Dynamic Audio Solutions

User’s Manual

P Series

AUDIO PROCESSOR

Revision A0-2025-01

Important Precautions

This symbol is used to alert the operator to follow
important operating and precautions detailed in documentation.

This symbol is used to warn operators that uninsulated“dangerous voltages” are present within the equipment that may pose a risk of electric shock.

1.Save the carton and packing materials even if the equipment is arrived in good condition. Should you ever need to ship the device (back to the factory), you can only use the original manufacturer’s packaging.

2.Read all documentation before operating your equipment. Retain all documentation for future reference.

3.Follow all instructions printed on unit chassis for proper operation.

4.Do not spill water or other liquids into or on the unit, or operate the unit while standing in liquid.

5.Make sure power outlets conform to the power requirements listed on the back of the unit.

6.Do not use the unit if the electrical power cord is frayed or broken. The power supply cords should be routed so that they are not likely to be walked on or pinched by items placed upon or against them, paying particular attention to cords and plugs, convenience receptacles, and the point where they exit from the appliance.

7.Always operate the unit with the AC ground wire connected to the electrical system ground. Precautions should be taken so that the means of grounding of a piece of equipment is not defeated.

8.Mains voltage must be correct and same as that printed on the rear of the unit. Damage caused by connection to improper AC voltage is not covered by any warranty.

9.Have gain controls on processors turned down during power-up to prevent amplifier or speaker damage if there are high signal levels at the inputs.

10.Power down and disconnect units from mains voltage beforemaking connections.

11.Never hold a power switch in the “ON” position if it won’t staythere itself.

12.Do not use the unit near stoves,heat registers,radiators,or other heat producing devices.

13.Do not block fan intake or exhaust ports. Do not operate equipment on a surface or in an environment which may impede the normal flow of air around the unit, such as a bed, rug, weather sheet, carpet, or completely enclosed rack. If the unit is used in an extremely dusty or smoky environment, the unit should be periodically “blown free” of foreign matter.

14.Do not remove the cover. Removing the cover will expose you to potentially dangerous voltages. There are no user serviceable parts inside.

15.Do not connect the inputs / outputs of processor or consoles to any other voltage source, such as a battery, mainssource, or power supply, regardless of whether the processor or console is turned on or off.

16.Non-use periods: The power cord of equipment should beunplugged from the outlet when left unused for a long period of time.

17.Service information: Equipment should be serviced by qualified service personnel when:

A.The power supply cord or the plug has been damaged;

B.Objects have fallen,or liquid has been spilled into the equipment;

C.The equipment has been exposed to rain;

D.The equipment does not appear to operate normally,or exhibits amarked change in performance;

E.The equipment has been dropped,or the enclosure damaged.

18.To obtain service, contact your nearest AUDIOCENTER service centre, distributor or dealer.

Table of Contents

◆ Introduction

The P48 Digital Audio Processor is a professional processor capable of implementing a variety of DSP functionalities. The P48 integrates features such as compressors, limiters, crossovers, dynamic equalizers, delays, equalizers, input/output FIR filters, and mixing matrix. Users can quickly calibrate and monitor speakers using an exquisite and intuitive PC software, providing a broad operational space for the construction and operation of professional sound reinforcement systems.

◆ Application

- Performance halls
- Live Shows
- Multi-function Halls
- Stadiums
- Auditoriums

◆ Technical Features

1. Designed with high-performance AD/DA converter chips, the 24-bit 96kHz sampling rate delivers exceptional audio quality and performance.
2. 4 analogue inputs and 8 analogue outputs.
3.Standard configuration includes Finite Impulse Response (FIR) filters, with 4x512 taps on input and 8x512 taps on output.
4. Each input channel is equipped with a three-band dynamic equalizer (DEQ).
5. Both input and output are equipped with classic Butterworth, Bessel, and Linkwitz crossover filters, featuring a steep slope of 48dB per octave.
6. Input and output delays are both 2000ms, with a precision of 0.01ms.
7. Each output channel is configured with dual protection comprising a compressor and a limiter.
8.The front panel control allows for immediate muting of any input or output channel, as well as the editing of all commonly used parameters and recall of presets.
9. Supports multiple connection control methods, with USB plug-and-play automatic software connection, TCP/IP support for centralized management and debugging of multiple devices, and RS232 interface for control system connections.

◆ Unpacking

Please inspect the audio processor carefully immediately after unpacking. If you find any damage, notify your supplier /dealer immediately. Only the shipper may file a damage claim with the carrier for damage incurred during shipping. Be sure to save the carton and all packing materials for the carrier's inspection. If you ever need to ship the unit back to AUDIOCENTER or an authorized service center, you should use only the original manufacturer's packaging.

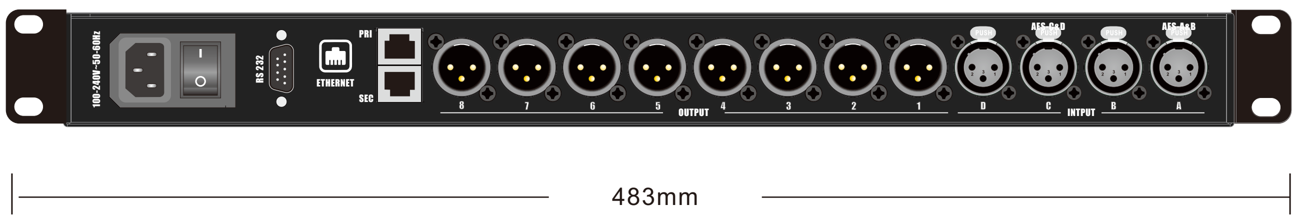
◆ Installation

P series is one-rack-space high. Four front-panel mounting holes are provided on each Audio Processor.
All mount in standard 19-inch racks.

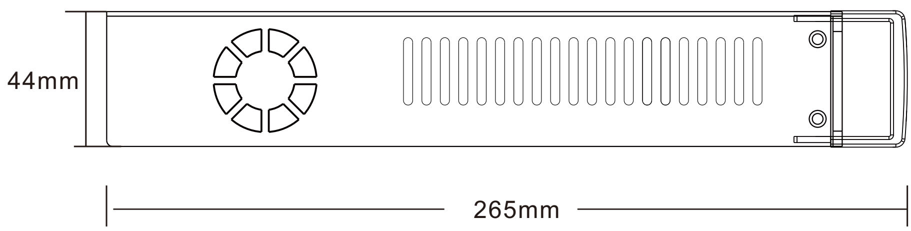
◆ Front Panel

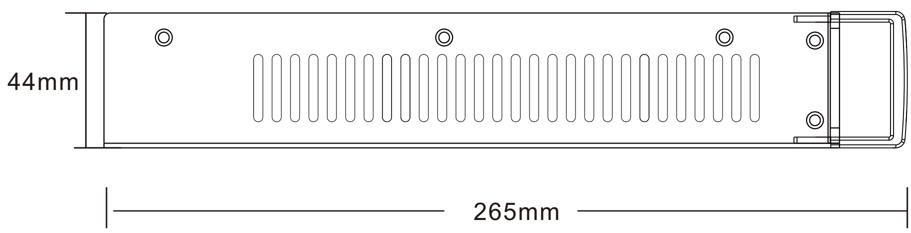
◆ Rear & Side Panel

Rear Panel

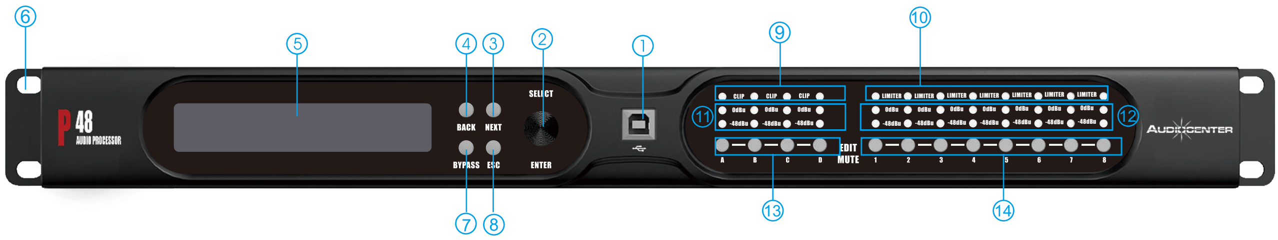


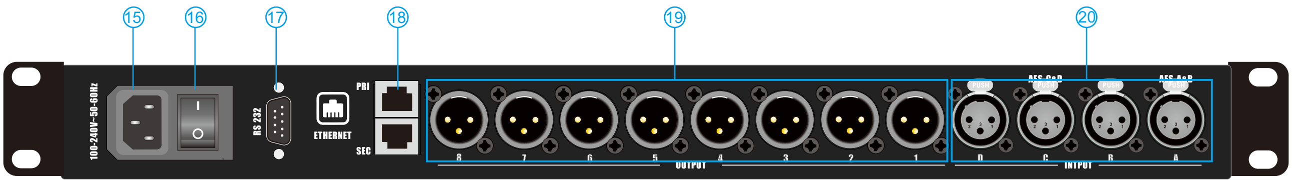
Side Panel





◆ Introduction of Front & Rear Panel Function





1.USB Port
The device can be connected to a computer control software via the USB port.

2.Select/Confirm Button
The rotary knob is used for browsing and modifying parameters. Pressing the knob is for confirming selections.

3.NEXT Button
Browse the menu forwards

4.BACK Button
Browse the menu backwards

5.Display Screen
A 2402-character display screen, showing the settings of the processor

6.Mounting Holes
There are two front panel mounting holes on each side of the device.

7.BYPASS Button
Bypass button: In some DSP function debugging menus, pressing this button can bypass this function (for example, in the delay function menu, pressing this button can turn on/off the delay of the channel).
Preset adjustment shortcut key: In the main interface state, pressing this button can directly enter the preset calling interface.

8. ESC Button
The return button

9.Clip Indicator (Input)
When the input signal exceeds the maximum input level of the processor and clipping distortion occurs, this light will light up. At this time, the input signal should be reduced.

10.Clip Indicator (Output)
When the output signal reaches the set clipping value, this light will light up. At this time, the input signal should be reduced.

11.Input Signal Indicator
When this light is on (blue), it indicates that there is a signal input to the channel.

12.Output Signal Indicator
When this light is on (blue), it indicates that there is a signal output from the channel.

13.EDIT/MUTE Button (Input)
Pressing briefly mutes the channel; holding for 2 seconds to put the channel into edit mode.

14.EDIT/MUTE Button (Output)
Pressing briefly mutes the channel; holding for 2 seconds to put the channel into edit mode.

15.Power Socket
Standard IEC plug connection

16.Power Switch
When not in use, please keep the power turned off.

17.RS232 interface
This port can be used to connect to a computer control software, or to connect to a central control device.

18.ETHERNET Interface (RJ 45 Dual Network Ports)
TCP/IP connection interface: This port can be used to connect to a computer control software.
Dante network audio transmission interface: For processors with Dante functionality (P48D), this port can be used for Dante network audio transmission.

19.Signal output
8-channel analog output, male XLR connectors, balanced output

20.Signal output
Analog input, female XLR connectors, balanced input
AES input, CHA/C channel input

Software operation instructions

The equipment management software “BraincoreNet” is designed for users to quickly interact with the parameters of one or multiple machines. It enables the storage of machine configuration parameters to disk files, providing a very convenient means for preset scene configuration and parameter switching and restoration for multiple machines or different usage venues.
The software can be downloaded from the official website of Audiocenter at https://audiocenter.com/software/.

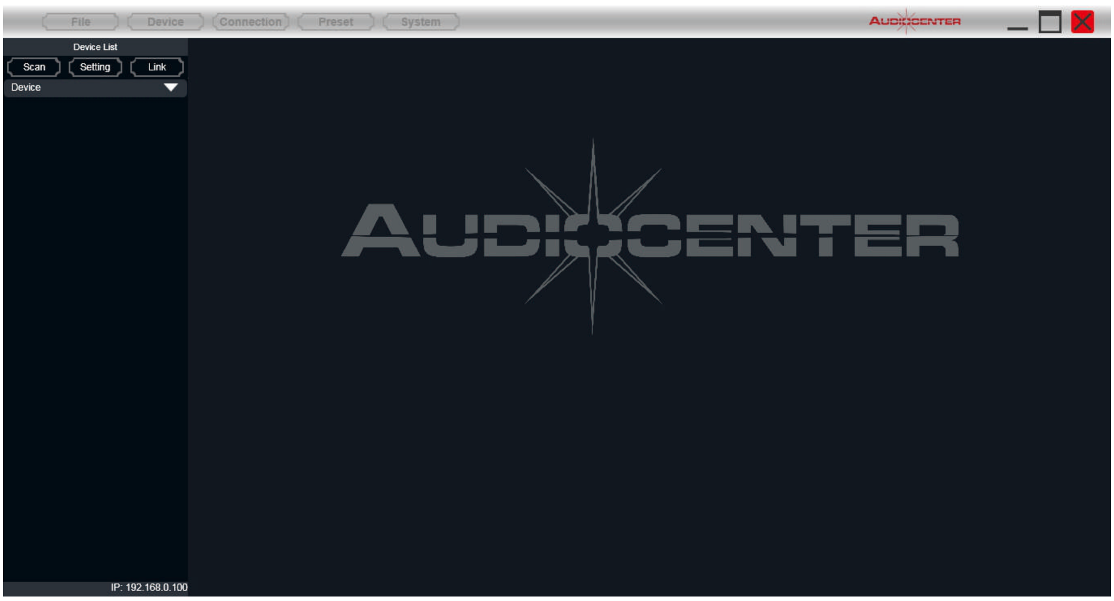
Operating environment

The software is compatible with any Windows operating system with x86/x64 architecture running WIN7/WIN8/WIN10 and equipped with the Microsoft.NET Framework 4.0 runtime library.

Software operation

Operating Steps:

Double-click the executable file to enter the main interface of the software, as shown in the figure below.



Precaution

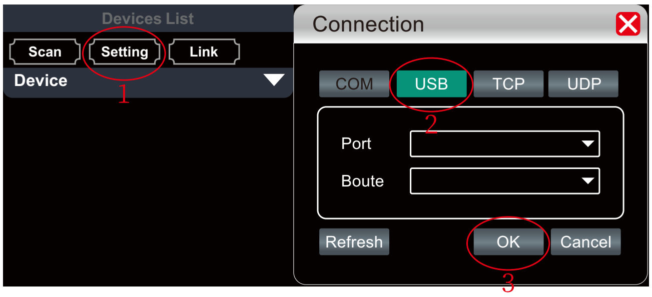
Some connection methods do not support multiple software to run simultaneously. Please ensure that only one software is open on each PC.

Software connection

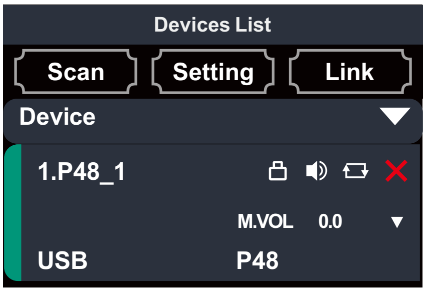
P series audio processor can be connected to “BrainCore Net” management software in a variety of ways.

USB connection

Users can quickly connect to the “BrainCore Net” management software through the USB interface of the processor panel. First of all, the user needs to set the connection port as "USB" port, and the software will scan the currently connected device and connect it.



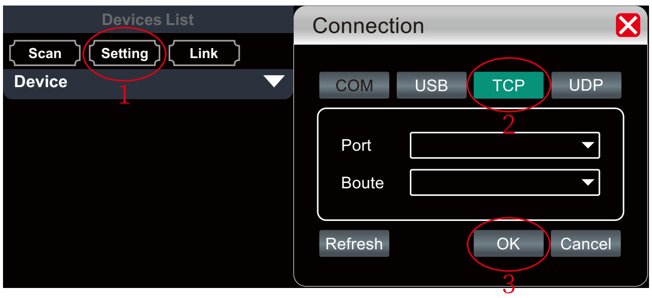
Note: 1, 2 and 3 are the sequence of operation steps



The green color indicates that "USB" of the current device has been connected.

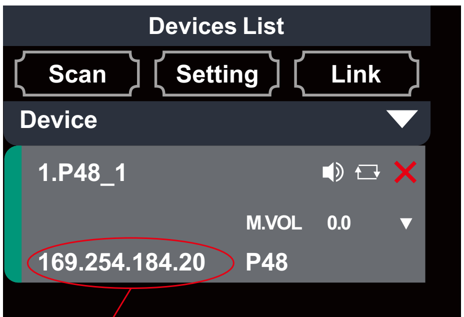
Ethernet connection

Users can connect to the "BrainCore Net" management software through the “Ethernet” interface(RJ45) on the back of the audio processor. First, the user needs to set the connection port as the "TCP"port, and the software will scan the currently connected device.



Note: 1, 2 and 3 are the sequence of operation steps

Then we need to change the IP address of the audio processor to be in the same IP segment with the computer, and there is no IP conflict in the LAN.



Device IP

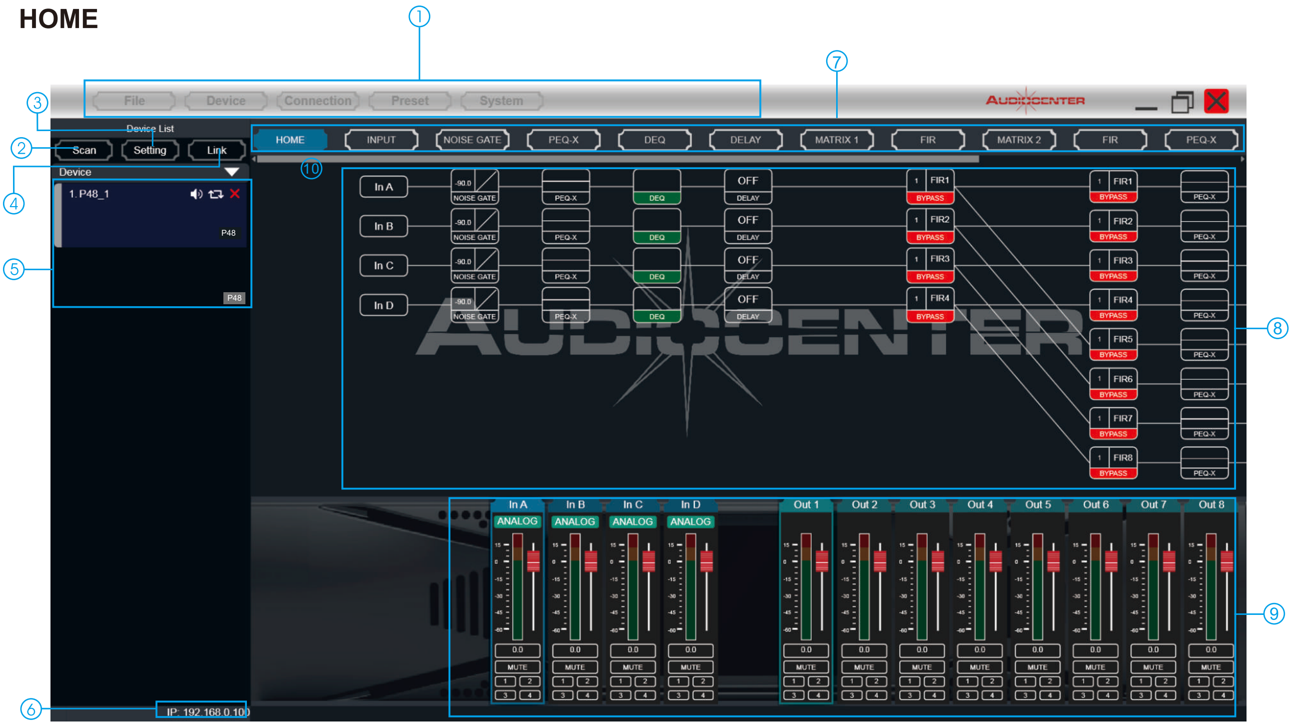
IP address of computer local connection

IP address of computer wireless network card

Note: If it is directly connected to the computer through the network cable, please change the IP of the audio processor to the local IP of the computer and connect it in the same IP segment.

If it is a wireless connection through a router, please change the IP of the audio processor to the IP of the computer wireless network card and connect it in the same IP segment.

HOME



(1)Menu (2)Scan (3)Setting

(4)Link (5)Device (6)Computer IP address

(7)Module (8)Module function (9)List of input and output channels

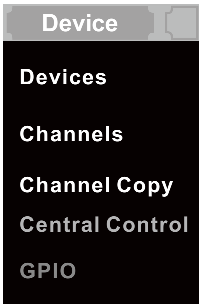
(10)Feature module drag bar

Menu column

File :

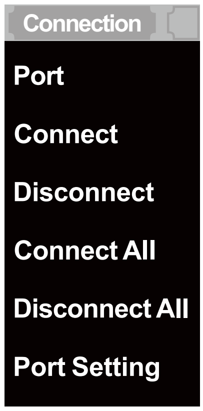
1.New project: the software can create the model of each device in this menu when the device is not connected.
2.Demo device: add virtual devices will not affect existing devices.
3.Open: open an existing device management project from the computer disk.
4.Save: save the current device management project in the computer disk.
5.Save As: save the current device management project as a file.

Device



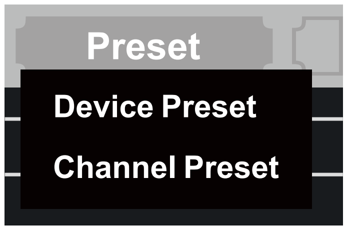
1.Devices: view or modify the firmware information, device name and IP address of the device,
2.Channels: output channel preset management,
3.Channel copy: copy the parameters of the same type of channel,
4.Central Control: users can click to view the central control protocol for quick configuration,

Connection



1.Port: set connection mode, port number and baud rate.
2.Connect: connect and download the device parameters.
3.Disconnect: disconnect the connected device.
4.Connect ALL: connect and download the parameters of all devices in the device list.
5.Disconnect ALL: disconnect all connected devices in the device list.
6.Port Setting: set the baud rate of the device.

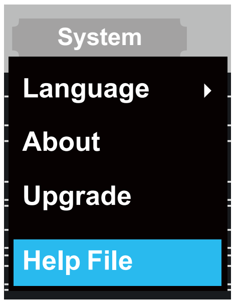
Preset



1.Device Preset: Click the preset management interface to manage the preset. For details, please refer to "Preset Management" on page 22.

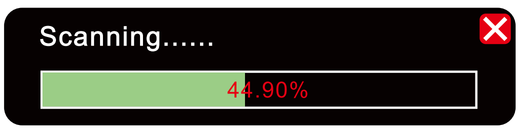
2.Channel Preset: Click the preset management interface for output channel, and the users can load Audiocenter standard presets or customized presets for the speakers through this function. For details, please refer to "Preset Management" on page 22.

System



1.Language: multi languages switch.
2.About: for the current firmware and device version information.
3.Upgrade: upgrade the firmware of the device.
4.Help File

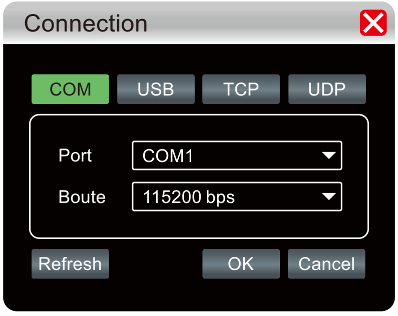
Scan



Click the "Scan" button to directly scan all devices currently in connection mode, and display the scanning progress as shown in the right figure.

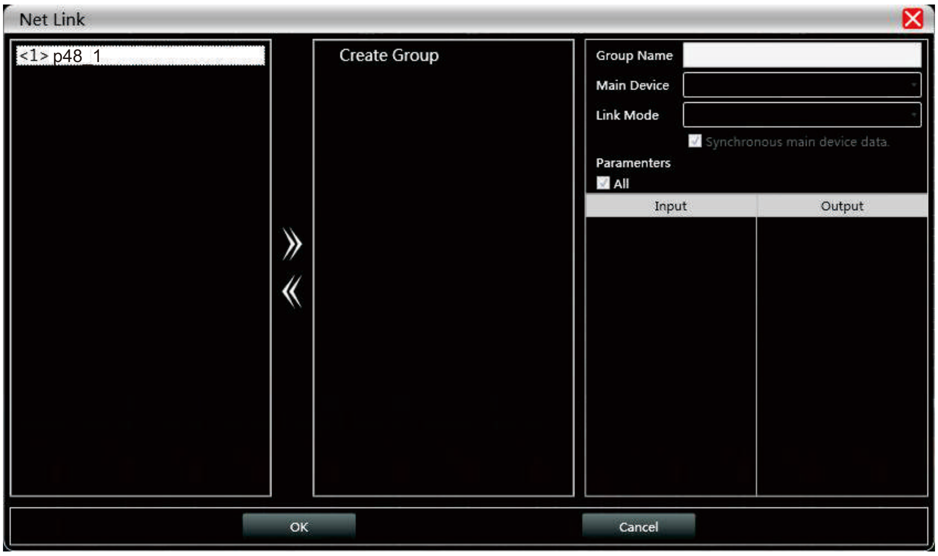
Setting

Set the connection mode of the scanning device and click the "Setting" button, and then the right figure will come out.
If the device port changes, click the "Refresh" button to update the port list instantly.



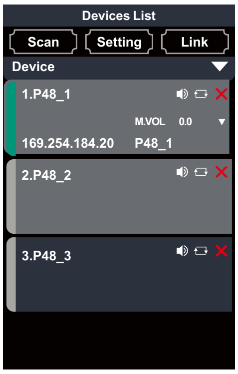
Link

To set the parameters of multiple devices at the same time, click the "Link" button, and the net link interface as shown in the following figure will come out. Select the devices to be set at the same time on the left, move to the create group in the middle, and then select the group setting parameters on the right. Finally, press the "OK" button to make the group function effective. You can also use the same operation to correct the net group settings.



Device

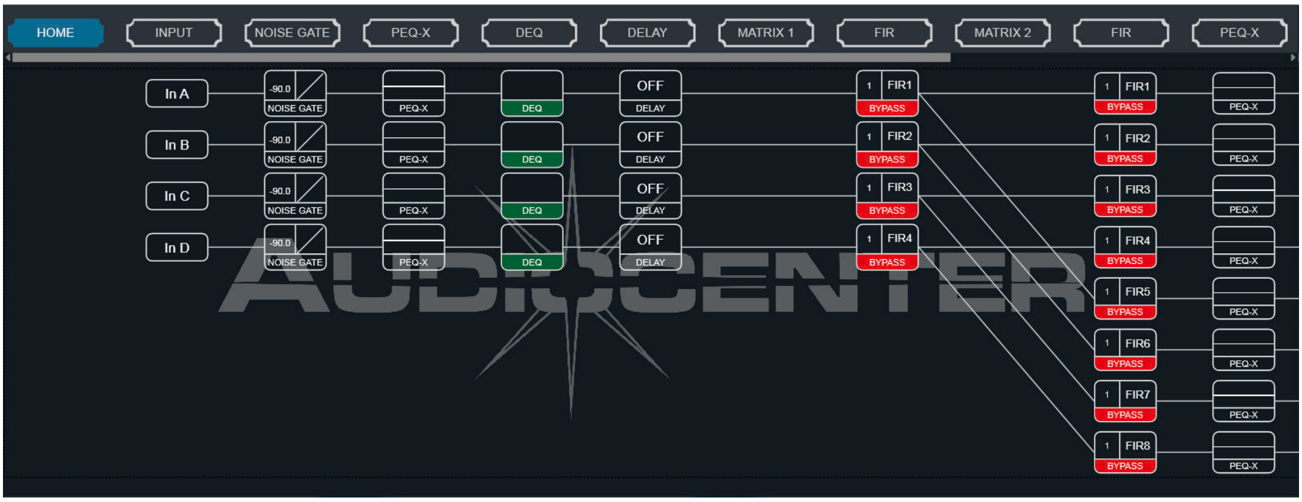
When the software scans or you add a simulation device, the corresponding device will be automatically added to the device list, which is convenient for users to interact with the required devices and operate multiple devices at the same time.



Local IP address

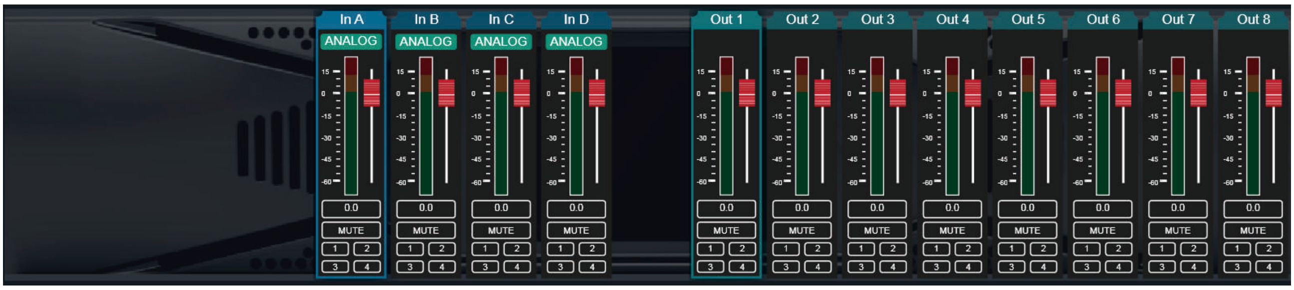
When the software is opened, it will automatically obtain the address of the network connection corresponding to the network adapter that has been in effect in the current computer system, which is convenient to manage the IP address of the device.

Function module control key



Access the corresponding operation page by clicking on the respective module.

List of input and output channels



The interface is capable of displaying various parameters for each channel, including level, gain, input mode, and channel name. It also allows for the control of the master switch for functions such as gain adjustment and muting for the respective channels. Additionally, it enables the activation of a grouped linking function for input and output channels, which significantly enhances operational convenience and power.

Introduction to function interface

INPUT

Click INPUT in the module button, and the channel input module shown on the right will pop up.

As shown in the diagram on the right, you can operate the polarity, mute, and input mode of the corresponding input channels:
As illustrated, you can manipulate the polarity, mute, and input mode for the respective input channels; within the input mode options, analog input, digital input, and test signal are three choices, and only one input mode can be selected for a single channel. The P48D supports four Dante inputs and four Dante outputs.

NOISE GATE

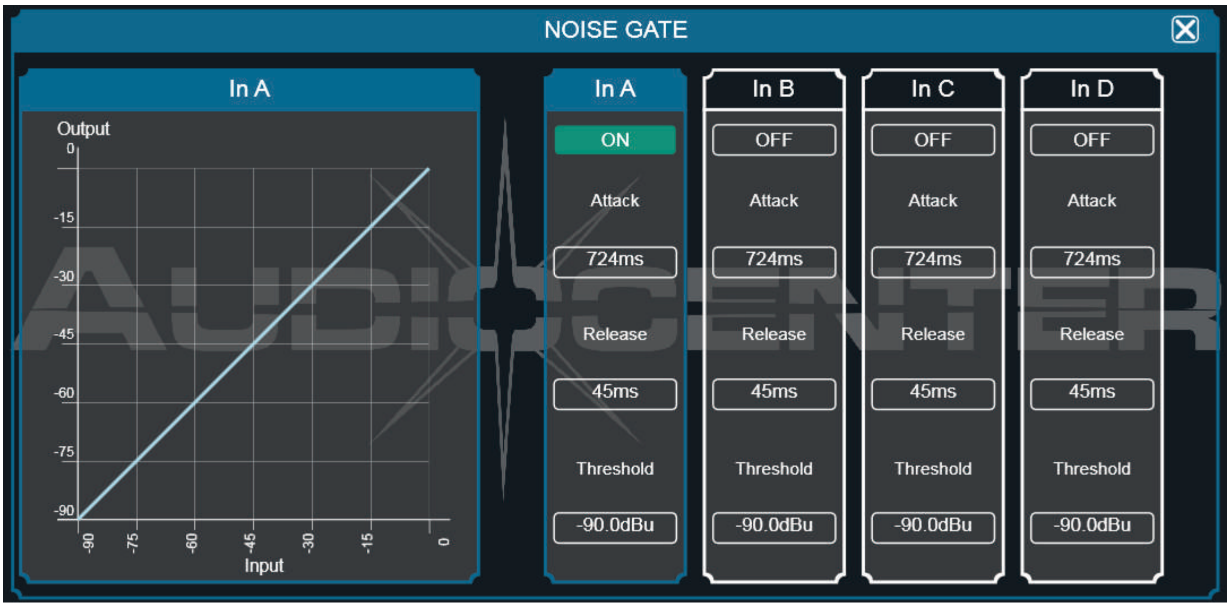
Click NOISE GATE in the module button to enter the input noise gate to set the module.

As shown in the figure above, click the noise gate switch ON to turn on or off the function of the noise gate of the channel. Green means turn on and red means turn off.

Attack time Attack 1.00 ms : Enter the corresponding value in the value box to control the Attack time.

Release time Release 1.00 ms : Enter the corresponding value in the value box to control the release time.

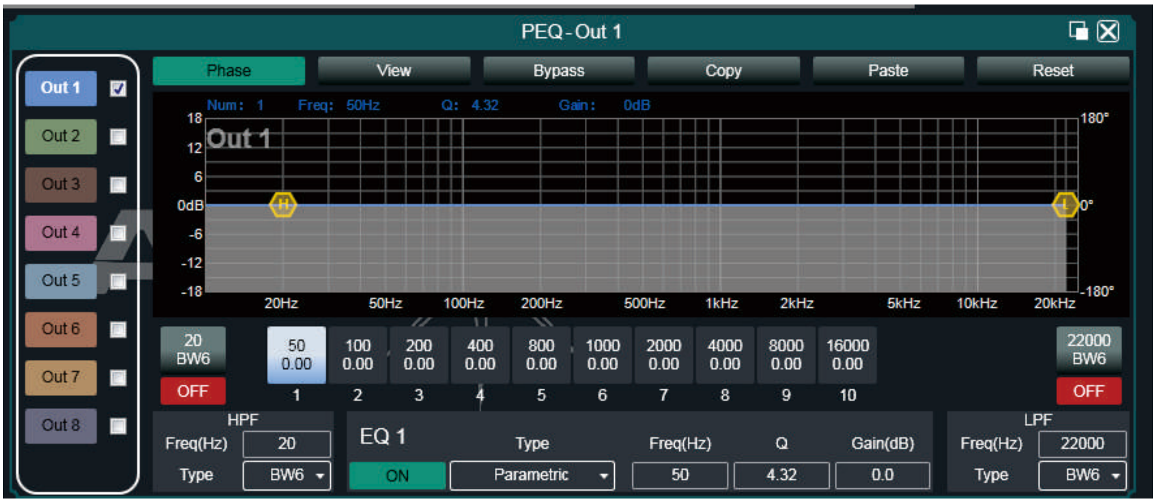
Threshold level Threshold -90.00 dBu : Enter the corresponding value in the value box to control the threshold level.



Equalizer (PEQ-X)

Click  in the module button, and then the PEQ setting interface as shown in the following figure will pop up.

The  button at the upper right corner of the module can enlarge the module, which can make the interface more clear.



Phase: display the phase curve of the current channel.

View: show or hide all equalization control points.

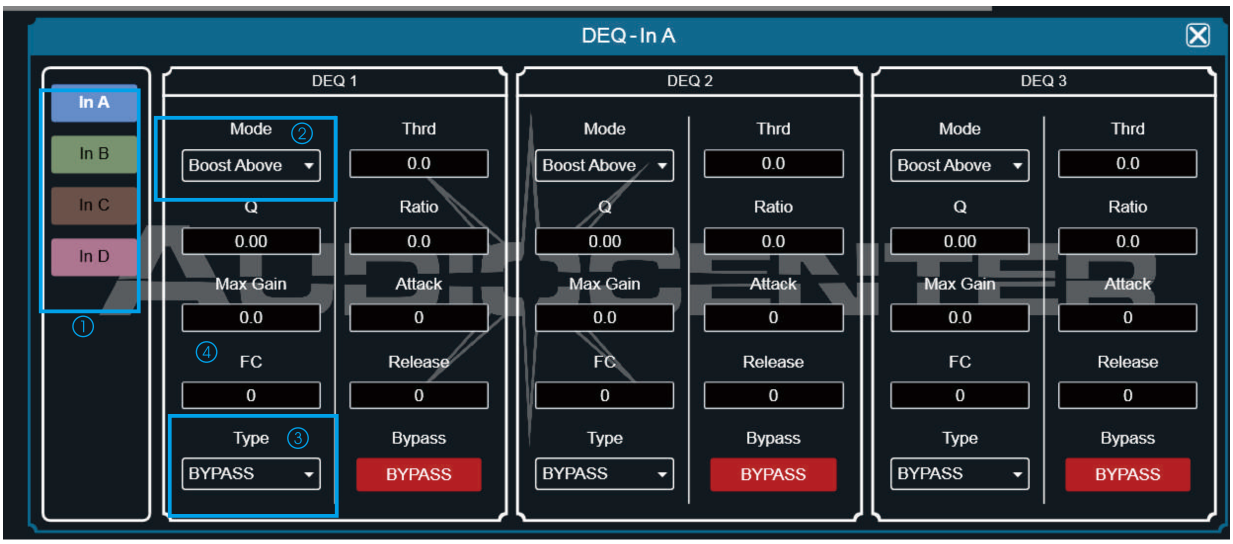
Bypass: turn on or off all EQ equalizers of the current channel at the same time.

Copy:copy the current equalizer parameter value and paste it into another input channel.

Paste: can be used together with the copy button, And can paste the equalizer parameter values copied by the copy function to the current channel.

Reset: reset the equalizer parameters to the default values.

Dynamic EQ (DEQ)

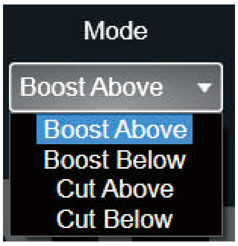


1. Click the input button to switch between input channels. (Figure 1)

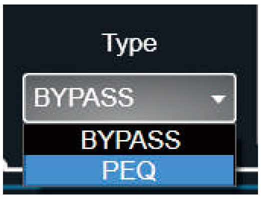
2. You can switch between four modes: gain increase above threshold, gain increase below threshold, gain decrease above threshold, and gain decrease below threshold. (Figure 2)

3. You can select the type of filter. (Figure 3)

Picture 1



Picture 2

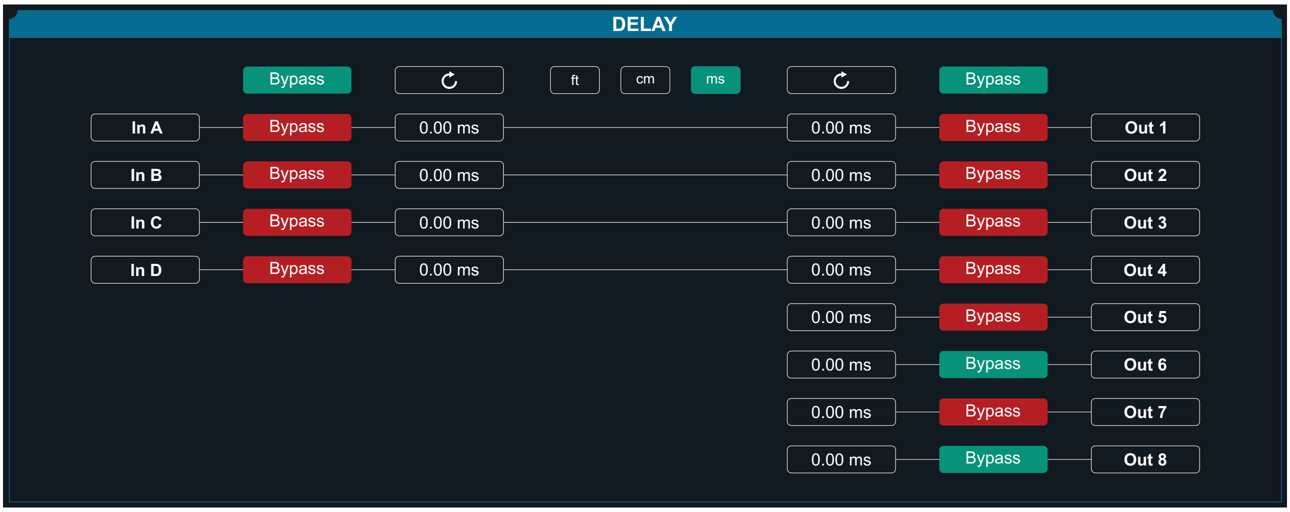


Picture 3

4. The control parameters for DEQ include threshold level, Q factor, ratio, maximum effect, attack time, frequency, and release time, which can be set by modifying the numerical values the corresponding input fields.

DELAY

Click in the module button to enter the I / O delay setting module.



As shown in the figure above, the time delay control contents of all input / output channels are listed.

Input the corresponding number in the value box , and the delay unit can be selected through

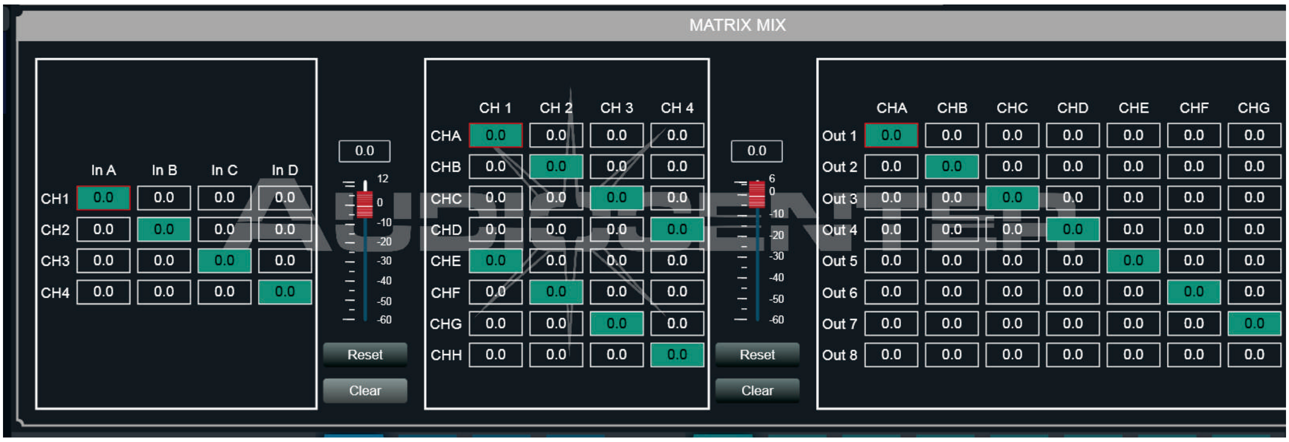
; is the switch of delay function: red means off, green means on; upper reset

button can directly reset the default value of channel delay.

MATRIX MIX

Click 

in the module button, and then the matrix mix setting module as shown below will pop up.

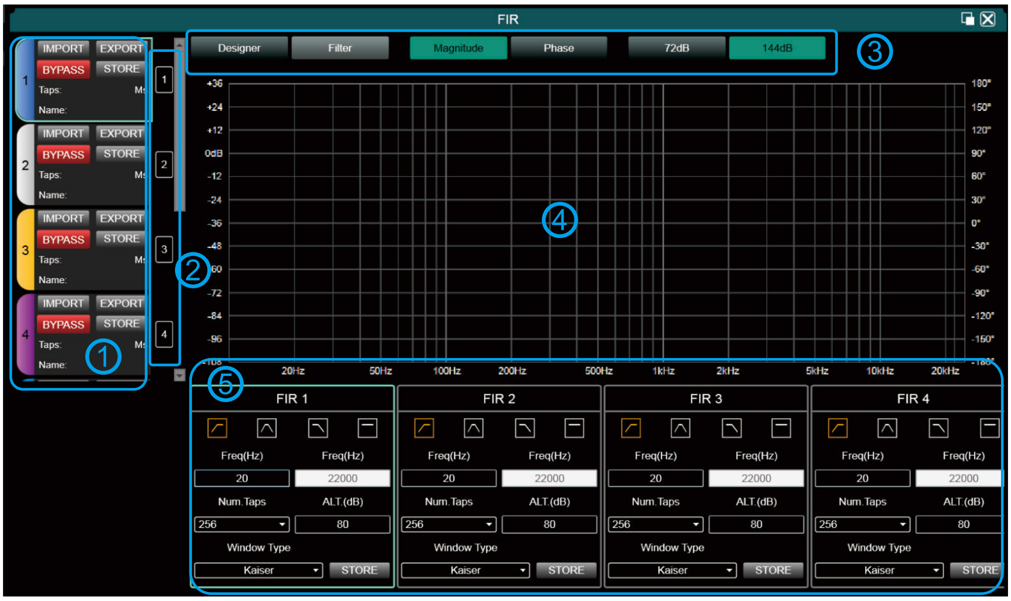


In the diagram above, the left side corresponds to the output channels, while the top side corresponds to the input channels. The numerical boxes with values are the mix keys for input and output channels. When the mix key is green (double-clicking the numerical box can toggle its state), the signal from this input channel and the signal to the output channel will be mixed.

The right section of the diagram includes the gain control, reset button, and clear button for the matrix mix. By clicking on the numerical box on the left and then dragging the slider of the matrix mix gain or entering a value in the numerical box, you can adjust the gain value within this matrix block. Clicking the reset button will revert the matrix mix function to its initial one-to-one state. Clicking the clear button will completely erase the matrix mix function, resulting in no correspondence between the device's inputs and outputs.

FIR Instructions

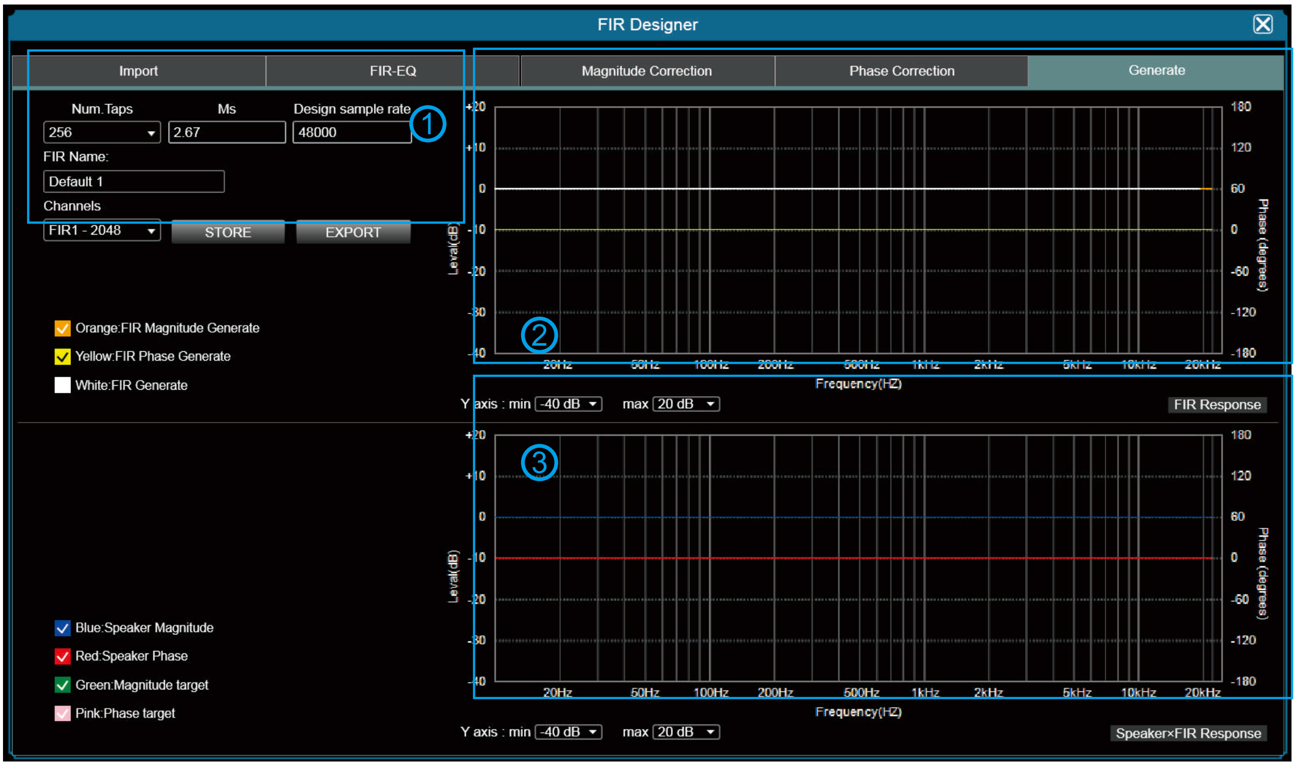
Double-clicking the function button  will bring up the FIR page shown in the image below. The button in the upper right corner of the module can be used to enlarge this interface for more convenient debugging.



The following table ①②③④⑤ indicates the corresponding function descriptions for the marked areas.

Settings ①	IMPORT	Import FIR preset document to current channel
	EXPORT	Export current channel FIR document
	BYPASS	Full bypass function, when activated, the channel EQ is set to a straight line. The default red status indicates it is on. When using FIR, it needs to be manually turned off.
	STORE	Store FIR parameters to the device
	Taps	Number of Taps for the current channel
	Ms	Latency of the current channel
	Name	Name of the current channel's FIR preset document
Channel ②	1234	Click to display the FIR curve for the corresponding channel, supports multiple lines at the same time
Interface ③	Design	Design interface switch button
	Magnitude	Interface displays amplitude curve
	Phase	Interface displays phase curve
	72dB	Display amplitude precision within 72dB
	144dB	Display amplitude precision within 144dB
Display④	-108~+36	Amplitude precision, when the amplitude curve is displayed,used as a reference value
	-180-+180°	Phase precision, when the phase curve is displayed,used as a reference value
	20Hz-20KHz	Frequency band
Design Area ⑤		High-pass
		band-pass
		Low-pass
		Flat
	Freq(Hz)	Start Frequency setting, left for High-pass, right for Low-pass
	Num.Taps	Taps number, a total of 8 stages. Options range from 256 to 2048
	ALT.(dB)	Depth, only available when using the Kaiser window
	Windows Type	Window type selection
	STORE	Save, after setting the parameters click to save to the device channel

FIR parameter generation interface



Page: After processing through the first 3 pages, it is necessary to proceed to the generation page, where you can select to generate to the corresponding channel or export and save the FIR parameters. Before generation, you can choose the number of taps and modify the FIR parameter name. Clicking the "Generate" button will immediately apply the channel parameters of the device, allowing for further testing confirmation or listening to the final effect.

①: Settings and Selection of FIR Data:

A. Num. Taps: Select the number of taps for the generated file;

B. mS Delay: The delay time corresponding to the current number of taps;

C. Sampling Rate: The sampling rate after generation (P series processors at 48K);

D. FIR Parameter Name: Can be modified according to your own requirements;

E. Channel: Generate to the selected channel;

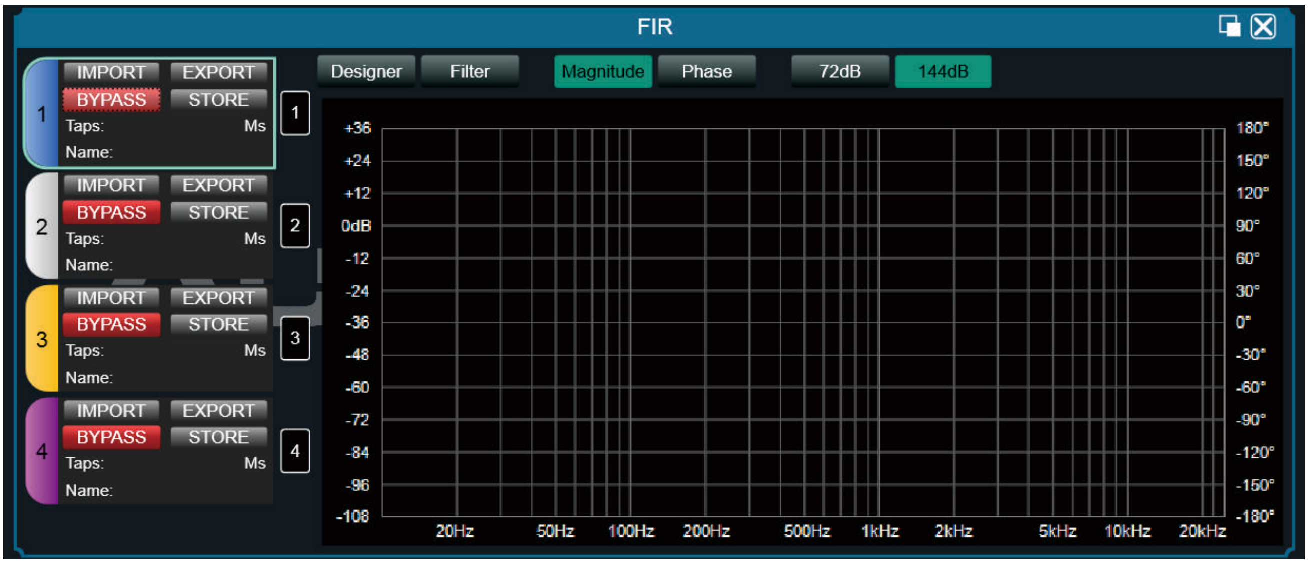
F. Save;

G. Export: Export and save the current FIR parameters to the computer.

②: FIR Response Graph

③: Speaker Response + FIR Response Graph

After generating the FIR curve, you need to return to the FIR interface and turn off the BYPASS button to enable the FIR function.



FIR Characteristic Description

The sampling rate divided by the number of taps can be used to calculate the frequency resolution of the filter. A higher number of taps implies a higher frequency resolution, which in turn means a narrower filter and a steeper roll-off.

Frequency Resolution = Sampling Rate (fs) / Taps

Lower Frequency Limit = Frequency Resolution × 3

For example:

Frequency Resolution = 48000 / 1024 = 46.875 Hz Lower Frequency Limit = 46.875 × 3 = 140.625 Hz

Explanation: For an FIR filter with a sampling rate of 48 kHz and 1024 taps, its frequency resolution is 46.875 Hz. To make a quick estimation based on this data, we can multiply the frequency resolution by 3 to obtain the lower frequency limit at which the filter can operate. 46.875 Hz × 3 ≈ 141 Hz, which means that an FIR filter with a sampling rate of 48 kHz and 1024 taps will be effective for frequencies above 141 Hz.FIR Delay: FIR filters have processing delays, which are related to the sampling rate and the number of taps. Delay = (1 / Sampling Rate × Taps) / 2 For example: Delay = (1 / 48000 × 1024) / 2 ≈ 10.7 ms Explanation: For an FIR filter with a sampling rate of 48 kHz and 1024 taps, its delay is 10.7 ms. To make a quick estimation based on this data, if the number of taps is 512, the delay would be 5.3 ms, and so on.

- FIR Delay: FIR filters have processing delays, which are related to the sampling rate and the number of taps.
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For example: Delay = (1 / 48000 × 1024) / 2 ≈ 10.7 ms

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Taps	48kHz	96kHz
256	2.67ms, LF 563Hz	1.33ms, LF 1125Hz
512	5.33ms, LF 279Hz	2.67ms, LF 558Hz
768	7.99ms, LF 188Hz	4.00ms, LF 375Hz
1024	10.67ms, LF 141Hz	5.33ms, LF 281Hz
2048	21.33ms, LF 70Hz	10.67ms, LF 141Hz

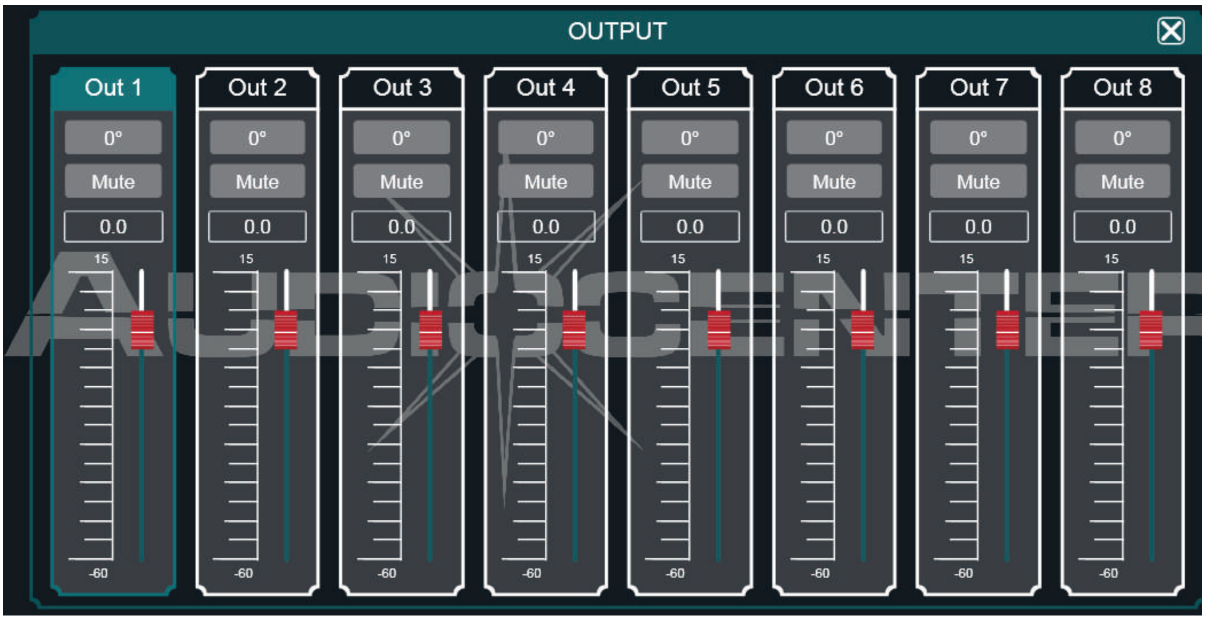
Example

LIMITERS

Click **LIMITERS** in the module button, and then the output setting module as shown in the following figure will pop up.

OUTPUT

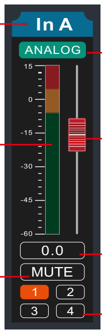
Click **OUTPUT** in the module button, and then the output setting module as shown in the following figure will come out.



As shown in the figure above, the polarity and mute setting of the corresponding output channel can be controlled.

Input channel

On the list of input and output channels in the software homepage, the left part is the input channels, as shown in the figure below.



Channel name

Input mode of the current channel

Input level status display

Input volume control

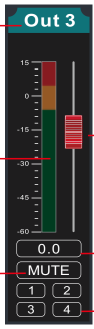
Display and control of input volume

Mute button

Group link of input channel (when the background color of corresponding number changes to yellow 1 , it means that the channel is in the input group of corresponding number)

Output channel

On the list of input and output channels in the software homepage, the right part is the output channels, as shown in the figure below.



Channel name

Output volume control

Output level status display

Display and control of output volume

Mute button

Group link of output channel (when the background color of corresponding number changes to yellow 1 , it means that the channel is in the output group of corresponding number)

The device list of the software homepage is shown in the figure on the right

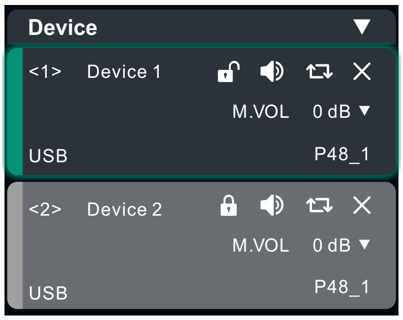
<1> is the number of the device connected

Device 1 is the device name (user can change the device name in the software)

P48_1 is the product name (user cannot modify, this example picture shows P48)

USB is the connection mode used by the device (USB, TCP, COM, this example picture shows USB)

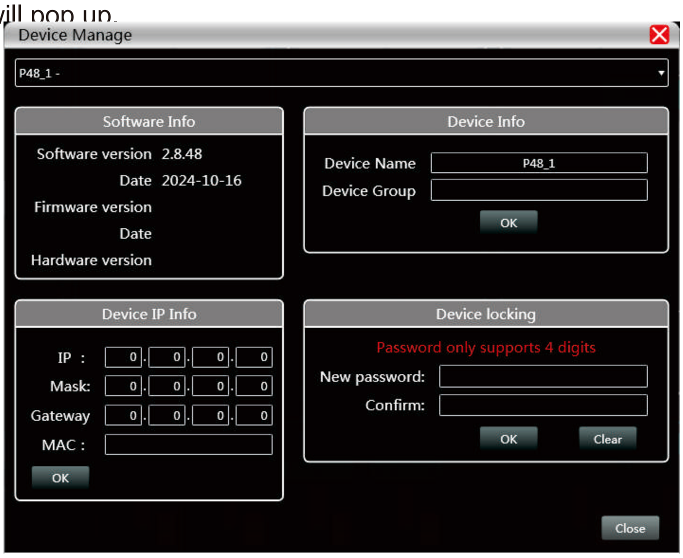

from left to right, shows device lock display, mute control state, refresh button and remove device button



If you need to debug different devices, you can click to select the target device, and the interface will be updated to the device function page.

Devices management

Click " Device "→" Device " in the menu bar of the main interface of the software, and then the interface of device management as shown in the figure below will pop up.



Software Info: display the version number and date of the upper and lower computer of the current device.

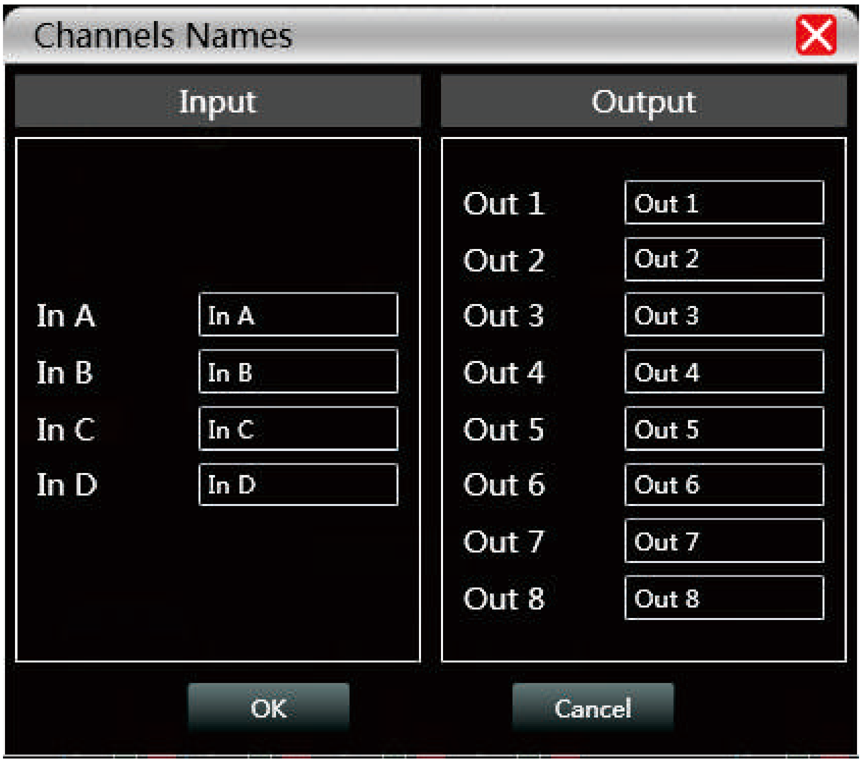
Device Info: display the current device name, device group and factory information. To display the factory name, you need to press the hide shortcut key Ctrl + Alt + F12, where a new name can be input in "Device Name" and "Factory Name", and then click button to save.

Device IP information: if the current device is connected with network information, the IP address, gateway and MAC address of the device will be displayed here. The IP and gateway can enter new information and click the OK button to save and restart the device network module. The newly entered network information will take effect in time.

Device Locking: can lock the device

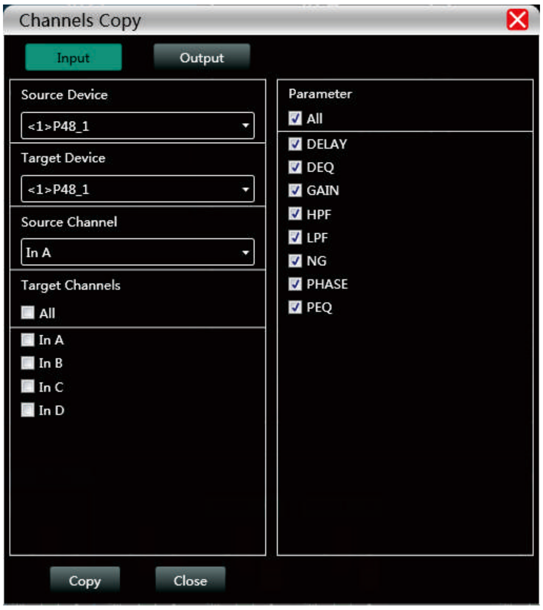
Preset management of output channel (Channels)

Click " Device "→" Channel Name" in the menu column of the main interface of the software, The channel name management will pops up as the following picture shows.



Channel copy

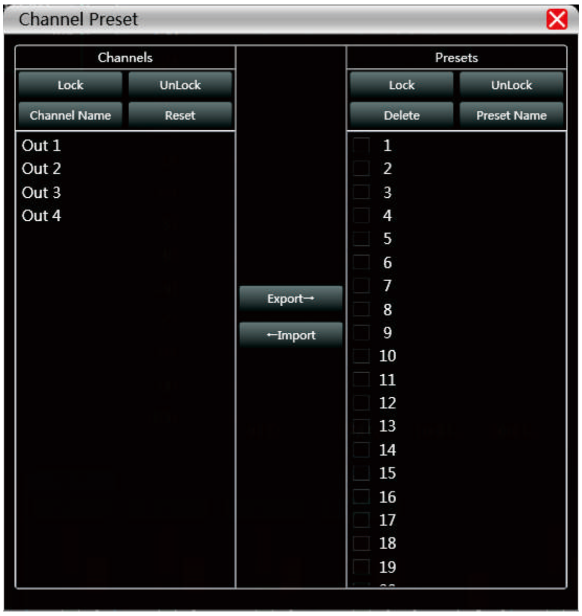
Click " Device "→" Channel copy " in the menu column of the main bar of the software, and then the interface of channel copy as shown in the figure below will pop up.



As shown in the figure above, channel copy is to select the channel parameters of one source device and copy them to the target channel of other target devices. Input channels and output channels cannot be copied with each other. The left side is the corresponding channel, and the right side is the copied parameters. The "Input" and "Output" buttons at the top of the interface can switch the channel type of the copy.

Output channel preset management(Channel Preset)

AUDIO PROCESSOR P series, according to the characteristics of Audiocenter speaker, is built-in rich speaker presets. Users can load a preset for each output channel separately according to the speaker equipped with the system. “Preset” →”Channel Preset”

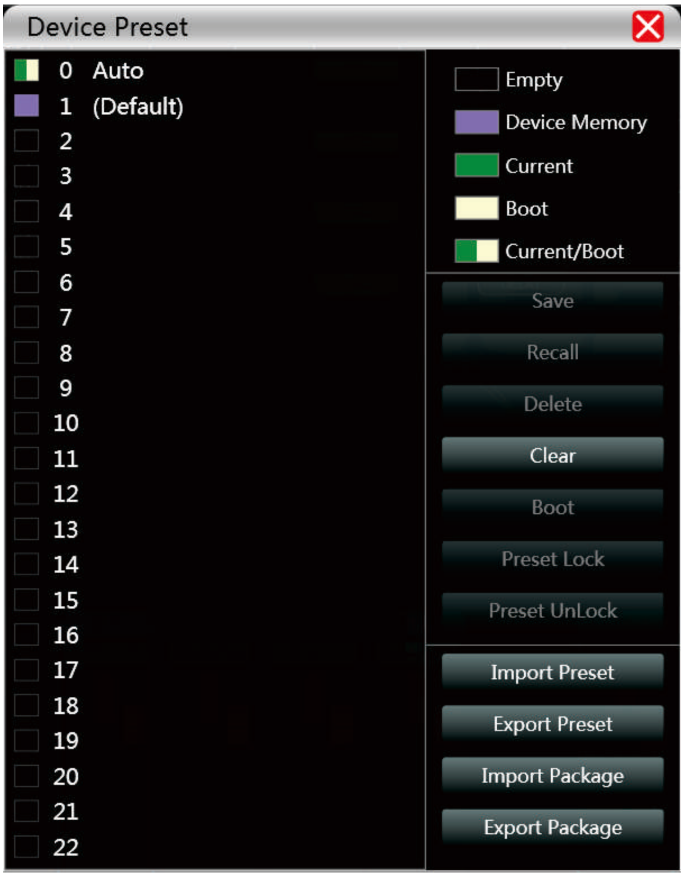


As shown in the picture, there are 8 output channels on the left, you can choose to save the preset for each channel through the"Export" key. You can also load the preset from the right output channel to any other output channel with the "Import" key. Encryption management is available for the presets (only for output EPQ and LIMIT encryption).

Whole device preset management

After debugging the system, user can save the presets of the whole device through "Preset", and a total of 30 preset gears are available.

As shown in the figure above, the left side of the archive interface shows gears, where "0 Auto" is the system gear and cannot be used directly; "1 (Default)" is the default gear of the device, which can only be recalled, but cannot be deleted or overwritten. After recalling, all the device parameters will be restored to the factory default parameters; other gears can be freely saved, recalled, deleted, etc.



The function buttons on the right side of the archive interface are as follows:

Save: save the existing device parameters to the corresponding selected archive.

Recall: recall the selected archive to the parameters of the current device.

Delete:delete the selected archive parameter.

Clear : clear all non system archived parameter records.

Boot: set the selected gear as the gear that will be called automatically to work when the device is turned on next time;

Preset lock: when preset is locked, relevant parameters cannot be changed and viewed.

Preset unlock: preset is unlocked.

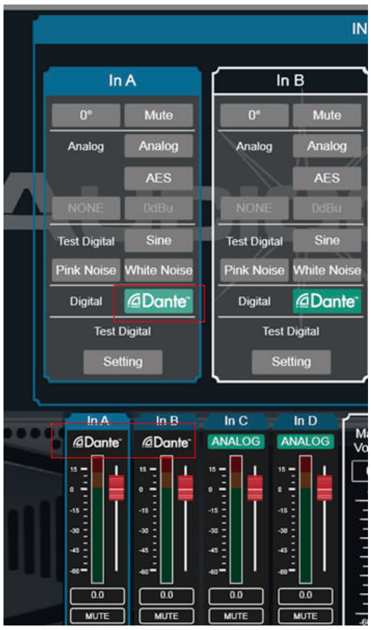
Import Preset: import a single device parameter file in the computer system to directly overwrite the existing parameter data.

Export Preset: save the parameters of the current device to the computer system, and generate a single device parameter archive file.

Import Package: import the archived parameter package in the computer system.

Export Package : export all gear parameters of the device archive to the computer system, and generate a parameter package file.

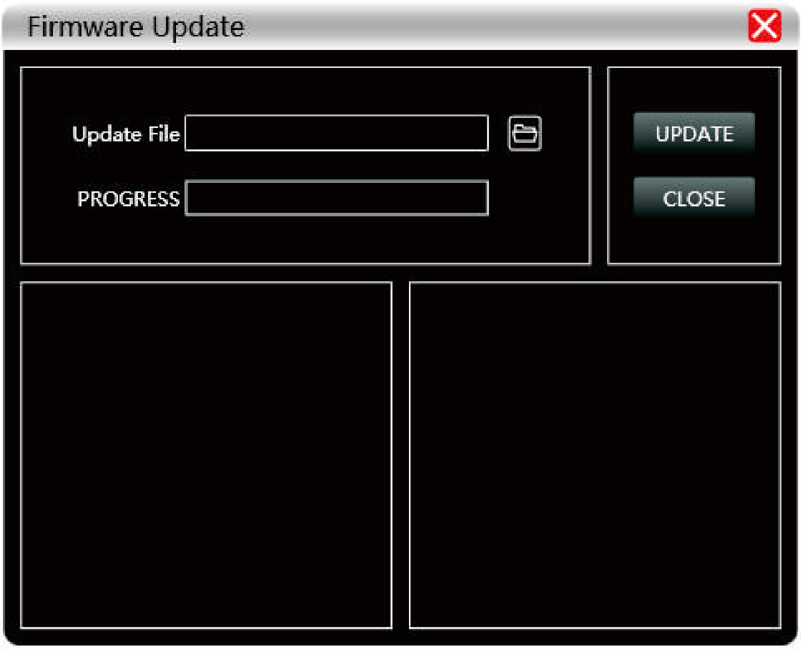
Dante Instructions



On the INPUT interface, set the channel input source to Dante mode.
A. There are two methods to connect to Dante: the first is to join an existing local area network with other Dante devices, and the second is to connect through a virtual sound card (Dante Virtual Soundcard, abbreviated as DVS). Both connection methods require a wired connection through a switch or router.
B. Regardless of the connection method used, it is necessary to configure the routing through the official Dante Controller software provided for computers. The Dante Controller software can be downloaded for free from the official website at www.audinate.com.

Firmware upgrade

Click the main interface menu of the software, you will access to the firmware upgrade interface as shown in the pictures.



When firmware is updated, you can access to the firmware upgrade interface once got the upgraded file.After selecting the corresponding upgrade file in the “Update File” column, click “UPDATE” in the upper right corner,the system will automatically transfer the upgrade file to the firmware for upgrade operation, and display the operation log in the progress box below. After the upgrade is completed, the machine system firmware will automatically restart or manually restart the machine to complete the upgrade.

◆Specification

SPECIFICATIONS		P48D	P48
Sampling Rate		96kHz	
AD/DA Converter		24bit	
Audio System Delay		<2.1ms (analog input- analog output)	
Input	Analogue	4-channel electronically balanced input	4-channel electronically balanced input
	AES	4-channel AES input	4-channel AES input
	Dante	4-channel Dante input	/
Input Interface		4x Female XLR	
Output	Analogue	8-channel electronically balanced output	8-channel electronically balanced output
	Dante	4-channel Dante output	/
Output Interface		8x Male XLR	
DSP	EQ	Each input 15-band EQ, each output 10-band EQ; EQ types: PEQ, High/Low Shelf,All Pass,All Pass2,VariQ High/Low Pass,Phase, Elliptic High/Low Pass, High/Low Pass,Band Pass, Notch	
	DEQ	Each input with a 3-band DEQ	
	Delay	Input delay: each channel 2000ms ; Output delay: each channel 2000ms	
	FIR	Input FIR :each channel 512Taps(@48kHz); Output FIR:each channel 512Taps(@48kHz)	
	Crossover	Butterworth, Linkwitz-Riley, Bessel:6 dB/oct to 48 dB/oct	
	Limiter	Compressor + Limiter	
Input Impedance		≥10kΩ	
Output Impedance		<100Ω	
Maximum Input/Output Level		≥+20dBu	
Requency Response		±0.3dB,20Hz-20kHz	
S/N Ratio		≥113dB @1kHz,A weighted	
THD+N		≤0.002% @1kHz,0dBu,A weighted	
Crosstalk		≥105dB @1kHz	
Connection Type		USB/RS232/TCP/IP	
AC Power Operating Range		100-240V/(±10%, 50-60Hz)	
Power consumption		＜20W	
Rack space		1U	
Dimension(WxHxD)		483x44x265mm	
Net weight		3.6kg	

Dynamic Audio Solutions

http://www.audiocenter.com E-mail:info@ac-pro.net