DSP-48DT

Network DSP FIR Processor 4x8

4x4 Dante channels

User manual

MENU

Chapter 1 Introduction

DSP-48DT is a 4in 8out FIR DSP audio processor, integrated with high performance DSP processor, Dynamic EQ, FIR automatic linear phase and other powerful functions. This audio processor supports a variety of input signals: Analog\AES3\Dante network audio. Standard FIR designer in the MusicAllDSP software provides automatic linear magnitude and phase function, easy to optimize speaker parameters.

With RJ45\USB and RS232 connectors, PC software MusicAllDSP provides users an easy way to control multiple devices. The RS232 connection supports control by a third-party system.

Applications

- Party room
- Live show
- Outdoor activities
- Sports stadium

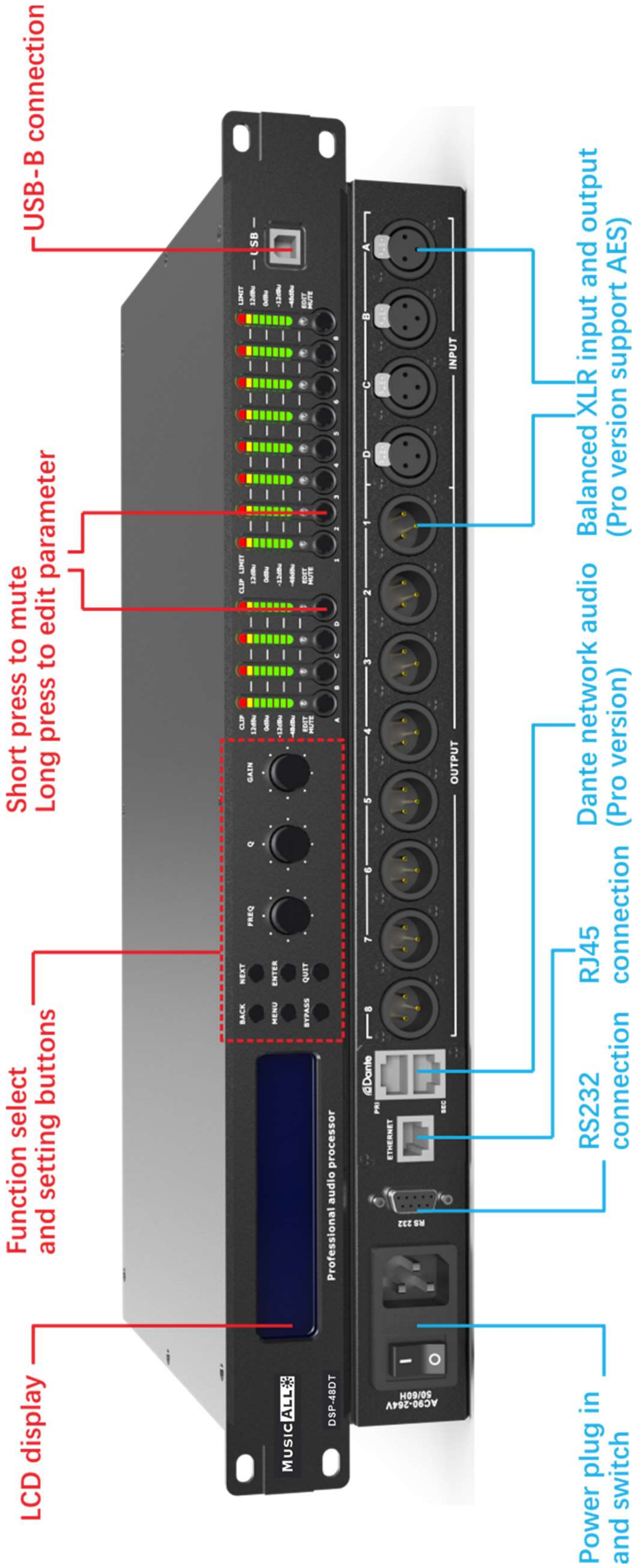
Features

- 4 analog inputs and 8 analog outputs.
- Each input can be switched to AES3 signal.
- 4 channels Dante network audio.
- High performance DSP processor, 96k 24bit sampling rate.
- Input with 15 bands PEQ, output with 10 bands PEQ.
- Supports HPF and LPF with Butterworth\Bessel\Linkwitz-Riley. Support LSLV and HSLV, ALL-PASS, Band pass, Band stop filters, Phase, Notch filters.
- Input with 3 bands Dynamic EQ.
- Input with 4 x 512Taps 48k FIR linear phase setting.
- Output with 4 x 512Taps 48k FIR linear phase setting.
- Standard FIR automatic linear phase function designer.
- Support presets archiving and locking, hide setting parameters.
- Control connections: USB or TCP/IP. Configured with RS232 central control connection.
- Nice GUI windows 7/8/10/11 software.
- LED VU indicator for input and output signals.
- High analytical power, wide dynamic audio performance, suitable for stage, bar and other high-end sound systems.

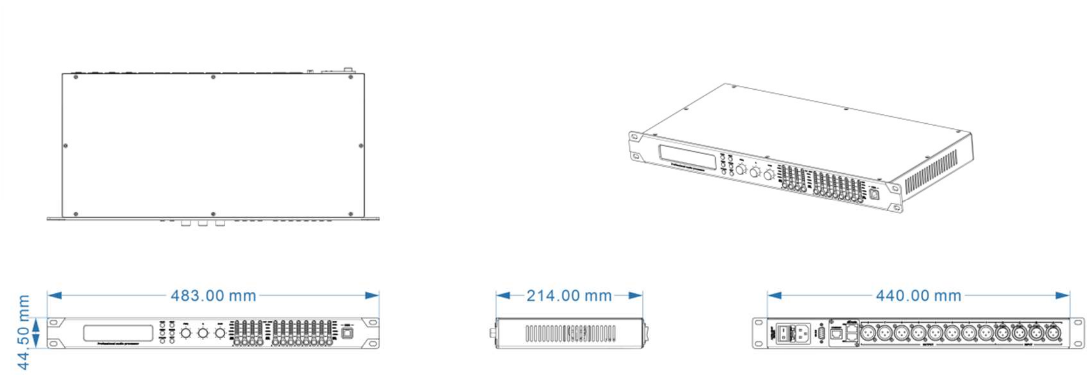
Chapter 2 Technical parameters

DSP-48DT	
1. DSP Processor	
Process:	ADI SHARC 21489 450MHz
System delay:	2.1ms
AD/DA:	24-bit 96KHz
2. Analog Audio Inputs and Outputs	
Input:	4 channels balanced.
Input interface:	XLR(Neutrik®)
Input impedance:	20KΩ
Max input level:	**20dBu/Line**
Dante in/out	4x4 channels
Output:	8 channels balanced. Line level
Output interface:	XLR(Neutrik®)
Output impedance:	50Ω
3. Audio Performance Specifications	
Frequency response:	20Hz-20kHz(+-0.5dB)/Line
THD+N:	-93dB(@0dBu,1kHz,A-wt)/Line
Ground noise:	20Hz-20kHz, A-wt, -93dBu
SNR:	113dB(@16dBu,1kHz,A-wt)/Line
4. Connection Ports and Indicators	
USB:	Type A-B
RS232:	Serial port communication
TCP/IP interface:	RJ-45
Indicator light:	Clip, level, edit, mute
5. Electrical and Physical	
Supply:	AC100V ~ 240V 50/60 Hz
Product Dimensions	483mmx215mmx44.5mm
Packaged Dimensions	537mmx343mmx77mm
Net Weight	3.8kg
Packaged Weight	4.2kg
Operating temperature:	-20℃ ~ 80℃

Chapter 3 Functions structure and panel operation



Dimensions (mm)



Operating front panel

①Press **MENU**, it will show menu list, using **NEXT** or **BACK** to select functions: GLOBAL MEMORY, INPUT SECTION, MATRIX, SYSTEM, press **QUIT** to exit.

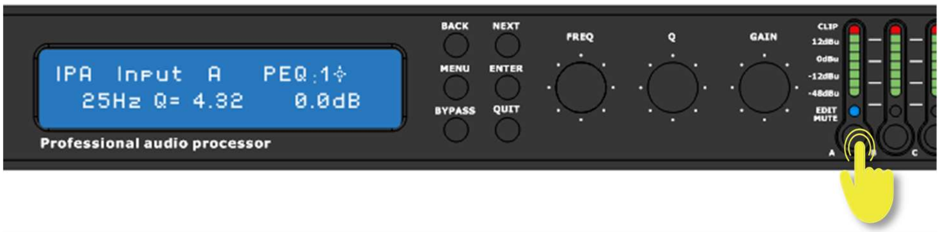
Functions in panel	Menu list	Remark
1.GLOBAL MEMORY	RECALL a Memory STORY a Memory DELETE a Memory	
2.INPUT SECTION	A ANALOG (DANTE) B ANALOG (DANTE) C ANALOG (DANTE) D ANALOG (DANTE)	
3.MATRIX	Routing Out.1=Input A*....	Long press such output channel button under LED to change.
4.SYSTEM	1 IP SET 2 RENAME 3 DSP VERSION	

②Press **BYPASS**, it will quickly show <u>RECALL a Memory</u> function, using **NEXT** or **BACK** to select presets and then press **ENTER** to enable one of presets.

Operation of buttons on front panel - Input & Output Channels

①When user need to quickly mute input or output channel, shortly press button under the LED of such channel, the light will turn red.

②When user need to set parameter of input or output channel, long press button under the LED of such channel, the light will turn blue.



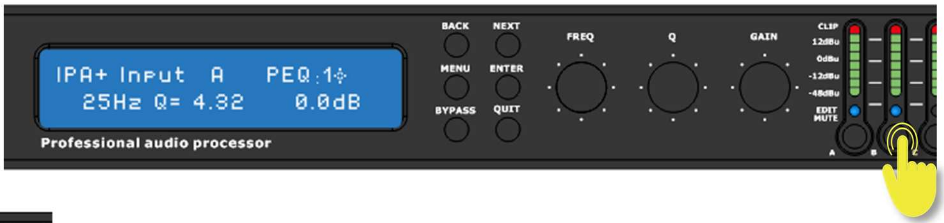
Functions - Input channels	Buttons for setting	Buttons for setting
IPX Input X Gain		
IPX Input X Polar	+,-	
IPX Input X PEQ:1...15	FREQ, Q, GAIN	BYPASS
IPX Input X HPF	FREQ, Q	BYPASS
IPX Input X LPF	FREQ, Q	BYPASS
IPX Input X Delay	GAIN	
IPX Input X Noise Gate	FREQ	BYPASS
IPX Input X Noise Gate	Q, GAIN	BYPASS
IPX Input X DEQ:1...3	FREQ, GAIN	BYPASS
IPX Input X DEQ:1...3	Q, GAIN	BYPASS
IPX Input X DEQ:1...3	Q, GAIN	BYPASS
IPX Input X DEQ:1...3	Q, GAIN	BYPASS
IPX Input X Fir		BYPASS

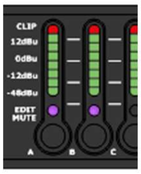
Functions - Output channels	Buttons for setting	Buttons for setting
OPX Output X Gain		
OPX Output X Polar	+,-	
OPX Output X PEQ:1...15	FREQ, Q, GAIN	BYPASS
OPX Output X HPF	FREQ, Q	BYPASS
OPX Output X LPF	FREQ, Q	BYPASS
OPX Output X Delay	GAIN	
OPX Output X Compress	FREQ, Q, GAIN	BYPASS

OPX Output X Compress	Q, GAIN	BYPASS
OPX Output X Limiter	Q, GAIN	BYPASS
OPX Output X Limiter	GAIN	BYPASS
OPX Output X Fir		BYPASS

Remark: **X**" means the No. of such channel user has selected. If find no effect after setting parameter, please check whether select BYPASS or not.

③When user need to link channels and then set their parameter, long press button under the LED of each channel, the light will turn blue. LCD will display "IPX+" or "OPX+", means the second channel or other channels will be set same with first channel.





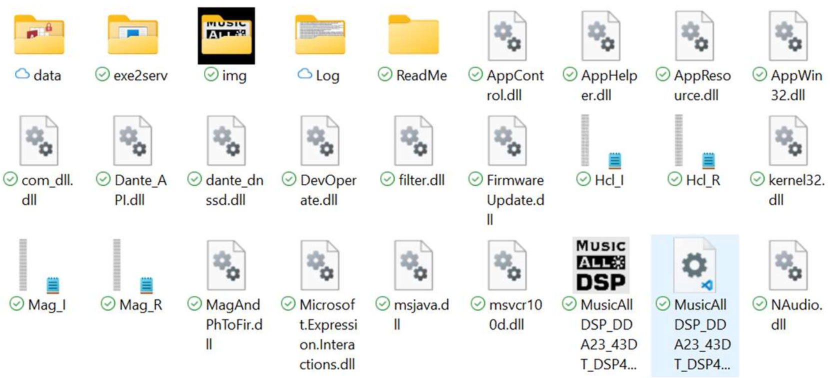
When user need to quickly mute channels in link setting, shortly press button under the LED of one of channels, all the lights will turn pink.

Chapter 4 Operation of control software - MusicAllDSP

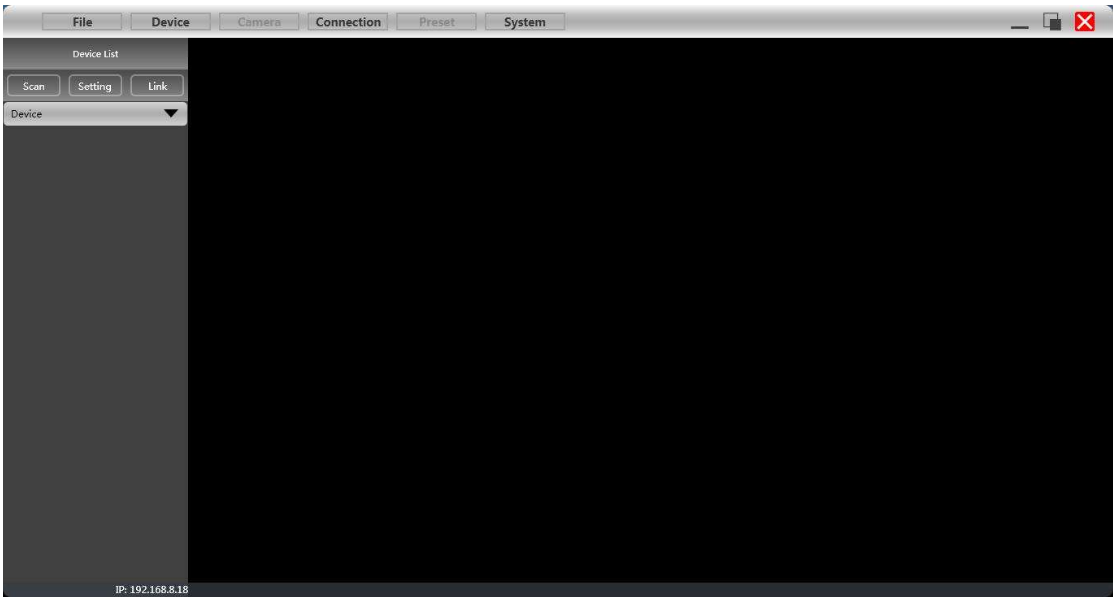
MusicAll provides user with a fast interaction to control one or more devices through multiple methods: TCP/IP, USB, common serial port (RS232). Easily set DSP functions of device and check central control codes. The configuration can be stored in presets, convenient for various applications.

4.1 Operating condition

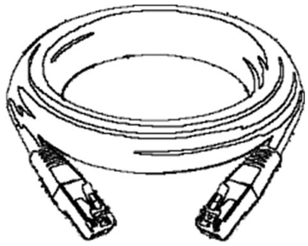
MusicAllDSP is suitable for Win7/8/10/11 x86/x64 PC systems with Microsoft .NET Framework 4.0 installed. Double click the file with the MusicAllDSP logo:



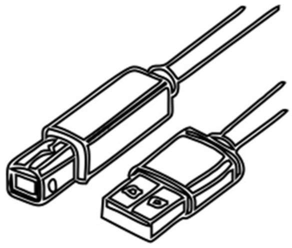
the main interface will pop up:



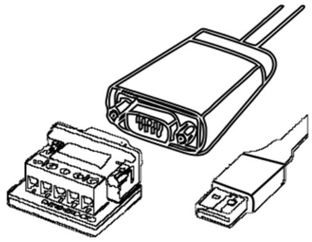
4.2 Connect to PC

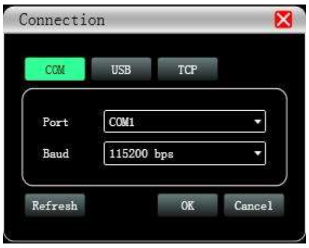


If you connect the device by using network cable, click Setting in Device List, choose TCP in Connection window.



If you connect the device by using USB A-B, click Setting in Device List, choose USB in Connection window.

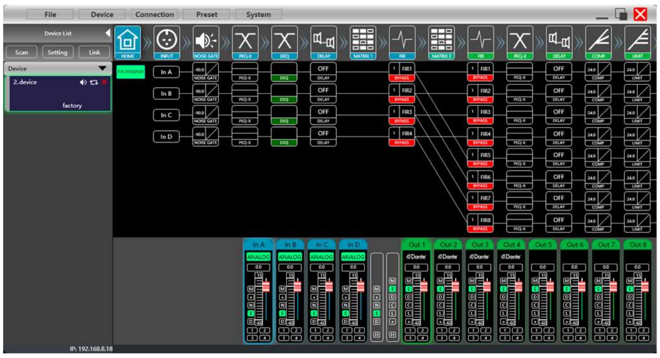


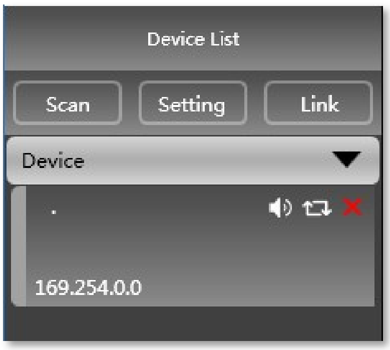


If you connect the device by using serial cable, click Setting in Device List, choose COM in Connection window. Please check port and baud rate carefully for 232 or 485 before setting.

The software will use the method set last time to check if the device is connected. If successfully connected, devices will be shown in device list.

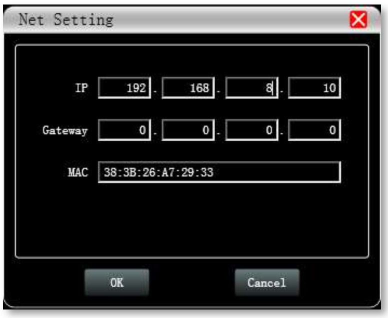
Users can mute devices, refresh connections, or delete devices in this window. Single click the device to load the function interface.





When using TCP control, there is a situation that only one point is displayed after scanning, but you can not connect the device. In this case change the IP address of the device to the same network range as the computer.

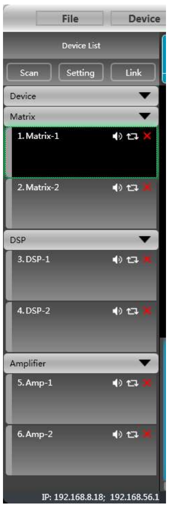
Right-click the device enclosure, a Net Setting window will show.

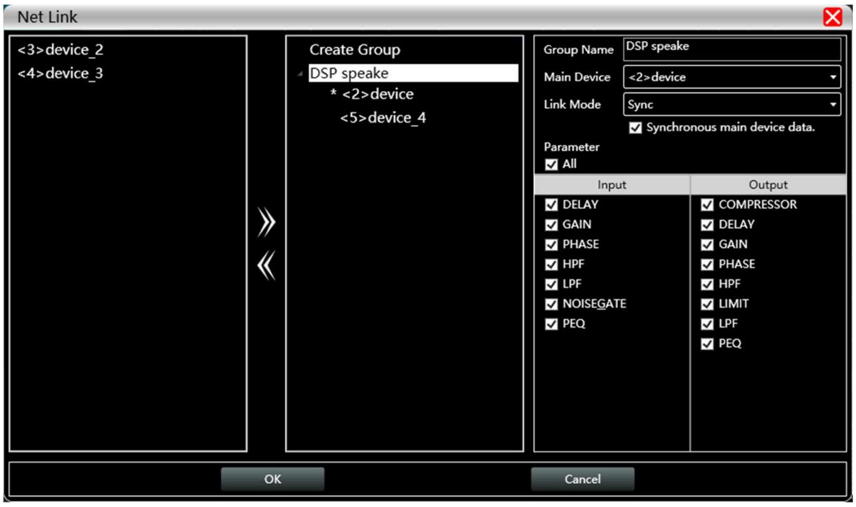


Set the IP address of the device. Refer to the IP showed in the bottom of the software. (the first 3 digits xxx.xxx.xxx. should be the same as the computer.

Successfully scanned and connected.

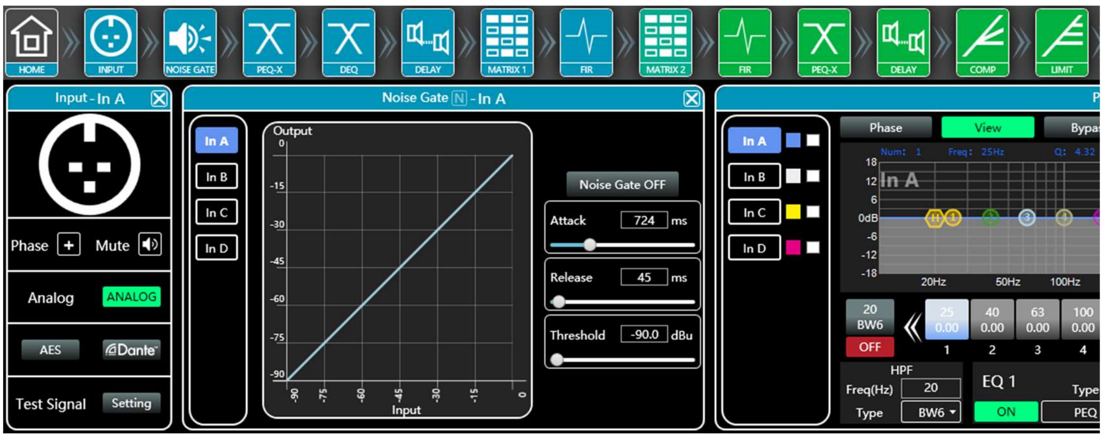
User can link a multiple of the same devices in groups by clicking the Link button, and then set group device, group name and main device, link mode and parameters according to needs.





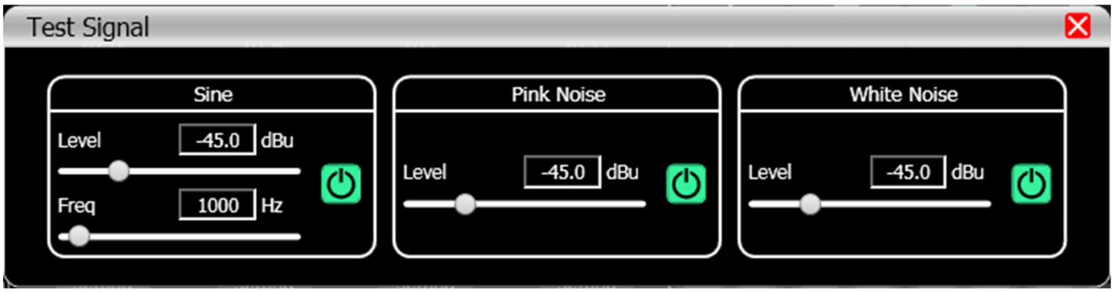
4.3 DSP functions setting

Double-click HOME icon to open all functional interfaces, or double-click a function icon separately to open the corresponding interface. When multiple function windows are opened, users can drag the window to switch function Settings.

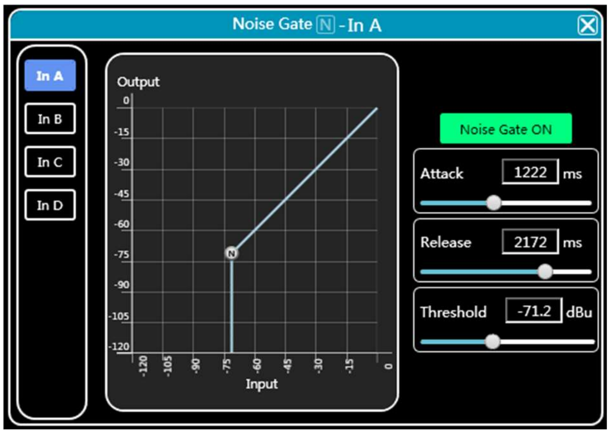


4.3.1 DSP functions setting - INPUT

- Set Phase of input;
- Set Mute of input;
- Select Analog\AES3\Dante input signal;
- When choosing test signal, user can select Sine/Pink Noise/White Noise for each input channel.

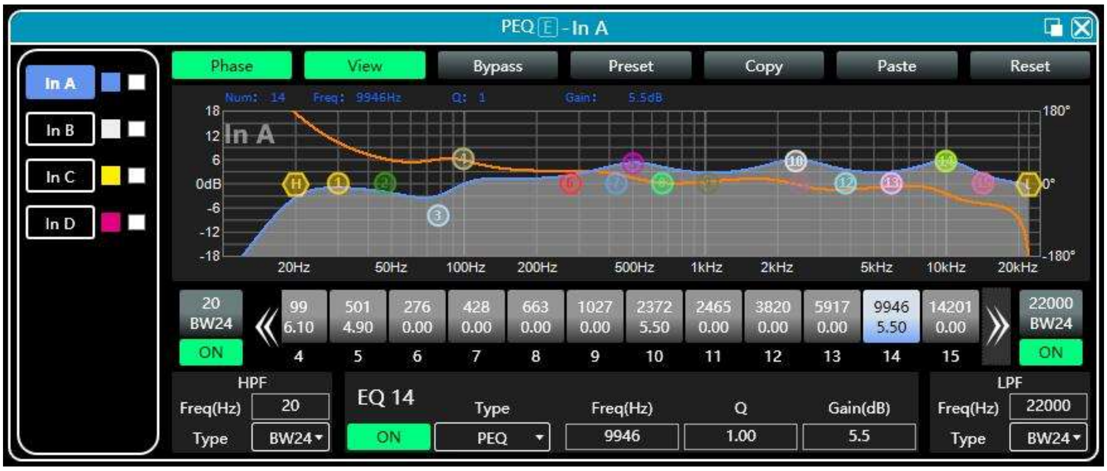


4.3.2 DSP functions setting - NOISE GATE



- Attack: 1 to 2895ms;
- Release: 1 to 2895ms;
- Threshold: -90 to 0dBu;
- Click  to enable this function.

4.3.3 DSP functions setting - PEQ-X (input and output)



High pass filter

enter value of frequency and select type, press  to enable this function:

Butterworth 6/12/18/24/36/48, Bessel 12/24/36/48, Linkwitz-Riley 12/24/36/48.

Low pass filter

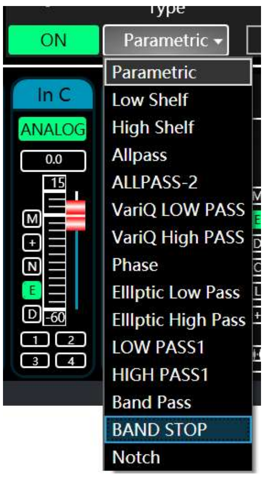
enter value of frequency and select type, press  to enable this function:

Butterworth 6/12/18/24/36/48, Bessel 12/24/36/48, Linkwitz-Riley 12/24/36/48.

PEQ 15 bands for input channel

Type: PEQ/LSLV/HSLV/ALLPASS-1/ALLPASS-2/3 kinds of high/low pass,Phase, Band pass,Band stop,Notch filter;

Freq(Hz) Q Gain(dB): input value or use mouse pulley to set value; Users can also drag the frequency dot on the curve to adjust.

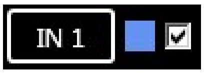


PEQ 10 bands for output channel

Type: PEQ/LSLV/HSLV/ALLPASS-1/ALLPASS-2/3 kinds of high/low pass,Phase, Band pass,Band stop,Notch filter;

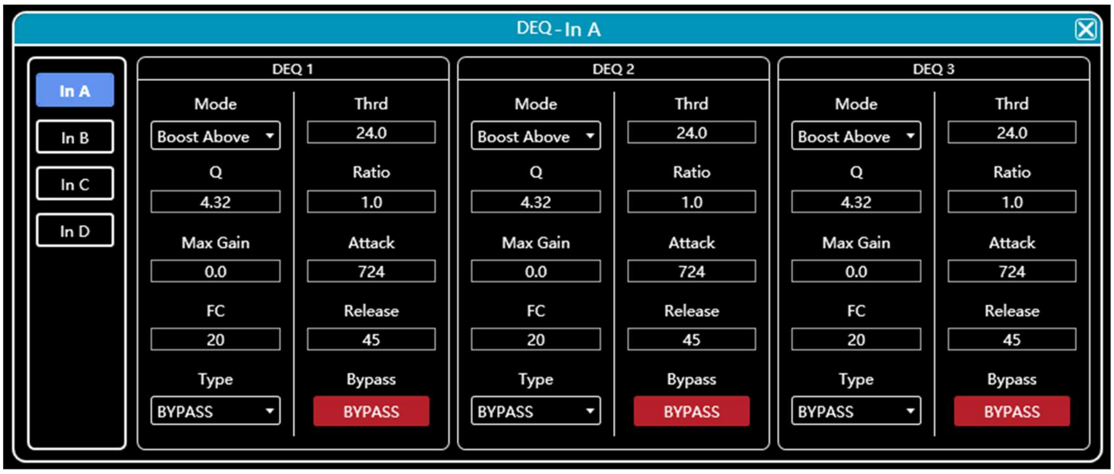
Freq(Hz) Q Gain(dB): input value or use mouse pulley to set value; Users can also drag the frequency dot on the curve to adjust.

Phase curve: display the phase curve of the current channel.
View: show or hide all balance control points.
Bypass: turn on or off all equalizer EQ of the current channel at the same time
Preset: save all the setting parameter of the equalizer of the current channel to the computer, and recall the channel equalizer parameter of the computer, which can be called across channels and devices.
Copy: copy the current channel equalizer parameter value, which can be pasted to other similar channels (such as input channel parameter can only be copied to other input channels).
Paste: used in combination with the copy button to paste the last copied equalizer parameter value to the current channel.
Reset: reset the equalizer parameter to the default parameter values.

As shown in the figure above, the left side 

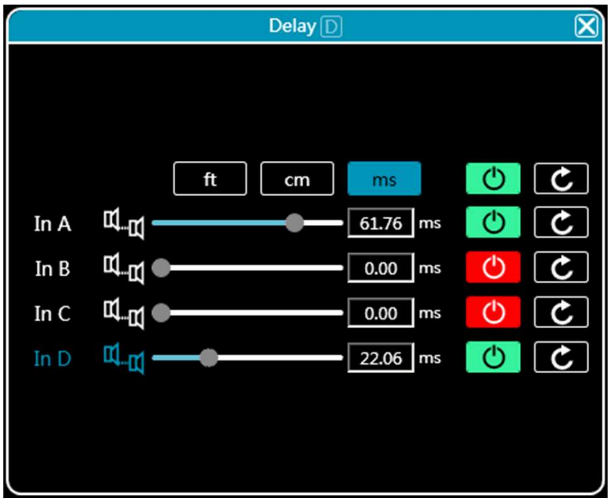
is the interface switching button for each channel. Click to switch the EQ channel, and the color is the currently selected channel.  is the curve color of the EQ channel.  For each channel's EQ curve display switch, check it to enable it to display the curves of other channels in the current channel interface.

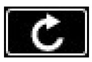
4.3.4 DSP functions setting - DYNAMIC EQ



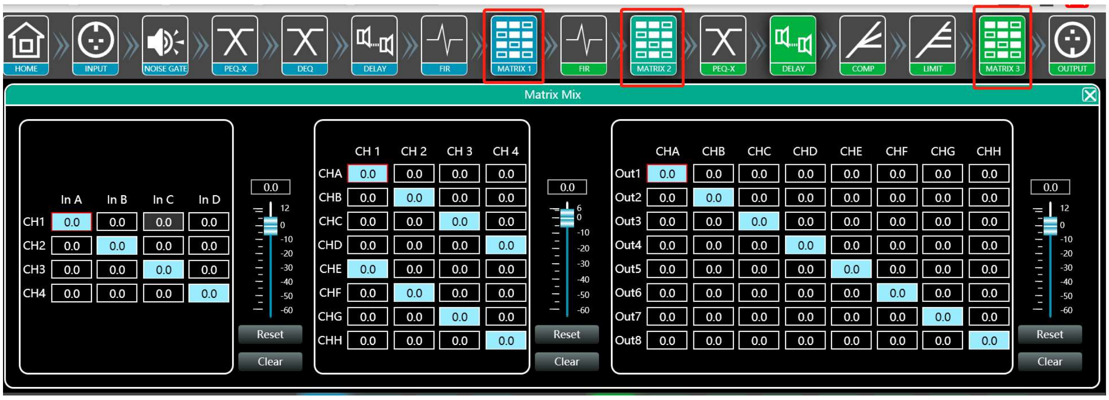
- Mode: Boost Above\Boost Below\Cut Above\Cut Below
- Threshold: -90 to 24.0dBu
- Q: 0.27 to 15
- Ratio: 1.0 to 100.0
- Max Gain: 0.0 to 12.0
- Attack: 1 to 2895ms
- Frequency: 20 to 22000Hz
- Release: 1 to 2895ms
- Type: BYPASS\PEQ
- Bypass button to switch

4.3.5 DSP functions setting - DELAY (input and output)



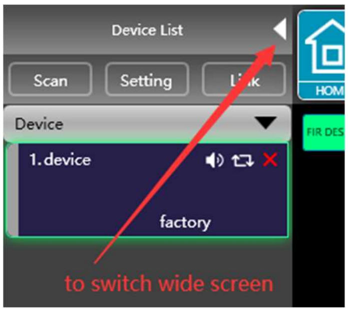
- Max 2000ms for input channel;
- Max 2000ms for output channel;
- Click  to enable this function;
- Click  to reset each channel;
- User can switch ft/cm/ms measurement for delay.

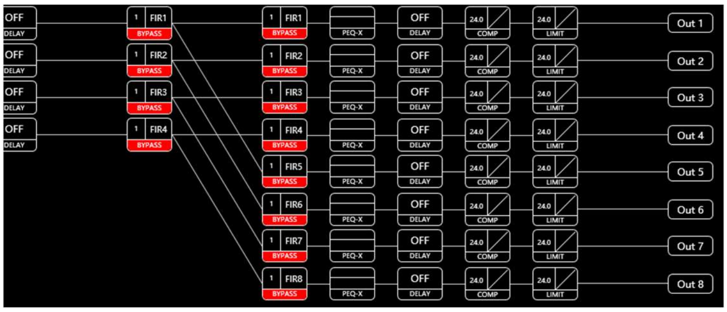
4.3.6 DSP functions setting - MATRIX MIX



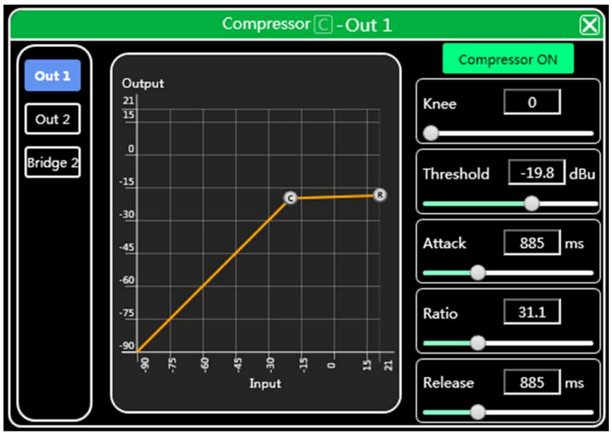
In the above figure, input channels (on top side) corresponds to output channels(on left side). The value box with a value is mixing key of channels. When the mixing key is green (double-click the value box to switch the state), the input channel and output channel signal realizes the mixing function.

The right part of the above figure contains the gain, reset button, and clear button of the matrix mix. Click the value box on the left, and then drag the sliding block of the matrix mix gain or enter a value in the value box to adjust the matrix block Click the reset button to reset the matrix mixing function to the initial one-to-one state; click the clear button to clear all the matrix mixing functions, and there is no correspondence between the input and output of the device.



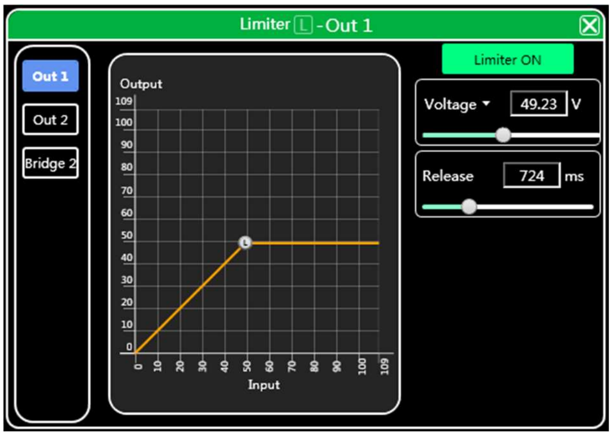


4.3.7 DSP functions setting - COMPRESSOR



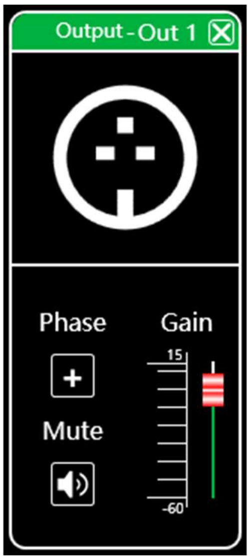
- Soft knee: 0 to 30;
- Threshold: -90.0 to 24.0 dB;
- Attack: 1 to 2895 ms;
- Ratio: 1.0 to 100.0;
- Release: 1 to 2895 ms;
- Click Compressor ON to enable this function;

4.3.8 DSP functions setting - LIMITER



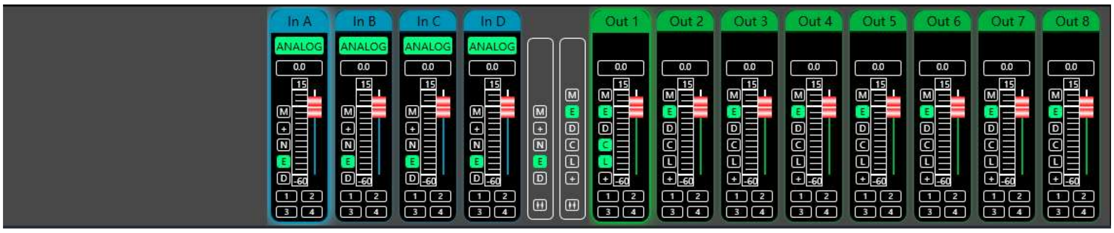
- Threshold: 0.0 to 24.0dBu;
- Release: 1 to 2895 ms;
- Click Limiter ON to enable this function;

4.3.9 DSP functions setting - OUTPUT



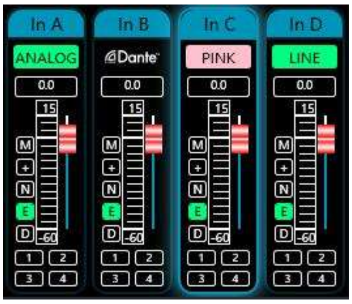
- Set phase of signal;
- Set mute of output channel;
- Set gain of output channel.

4.4 Monitoring and setting of channels



User can monitor gains level of input and output channels.

4.4.1 Channel gain level

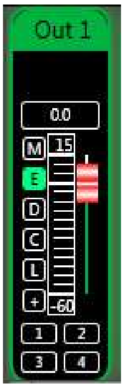


There are 4 kinds of input signal in some products: ANALOG, DANTE network audio, AES digital audio, testing signal. It will show a label for user.

Input value, drag gain fader or use mouse pulley to set value of gain.

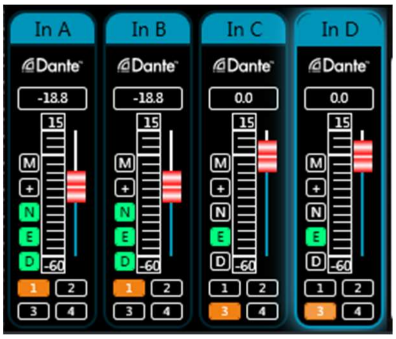
4.4.2 Quick buttons of DSP in channels

M Mute
+ Phase
N Noise Gate
E PEQ
D Delay



M Mute
E PEQ
D Delay
C Compressor
L Limiter
+ Phase

4.4.3 Group and channels link



User can quickly set channels in groups for opening or closing mute, phase, noise gate, PEQ and delay function.

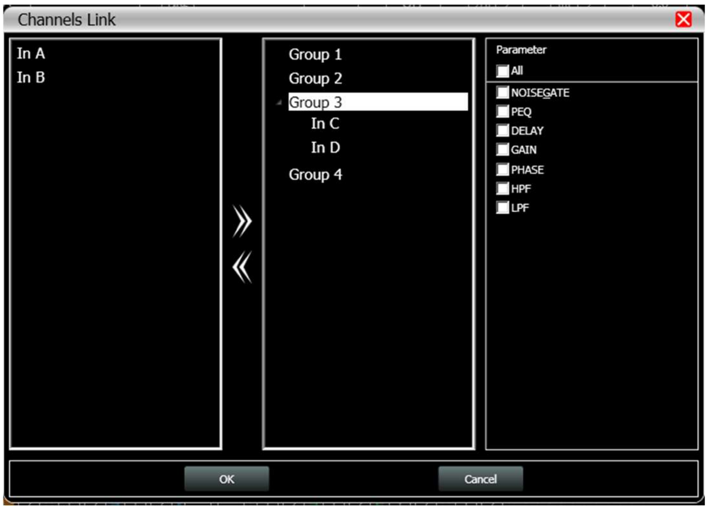
M Mute
+ Phase
N Noise Gate
E PEQ
D Delay

Channels link for input

M Mute
E PEQ
D Delay
C Compressor
L Limiter
+ Phase

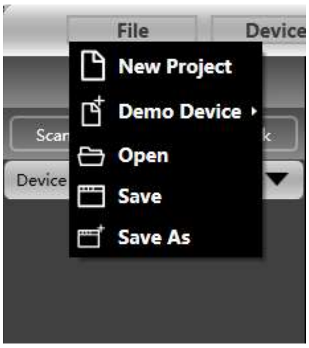
Channels link for output

When click link button, Channels Link window would show as below:



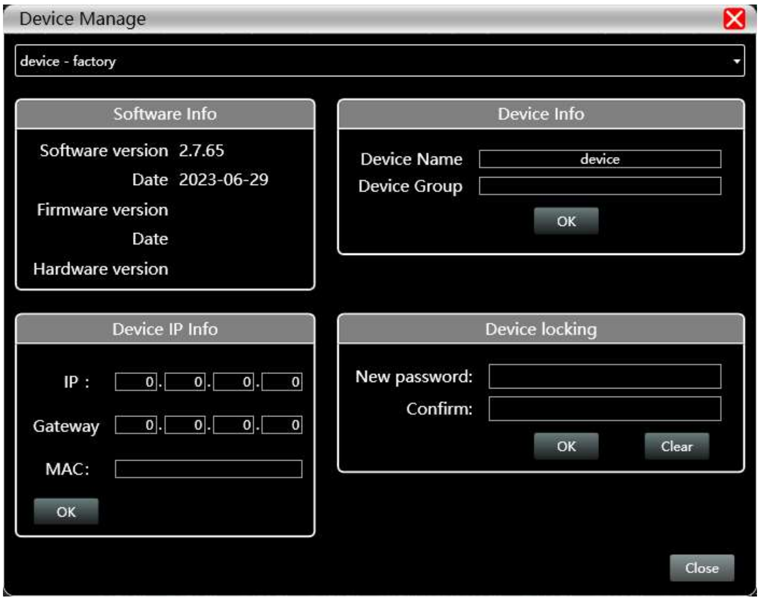
Select the corresponding channels to link, they will be in group for user to set parameter.

4.5 Menu - File

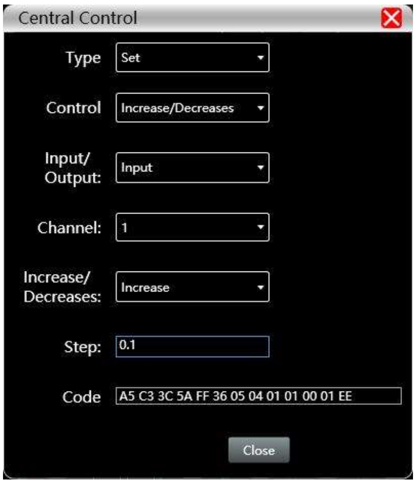


New project: the project is restored to the initial open state.
Demo Device: user can view all the functions of the device without affecting the specific device connected.
Open: open an existing device management project from the computer disk.
Save: save the current equipment management project in the computer disk.
Save as: save the current equipment management project to the computer disk.

4.6 Menu - Device (including Device lock)



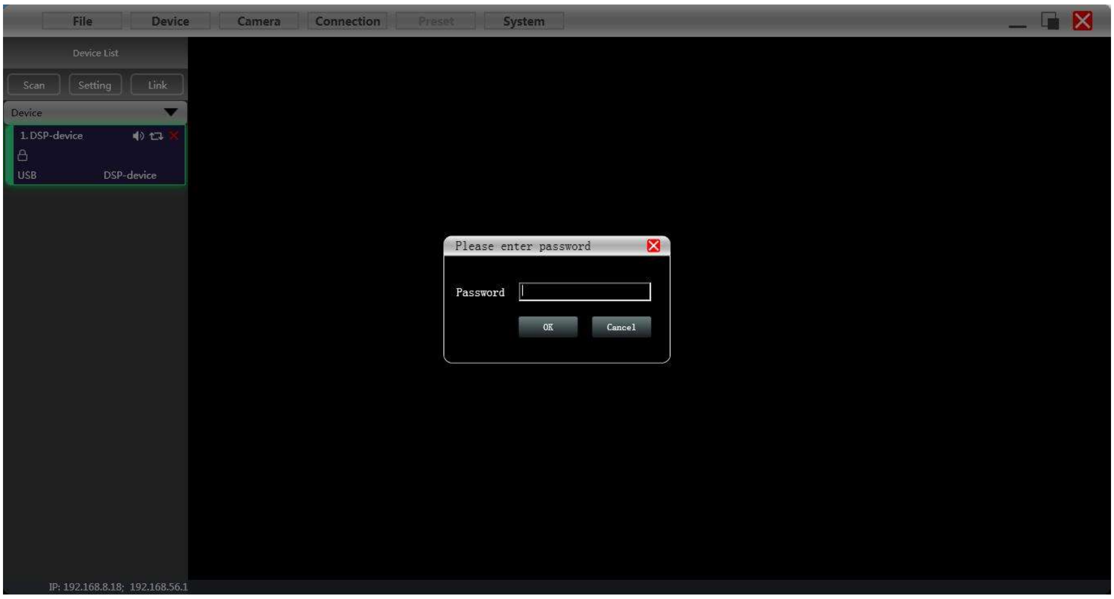
Devices: view or modify the software version information, device name and device IP address of the upper and lower computer of the device. Set password of device.
Channel name: set the name of each input and output channel, with memory function.
Channel copy: copy device input and output channel parameter, can realize cross-device copy parameter (Note: the same type of device is required).
Central control: provides user a quickly way to inquiry code of Center Control setting. More details, please refer to another user manual <Center Control Code User Manual>, it provides whole guide and codes for user to match every specific system.



Device locking User can set his own password of this device to protect audio project after setting parameters. After unlock the device in software, user can clear password or reset the password.

The password can be in four-digit format (**0,1,2...9**), so that user can use the control software or the front panel of device to unlock the password. If the device is locked, there is a icon showed in software and LCD display, as per Figure 7.18.1

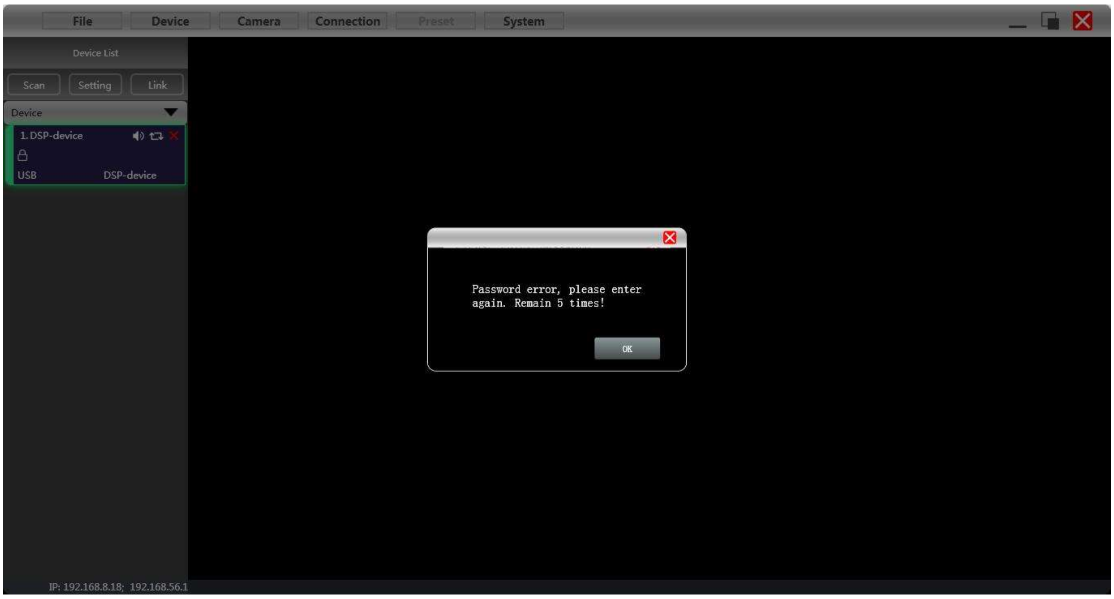
Figure 7.18.1



Input the password in software to unlock the device

Input the password in LCD display to unlock the device

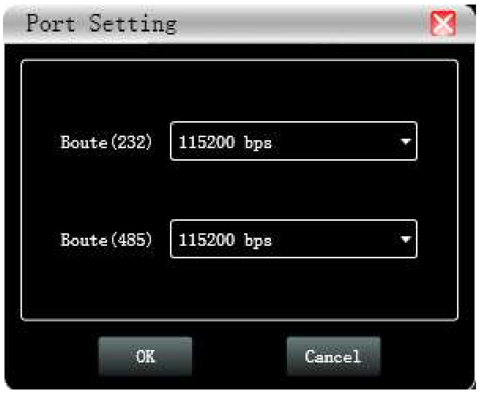
When device is locking, user can press mute button of each channels in front panel. Press either of BACK, NEXT, MENU, ENTER, BYPASS, QUIT button, LCD will show interface to input password, press GAIN button to select digit, and press BACK or NEXT to select digit position. Then select “OK” and press ENTER to unlock device.



If wrong input, software will remind user there is only 5 times to input right password. More than 5 times, device can’t be unlock any more, user have to return this device back to dealer or factory. **The dealer or factory will reset the device and clear all parameter settings.**

Successfully input password and enter main interface

4.7 Menu - Connection



Port: set the connection mode, port number and baud rate, confirm the connection mode and then select the corresponding port.
Connect: connect and download the device parameter.
Disconnect: disconnect the connected device.
Connect all: connect and download the device parameter of all devices in the device list.
Disconnect all: disconnect all connected devices in the device list.

4.8 Menu - Preset

Save: select the saved gear, save all the parameter of the current automatic gear of the machine to the device preset (2~30 Preset bit).
Recall: call the device preset to the current automatic gear position.
Delete: delete the existing preset, the default file cannot be deleted, over written or saved.
Clear: delete all presets in the device.
Boot: select a certain preset, after setting it as the boot file, each time the device is powered on, it will automatically call the save the parameter; the last set parameter need

to be automatically saved, please set the automatic file to the boot file.
Import preset: import a single preset file on the computer.
Export the preset: export all the parameter of the current state to the computer, and generate a single preset file.
Import preset package: import the preset package file containing multiple presets on the computer.
Export preset package: pack multiple presets in the machine's preset into one preset package and export it to the computer.

4.9 Menu - System

Language: multi-language switching, supports simplified, traditional, and ENGLISH.
About: current control software and device firmware version information.
Upgrade: use can upgrade the firmware by using this function, a upgrade *.bin* file should be needed from seller or speaker factory. In general, no need to upgrade the firmware in device. Only there is a bug or new function in software, upgrade function will be used.

4.10 FIR filter and function

4.10.1 FIR filter and applications

When user uses PEQ to adjust audio signal and set a linear magnitude, he can find the phase of signal changed, due to IIR filter. However, DSP products provide user a useful tool FIR filter to adjust audio signal with a linear phase.

Some calculation:

Frequency resolution = Sampling/Taps

Available min. frequency ≈ Frequency resolution*3

Means when use adjust audio signal with 48kHz, 1024 taps, FIR filters will take effect in frequency above 141Hz of audio signal. The taps value more high, the FIR filter curve more steep.

FIR filter processing audio signal will produce a certain delay:

Delay = (1/Sampling Hz)*Taps/2

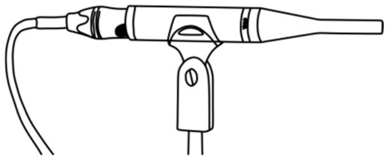
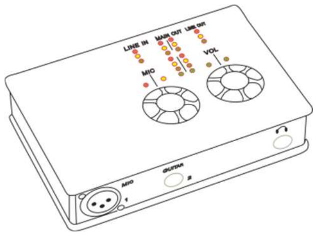
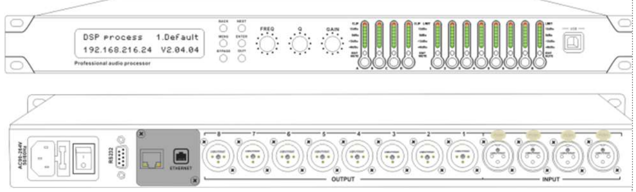
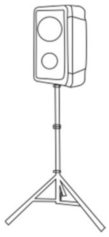
Taps / Sampling	48kHz	96kHz
256	2.67ms, LF 563Hz	1.33ms, LF 1125Hz
512	5.33ms, LF 279Hz	2.67ms, LF 558Hz
768	7.99ms, LF 188Hz	4.00ms, LF 375Hz
1024	10.67ms, LF 141Hz	5.33ms, LF 281Hz
2048	21.33ms, LF 70Hz	10.67ms, LF 141Hz

Applications:

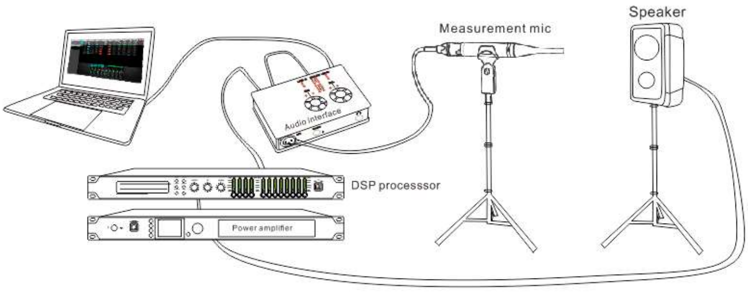
- Linear of the phase curve of the speaker;

- Match the phase and magnitude of different speaker models within the same product line, as well as different speaker models in the installation project to make it easier to debug speaker groups and arrays;
- Dealing with linear array systems (for audience area coverage optimization);
- Frequency division optimization to improve the consistency of frequency response of multi-division speakers over their coverage Angle range.

Devices required:

Measurement Microphone	×1	
Audio Interface	×1	
Windows PC (installed software including Smaart, rePhase or FIR Designer, MusicAllDSP)	×1	rephase1. 4. 2
FIR audio processor or DSP network power amplifier	×1	
Speaker	×1	

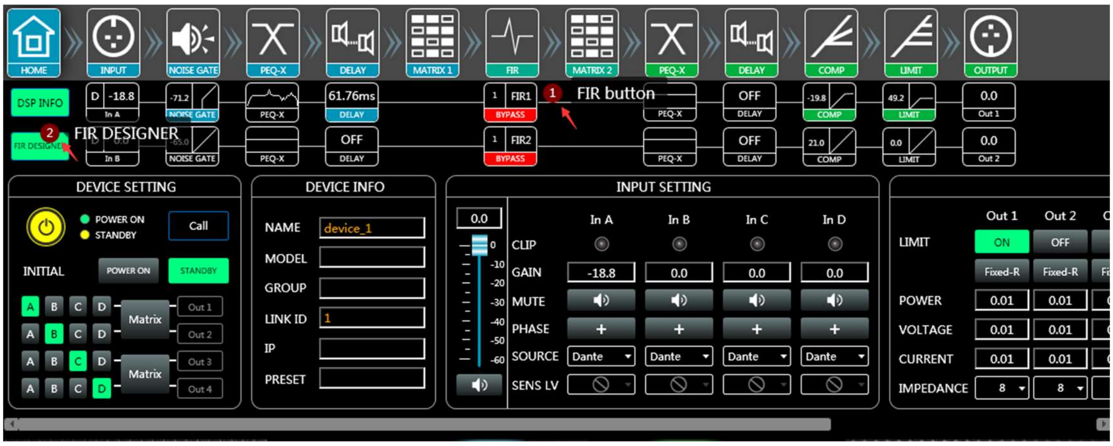
Connection schematic diagram:

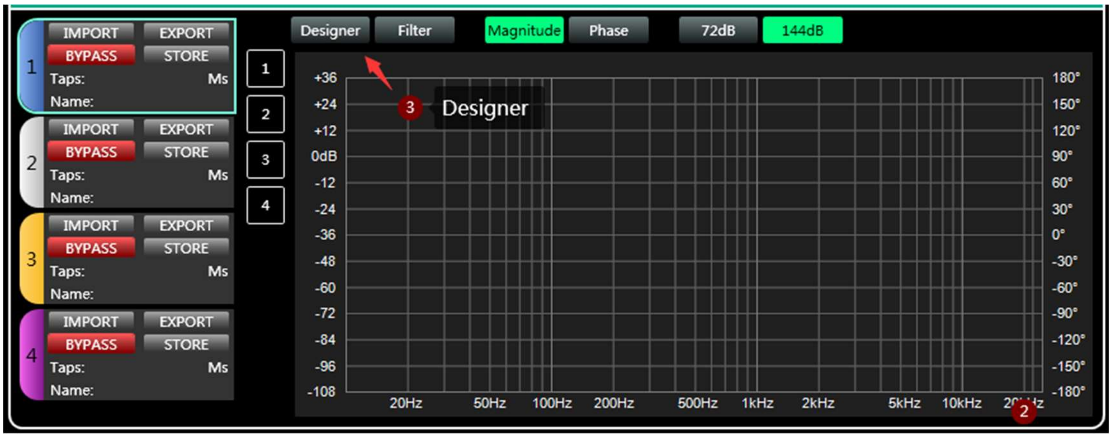


4.10.2 Using FIR DESIGNER in Mconsole to adjust FIR magnitude and phase

Beside using third party software, Mconsole provides user a more convenient way to adjust FIR magnitude and phase of each channels.

There are two ways to open FIR DESIGNER interface:



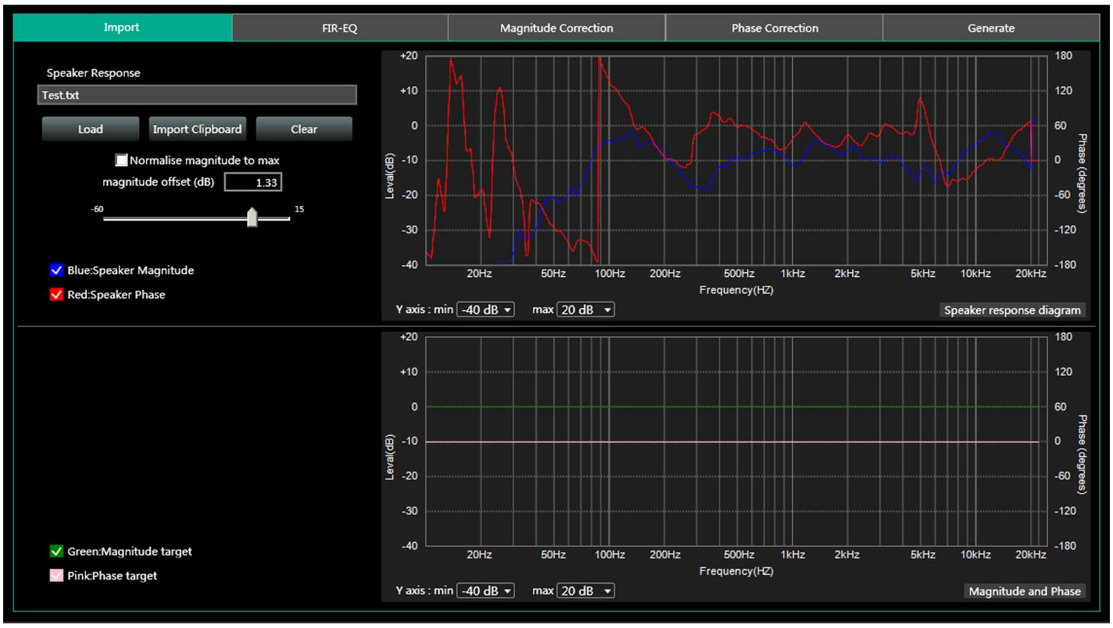


① Click “FIR” - “Designer” button to enter FIR automatic linear magnitude and phase function interface.

② Or click “FIR DESIGNER” in main interface to enter FIR automatic linear magnitude and phase function interface, which can quickly help user return to the page he set last time.

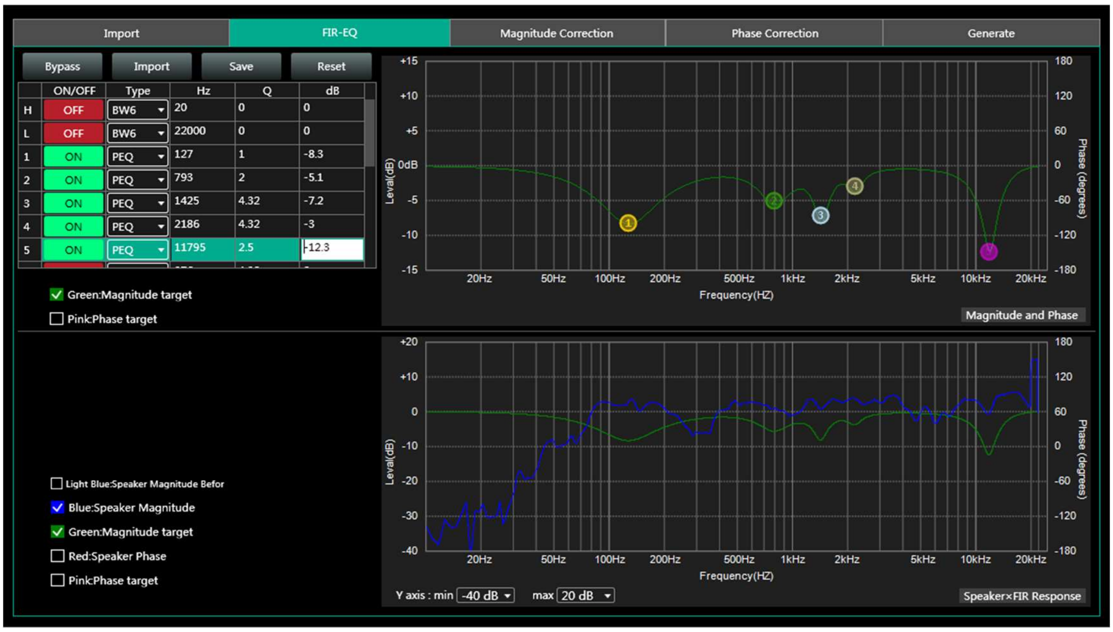
Let’s begin to set:

4.10.2.a FIR DESIGNER - Import



- **Load**: load speaker measurement file from Smaart, usually it's a *.txt* file.
- **Import Clipboard**: load ASCII data directly from Smaart.
- **Clear**: clear measurement data.
- **Normalise magnitude to max** or **Magnitude offset (dB)**: this can help user to adjust a certain dB of magnitude, in order to adjust magnitude curve as little as possible.

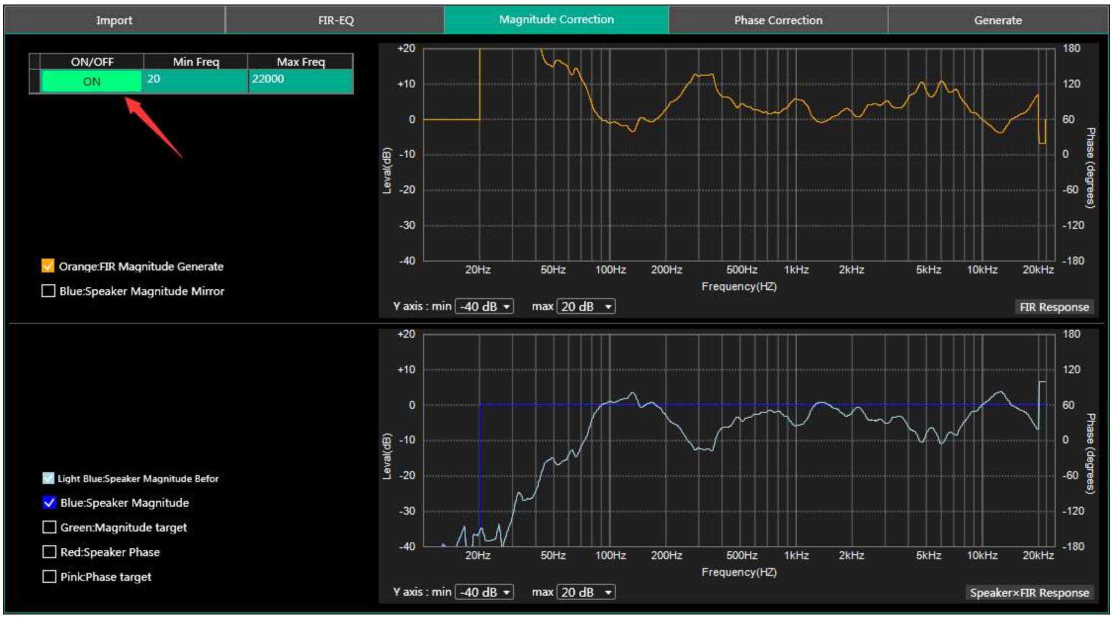
4.10.2.b FIR DESIGNER - FIR-EQ



There are High pass filter and low pass filter for setting frequency divider, and 15 bands of PEQ \ LSLV \ HSLV to adjust magnitude. Try to set a linear magnitude of target speaker. Mark: changing FIR magnitude doesn't effect its phase.

4.10.2.c FIR DESIGNER - Magnitude Correction and Phase Correction

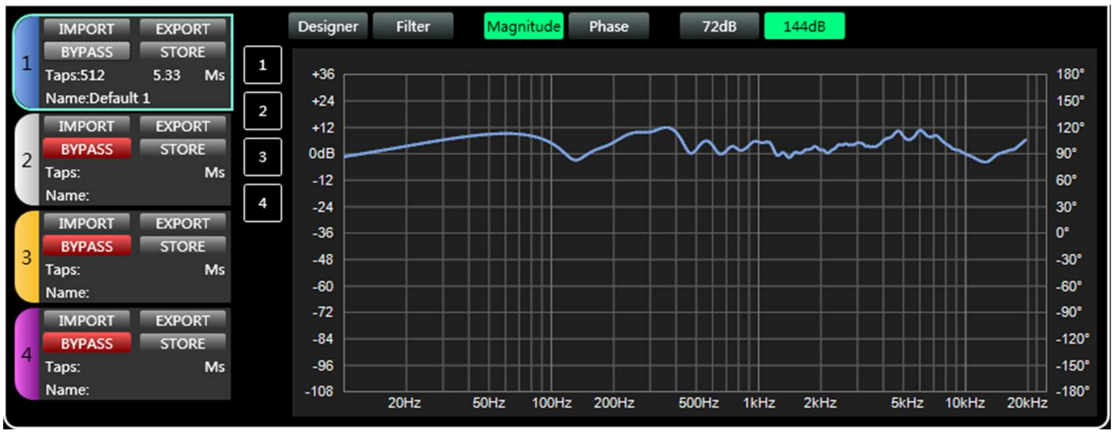
Of course, if there are too many speakers to be adjust, user has to spend a long time manually adjusting their magnitude. In this case, Magnitude Correction will be more useful. Just enable ON button for frequency.



After adjusting magnitude, set linear phase of speaker.

4.10.2.d FIR DESIGNER - Generate

Select **Taps** (such as 512) of this adjustment, and store it in a FIR channel. User can also name this FIR adjustment and export it to a *.KF* file. After finish all setting, return back to FIR interface. Cancel BYPASS button to make it work.



4.11 Filters and application

In parameter EQ, this processor provides a variety of useful filters, including kinds of shelf, pass, and phase filters. Users can make full use of them in actual acoustic engineering.

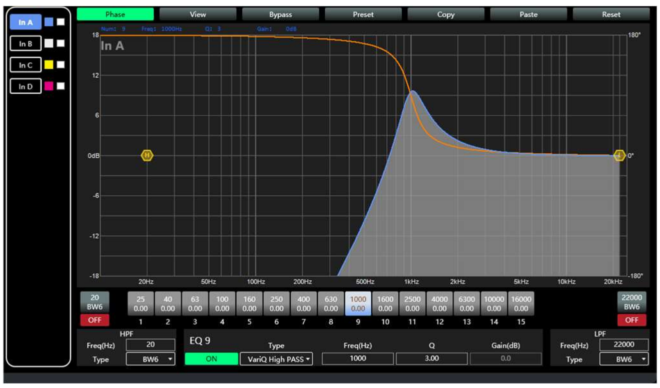
4.11.1 Low shelf / high shelf filter

Low shelf in freq 1kHz, Q 1.00 Gain 5dB
(yellow curve: phase)
Application: musical instruments

Using a shelf equalizer will allow more frequencies to remain intact, but still decay to a more acceptable level. This helps user retain the original sound, as well as the overall sound.

- ❖ Freq (Hz): 20 to 22kHz
- ❖ Q: 0.25 to 1.00
- ❖ Gain: -15.0 to 15.0 dB

4.11.2 LPF / HPF / Variable Q pass / Elliptic pass / Band pass / LP-1 / HP-1 filter



VariQ high pass in freq 1kHz, Q 3.0
(yellow curve: phase)

Using a shelf equalizer will allow more frequencies to remain intact, but still decay to a more acceptable level. This helps user retain the original sound, as well as the overall sound.

- ❖ Freq (Hz): 20 to 22kHz
- ❖ Q: 0.40 to 128.00

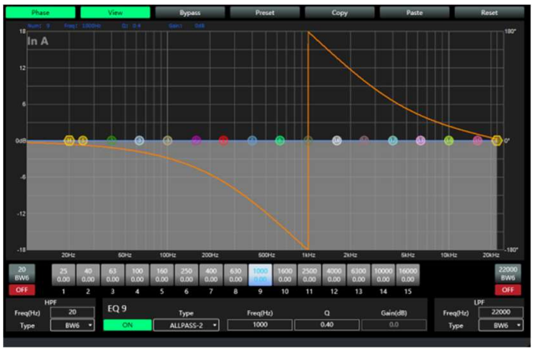
4.11.3 Allpass-1 / Allpass-2 / Phase filter

Phase filter in freq 1kHz, Q 90 degree
(yellow curve: phase)

Phase filter is a very useful filter type, without changing the frequency response, the user can adjust the original phase curve, optimize the phase coupling of part of the frequency band.

❖ Freq (Hz): 20 to 22kHz
❖ Q: 0.25° to 179.00 °
❖ Q value for setting degree

Allpass-1 *in freq 1kHz*
(yellow curve: phase, 90 degree)
Application: different speaker



Allpass-2 *in freq 1kHz, Q 4.32*
(yellow curve: phase, 180 degree)
Application: different speaker

4.11.4 Band stop - Notch filter

Using a shelf equalizer will allow more frequencies to remain intact, but still decay to a more acceptable level. This helps user retain the original sound, as well as the overall sound.

❖ Freq (Hz): 20 to 22kHz
❖ Q: 0.25 to 128.00

Notch filter in freq 1kHz, Q 4.32
(yellow curve: phase)