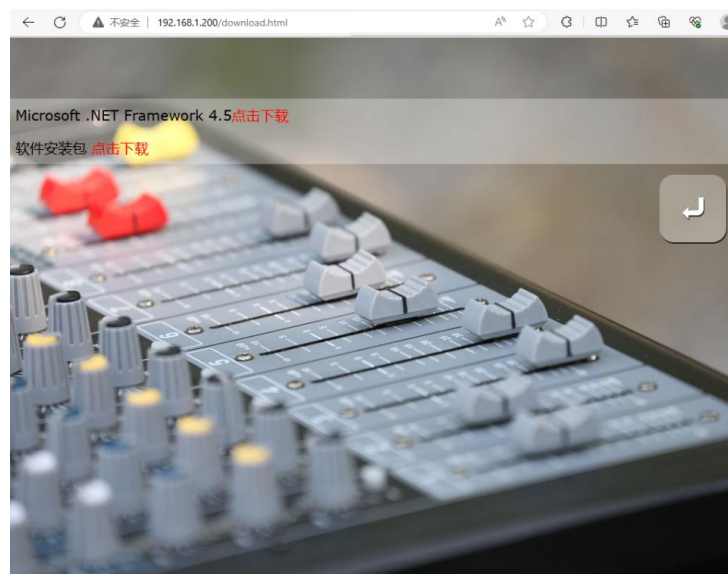
1.Product Overview .....	2
2.Specification Parameters .....	2
3.Interface Description .....	3
4.Operation Instruction .....	4
4.1Main Interface .....	4
4.2Menu bar and statu bar .....	5
Log in .....	5
Setting .....	6
Scene Setting .....	错误！未定义书签。
4.3 Pre-processing .....	7
Input Setting .....	7
Parametric Equalizer .....	错误！未定义书签。
Compressor .....	错误！未定义书签。
Evader .....	错误！未定义书签。
Feedback/Echo/Noise .....	错误！未定义书签。
4.4 Matrix Mixing .....	11
4.5 Post-processing .....	13
Graphic Eualizer .....	13
Limiter .....	14
Output Setting .....	15
6. Warranty Regulations .....	16

## Software Download

The installation software source file is embedded in the device, download the software only by entering the device factory default IP address (default IP: 192.168.1.200) information in the browser url address bar, enter to navigate to the download interface, according to the content of the web interface information click to download the software can be, in addition to the need to pay attention to the installation of the PC side of the software, please make sure that the PC client has installed the following Net Framework3.5 or above.

**Note: When downloading the software, make sure that the PC client and the IP address of the device (default IP: 192.168.1.200 subnet mask: 255.255.255.0) are in the same network segment, or else you will not be able to access it;**

Download interface:



## 1.Product Overview

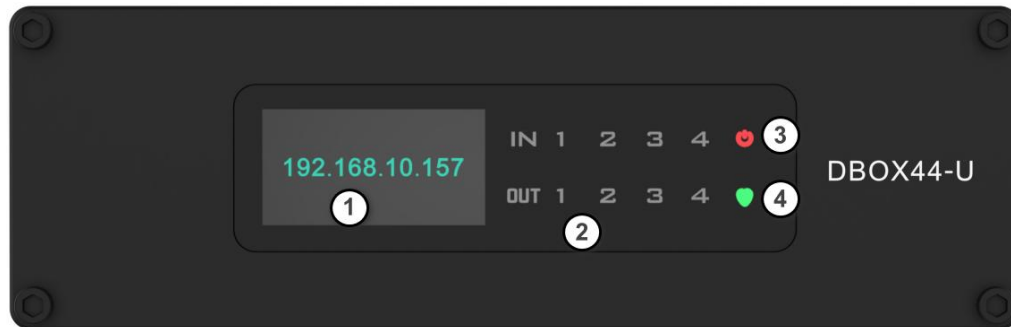
This equipment supports analog channel 4 in 4 out, USB playback and recording, comprehensive matrix mixing function, equipped with high-performance A/D D/A converter and 32-bit floating-point DSP processor, support 24bit/48KHz sampling frequency, restore high-quality sound, user-friendly operation software interface, easy to operate and powerful performance. Mainly used in a variety of large places, can meet the theater, concert halls, remote video conferencing, stadiums, churches, conference centers, theme parks, public sound reinforcement systems and other aspects of the application needs.

## 2. Specification parameters

Parameter Indicators	
Analog	4 balanced inputs + 4 balanced outputs
DANTE	4 inputs+4 outputs
USB Sound Card	1input+1output
RS232	1
RS484	1
★Input gain amplification	36dB
Quantile (math.)	24bit
Sampling Rate	48kHz
Phantom power (per input)	48V
Frequency Response	20Hz~20KHz, ±0.15dB
THD+N	≤0.003% @ 4dBu
Input Impedance (Balanced)	20KΩ
Output Impedance (Balanced)	100Ω
Channel Isolation	100dB@1kHz, 4dBu
Common Mode Rejection	52dB@60Hz
Maximum Input Level	18dBu
Maximum Output Level	18dBu
Background Noise	-89dBu
Operating Temperature	-10-40°C
★Operating power	AC110V-220V,50Hz/60Hz; DV12V; PoE Power Supply
Dimensions (W*D*H)	142X157*46mm
Input Impedance (Balanced)	20KΩ
Output Impedance (Balanced)	100Ω

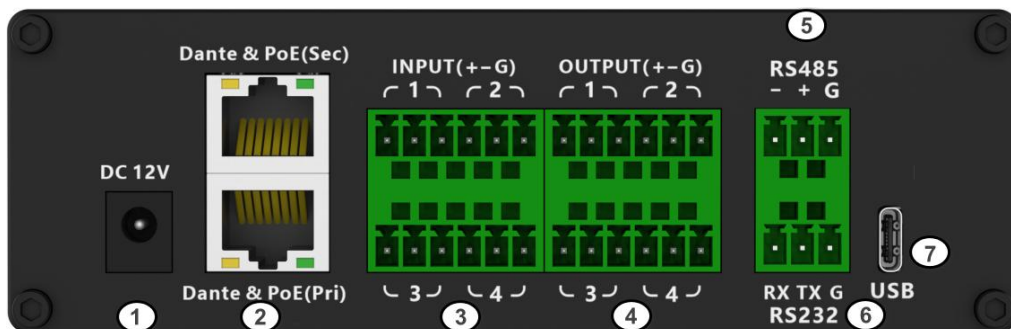
### 3. Interface Description

#### Front Panel



- 1.OLED display: after normal operation of the system, the current IP address information of the device will be displayed;
- 2.Channel indicator: when the analog IN/OUT channel flows into a very small signal, the indicator is not lit; when the analog IN/OUT channel flows into a very large signal, the indicator is red; in other cases, the indicator is green;
- 3.Power indicator: when the device is powered on, the power indicator is always on;
- 4.System indicator: when the device is running normally, the indicator shows flashing status.

#### Rear Panel



1. DC 12V: DC power supply interface. 2;
2. Dante POE network power supply interface: this interface can be used for Dante digital signal transmission, connecting to PC interactive control software and as a direct power supply to the device;

3. INPUT signal input interface: can be connected to microphone, DVD and other devices. 4;
4. OUTPUT signal output interface: can be connected to amplifier, active speakers and other devices;
5. RS485 interface: RS485 balanced interface;
6. RS232 interface: RS232 balanced interface;
7. USB interface: support USB sound card function.

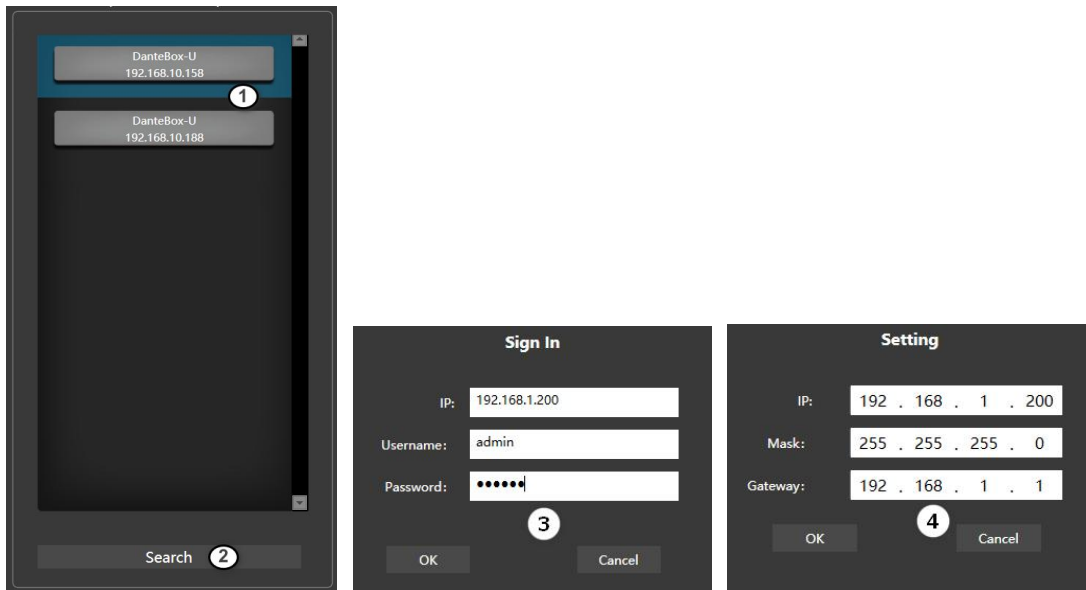
## 4. Operation Instruction

### 4.1 Main Interface



## 4.2 Menu bar and Status bar

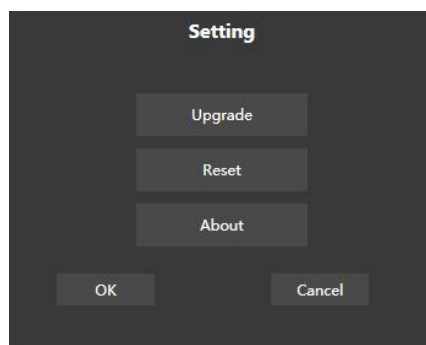
Log in



a) Click [Search] button, the device list will show the online device name and IP address information. b) Double click the device name in the device list, the Sign IN login window will pop up;

b) Double-click the device name in the device list, the Sign IN login window will pop up, enter the user name and password: (default user name: admin, password: 123456) Click [OK] button to enter; if you need to modify the device IP address information, right-click the name of the device, the Setting window will pop up, according to the needs of the environment for changes can be made.

Setting

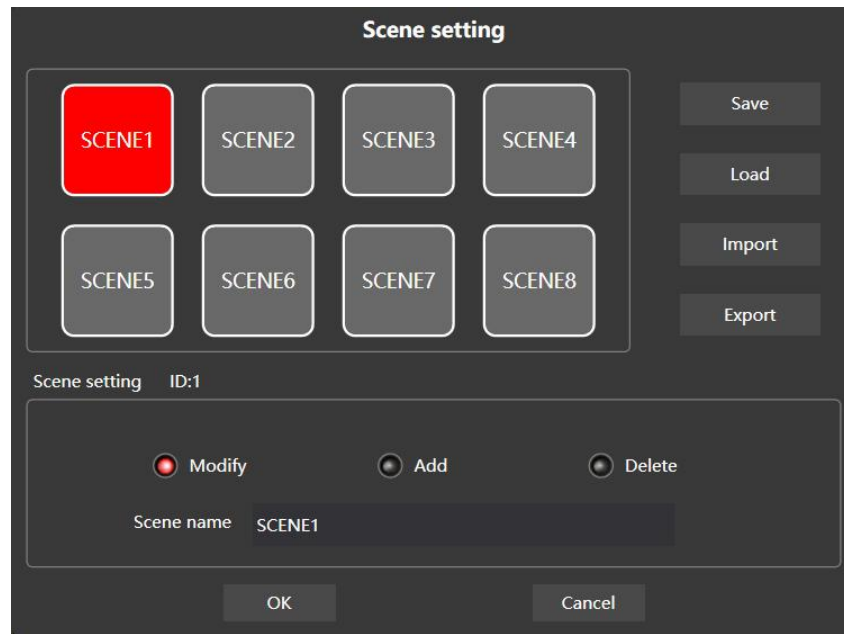


Upgrade: Device firmware upgrade;

Reset: Restore Factory Defaults;

About: View device serial number information.

## Scene Setting



**Save:** Saves the current parameters to the selected scene;

**Load:** Load the selected scene;

**Import:** Importing a saved scene file;

**Export:** Exporting Scenes to Local;

**Modify:** Modify the scene name;

**Add:** Add new scenes, up to 8 scenes;

**Delete:** Delete Scene.

### 4.3 Pre-processing

#### Input Setting



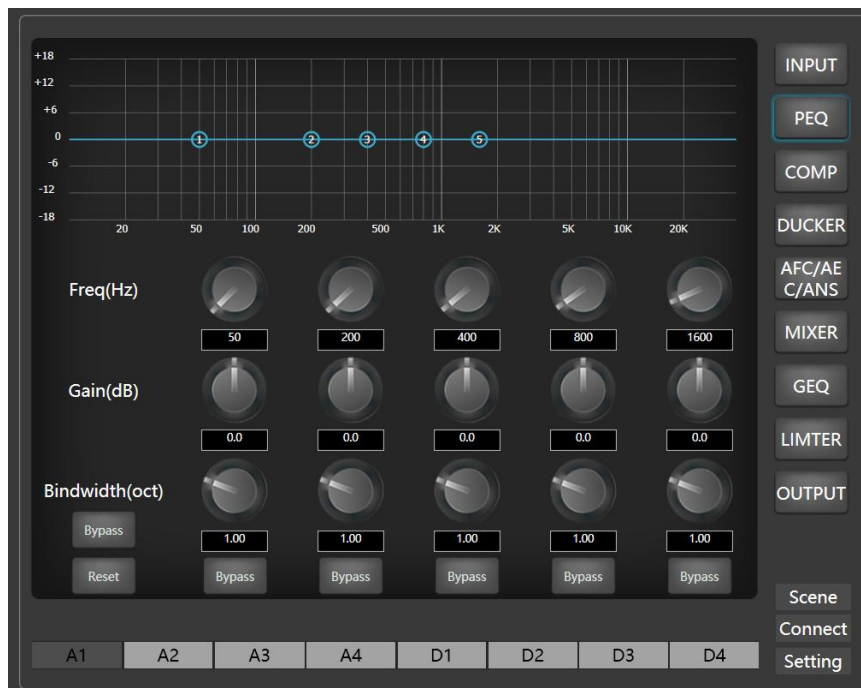
Indicator light: when there is no signal into the input channel, the indicator light is gray; when there is a small signal into the input channel, the indicator light is green; when there is a large signal into the input channel, the indicator light is red;

48V: analog channel 48V phantom power supply on or off, non-condenser microphone please do not turn on, to prevent burning;

Mute: mute button;

Channel fader: channel gain value range (0~-72dBFS), can be controlled by the fader.

## Parametric Equalizer



Bypass/Active: enable/disable the equalizer;

Freq Bypass/Active: enable/disable the current band equalizer;

Reset: restores the current parameters to the default state;

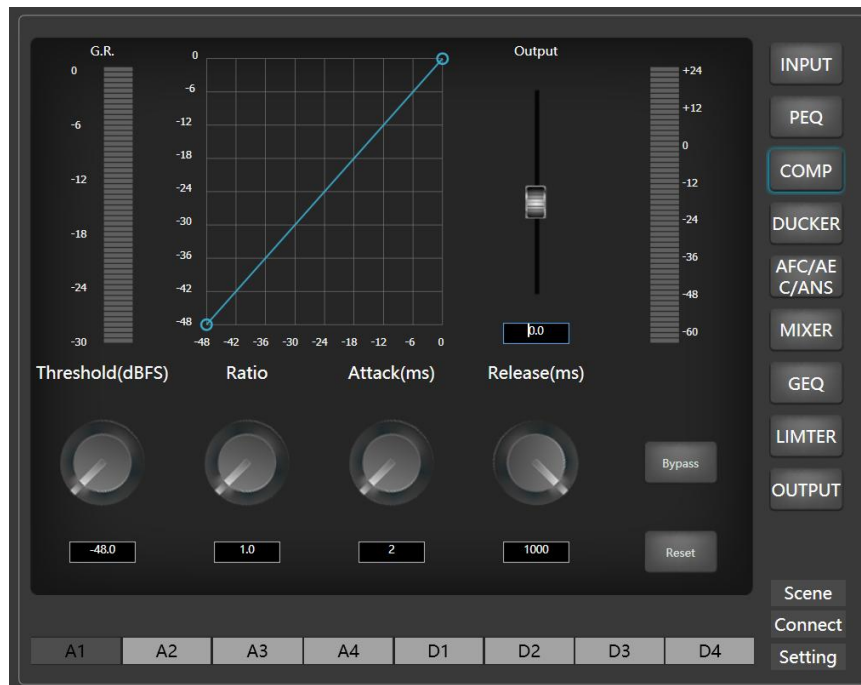
Freq: the center frequency to be equalized;

Gain: the gain/attenuation value at the center of the frequency;

Bandwidth: the influence range of the band around the center frequency, the larger the value, the larger the bandwidth and the larger the influence range.

## Compressor

The compressor is used to reduce the dynamic range of a signal above a user determined threshold. Signal levels below the threshold remain unchanged.



Threshold: the threshold value of the compressor;

Ratio: the input/output compression ratio of the compressor;

Attack: startup time of the compressor;

Release: the recovery time of the compressor;

Reset: reset to default parameters;

Bypass/Active: enable/disable the compressor.

## Evader



Paging Input: control signal input channel;

Background Input: controlled signal input channel;

Threshold: when the control signal level is higher than the threshold value, the controlled signal will be blocked;

Depth: attenuation value of controlled signal;

Attack Time: the effective time of dodger algorithm;

Hold Time: when there is no signal input from the control signal input channel, the algorithm will hold time;

Release Time: release time of dodger algorithm;

Output Selection: output channel;

Reset: reset as default parameter;

Bypass/Active: enable/disable the compressor.

## Feedback/Echo/Noise



AFC: selects the signal that needs to be processed through the feedback eliminator, and the processed signal selects the output channel in the mixer;

AEC: sets the signal that needs to be processed through the echo canceller, and the processed signal selects the output channel in the mixer;

Input: local MIC input, i.e. the signal that needs to be processed by echo canceller;  
Remote: reference signal.

ANS: select the signal that needs noise cancellation processing, and the processed signal selects the corresponding channel output in the mixer.

## 4.4 Matrix Mixing

Clear route All route Default route

	Input 1	Input 2	Input 3	Input 4	Dante 1	Dante 2	Dante 3	Dante 4	USB Play
Output 1	0.0	0.0	0.0	0.0	0.0	0.0	0.0	0.0	0.0
Output 2	0.0	0.0	0.0	0.0	0.0	0.0	0.0	0.0	0.0
Output 3	0.0	0.0	0.0	0.0	0.0	0.0	0.0	0.0	0.0
Output 4	0.0	0.0	0.0	0.0	0.0	0.0	0.0	0.0	0.0
Dante 1	0.0	0.0	0.0	0.0	0.0	0.0	0.0	0.0	0.0
Dante 2	0.0	0.0	0.0	0.0	0.0	0.0	0.0	0.0	0.0
Dante 3	0.0	0.0	0.0	0.0	0.0	0.0	0.0	0.0	0.0
Dante 4	0.0	0.0	0.0	0.0	0.0	0.0	0.0	0.0	0.0
USB Rec	0.0	0.0	0.0	0.0	0.0	0.0	0.0	0.0	0.0

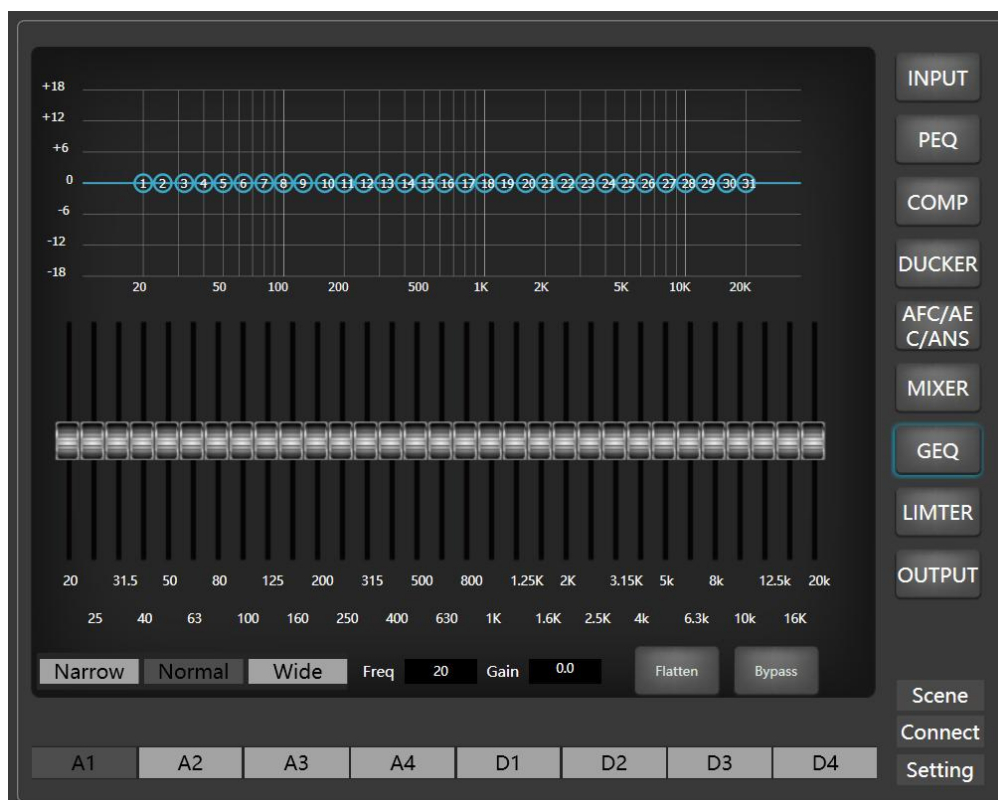
INPUT  
PEQ  
COMP  
DUCKER  
AFC/AE  
C/ANS  
MIXER  
GEQ  
LIMTER  
OUTPUT

Scene  
Connect  
Setting

A1 A2 A3 A4 D1 D2 D3 D4

Columns: Input channels Rows: Output channels

## 4.5 Post-processing Graphic Equalizer



31 bands of frequency points can be individually adjusted gain, so as to achieve the purpose of strengthening, weakening certain frequency points, to achieve different effects;

Bypass/Active: enable/disable the equalizer;

Gain: the gain/attenuation of the frequency center point;

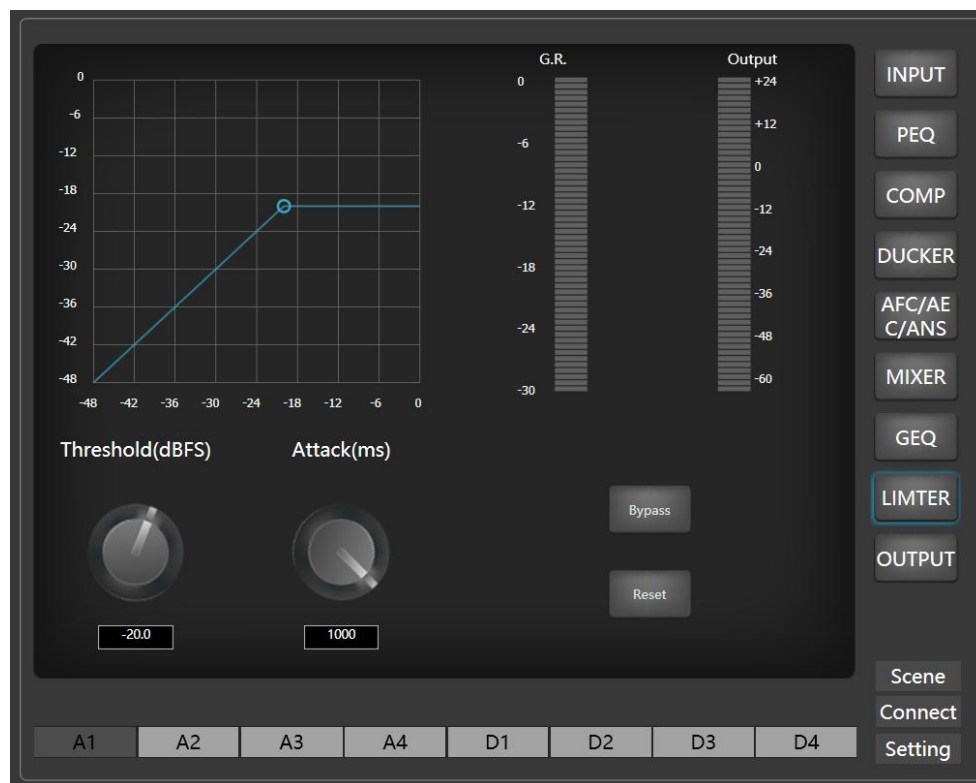
Flatten: restores all frequency bands to 0dB gain;

Narrow: a kind of bandwidth, the bandwidth is lower than the normal bandwidth;

Normal: the common normal bandwidth;

Wide: the largest bandwidth.

## Limiter



Bypass/Active: enable/disable the limiter;

Reset: reset to default parameter;

Threshold: the starting level of the limiter, when the signal is higher than this limit value, the limiter processing module will be activated;

Attack: when the input signal is lower than this value, the sound channel will not be shut down immediately, but will delay the shutdown time according to this value. During this time, as long as there is a signal above the "Threshold" limit value, the sound channel can be turned on continuously.

G.R.: The difference between the signal processed by the limiter and the input signal.

## Output Setting

Muting and inverting the outputs as well as the output audio gain can be set.



Indicator light: when there is no signal into the output channel, the indicator light is gray; when there is a small signal into the output channel, the indicator light is green; when there is a large signal into the output channel, the indicator light is red;

INVERT: invert button;

Mute: mute button;

Channel fader: channel gain value range (0~-72dBFS), can be controlled by fader.

## **6.Warranty Regulations**

In mainland China, the warranty period of this product is 1 year.

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

Warranty card stamped by the sales unit to take effect. The warranty card will not be valid if it is altered!

**The following conditions (including but not limited to) are not covered by the three guarantees:**

1. No warranty card or missing valid invoice or the date has exceeded the validity period of the three guarantees;
2. Damage caused by failure to use, maintain and manage the product in accordance with the requirements of the instruction manual;
3. the product model or code on the warranty voucher does not match with the physical goods;
4. Damage caused by dismantling and repairing by non-authorized service providers;
5. Normal discoloration, wear and tear and consumption during the use of the product are not covered by the warranty;
6. the product can not be used due to the user's own network reasons, please consult the customer service staff.