



DriveSYS series

Audio Matrix Processor

User Manual

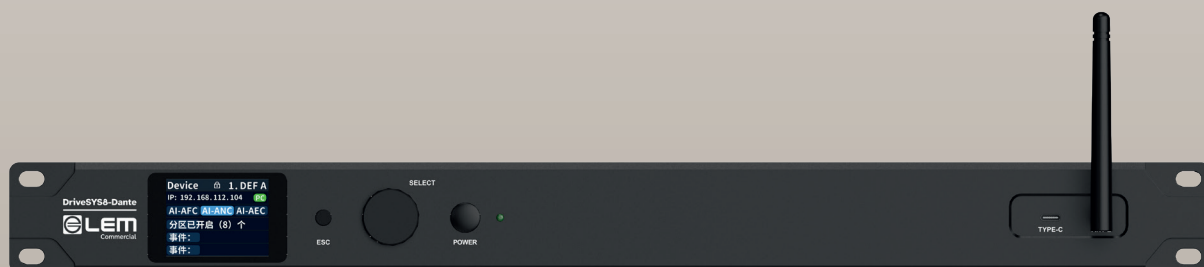


Table of Contents

Chapter 1 Introduction	5
Chapter 2 Technical Specifications	7
Chapter 3 Functional Structure	9
Chapter 4 WEB Interface Overview	12
Chapter 5 WEB Interface Usage	12
5.1 Operating Environment	12
5.2 Operation Steps.....	12
Chapter 6 WEB Interface Description	13
6.2 Control Panel	16
6.2.1 Device Name, Current Preset and Master Volume	16
6.2.2 Device Status and Local/Network Audio IP Address	16
6.2.3 Input Channel Interface	17
6.2.5 Output Channel Interface	19
6.3 DSP Flowchart	22
6.4 Signal Source Input	
6.5 DSP Input	23
6.5.1 Test Signal	23
6.5.2 Noise Gate	23

6.5.3 Compressor	25
6.5.4 Expander	26
6.5.5 PEQ/High-Low Pass	27
6.5.6 DEQ (Dynamic Equalizer)	43
6.5.7 Input Delay	29
6.6 FIR	
6.7 Matrix Mixing	35
6.8 DSP Output	36
6.8.1 Delay	39
6.8.2 Output PEQ	39
6.9 Speaker Parameters	41
6.9.1 Speaker Parameters - FIR	41
6.9.2 Speaker Parameters - PEQ/High-Low Pass	42
6.9.3 Speaker Parameters - Delay	44
6.9.4 Speaker Parameters - Gain	41
6.9.5 Speaker Parameters - Compressor	44
6.9.6 Speaker Parameters - Limiter	45
6.10 Presets	46
6.10.1 Scene Presets	46
6.10.2 Speaker Presets	47

6.11 Settings50

6.11.1 Device Information50

6.11.2 Network Information50

6.11.3 Security Settings51

6.11.4 Gain Limit52

6.11.5 Cloud Drive53

6.11.6 GPIO.....53

Chapter 1 Introduction

This series have 8x8 or 16x16 auto-mixing media matrix with 8/16 analog inputs and 8/16 analog outputs, AES67 32 receives and 32 outputs. It features DSP including feedback suppression, auto-mixing, matrix mixing, zone, EQ, crossover, compressor, AGC, AEC, AI ANC. It connects via USB (driver-free) and supports RS232, RS485, TCP/IP for remote control.

Applications

- ▲ Performance Hall
- ▲ Stadium
- ▲ Professional Performance
- ▲ Auditorium
- ▲ Multi-function Hall
- ▲ Conference System

Features

1. 16 balanced analog inputs and 16 balanced analog outputs, max 20dBu (7.746Vrms) without distortion
2. Each input channel with 48V phantom power
3. Each input channel with mic preamp, 0~40dB gain, 1dB steps
4. Stereo USB audio interface, supports playback and recording

- 5、 USB interface, USB audio transmission
- 6、 RS232 interface, central control
7. RS485 interface, central control
8. Network interface, central control, manage multiple devices via network
9. External 8-channel GPIO
10. LCD screen for device name, presets, IP, input/output volume, input mode, device version
11. DSP: AFC, AEC, ANC, AGC, AUTO MIX, MATRIX MIX, zone, speaker library, noise gate, PEQ, delay, PHASE filter, high/low pass, compressor, limiter
12. Input 8-band PEQ, output 8-band PEQ, speaker 15-band PEQ. 14 filter types: Parametric, Low Shelf, High Shelf, Allpass (180), ALLPASS-2 (360), VariQ Low Pass, VariQ High Pass, Phase, Elliptic Low Pass, Elliptic High Pass, LOW PASS1, HIGH PASS1, Band Pass, Notch
13. Each speaker with 3-band DEQ (PEQ/Low Shelf/High Shelf)
13. Each channel input 2000ms delay, output 2000ms delay, speaker preset 2000ms delay
14. Input voice activation (camera tracking), supports PELCO-D, PELCO-P, VISCA. 15. 80 preset memory positions
16. Supports wired remote control box
17. WEB supports mobile phone control
18. Supports preset locking, speaker preset locking, hiding settings for project security

Chapter 2 Technical Specifications

DSP Chip	
Signal Processing	32bit float point dsp 1GHz
Audio System Latency	3.0ms
D/A Conversion	24-bit
Sampling Rate	48kHz

Analog Audio Inputs and Outputs	
Input Channels	16 balanced analog inputs, Mic/line level
Audio Connector	3.81mm Phoenix connector, 12-pin
Input Impedance	7k Ω
Maximum Input Level	20dBu (7.746Vrms)/Line; 0dBu (0.774Vrms)/Mic @20dB gain
Phantom Power	+48VDC, 5.5mA, per channel
Output Channels	16 balanced analog outputs, line level
Output Impedance	87 Ω

Audio Performance Specifications	
Frequency Response	20Hz-20kHz (+-0.5dB)/Line, input 0dBu; 20Hz-20kHz (+-0.8dB)/Mic, 20dB gain
Noise Floor	20Hz-20kHz, A-wt,-93dBu /15.5uV
THD+N	-90dB(@0dBu,1kHz,A-wt)/Line -76dB (@0dBu, 1kHz, A-wt)/Mic, 20dB gain
Signal-to-Noise Ratio	114dB(@20dBu,1kHz,A-wt)/Line 105dB (@0dBu, 1kHz, A-wt)/Mic, 20dB gain
Channel Isolation	-110dB(@0dBu,1kHz)/Line -120dB (@0dBu, 1kHz) Mic, 20dB gain
Maximum Input	20dBu/7.74Vrms 0dBu/0.775Vrms/Mic, 20dB gain

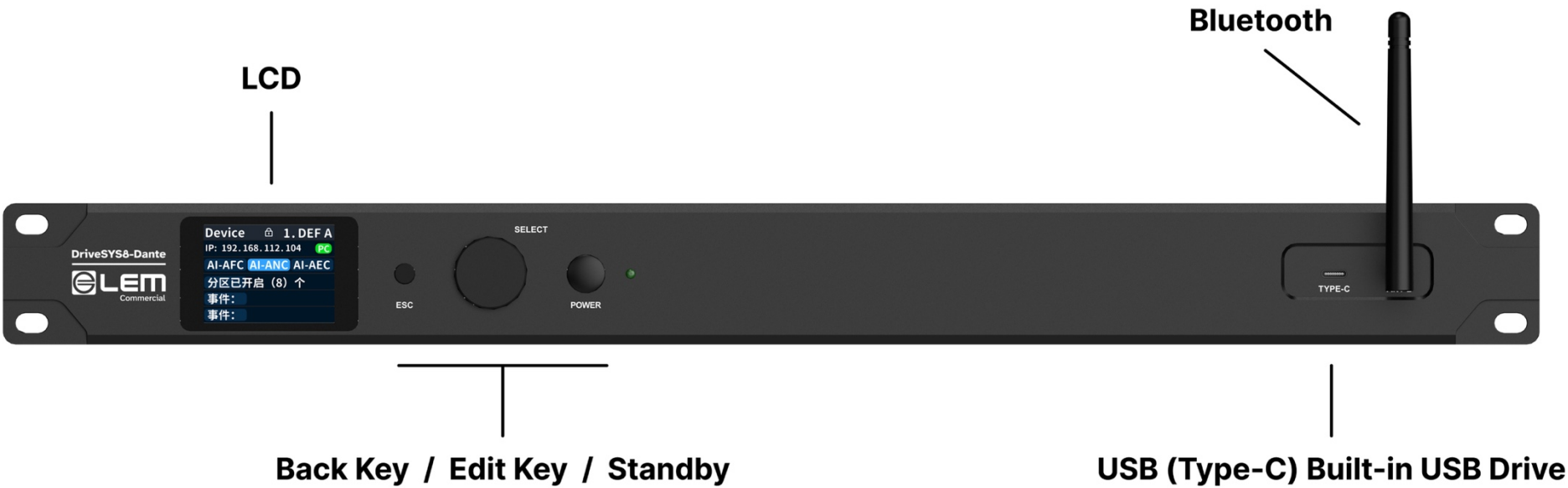
Communication Ports and Indicators	
USB	Built-in USB drive 31MB
RS232	Serial port communication
TCP/IP Network Port	RJ-45
Indicators	Power LED

Electrical	
Power Supply	AC100V---240V 50/60 Hz
Operating Temperature	-20°C~60°C

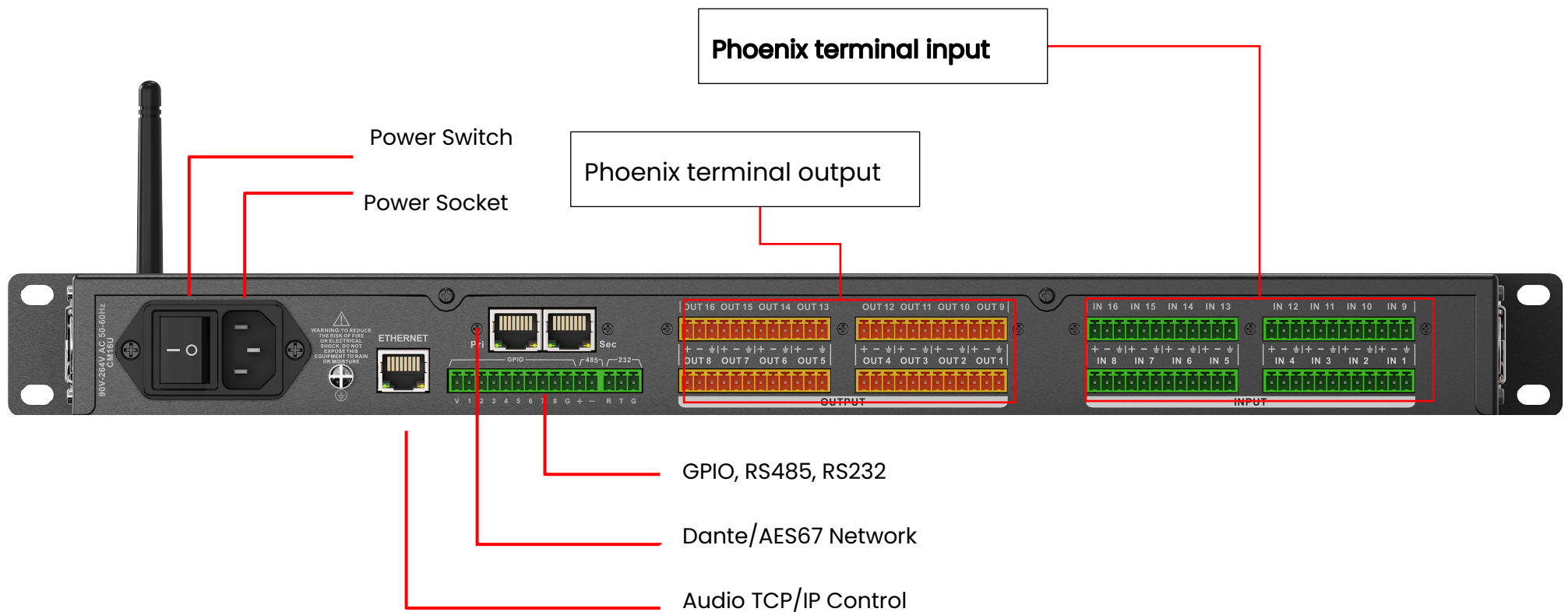
Physical	
Net Weight	3.6kg
Gross Weight	4.6kg
Package Weight (5 units)	23.1kg
Device Dimensions (W*D*H)	483*276*44.5mm
Package Dimensions (W*D*H)	542*393*77mm
Shipping Box Dimensions (W*D*H)	560*415*425mm

Chapter 3 Functional Structure

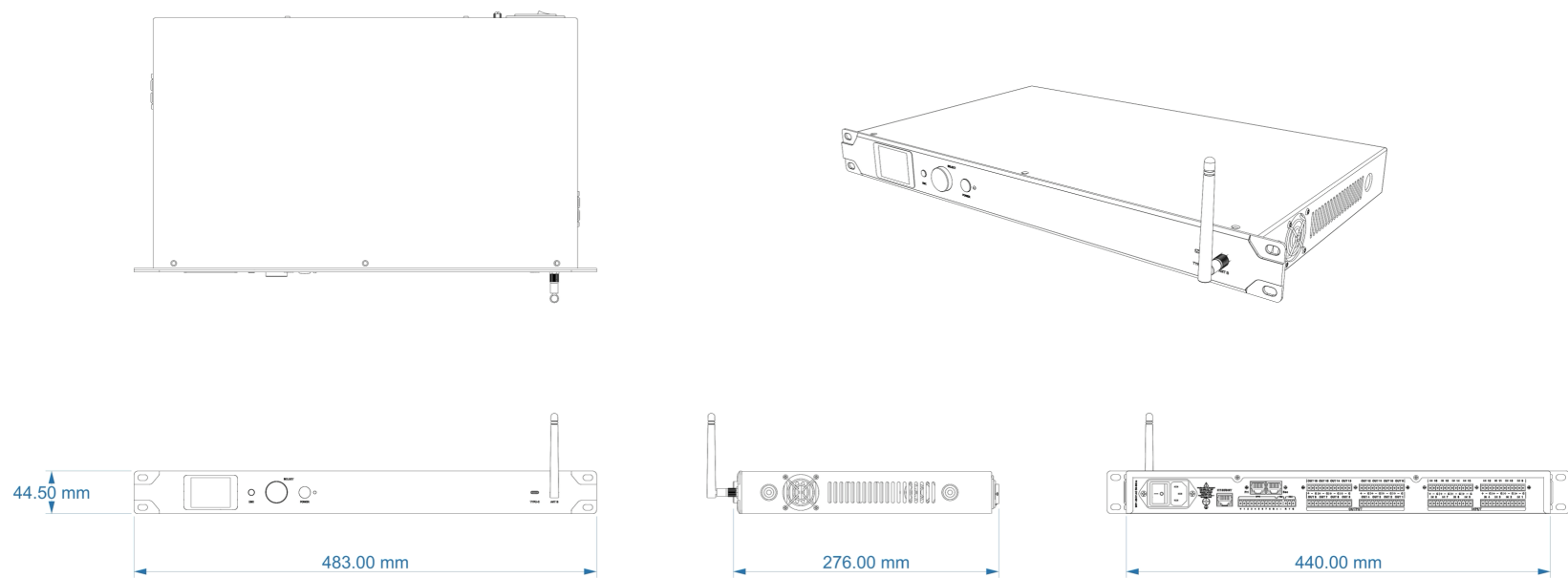
Front Panel



Rear Panel



Machine Dimensions



Chapter 4 WEB Interface Overview

The WEB interface allows quick interaction with parameters of one or multiple devices. It saves configuration parameters to disk files for easy preset scene configuration and parameter switching/restoration. The UI uses a self-developed control library and can be customized per customer needs.

Chapter 5 WEB Interface Usage

5.1 Operating Environment

Use mainstream browsers like Google Chrome or Microsoft Edge. Internet Explorer is not supported.

5.2 Operation Steps

Enter the device IP to directly access the interface for debugging.

Notes:

Default IP allocation is DHCP. Without a router, initial IP is 169.254.X.X, mask 255.255.0.0. Set your computer IP to the same subnet. Device IP can be viewed on screen after power-on.

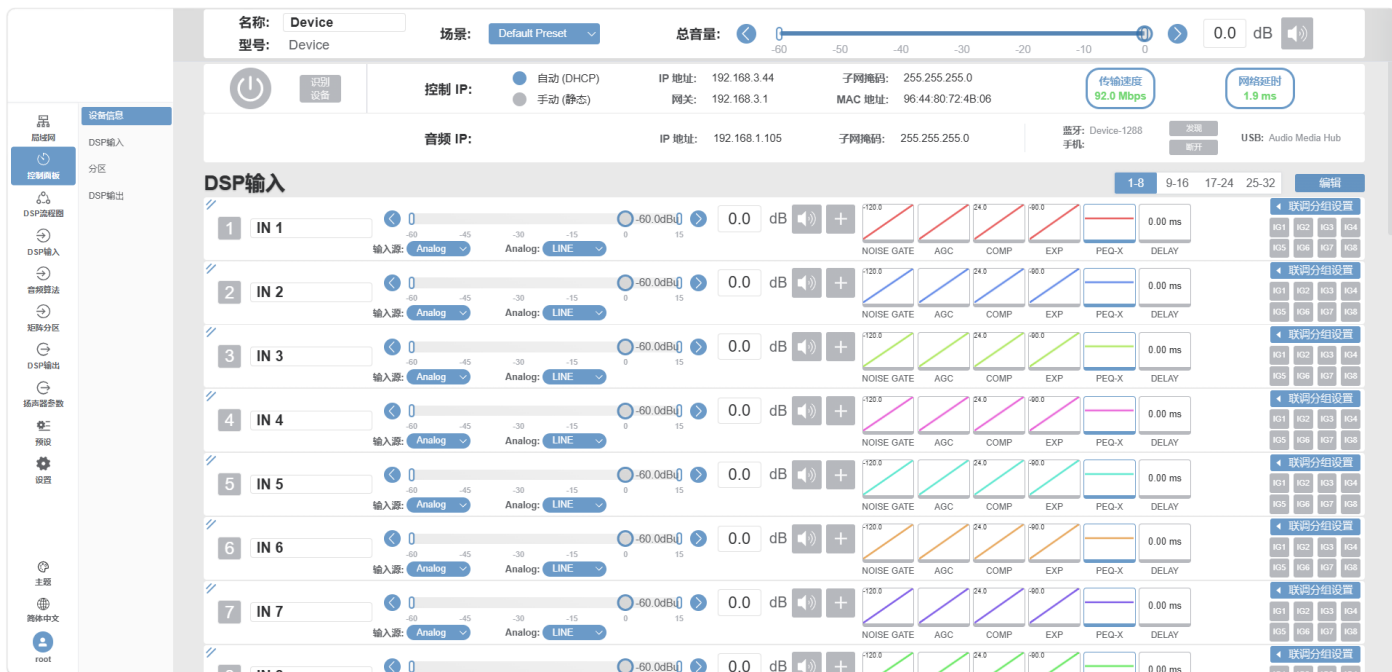


Figure 1.1

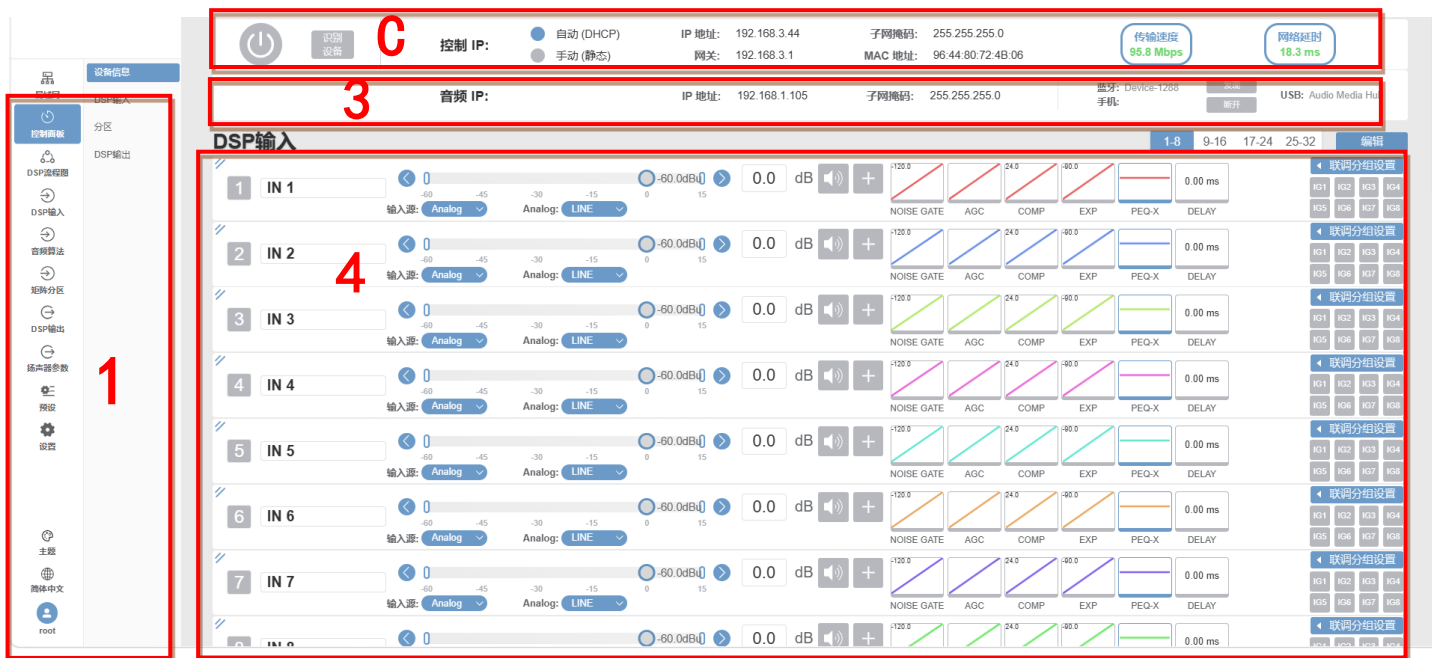


Figure 1.2

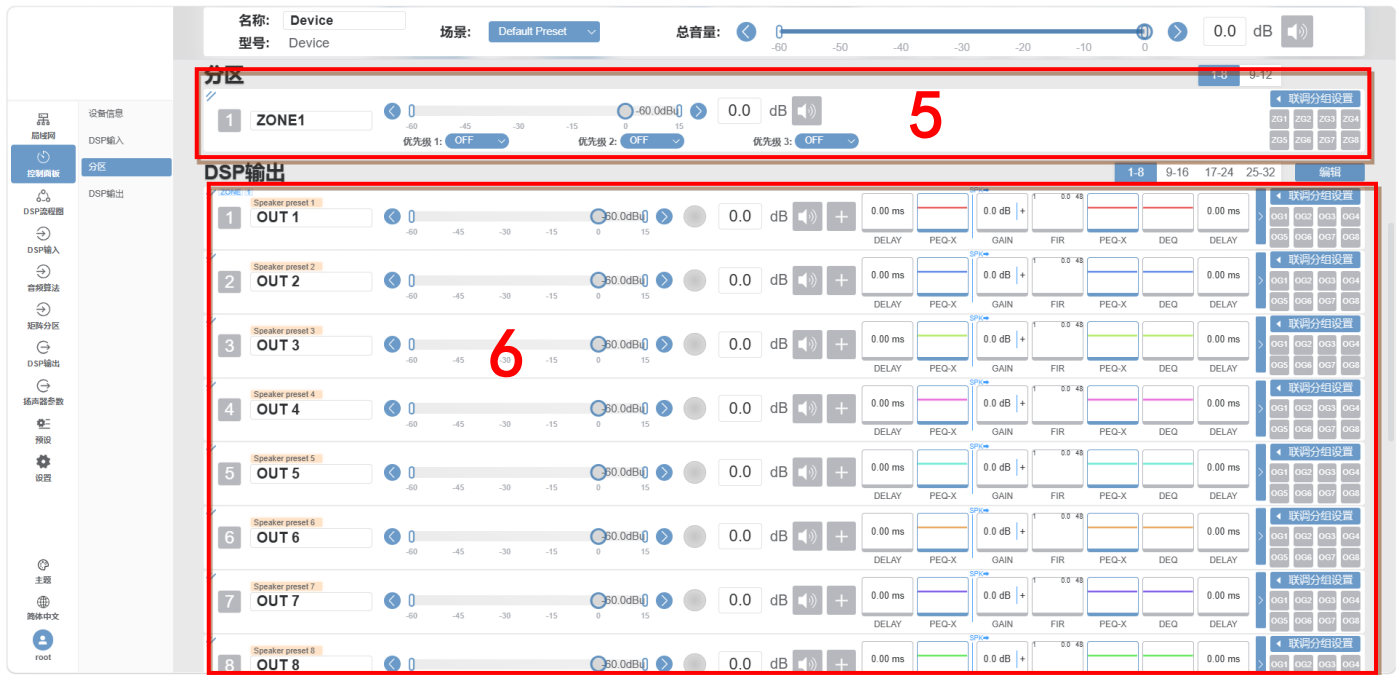


Figure 1.3

As shown in Figures 1.2 and 1.3, the modules corresponding to red text positions in the main interface are as follows:

- (1) Menu Bar
- (2) Device Name, Current Preset and Master Volume Control
- (3) Device Status and Local/Network Audio IP Address
- (4) Input Channel Interface
- (5) Zone Channel Interface
- (6) Output Channel Interface

6.1 LAN

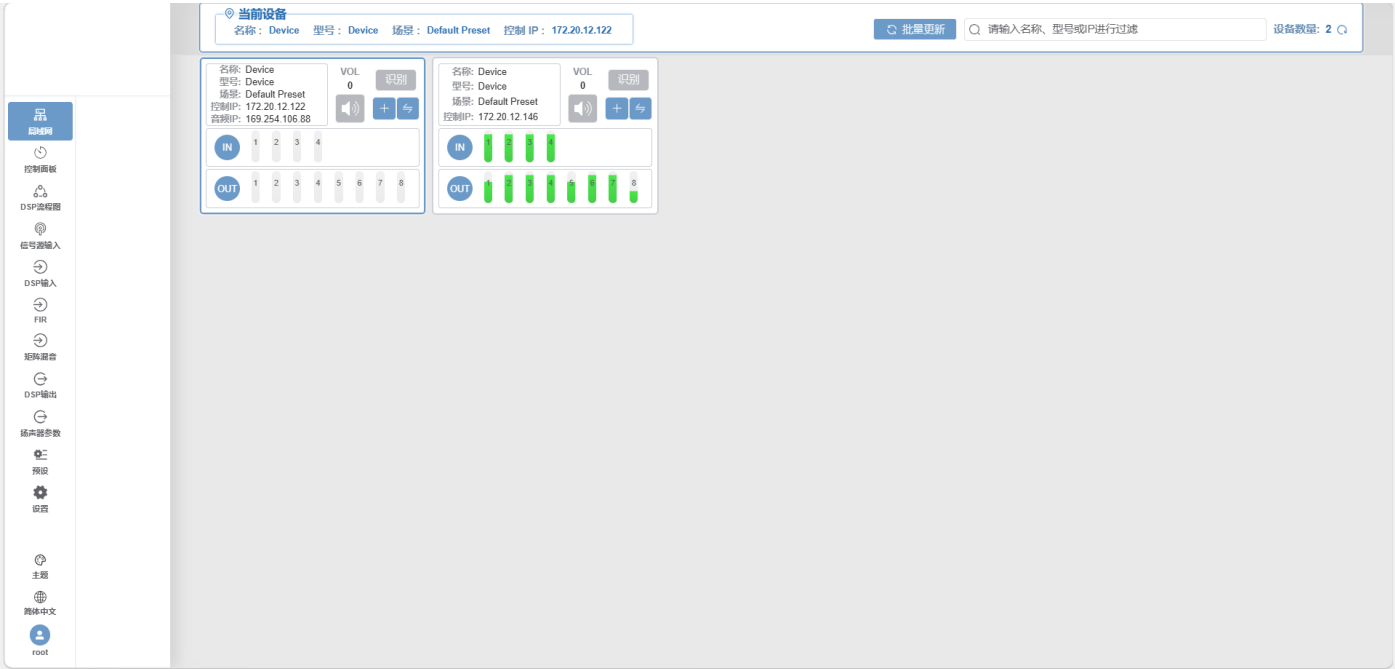


Figure 1.4

When multiple devices are on the same LAN, the WEB LAN page shows all devices. Here you can see basic info of the current device with a blue border. Batch upgrade allows simultaneous firmware upgrade of same-model devices. Entering IP in the text box filters devices. The left represents 'Add Page', the right represents 'Replace Current Page'.



6.2 Control Panel

6.2.1 Device Name, Current Preset and Master Volume



Figure 1.5

As shown in Figure 1.5, click the device name to rename the device. 名称: **DEVICE**
型号: device

Click Scene to quickly recall the desired preset. 场景: **Default Preset** ▼

Here you can adjust the master volume by sliding, mute/unmute directly, or enter a value in the text box. 总音量: 0.0 dB

6.2.2 Device Status and Local/Network Audio IP Address

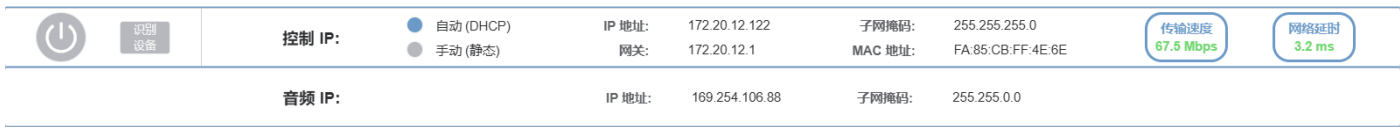


Figure 1.6

- ① As shown in Figure 1.6, click to switch between active and standby. Gray = active, blue = standby.
- ② Click and the blue and gray colors will flash alternately, making the connected device's display and status light flash, allowing you to quickly locate your device from the 'device stack'.

- ③ Figure 1.6 shows whether the IP allocation is manual (static) or automatic (DHCP), and the network audio IP address. The transmission speed and network latency on the right allow real-time monitoring of network connection signal stability.

6.2.3 Input Channel Interface

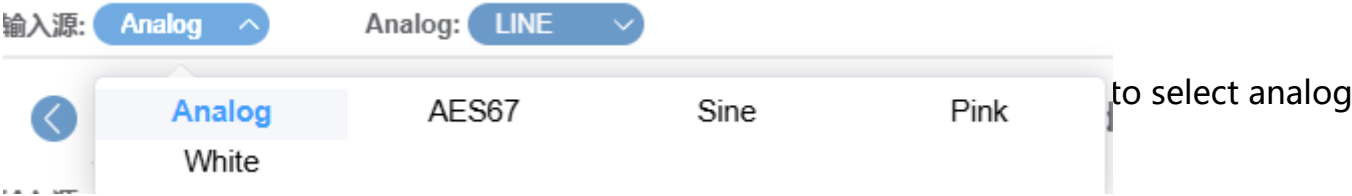


Figure 1.7

Figure 1.7 shows the input channel list for phase, mute and volume adjustment, with sensitivity, phantom power and group adjustment functions, and displays input signal names.

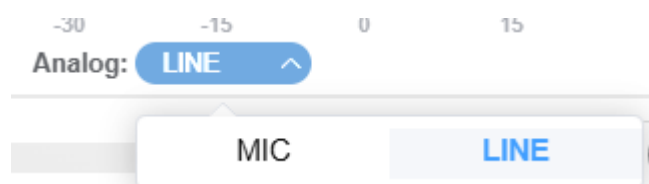
Click to modify the channel name, supports Chinese characters. **IN 1**

Click the input source



input, digital input, or test signal. One channel can only have one input mode.

Click to switch between LINE mode and MIC mode.



After selecting MIC mode for analog input



Mic input sensitivity (0~40dB), can also set 48V phantom power (OFF/ON).

Here you can perform DSP input group adjustment



click Group

Adjustment Settings to select parameters for different groups.

Here you can see the input signal level. The circle represents channel gain. Set channel volume, mute and phase. Gray = unmuted/normal phase, red = muted, blue = inverted phase. DSP input parameters are displayed. Click a module to jump to its settings. Blue = function enabled.



Note: AGC, COMP and EXP can only have one enabled at a time. Enabling COMP after AGC will disable AGC.

Here you can switch between different channel interfaces and display corresponding parameters.



6.2.4 Zone Channel Interface



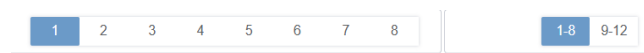
1 **ZONE1** This is Zone 1. Click to modify the name. Adjust master volume by sliding, mute/unmute, or enter value in text box.



优先级 1: OFF 优先级 2: OFF 优先级 3: OFF Priority: Smaller number = higher priority (e.g., Ch1 > Ch2 > Ch3). When multiple priority signals trigger, only the highest priority channel takes effect.



Click Group Adjustment Settings to select parameters for different groups.



There are 12 zone channels. Click to switch channels.

6.2.5 Output Channel Interface

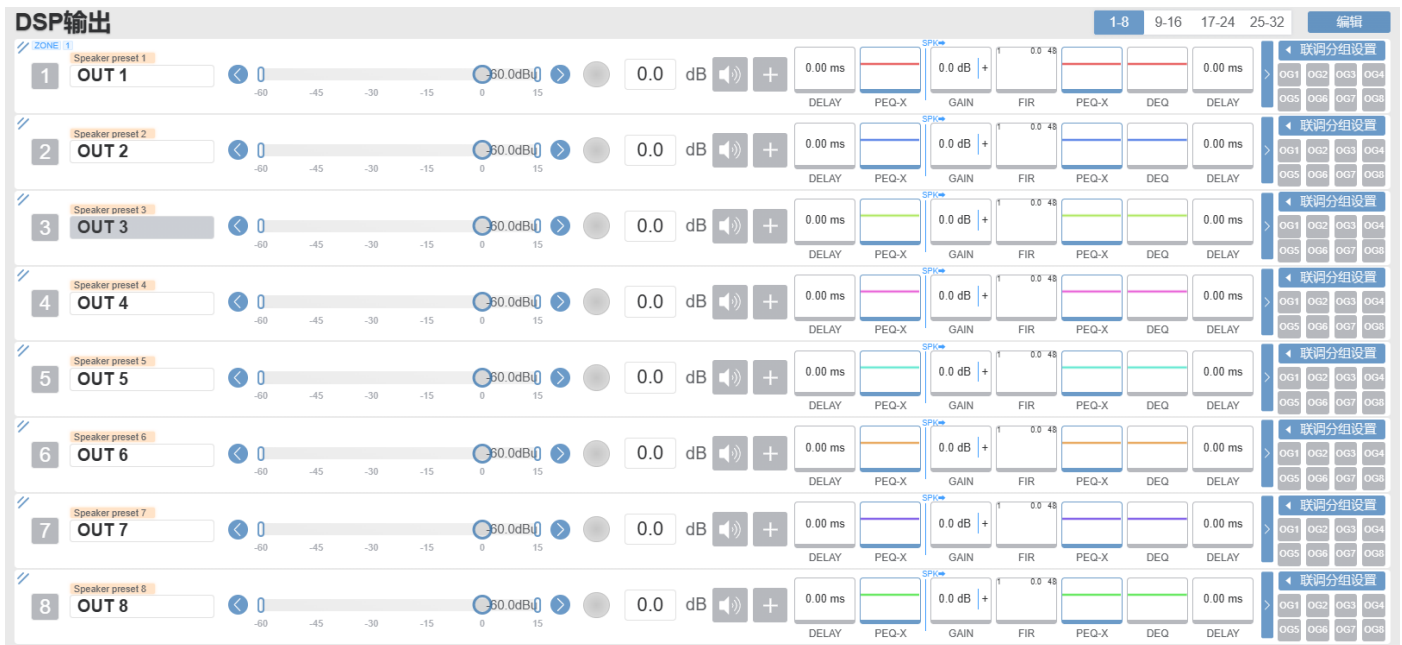


Figure 1.8

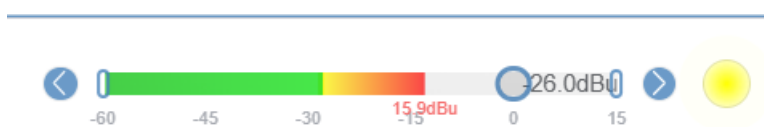
Figure 1.8 shows the output channel list for phase, mute and volume adjustment, with gain settings and DSP output group adjustment.

Click to modify the channel name, supports Chinese characters.

OUT 1

Shows the current channel's limiter threshold, compression threshold and actual output level. Yellow when limiting is active. Red on level bar = compression amount.

Yellow on level bar = limiter threshold position.



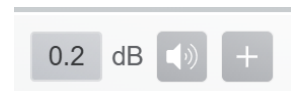
Here you can perform DSP output group adjustment. Click Group Adjustment

Settings to select parameters.

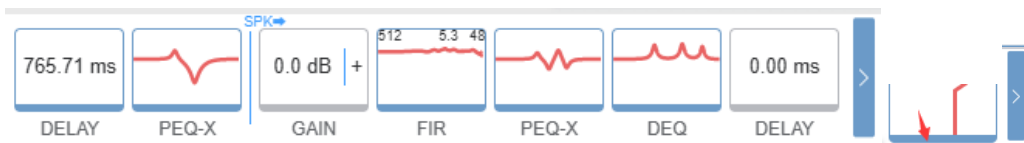


Here you can see the output signal level. The circle represents channel gain. Set DSP output channel volume, mute and phase. Gray = unmuted/normal, red = muted, blue =

inverted.



Displays DSP output parameters and speaker parameters. Click a module to jump to settings. Blue = function enabled. Click to view other speaker parameters.



6.3 DSP Flowchart



Figure 1.9



Figure 1.10

Figures 1.9 and 1.10 show the entire DSP amplifier flowchart. Click a DSP function module to jump to its specific location.



6.4 DSP Input

6.4.1 Test Signal

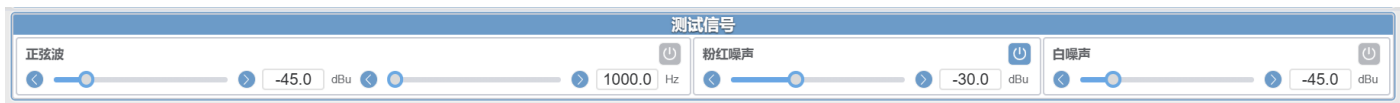


Figure 1.11

Select Sine Wave, Pink Noise or White Noise from the input source, then set parameters in test signal settings for channel testing.

All three test signals can have their gain values set. The switch controls whether active (blue = on, gray = off). Sine wave frequency can also be set.

6.4.2 Noise Gate

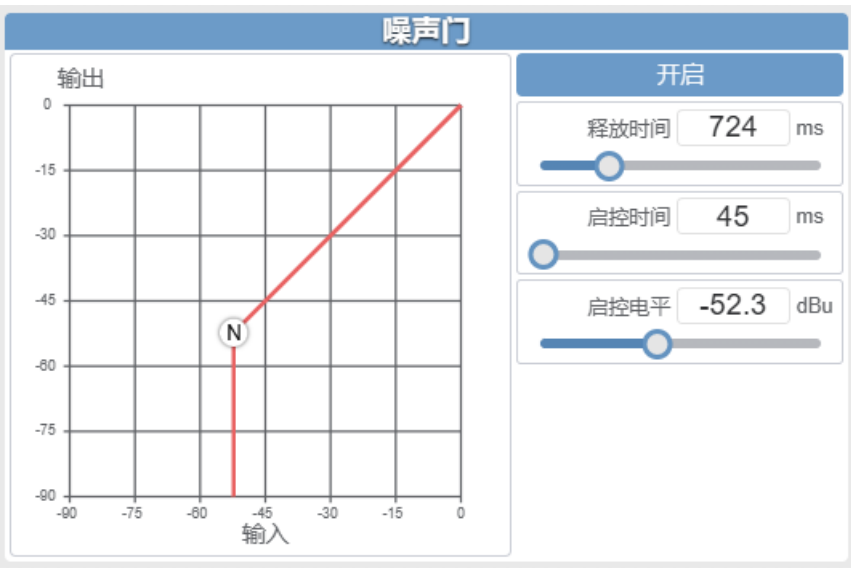


Figure 1.12

As shown in Figure 1.12, click the noise gate switch to enable/disable. Blue = on, red = off. Attack time, release time, and threshold can be adjusted by slider or numeric input.



When noise gate is enabled, the point in the left chart can be dragged diagonally to

adjust values. 

6.4.3 AGC

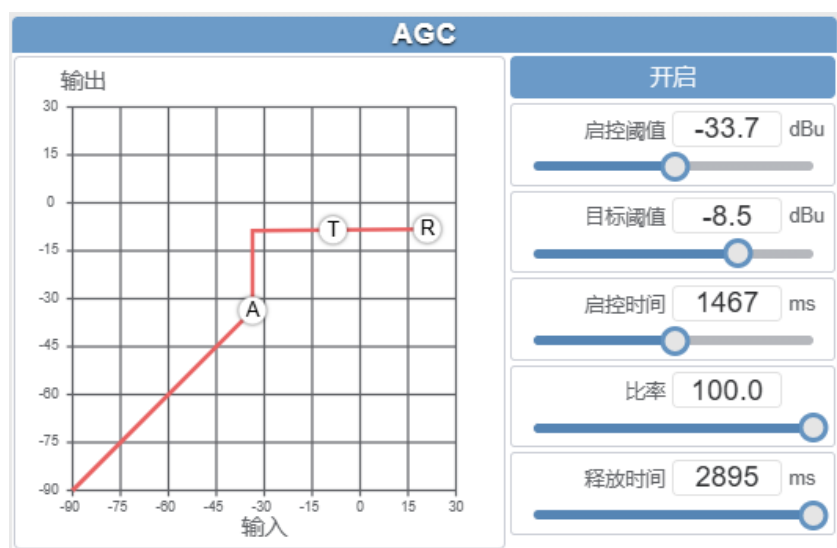


Figure 1.13

As shown in Figure 1.13, click the switch to enable/disable AGC. Blue = on, gray = off.

开启

Threshold: When input level is below this value, AGC does not intervene. Input and output change linearly 1:1. When exceeded, AGC starts gain compression.

Target Threshold: In the compression transition zone, output level rise is gradually limited and locked to an upper limit, preventing overload distortion.

Attack Time: Response time for AGC to switch from bypass to compression after signal exceeds threshold.

Ratio: The ratio of input level increment to output level increment. Higher values = stronger compression, smoother output volume.

Release Time: Time for AGC to recover from compression to normal bypass after high peak signals disappear.

6.4.4 Compressor

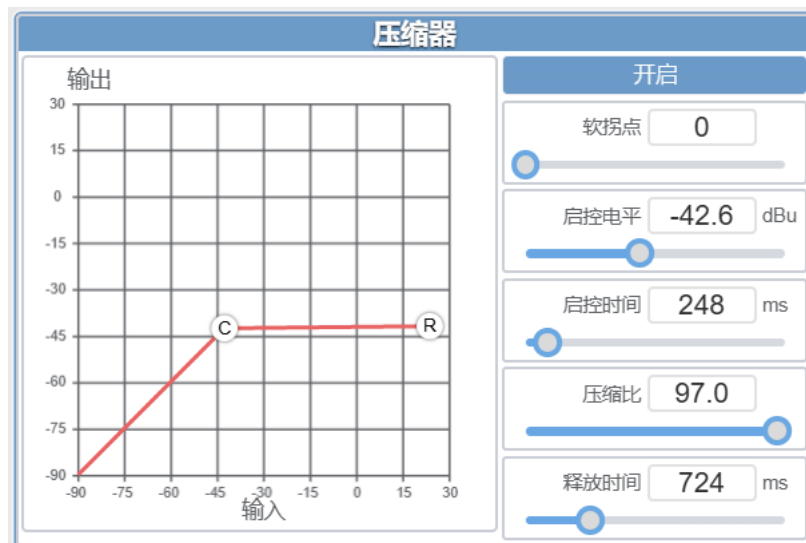


Figure 1.14

Soft Knee: Instead of immediately applying compression at threshold, there is a gradual transition zone where the ratio changes from 1:1 to the set ratio.

Threshold: When signal level exceeds this value, the compressor/limiter starts reducing gain. The more the signal exceeds threshold, the more it is attenuated.

Ratio: Determines how much overshoot is attenuated toward threshold. With 2:1 ratio, if input exceeds threshold by 2dB, output only exceeds by 1dB. 1:1 means no compression. Adjustable range: 1~20.

Attack Time, Release Time: To preserve natural attack, the initial level should pass through the compressor unaffected. Slower response prevents pumping effects. Attack time determines gain reduction speed, release time determines gain recovery speed.

Output Gain: Also called makeup gain. If the compressor significantly reduces signal level, increase output gain to maintain volume. Applied uniformly regardless of other compressor settings.

6.4.5 Expander

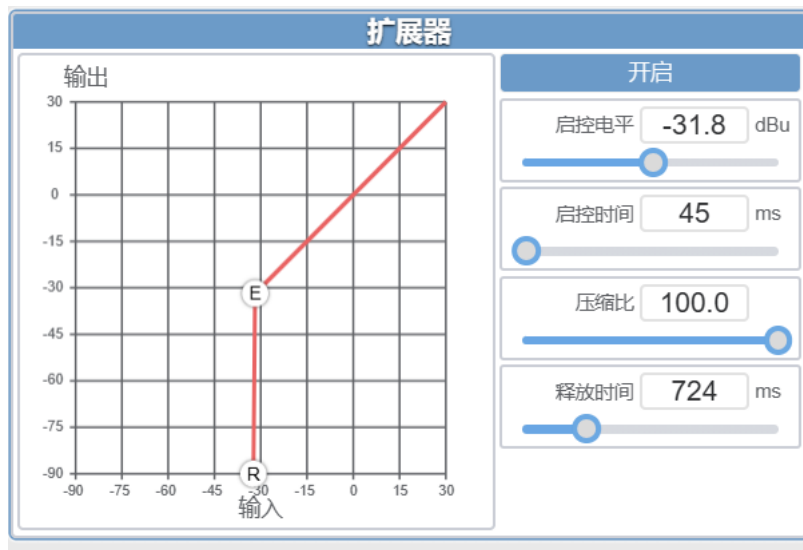


Figure 1.15

Threshold: Signal must exceed this level to open the expander. Generally set to ambient noise level.

Attack Time: Duration signal must be above threshold before expander opens. Faster attack allows quicker transient opening.

Ratio: Slope of gain curve below threshold. Higher ratio approaches gate-like behavior.

Release Time: Time for gain to recover after signal falls below threshold.

Note: AGC, COMP and EXP can only have one enabled at a time. Enabling COMP after AGC will disable AGC.

6.4.6 PEQ/High-Low Pass

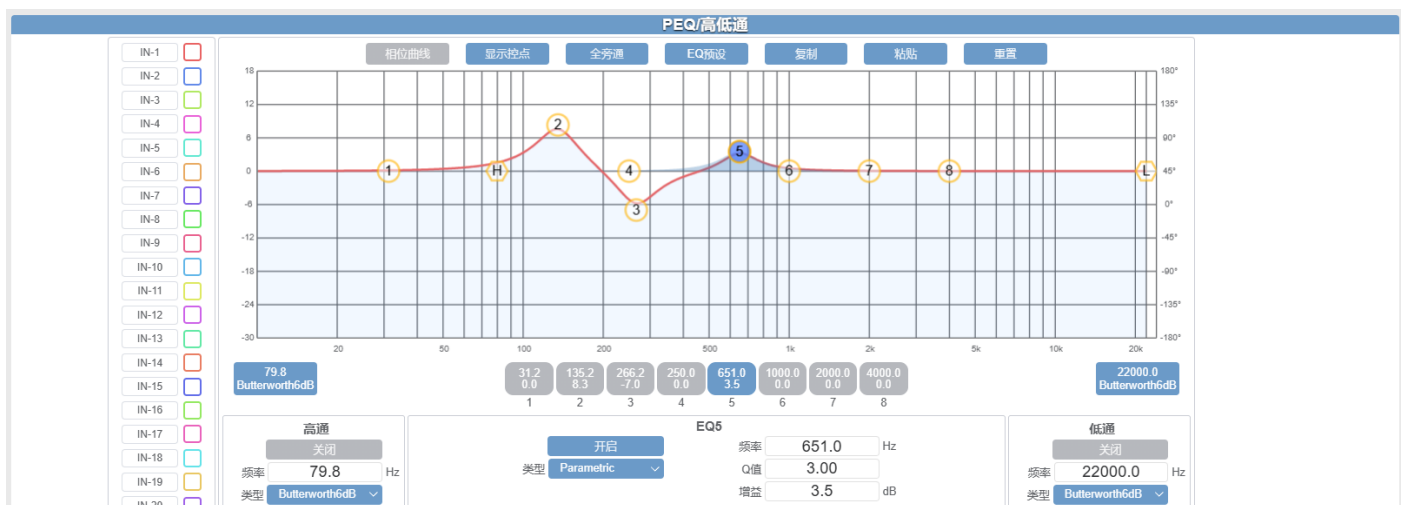


Figure 1.16

Top button functions:

Phase Curve: Displays the phase curve of the current channel.

Show Handles: Shows or hides all EQ control points.

All Bypass: Enables or disables all EQ bands simultaneously.

EQ Save: Saves current EQ settings to computer for recall.

Copy: Copies current EQ parameter values for pasting into other channels.

Paste: Used with Copy button to paste copied EQ values into current channel.

Reset: Resets EQ parameters to default values. Reset password: 8888

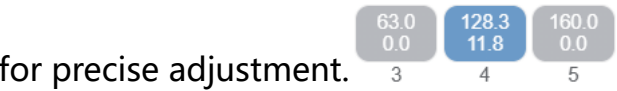
Multi-channel EQ Curve Display



As shown in Figure 1.16, the top shows the EQ curve display switch for each channel. When enabled, the corresponding curve is drawn.

EQ Controls

As shown in Figure 1.16, below the curve chart, all EQ controls and their values are displayed



EQ Parameter Adjustment Items



As shown in Figure 1.16, below EQ controls are parameter adjustment items for precise control of on/off, type, frequency, Q, gain, etc.

High/Low Pass Filter

As shown in Figure 1.16, the high-pass or low-pass sections control on/off, frequency



and type.

EQ Save Button

As shown in Figure 1.16, clicking the EQ preset button pops up Import and Export.

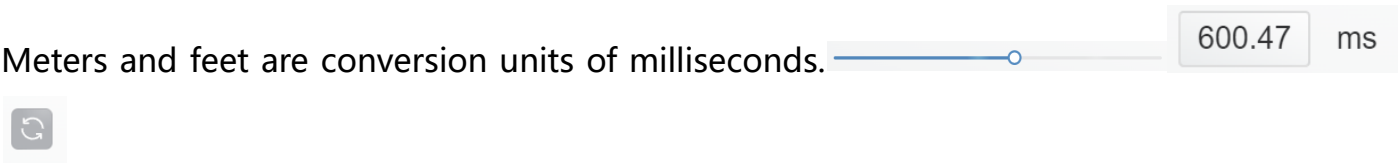


6.4.7 Input Delay



Figure 1.17

As shown in Figure 1.17, all input channel delay controls are listed. Drag the left slider or enter a value. Right side = delay switch (gray = off, blue = on). Reset button restores defaults.



6.5 Algorithms

6.5.1 AFC



Figure 1.18

AFC Algorithm: Traditional feedback suppression has 2 levels, frequency shifting at 3Hz and 6Hz.

6.5.2 ANC (AI Noise Cancellation)

ANC

名称ANC

AFC等级2

AI AFC

AI ANC

输入

IN 1: IN 1	IN 7: IN 7	IN 13: IN 13	IN 19: BT-L	IN 25: IN 25	IN 31: IN 31
IN 2: IN 2	IN 8: IN 8	IN 14: IN 14	IN 20: BT-R	IN 26: IN 26	IN 32: IN 32
IN 3: IN 3	IN 9: IN 9	IN 15: IN 15	IN 21: IN 21	IN 27: IN 27	AMX 1: AUTO MID
IN 4: IN 4	IN 10: IN 10	IN 16: IN 16	IN 22: IN 22	IN 28: IN 28	AMX 2: AUTO MID
IN 5: IN 5	IN 11: IN 11	IN 17: USB-L	IN 23: IN 23	IN 29: IN 29	
IN 6: IN 6	IN 12: IN 12	IN 18: USB-R	IN 24: IN 24	IN 30: IN 30	

输出

OUT 1: OUT 1	OUT 7: OUT 7	OUT 13: OUT 13	OUT 19: OUT 19	OUT 25: OUT 25	OUT 31: OUT 31
OUT 2: OUT 2	OUT 8: OUT 8	OUT 14: OUT 14	OUT 20: OUT 20	OUT 26: OUT 26	OUT 32: OUT 32
OUT 3: OUT 3	OUT 9: OUT 9	OUT 15: OUT 15	OUT 21: OUT 21	OUT 27: OUT 27	
OUT 4: OUT 4	OUT 10: OUT 10	OUT 16: OUT 16	OUT 22: OUT 22	OUT 28: OUT 28	
OUT 5: OUT 5	OUT 11: OUT 11	OUT 17: USB-L	OUT 23: OUT 23	OUT 29: OUT 29	
OUT 6: OUT 6	OUT 12: OUT 12	OUT 18: USB-R	OUT 24: OUT 24	OUT 30: OUT 30	

Figure 1.19

After selecting input/output channels, enable AI AFC and AI ANC for AI noise cancellation. Blue = on, gray = off. ANC frequency shift: 0 = AFC off, 1 = 3Hz, 2 = 6Hz.

AFC等级

2

0

1

2

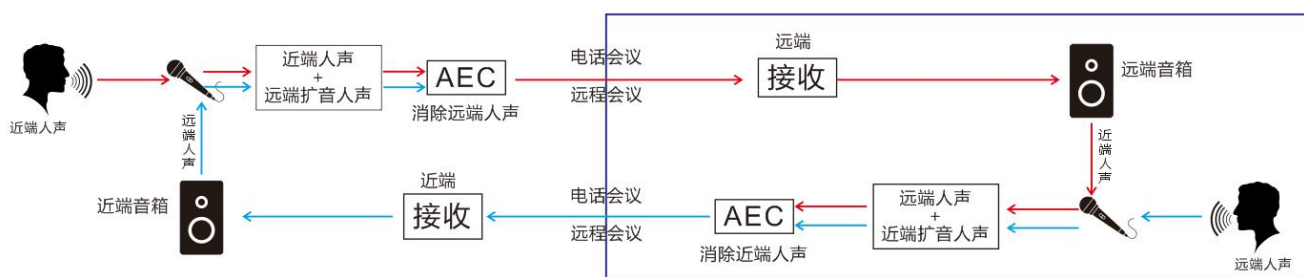
Input channels can select 1-32 microphones or AES67 input. AUTO AMX signal can be input to ANC local.

Output enters ANC processing to the corresponding channel.

AI AFC: Adaptive feedback suppression algorithm built on Kalman optimal estimation theory, providing real-time tracking and canceling of acoustic feedback loops with superior convergence speed and stability.

AI ANC: Intelligent noise reduction engine based on deep neural networks (RNN), capable of distinguishing human voice from environmental noise in real time, maintaining clear speech in complex non-stationary noise scenarios.

6.5.3 AEC (Acoustic Echo Cancellation)



AEC原理图

Figure 1.20

Figure 1.20: Near-end voice is transmitted to far end. Due to reflections, echo re-enters through the microphone. AEC processing cancels the near-end amplified voice.

AEC

名称
噪声等级
AEC

AEC
0
AEC

本地信号

IN 1: IN 1	IN 7: IN 7	IN 13: IN 13	IN 19: BT-L	IN 25: IN 25	IN 31: IN 31
IN 2: IN 2	IN 8: IN 8	IN 14: IN 14	IN 20: BT-R	IN 26: IN 26	IN 32: IN 32
IN 3: IN 3	IN 9: IN 9	IN 15: IN 15	IN 21: IN 21	IN 27: IN 27	AMX 1: AUTO MIX
IN 4: IN 4	IN 10: IN 10	IN 16: IN 16	IN 22: IN 22	IN 28: IN 28	AMX 2: AUTO MIX
IN 5: IN 5	IN 11: IN 11	IN 17: USB-L	IN 23: IN 23	IN 29: IN 29	
IN 6: IN 6	IN 12: IN 12	IN 18: USB-R	IN 24: IN 24	IN 30: IN 30	

远程信号

IN 1: IN 1	IN 7: IN 7	IN 13: IN 13	IN 19: BT-L	IN 25: IN 25	IN 31: IN 31
IN 2: IN 2	IN 8: IN 8	IN 14: IN 14	IN 20: BT-R	IN 26: IN 26	IN 32: IN 32
IN 3: IN 3	IN 9: IN 9	IN 15: IN 15	IN 21: IN 21	IN 27: IN 27	
IN 4: IN 4	IN 10: IN 10	IN 16: IN 16	IN 22: IN 22	IN 28: IN 28	
IN 5: IN 5	IN 11: IN 11	IN 17: USB-L	IN 23: IN 23	IN 29: IN 29	
IN 6: IN 6	IN 12: IN 12	IN 18: USB-R	IN 24: IN 24	IN 30: IN 30	

输出信号

OUT 1: OUT 1	OUT 7: OUT 7	OUT 13: OUT 13	OUT 19: OUT 19	OUT 25: OUT 25	OUT 31: OUT 31
OUT 2: OUT 2	OUT 8: OUT 8	OUT 14: OUT 14	OUT 20: OUT 20	OUT 26: OUT 26	OUT 32: OUT 32
OUT 3: OUT 3	OUT 9: OUT 9	OUT 15: OUT 15	OUT 21: OUT 21	OUT 27: OUT 27	
OUT 4: OUT 4	OUT 10: OUT 10	OUT 16: OUT 16	OUT 22: OUT 22	OUT 28: OUT 28	
OUT 5: OUT 5	OUT 11: OUT 11	OUT 17: USB-L	OUT 23: OUT 23	OUT 29: OUT 29	
OUT 6: OUT 6	OUT 12: OUT 12	OUT 18: USB-R	OUT 24: OUT 24	OUT 30: OUT 30	

Figure 1.21

AEC is a core module for conference audio processing. It uses adaptive filtering to eliminate acoustic echo and suppress ambient noise, ensuring bidirectional voice clarity in full-duplex communication.

As shown in Figure 1.21, AEC: gray = off, blue = on.

AUTO AMX: When enabled, auto-mix signal is input to AEC local signal.

Echo Level: 0~5, 0 = off, 5 = highest. Configure ambient noise suppression. Select the

lowest effective level to avoid voice distortion.



Local Signal: Multiple local input channels (IN1-IN32, USB-L/R, BT-L/R, AMX 1-2, etc.). Select channels needing AEC processing.

Far-end Signal: Remote input channels corresponding to local signal area, used to select far-end signal channel as echo reference.

Output: Multiple output channels (OUT1-OUT32, USB-L/R, etc.). Select AEC-processed signal output channel. Selected channel is marked blue with speaker icon.

6.5.4 AMX (Auto Mix)

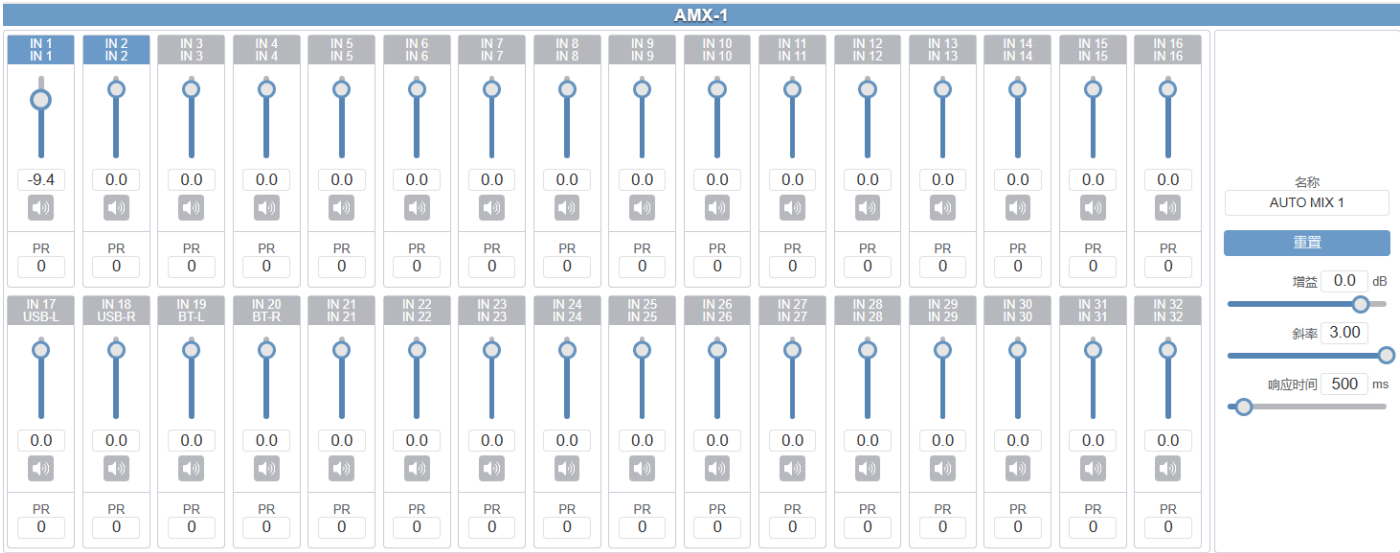
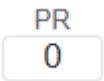


Figure 1.22

As shown in Figure 1.22, auto-mixing automatically controls gain in multi-mic scenarios, boosting speaking mics and attenuating non-speaking mics to reduce background noise and feedback risk.

1. Controls manual gain of corresponding channel. In Figure 1.22, IN1 is set to -9.4dB, others at 0dB. The fader determines base volume; auto-mix dynamically adjusts.

2.  Channel mute switch. Click to mute the channel output.

3.  Channel priority. Higher values = higher priority to seize gain when speaking, used for main speaker mic.

4.  Total output gain of auto-mix. Adjusts overall volume after mixing. 0dB = no additional amplification/attenuation.

5.  Controls the distribution intensity of auto-mix, similar to expansion ratio. Higher values = speaking mics get more gain, non-speaking mics are attenuated more. Value of 1 = all channels share gain equally. Common setting: 2~3.

6.  The time from 'speech detected' to 'gain adjusted' determines response speed. Smaller values = faster response. Larger values = smoother response. In Figure 1.22, 500ms is a medium-slow setting.

7. Reset: One-click to restore all faders, mute, and priority to default state.

8. Name (AUTO MIX 1): Modify the auto-mix scene name for saving and recalling presets (e.g., 'Conference Room Mode', 'Lecture Hall Mode').

6.6 Matrix Mixing

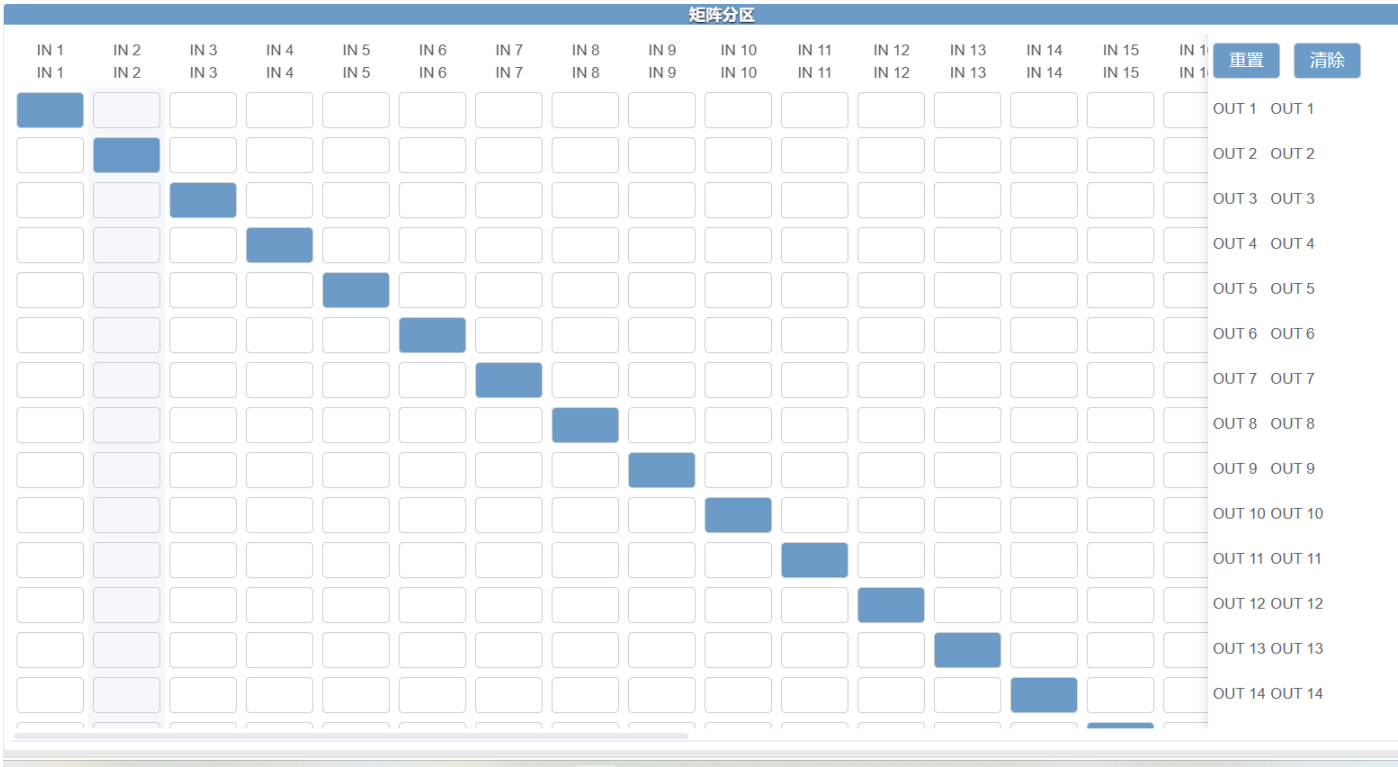


Figure 1.23

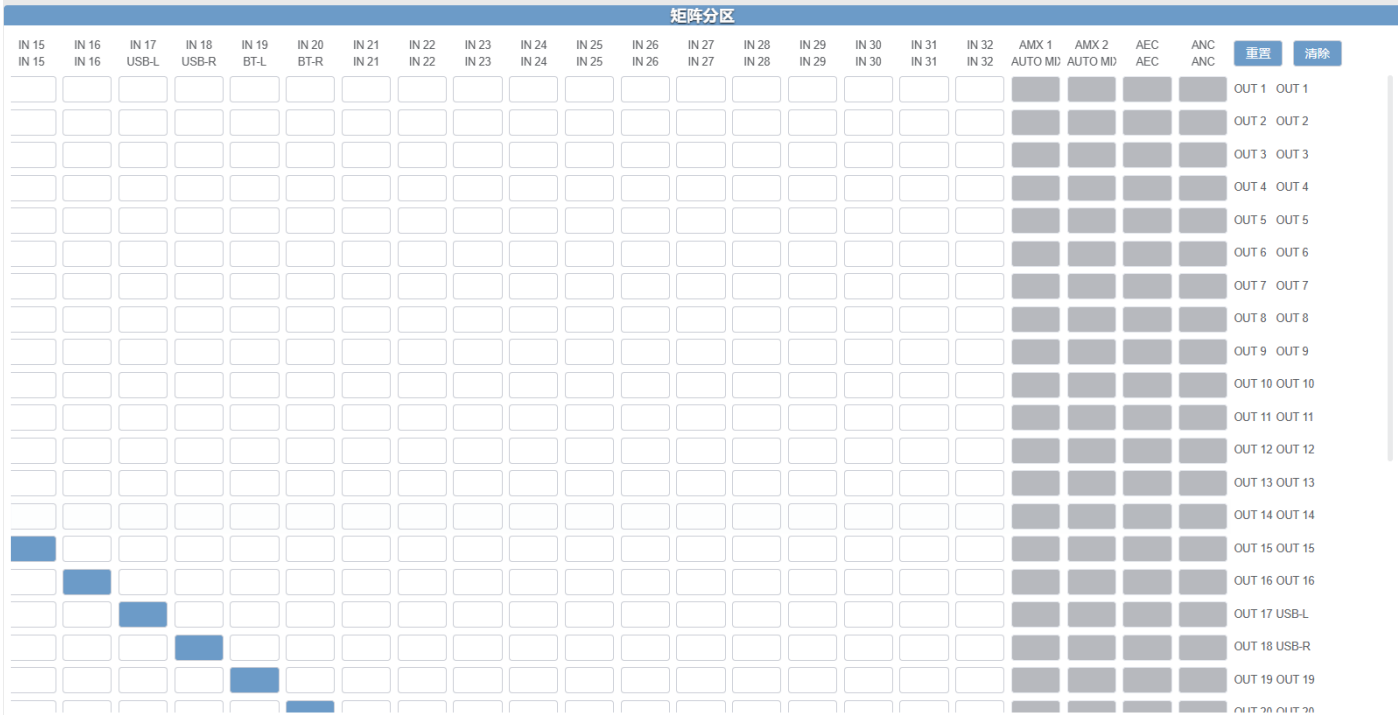


Figure 1.24

In Figures 1.23 and 1.24, the right side = output channels, top side = input channels.

When the mix button is blue (click to toggle), input and output signals are mixed.

The right side includes Reset and Clear buttons. Reset restores initial one-to-one state. Clear removes all matrix connections.



Note: When algorithms are enabled, corresponding channels cannot be manually defined and appear grayed out.

6.6.1 Zone

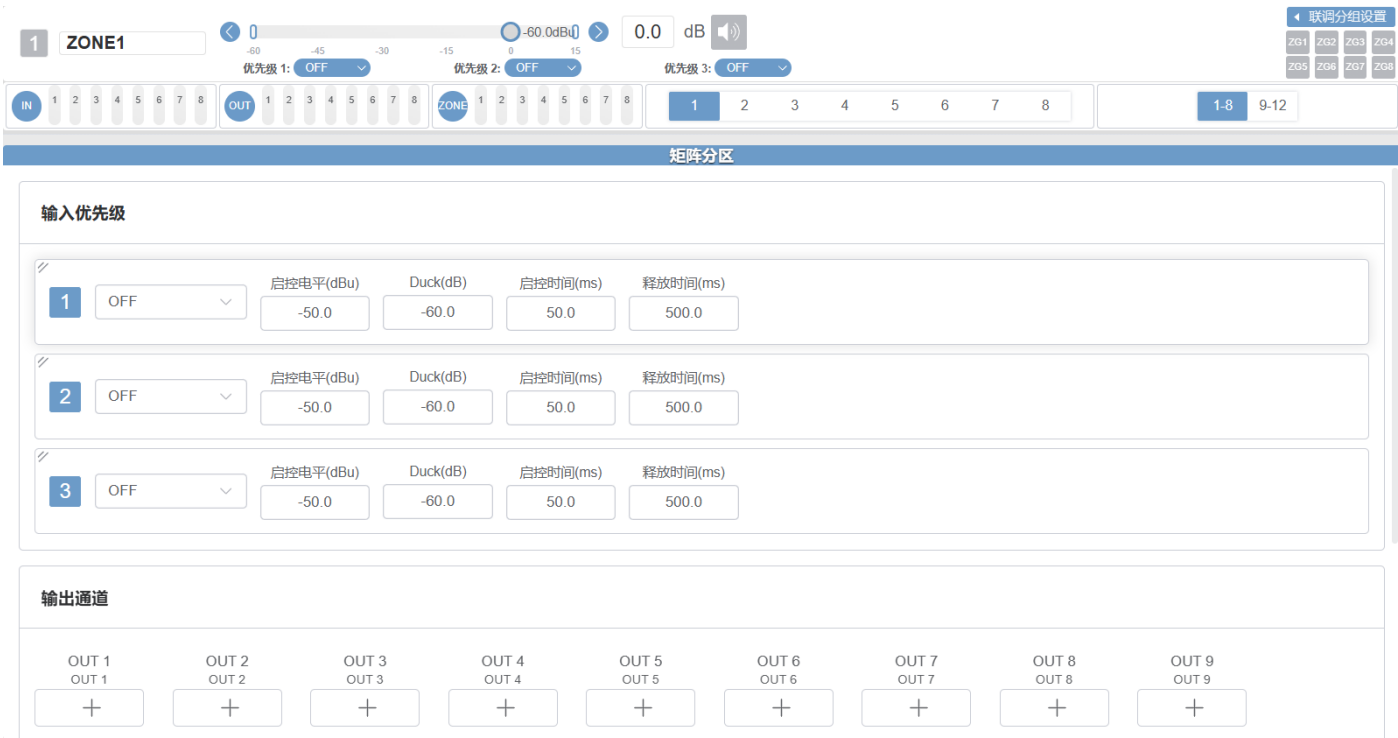


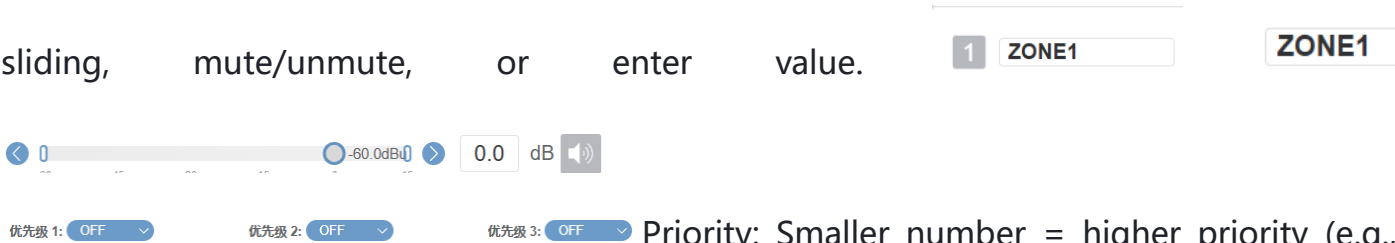
Figure 1.25

Matrix Zone is a core module for multi-input priority management and multi-output zone routing, used in broadcasting, PA, conference rooms, background music, etc.

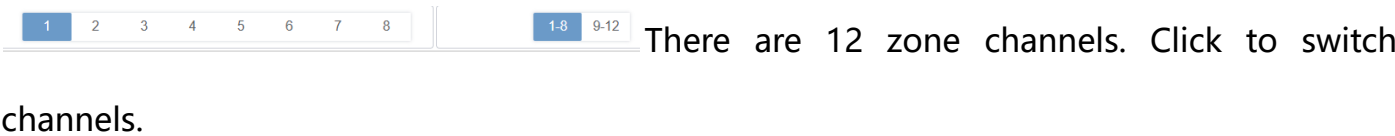
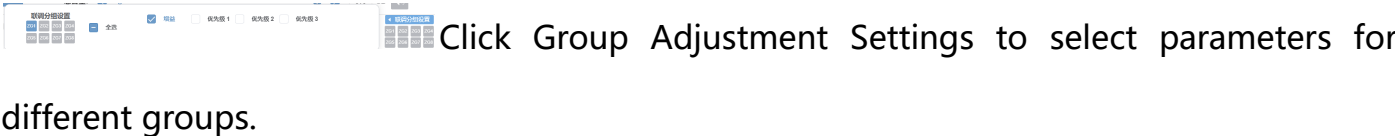
Input Priority: Implements audio ducking. When high-priority signal activates, it attenuates low-priority signals to ensure voice clarity.

Output Channel: Distributes processed audio to different output zones. Each zone independently controls source and parameters.

As shown in Figure 1.25, this is Zone 1. Click to modify name. Adjust master volume by

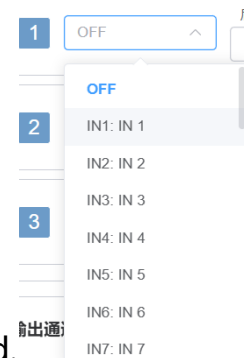


sliding, mute/unmute, or enter value. Priority: Smaller number = higher priority (e.g., Ch1 > Ch2 > Ch3). When multiple priority signals trigger, only the highest priority channel takes effect.



Input Priority Module Details

1. Priority Switch: Controls whether ducking is enabled. When OFF, priority is not used.



When selecting other input channels, priority control is enabled.

2. Priority: Smaller number = higher priority (e.g., Ch1 > Ch2 > Ch3). When multiple priority signals trigger, only the highest priority channel takes effect.

3. Threshold (dBU): Trigger threshold for ducking. When input signal exceeds this value, the system determines it as a valid signal and triggers ducking. Lower values are easier to trigger.

4. Duck (dB): Ducking depth. Smaller values = greater attenuation. -60dB nearly mutes background, suitable for emergency. -15dB suitable for daily paging.

5. Attack Time (ms): Response time for triggering ducking. Smaller values = faster response, avoiding 'background sound hasn't dropped when voice starts'.

6. Release Time (ms): Recovery time for ducking release. Larger values = smoother recovery, avoiding abrupt 'background sound suddenly bursts back'.

Output Channel Module Details

The interface displays OUT 1~OUT 32, a total of 32 output channels. Each button configures audio source routing.

1. Core function: Distribute audio signals processed by input priority to different output zones. For example:

- OUT1: Assigned to 1st floor hall speakers
- OUT2: Assigned to 2nd floor corridor speakers
- OUT3: Assigned to conference room speakers

2. Extended Control: After clicking the + button, you can further set:

Volume, mute, EQ, whether the zone responds to a certain priority level, zone switch.

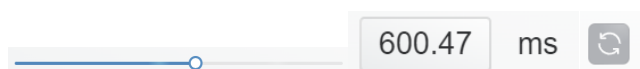
6.7 DSP Output

6.7.1 Delay



Figure 1.26

As shown in Figure 1.26, all output channel delay controls are listed. Drag slider or enter value. Right side = delay switch (gray = off, blue = on). Reset button restores defaults.



6.7.2 Output PEQ

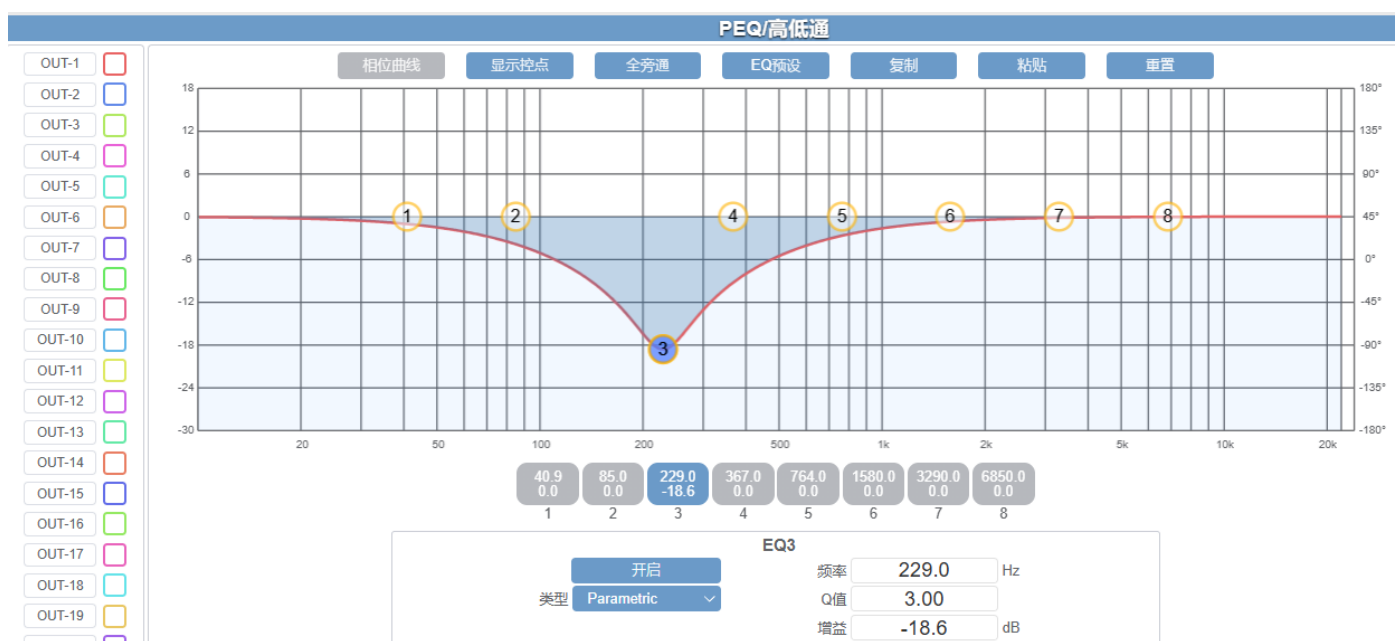


Figure 1.27

Top button functions:

Phase Curve: Displays the phase curve of the current channel.

Show Handles: Shows or hides all EQ control points.

All Bypass: Enables or disables all EQ bands simultaneously.

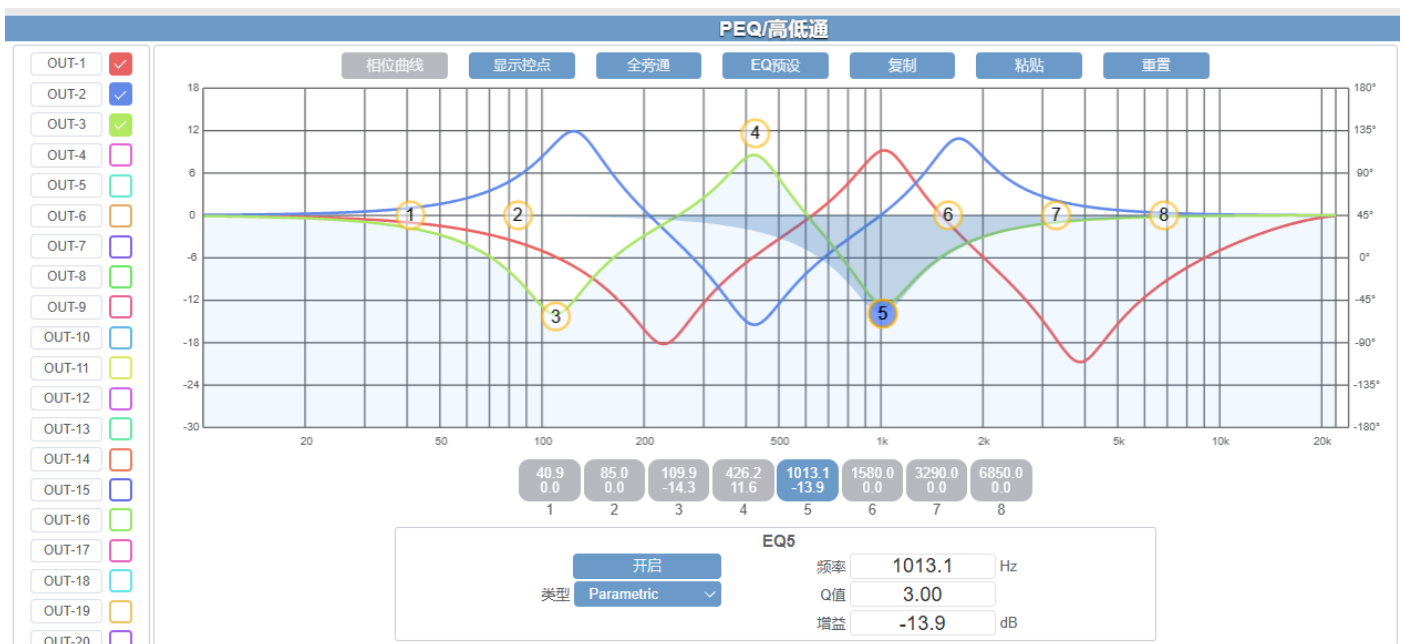
EQ Save: Saves current EQ settings to computer for recall.

Copy: Copies current EQ parameter values for pasting into other channels.

Paste: Used with Copy button to paste copied EQ values into current channel.

Reset: Resets EQ parameters to default values. Reset password: 8888

Multi-channel EQ Curve Display



As shown in Figure 1.27, the top shows the EQ curve display switch for each channel.

EQ Controls

As shown in Figure 1.27, below the curve chart, all EQ controls and values are displayed

for precise adjustment.

100.0 0.0	200.0 0.0	181.6 14.3
2	3	4

EQ Parameter Adjustment Items

EQ4

开启

类型 Parametric

频率

181.6

Hz

Q值

4.32

增益

14.3

dB

As shown in Figure 1.27, below EQ controls are parameter adjustment items for precise control.

EQ Save Button

As shown in Figure 1.27, clicking the EQ preset button pops up Import and Export.

EQ预设

导入

导出

6.8 Speaker Parameters

6.8.1 Speaker Parameters - Gain

OUT SPK 1

0.0

dB

联调分组设置

SG1

SG2

SG3

SG4

SG5

SG6

SG7

SG8

Figure 1.28

Click to modify channel name. Adjust gain by slider or text box. Set phase and mute. Speaker parameter group adjustment, same as DSP input/output group adjustment.

OUT SPK 4

-11.4

dB

联调分组设置

SG1

SG2

SG3

SG4

6.8.2 Speaker Parameters - FIR

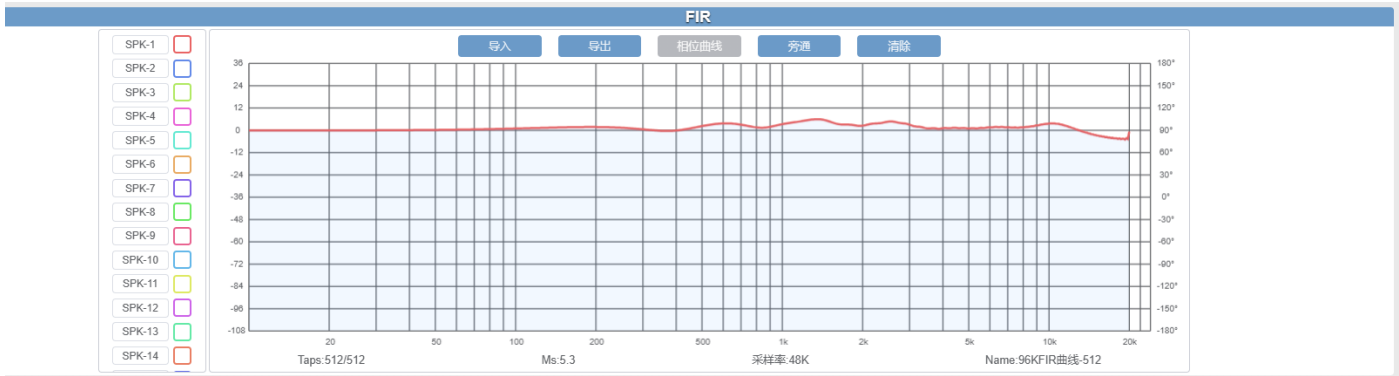


Figure 1.29

Import: Import the configured 48kHz sampling rate curve into the device

Export: Export and save the FIR curve

Phase Curve: Displays the FIR phase curve

Bypass: Default state is bypass. Turn off bypass to activate the curve effect.

Clear: Clear the FIR curve

6.8.3 Speaker Parameters - PEQ/High-Low Pass

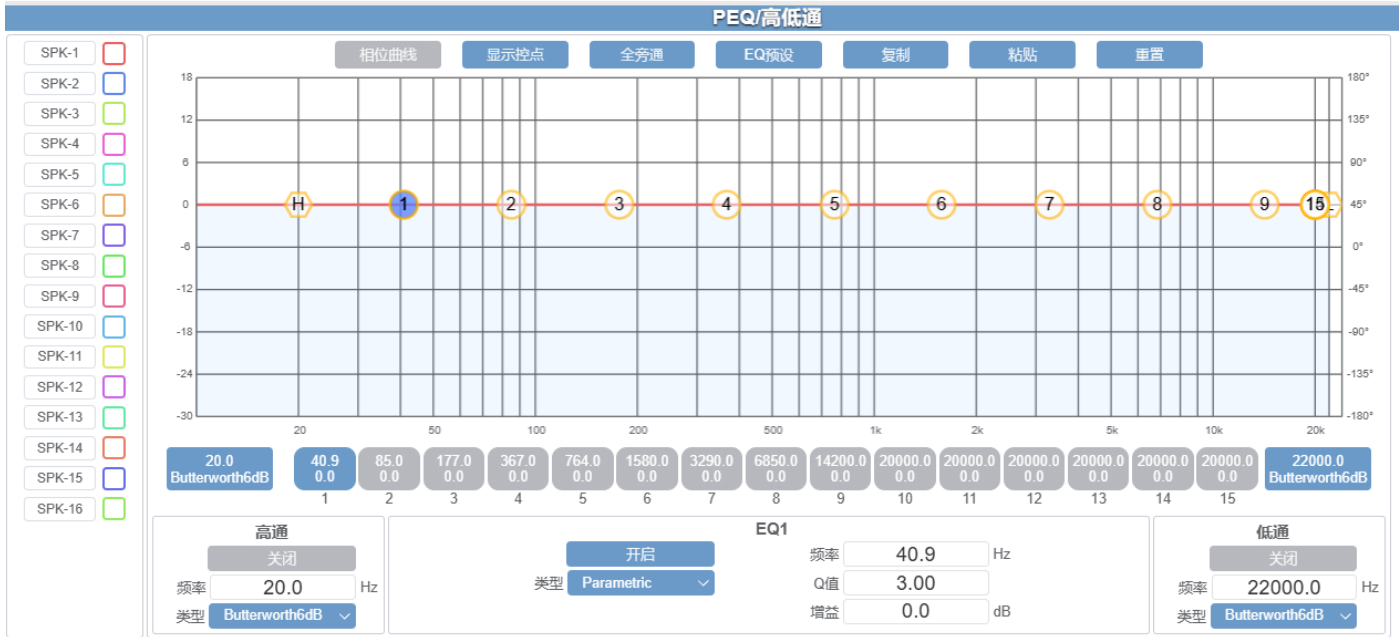


Figure 1.30

Functions and operation are the same as 6.4.6 Input PEQ.

6.8.4 DEQ (Dynamic Equalizer)

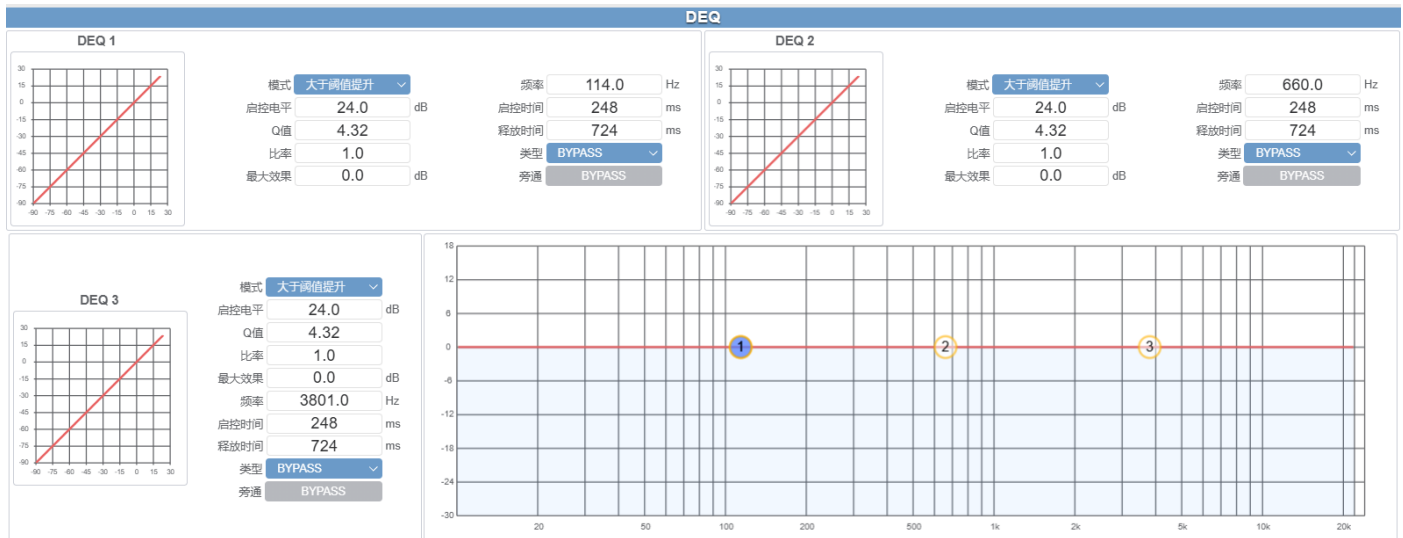
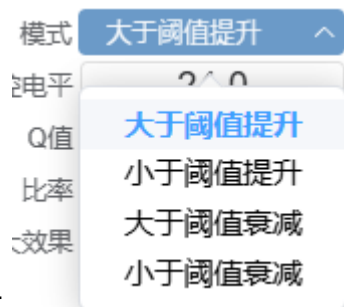


Figure 1.31

Click to switch channels. Channels 1-16 have speaker parameter libraries.



Supports 4 modes: Above Threshold Boost, Below Threshold Boost, Above Threshold Cut, Below Threshold Cut.



Below Threshold Cut.

Click to close DEQ, cancel bypass, enable DEQ. 旁通 BYPASS

Threshold: -90~24dBu. When the frequency signal reaches the threshold, the function activates.

Q Value: 0.27~15, Q value range of the variable frequency point.

Ratio: 1~100, controls dynamic adjustment proportion. Larger = more noticeable.

Max Effect: 0~12, limit value for boost or cut based on selected mode.

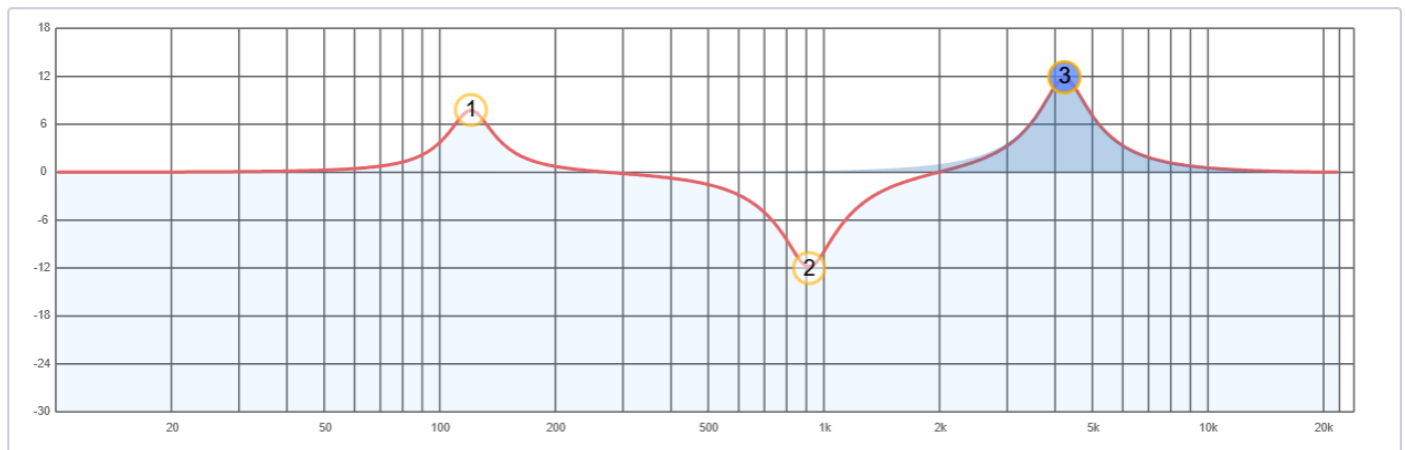
Attack Time: 1~2895, delay before boost or cut takes effect.

Frequency: Sets the center frequency point.

Release Time: Duration of the effect.

Type: Select PEQ, LSLV, HSLV as effect types. BYPASS disables the effect.

The above are all DEQ parameters. Set by modifying the numeric boxes. Parameters can be displayed in the curve.



6.8.5 Speaker Parameters - Delay



Figure 1.32

Functions and operation are the same as 6.5.7 Input Delay.

6.8.6 Speaker Parameters - Compressor

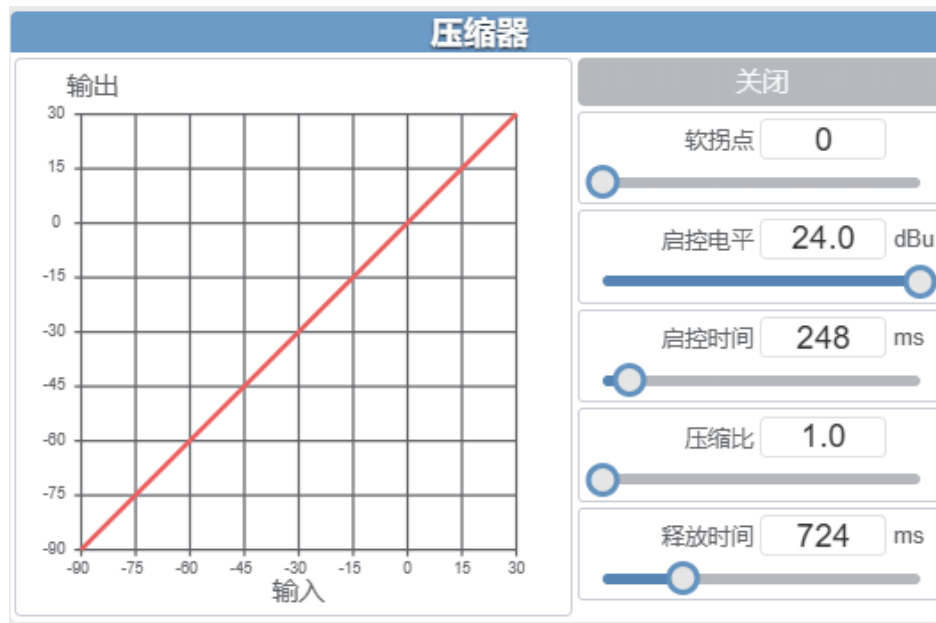


Figure 1.33

Functions and operation are the same as 6.4.4 Input Compressor.

6.8.7 Speaker Parameters - Limiter

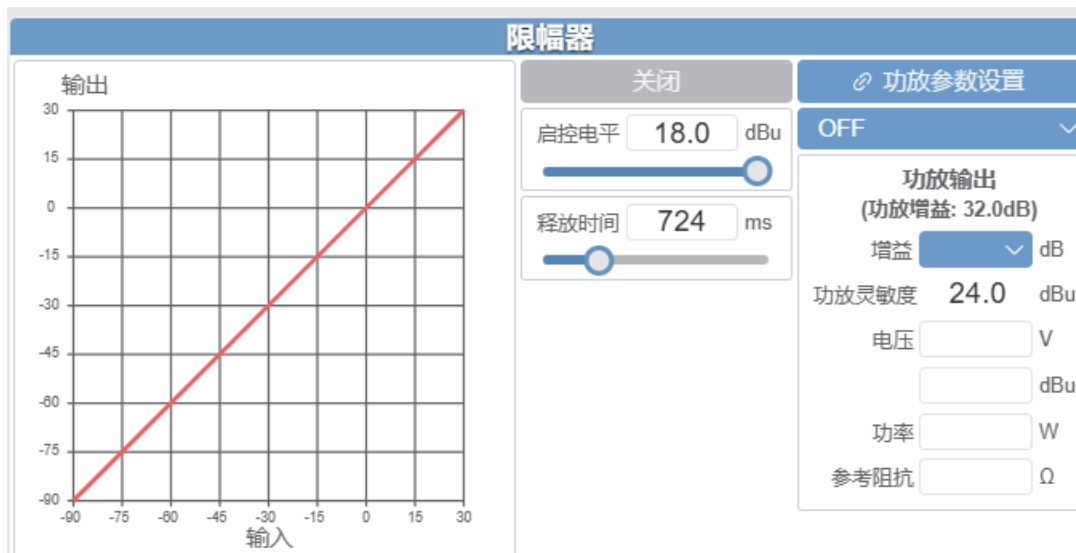


Figure 1.34

Threshold: When signal level is below threshold, the limiter does not process the signal.

When it reaches or exceeds threshold, the limiter changes signal amplitude.

Release Time: Time for gain to recover after signal falls below threshold.

Amplifier Parameter Settings: Click to jump to section 6.9.3.

Parameter Selection: Dropdown to select configured amplifier parameters.



Amplifier Output: Displays corresponding amplifier parameters. Output level changes according to selected gain range. 增益

6.9 Presets

6.9.1 Scene Presets

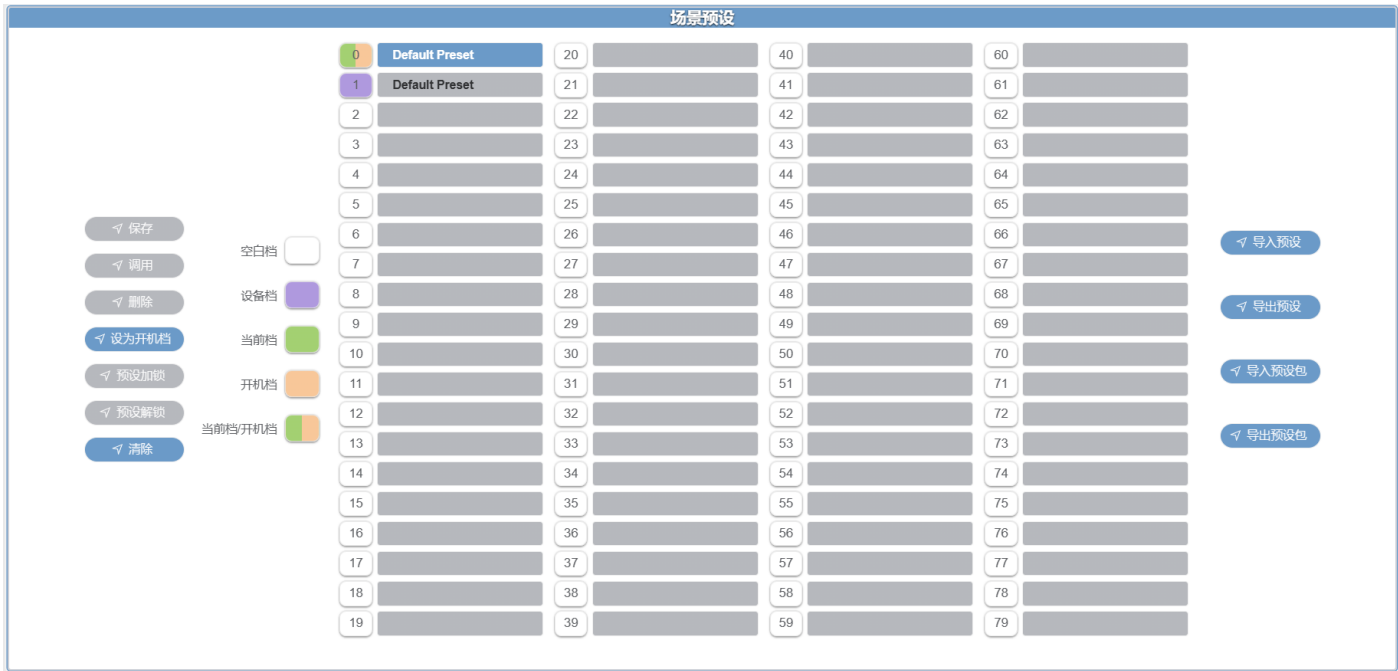


Figure 1.35

As shown above, the left side shows preset slots. '0 Auto' is a system preset and cannot be used directly. '1 (Default)' is the device's default preset, which can only be recalled, not deleted or overwritten.

The function buttons on the right side are as follows:

Save: Save current device parameters to the selected preset.

- Recall:** Load the selected preset into current device parameters.
- Delete:** Delete the selected preset parameters.
- Clear:** Clear all non-system preset parameter records.
- Set as Power-on Preset:** The device will automatically recall this preset when powered on.
- Preset Lock:** Set a password to lock the saved preset.
- Preset Unlock:** Enter the password to unlock the preset.
- Import Preset:** Import a single device parameter file from computer, overwriting existing data.
- Export Preset:** Save current device parameters to computer as a single preset file.
- Import Preset Package:** Import a package of multiple presets from computer.
- Export Preset Package:** Export all preset slot parameters to computer as a package file.

6.9.2 Speaker Presets

扬声器预设

恢复默认

全部恢复默认

导入电脑预设

联调分组设置

SG1 SG2 SG3 SG4

SG5 SG6 SG7 SG8

联调分组设置

SG1 SG2 SG3 SG4

SG5 SG6 SG7 SG8

联调分组设置

SG1 SG2 SG3 SG4

SG5 SG6 SG7 SG8

联调分组设置

SG1 SG2 SG3 SG4

SG5 SG6 SG7 SG8

联调分组设置

SG1 SG2 SG3 SG4

SG5 SG6 SG7 SG8

联调分组设置

SG1 SG2 SG3 SG4

SG5 SG6 SG7 SG8

联调分组设置

SG1 SG2 SG3 SG4

SG5 SG6 SG7 SG8

通道号

类型

通道名称

预设名称

SPK-1

OUT SPK 1

Speaker preset 1

SPK-2

OUT SPK 2

Speaker preset 2

SPK-3

OUT SPK 3

Speaker preset 3

SPK-4

OUT SPK 4

Speaker preset 4

SPK-5

OUT SPK 5

Speaker preset 5

SPK-6

OUT SPK 6

Speaker preset 6

SPK-7

OUT SPK 7

Speaker preset 7

创建预设

导入预设

预设列表

导入

导出

编辑

删除

预设

Figure 1.36

- Restore Default:** Select a channel to restore default settings.
- Restore All Defaults:** Restore all channels to default settings.
- Create Preset:** Select a channel to create, save and protect data.
- Import Preset:** Import the configured preset into the channel.
- Import:** Import multiple preset files into the preset list simultaneously.
- Export:** Export saved speaker presets. Multiple can be selected.
- Edit:** Click the preset to edit. Note: Editing a locked preset requires the lock password.
- Delete:** Delete an existing preset.

6.9.3 Amplifier Parameter Settings

功放参数设置

Q 按名称滚动

确定

导入

导出

清除

ID	开启	名称	功放增益	使用选择	电阻	电压	电流	功率	灵敏度/增益档位是否开放选择	
1	☒	default amp	25.0 (dB) 17.8 (倍)	☐ ☐ ☒	2.0 4.0 8.0	70.71/39.2 100.00/42.2 109.54/43.0	35.36 25.00 13.69	2500.0 2500.0 1500.0	☒ 26dB ☒ 32dB ☒ 38dB ☒ 44dB ☒ 27dB ☒ 33dB ☒ 39dB ☒ 28dB ☒ 34dB ☒ 40dB	

新增

Figure 1.37

This is the parameter configuration interface for professional digital amplifiers, supporting multi-channel preset management for different loads (2/4/8 Ohm).

Top Function Buttons

Search Box: Enter name and click OK to display corresponding amplifier parameters.

Import: Import amplifier parameter presets from local files (batch configuration).

Export: Export all configured parameters to a file (backup/reuse).

Clear: Reset current channel/all parameters to defaults (use with caution).

Amplifier Gain Settings

Amplifier gain is a common representation. Multiplier directly represents the amplification factor of output vs input voltage.

Load Configuration

Each row corresponds to a speaker load impedance (2/4/8 Ohm). After checking 'Use Selection', the parameters for that impedance will be enabled.

Sensitivity/Gain Levels

The 26dB~44dB options are input sensitivity levels. After checking, corresponding parameters can be selected in the limiter.

Higher dB = higher input sensitivity, more output power with same input. 44dB drives speakers more easily than 26dB but is more prone to distortion.

Usage Logic: Amplifier gain level = fixed voltage gain - input sensitivity. Higher dB = lower sensitivity, needs larger input signal for full power.

Setup Recommendations:

For regular performances, select 26~36dB to avoid noise, feedback or distortion.

For low-sensitivity speakers (line arrays, subwoofers), higher levels (38~44dB) can be used with processor limiting output level.

After debugging, keep only 1-2 commonly used levels to avoid on-site misoperation.

6.10 Settings

6.10.1 Device Information

设备信息			
设备名称: Device	软件版本: 1.1.17 固件版本: 1.1.17 硬件版本: 1.0.0	日期: 2025-12-19 日期: 2025-12-19	备份 更新 导入备份

Figure 1.38

This screen allows modifying device name and viewing program version.

控制面板

☒ 设备名称

☒ 总音量

☒ 待机状态

☒ 输入输出通道数配置

☒ 联调分组设置

预设

☒ 场景预设

☒ 扬声器预设

☒ 功放预设

设置

☒ 增益上下限设置

☒ GPIO

导出

Select info to back up, check boxes, click Export.

Import Backup imports previously exported files. Update is for firmware upgrade: import firmware, wait for 100%, then 5 seconds for device to restart.

导入备份

6.10.2 Network Information

网络设置			
IP 控制: 手动 (静态) ☐ 自动 (DHCP) ☒	IP 地址: 172.20.12.122	网关: 172.20.12.1	更新
子网掩码: 255.255.255.0	MAC 地址: FA:85:CB:FF:4E:6E		

Figure 1.39

Set device to DHCP or static IP. In DHCP mode, connect to a router that assigns IPs. If not connected, system assigns 169.254.X.X. Click 'Update' for changes to take effect.

6.10.3 Security Settings

安全设置

用户设置 刷新 默认登录用户: root

Root

个人信息

用户名root

电话

邮箱

确定

Root

账号 & 密码

账号root

旧密码请输入3-15个数字或字母

新密码请输入3-15个数字或字母

确认密码请输入3-15个数字或字母

确定

Admin

Admin 开启: Admin 界面定制: 账号密码登录:

个人信息

用户名admin

电话

邮箱

确定

Admin

账号 & 密码

账号admin

旧密码请输入3-15个数字或字母

新密码请输入3-15个数字或字母

确认密码请输入3-15个数字或字母

恢复默认 确定

User

User 开启: User 界面定制: 账号密码登录:

个人信息

用户名user

电话

User

账号 & 密码

账号user

旧密码请输入3-15个数字或字母

屏幕锁

密码: 请输入4个数字

确认密码: 请输入4个数字

取消锁 确定

验证码

旧密码: 请输入4个数字

新密码: 请输入4个数字

确定

识别码

旧密码: 请输入0-8位数字

新密码: 请输入0-8位数字

确定

Figure 1.40

Figure 1.40 shows three-level user, screen lock and verification code settings. Root has highest authority. Blue = enabled, gray = disabled. Default accounts: root/admin/user,

default password: 8888.

默认登录用户: root

账号密码登录: ☒

账号 & 密码

账号root

旧密码请输入3-15个数字或字母

新密码请输入3-15个数字或字母

确认密码请输入3-15个数字或字母

确定

个人信息

用户名root

电话

邮箱

确定

Admin 界面定制: ☒

Root

Admin

User

验证码

旧密码: 请输入4个数字

新密码: 请输入4个数字

确定

识别码

旧密码: 请输入0-8位数字

新密码: 请输入0-8位数字

确定

When the device leaves factory without a manually set identification code, the status is 'None'. Devices without identification codes can directly connect and communicate without restrictions.

1. When exporting configuration or backup files, the file includes the device's customer identification code.
2. Importing a backup file into another machine has two scenarios:
- Scenario 1: Target machine has no identification code. Import proceeds normally. Machine inherits the code from the backup file.
 - Scenario 2: Target machine has a different identification code. Import fails. Modify both codes to match before importing.

6.10.4 Gain Limit



Figure 1.41

Shows upper and lower gain limits for input/output channels and master volume.
Master volume: -60dB to 0dB. I/O channels: -60dB to 15dB.

6.10.5 Cloud Drive

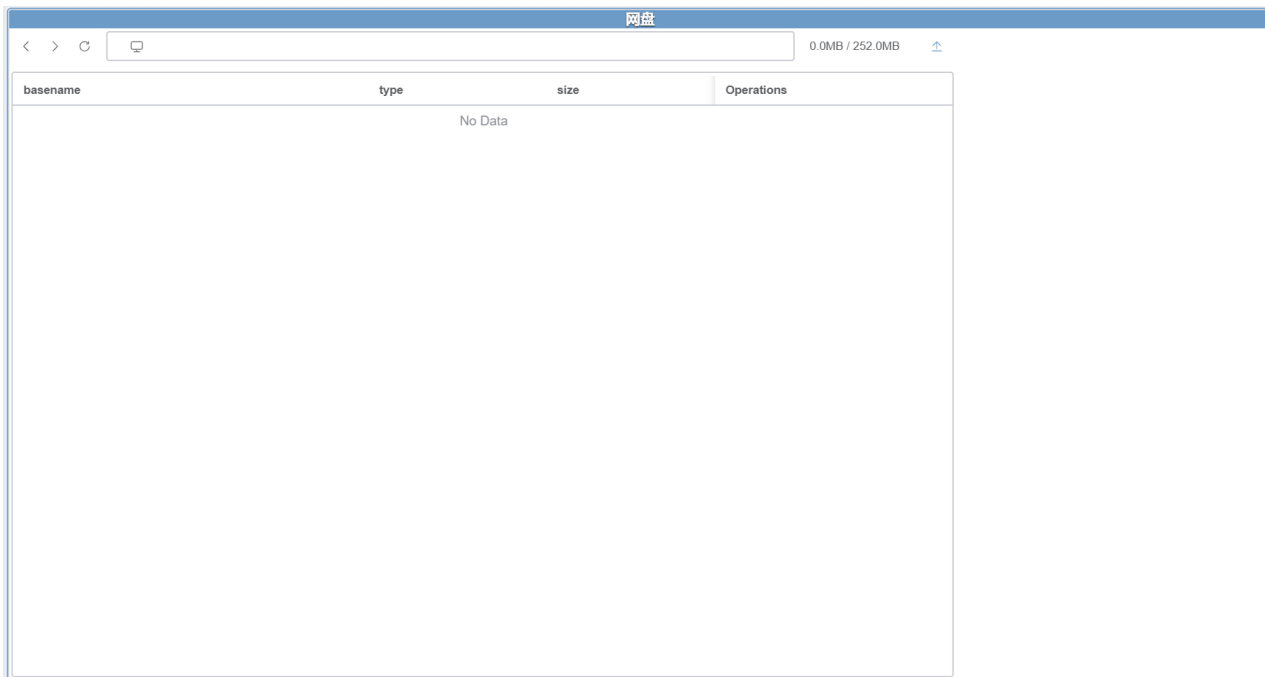


Figure 1.42

Cloud drive has 252MB memory. The virtual USB drive via USB port and WEB interface share the same storage.

6.10.6 GPIO

GPIO

1

名称

GPIO方向:

OFF

重置

保存

2

名称

GPIO方向:

OFF

重置

保存

3

名称

GPIO方向:

OFF

重置

保存

4

名称

GPIO方向:

OFF

重置

保存

5

名称

GPIO方向:

OFF

重置

保存

6

名称

GPIO方向:

OFF

重置

保存

7

名称

GPIO方向:

OFF

重置

保存

8

名称

GPIO方向:

OFF

重置

保存

Figure 1.43

The device has eight GPIO ports. Set GPIO name and port functions here. Click 'Save' after configuration.

6.10.7 Accessory Settings

已连接配件

名称	状态	型号	主机地址	IP 地址	MAC 地址	控制	操作
WCP01	在线 匹配	WCP01	192.168.31.109	192.168.31.237	9E:53:FA:A4:EE:5A	打开链接	识别 匹配 删除
WCP01	在线 匹配	WCP01	192.168.31.109	192.168.31.30	CA:4F:BD:54:2D:69	打开链接	识别 匹配 删除

局域网配件

名称	状态	型号	主机地址	IP 地址	MAC 地址	控制	操作

Figure 1.44

As shown in Figure 1.44, the device connected to a DriveControl-P wall panel displays connection status.

已连接配件						
名称	状态	型号	主机地址	IP 地址	MAC 地址	操作
No Data						

局域网配件						
名称	状态	型号	主机地址	IP 地址	MAC 地址	操作
WCP01	已占用	WCP01	192.168.31.109	192.168.31.30	CA:4F:BD:54:2D:69	识别 匹配
WCP01	未匹配	WCP01		192.168.31.237	9E:53:FA:A4:EE:5A	识别 匹配

Figure 1.45

As shown in Figure 1.45, the device displays connection status of connected and unconnected DriveControl-P wall panels on the LAN.

6.10.8 Camera Settings

摄像机设置

预置点控制

预置点

1

调用

保存

清除

串口号

RS232

摄像机地址号

1

协议

PELCO-D

云台转速

10

<

↑

↓

>

变焦放大

变焦缩小

调焦-近

调焦-远

光圈-大

光圈-小

语音跟踪

摄像机跟踪

默认麦克风

1

跟踪阈值(dBu)

-32.0

语音间隔(s)

0.1

轮换时间

0.0

轮换周期

0.0

麦克跟踪设置

摄像机跟踪

禁用

麦克风编号

1

优先级

1

设置通道

In1

串口号

RS232

摄像机地址号

1

协议

PELCO-D

预置点号

1

应用

Figure 1.46

Camera settings is a camera debugging interface. Adjust camera position before tracking starts. Parameters will be saved to the camera.

55 / 59

First configure serial port (232, 485). Set camera address and protocol type. Preset point number is user-defined. Adjust up/down/left/right, focus, aperture. Click 'Save' to save to camera.

Note: One camera address can contain multiple preset points, but one preset point corresponds to only one camera address.

1. Preset Point Control Module (Upper Left)

Function: Save multiple fixed monitoring positions for PTZ camera with one-click recall.

Preset Point Dropdown: Select from 1-32 numbered positions.

Recall: Select a preset point number and click. PTZ automatically rotates to the corresponding angle.

Save: Manually rotate camera to desired position, select preset number, click Save.

Clear: Select a preset point number and click to delete stored position information.

2. PTZ Communication Serial Port Parameters (Upper Middle)

Configure PTZ 485/232 control communication parameters. Check here first if PTZ cannot rotate.

Serial Port: Hardware serial port controlling PTZ. Default RS232. Switch based on wiring.

Camera Address: PTZ device address code. Must be unique on same bus. Default: 1.

Protocol: PELCO-D, PELCO-P, VISCA, or custom. Mismatched protocol = no PTZ response.

PTZ Speed: Rotation speed. Higher = faster. Range: 1~100. Default: 10.

3. PTZ Manual Control Area (Upper Right)

Real-time manual control of camera PTZ and lens for adjusting image position.

Direction Control: Up/down/left/right arrow buttons. Press and hold for continuous rotation.

Lens Adjustment Buttons:

Zoom In / Zoom Out: Controls lens zoom.

Focus Near / Focus Far: Manual focus. Adjust when image is blurry.

Aperture Open / Close: Adjust light intake. Close in bright conditions, open in dark.

4. Voice Tracking - Camera Tracking Parameters (Lower Left)

Basic parameters for microphone-linked PTZ tracking. Sound triggers PTZ to follow sound source.

Default Mic: Select microphone number. Default: 1.

Tracking Threshold (dBu): Sound source trigger threshold (-32.0). Higher = more sensitive.

Voice Interval (s): Delay duration for continued tracking after sound detected. 0.1s = stop 0.1s after sound disappears.

Queue Time / Queue Cycle: Multi-mic tracking switching wait time. 0.0 = cycling disabled.

5. Microphone Tracking Settings (Lower Right)

Complete configuration of mic-linked PTZ tracking. Current: disabled. When enabled, sound controls PTZ.

Camera Tracking: Enable/Disable. Gray = disabled, blue = enabled.

Microphone Number: Bind pickup mic number. Default: 1.

Priority: Higher number = higher priority. Preferentially follows high-priority mic source.

Right-side Communication Parameters (must match PTZ serial port parameters above):

Serial Port: RS232/RS485, must match PTZ wiring.

Camera Address: PTZ address code, must match PTZ address above.

Protocol: PELCO-D, PELCO-P, VISCA, or custom. Must be consistent.

Preset Point: PTZ auto-return position when no sound source. Default: 1.

6. Configuration Save Operation

After modifying all parameters, click Apply in the lower right corner. Switching pages without Apply will lose changes.

Example Operation Procedures

Scenario 1: Manually Setting Monitoring Preset Points

Use upper right direction and zoom buttons to adjust camera to desired view.

Select preset point number from upper left dropdown (e.g., 1).

Click Save to store the position.

To recall, select the corresponding number and click Recall.

Scenario 2: Enabling Automatic Sound Tracking PTZ

Change Camera Tracking option to Enable.

Verify serial port, address, and protocol match PTZ parameters exactly.

Adjust tracking threshold in lower left to match on-site noise.

Click Apply in lower right corner to save.

When mic detects human voice, PTZ automatically turns toward sound source.

6.10.9 Port Settings



Figure 1.46

Configure baud rates for two independent serial ports: RS232 and RS485. Parameters

can be set independently.

Baud rate options: 1200, 2400, 4800, 9600, 19200, 38400, 57600, 115200, 230400, 460800.

Default: 115200.

