LM4080A

PROFESSIONAL AUDIO PROCESSOR



OPERATING MANUAL

ATTACK
AUDIO SYSTEM
HEAR**THE**DIFFERENCE

PRESENTATION

Thank you for choosing our audio equipment. Attack, always committed to its customers, offers products at a fair price, with high quality and reliability.

You have just acquired a device that has been designed to provide you with years of satisfaction. For this reason, please read this manual carefully before turning on your unit. Follow the connection and operation instructions contained in this manual correctly. If you have any questions, please contact us. Our address and phone number can be found on the last page.

Our products undergo strict quality control up to the final packaging stage. It is very unlikely that you will encounter any issues, but if this happens, please notify your dealer immediately. Attack will make every effort to resolve any problem that is its responsibility. Customer satisfaction is our top priority.

UNPACKING THE PRODUCT

Upon receiving your unit, open the package and check that everything is in order. If you find any irregularities, immediately notify the carrier or your dealer. Such damages are most likely caused by transportation or storage issues.

Keep the original packaging of this unit, along with all its accessories, and use it whenever you need to transport it. This packaging ensures better protection for the product, preventing further damage to the unit.



The “exclamation mark” symbol inside the triangle is intended to alert the user to the presence of operating and maintenance instructions.



The “lightning bolt” symbol inside the triangle warns of the presence of uninsulated components with dangerous voltages.



The “droplet” symbol inside the triangle warns of the risk of fire or electric shock if the equipment is exposed to rain or highly humid environments.



The “fan” symbol inside the triangle warns of possible operational issues if the ventilation openings are obstructed.



The “AC connector” symbol inside the triangle warns about the need to check the mains voltage before turning on the equipment.



The “hot surface” symbol inside the triangle alerts to the presence of a high-temperature surface that must never be touched.

• To prevent electric shock, do not open this equipment. In case of problems, contact authorized service agents.

• To prevent the risk of fire or electric shock, do not expose this equipment to rain or extremely humid environments.

• After turning off your equipment, wait 30 seconds before handling input or output connections.

• Always check the mains voltage before turning on this equipment.

• Always use the grounding (earth) pin on the AC connector. This is a fundamental safety feature of the equipment.

• Do not subject this product to extreme temperature conditions. Never leave it exposed to direct sunlight or near heat sources such as stoves, radiators, heaters, etc.

• Avoid impacts. This equipment has a robust mechanical construction; however, strong impacts may damage internal components or even the external enclosure.

• For cleaning, use clean, slightly damp cloths. Never use solvents, as they may damage the product’s finish, including paint and plastic parts.

• Use high-quality cables and connections. We recommend those manufactured by Wireconex (www.wireconex.com).

INTRODUCTION

The LM4080A DSP FIR audio processor features 4 inputs and 8 outputs, integrating a high-performance DSP, Dynamic EQ, automatic linear phase correction via FIR, and other advanced functions. The processor includes both analog and digital (AES3) inputs.

The standard FIR Designer, integrated into the LMControl+ software, provides automatic magnitude and phase correction, allowing users to configure transducer parameters in a simple and efficient way.

With RJ45, USB, and RS232 connections, the LMControl+ software for Windows enables easy control of multiple devices.

Applications

- Party venues
- Live shows
- Outdoor events
- Sports stadiums

Key Features

- 4 analog inputs and 8 analog outputs
- Each input can be configured for AES3 signal
- High-performance DSP with 96 kHz / 24-bit sampling rate
- 15 bands of PEQ on inputs and 10 bands of PEQ on outputs
- HPF and LPF filters: Butterworth, Bessel, and Linkwitz-Riley
- LSLV, HSLV, All-Pass, Band-Pass, Band-Stop, Notch, and Phase 1 and 2 filters
- 3 bands of dynamic EQ on inputs
- Linear phase FIR: 512 taps (48 kHz) on inputs and outputs
- FIR Designer with automatic linear phase correction
- Presets with storage and lock functions
- Control connections: USB, TCP/IP, and RS232
- LMControl+ software compatible with Windows 7/8/10/11
- LED indicators for level, clip, mute, and edit

TECHNICAL SPECIFICATIONS

Item	Specification
DSP Processor	
Processor	ADI SHARC 21489 – 450 MHz
System latency	2.1 ms
AD/DA Conversion	24 bits / 96 kHz

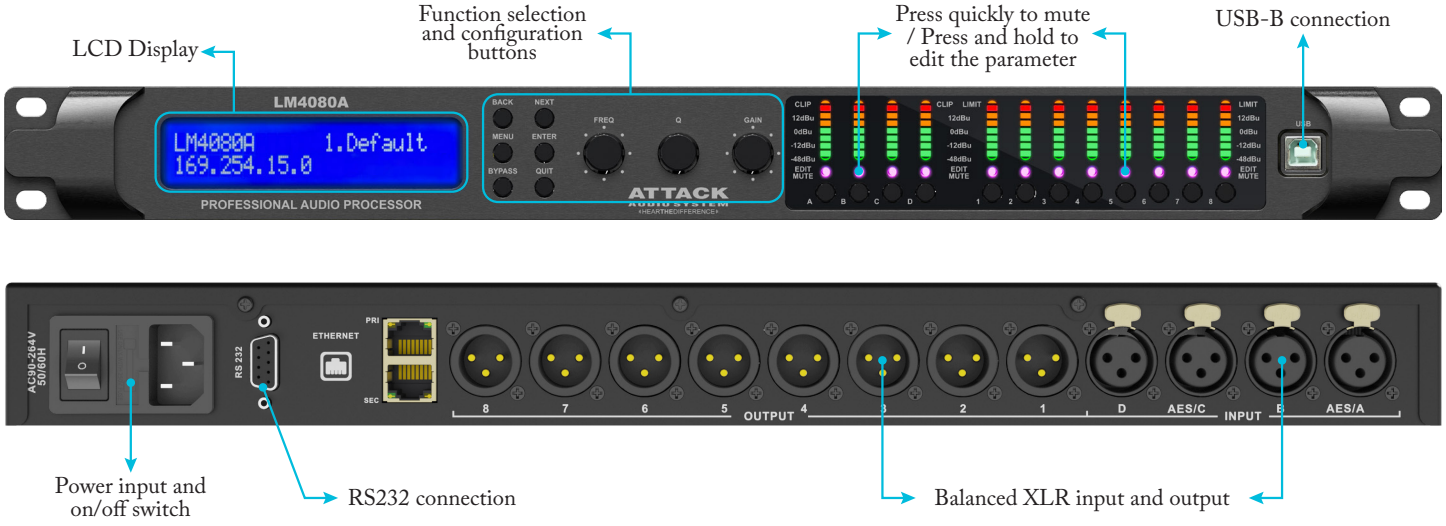
Analog Audio Inputs and Outputs	
Input	4 balanced channels
Input interface	XLR
Input impedance	20 kΩ
Maximum input level	+20 dBu / Line
Output	8 balanced channels
Output interface	XLR
Output impedance	50 Ω

Audio Performance Specifications	
Frequency response	20 Hz – 20 kHz (±0,5 dB) / Line
THD+N	–93 dB (@0 dBu, 1 kHz, A-wt) / Line
Noise floor	20 Hz – 20 kHz, A-wt, -93 dBu
Signal-to-noise ratio (SNR)	113 dB (@16 dBu, 1 kHz, A-wt) / Line

Connection Ports and Indicators	
USB	Type A–B, no driver required
RS232	Communication via serial port
TCP/IP interface	RJ-45
LED indicators	Clip, Level, Edit, Mute

Electrical and Mechanical Characteristics	
Power supply	AC 100 V – 240 V / 50–60 Hz
Product dimensions	483 × 265 × 44.5 mm
Package dimensions	537 × 343 × 77 mm
Net weight	3.8 kg
Gross weight	4.2 kg
Operating temperature	–20 °C to 80 °C

FRONT AND REAR PANEL FUNCTIONS



Front operation panel



1- Press MENU to display the list of functions; use NEXT or BACK to select the functions: GLOBAL MEMORY, INPUT SECTION, MATRIX, SYSTEM; press QUIT to exit.

Panel Functions	Menu List	Notes
1.GLOBAL MEMORY	RECALL a Memory	
	STORE a Memory	
	DELETE a Memory	
2.INPUT SECTION	A ANALOG-AES	
	B ANALOG-AES	
	C ANALOG-AES	
	D ANALOG-AES	
3.MATRIX	Routing Out.1=Input A*....	Press and hold the corresponding output channel button below the LED to change.
4.SYSTEM	1 IP SET	
	2 RENAME	
	3 DSP VERSION	

2- Press BYPASS to quickly display the RECALL a Memory function. Use NEXT or BACK to select the presets, then press ENTER to activate one of the presets.

Front Panel Button Operation – Input and Output Channels



2- When the user needs to adjust the parameters of an input or output channel, press and hold the button located below the LED corresponding to that channel; the LED will turn blue.



Input Channel Functions	Configuration Buttons	Configuration Buttons
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

Output Channel Functions	Configuration Buttons	Configuration Buttons
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

Note: "X" indicates the channel number selected by the user. If there is no effect after adjusting a parameter, check whether the BYPASS function is enabled.

3- When the user needs to link channels and adjust their parameters, press and hold the button located below the LED of each channel; the LED will turn blue. The LCD display will show "IPX+" or "OPX+", indicating that the second channel or additional channels will be adjusted with the same parameters as the first channel.



When the user needs to quickly mute channels in link mode, briefly press the button located below the LED of one of the channels; all LEDs will turn pink.



LMControl+ Software Operation

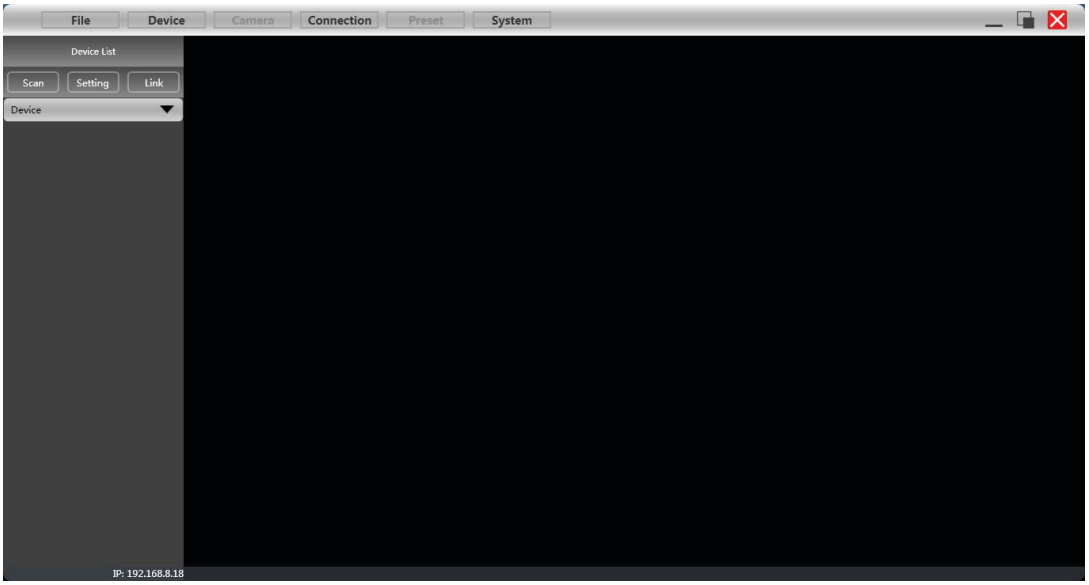
The **LMControl+** software provides users with fast interaction to control one or more devices through various methods: TCP/IP, USB, and standard serial port (RS232). It allows easy configuration of the device's DSP functions and enables monitoring of central control codes. Configuration parameters can be stored as presets, making it convenient for use in different applications.

Operating Conditions

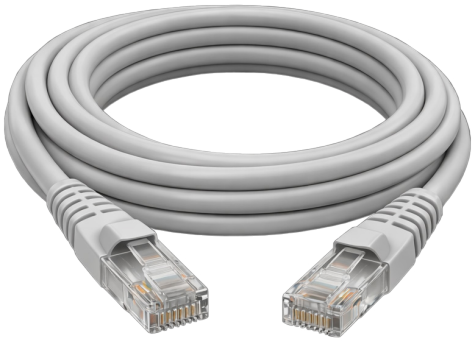
LMControl+ is compatible with Windows 7 / 8 / 10 / 11 (x86/x64) and requires Microsoft .NET Framework 4.0.

When connecting the device via USB, it will be automatically recognized as a storage drive; the user can extract the software directly in Windows without the need for installation.

Double-click **LMControl+.exe** to launch the main interface.



Connection Setup





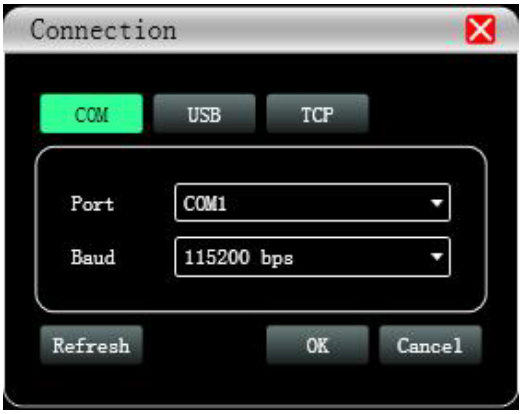
If the device is connected via network cable, click Setting in the Device List and select TCP in the Connection window.





If the device is connected via USB B, click Setting in the Device List and select USB in the Connection window.





If the device is connected via serial port, click Setting in the Device List and select COM in the Connection window. Carefully check the port and the transmission rate (baud rate) for RS232 or RS485 before confirming the configuration.

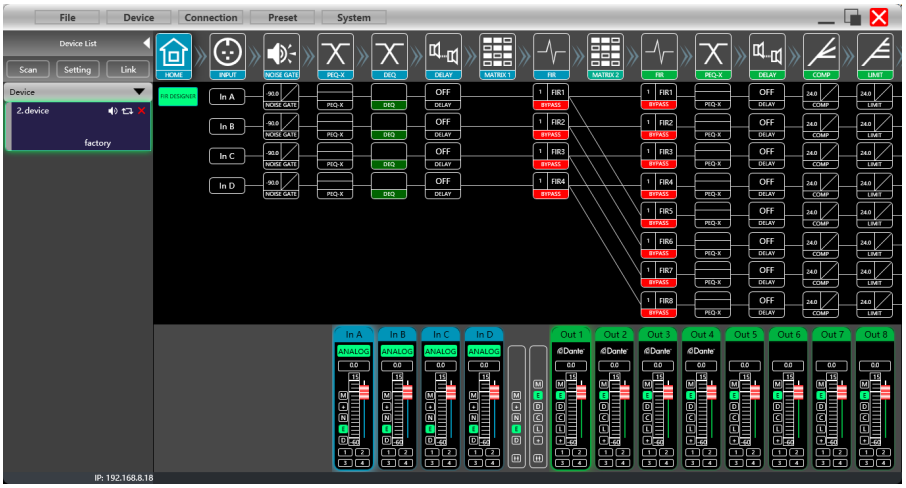
The software will search for the device using the previously configured connection method to verify whether the equipment is connected. If the connection is successful, the devices will be displayed in the device list.

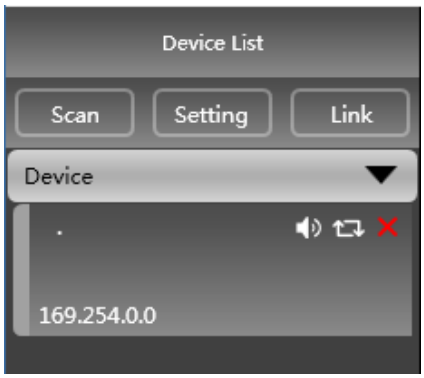


In this window, the user can mute the device, refresh the connection, or remove the device from the list.

By clicking once on the device, the function interface will be loaded.





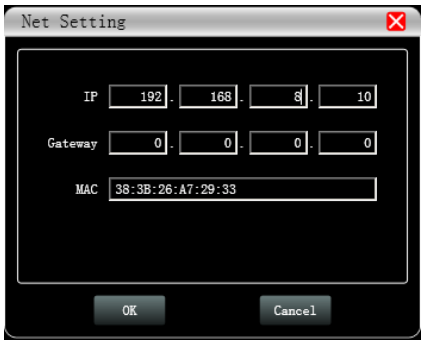


When using TCP control, it may happen that only one point is displayed after scanning, without being able to connect to the device. In this case, it is necessary to adjust the device's IP address so that it is on the same network segment as the computer.

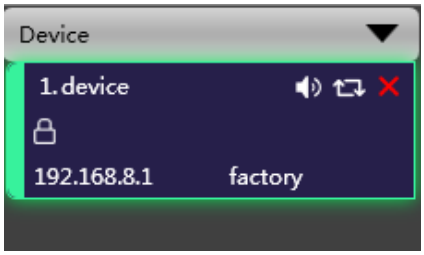
Right-click on the device; the Net Setting window will be displayed. Select Change IP.



Configure the device's IP address based on the IP shown at the bottom of the software. The first three blocks of the IP address must be the same as those of the computer's IP.



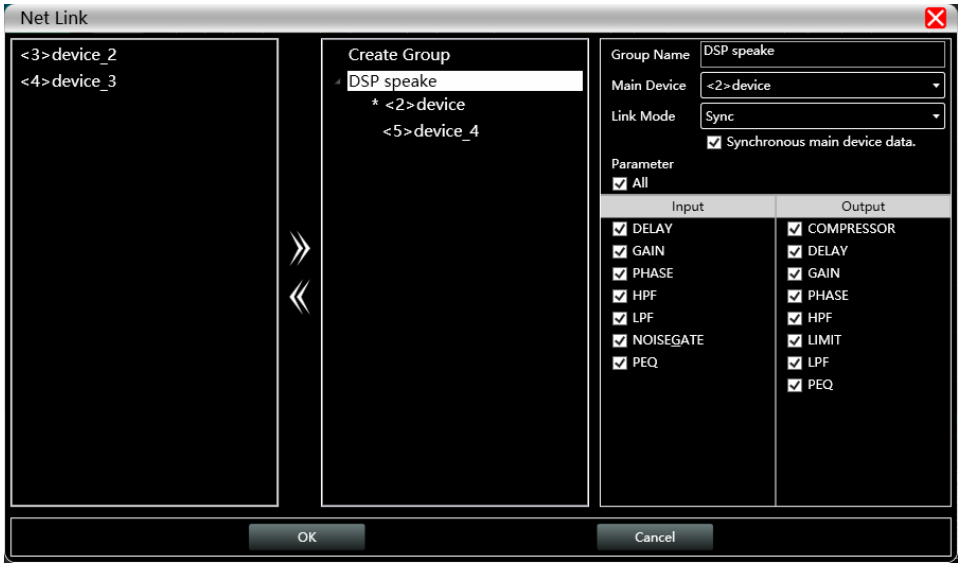




Scan completed successfully and device connected.

The user can link multiple devices of the same model into a group by clicking the Link button, and then define the group device, group name, main device, link mode, and parameters as needed.



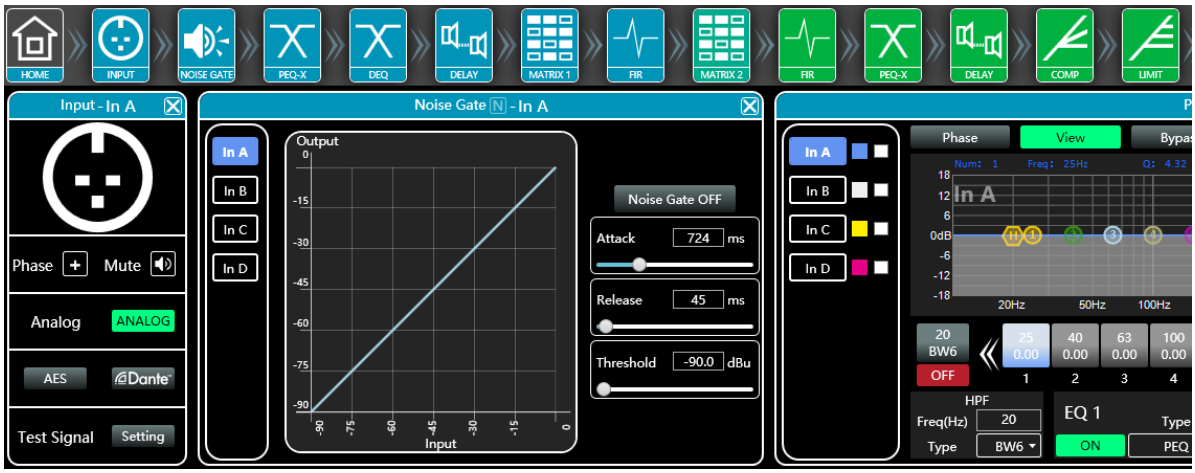


DSP Function Configuration



Double-click the HOME icon to open all functional interfaces, or double-click a specific function icon to open only its corresponding interface.

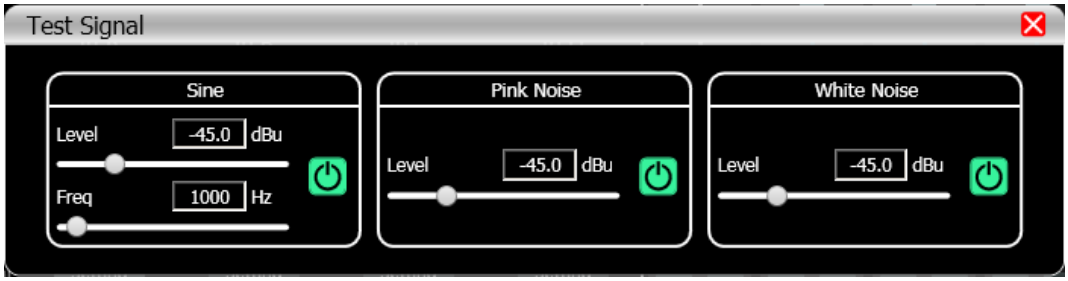
When multiple function windows are open, the user can drag the windows to switch between function settings.



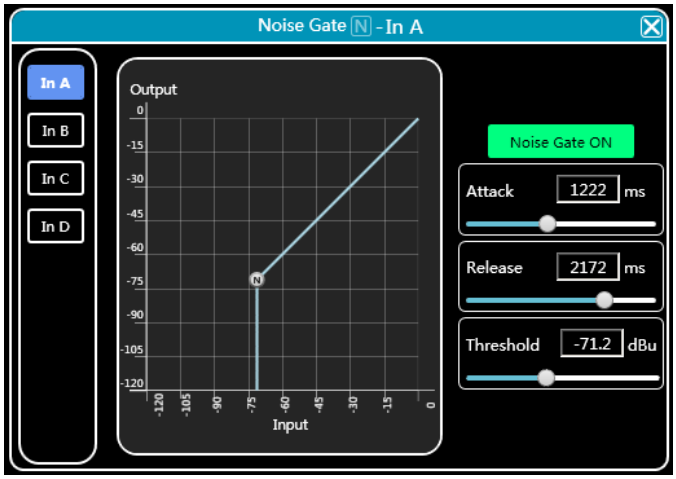
DSP Function Configuration – Input



- Adjust the input phase;
- Enable or disable input mute;
- Select the input signal: Analog / AES3;
- When selecting the test signal, the user can choose Sine, Pink Noise, or White Noise for each input channel.

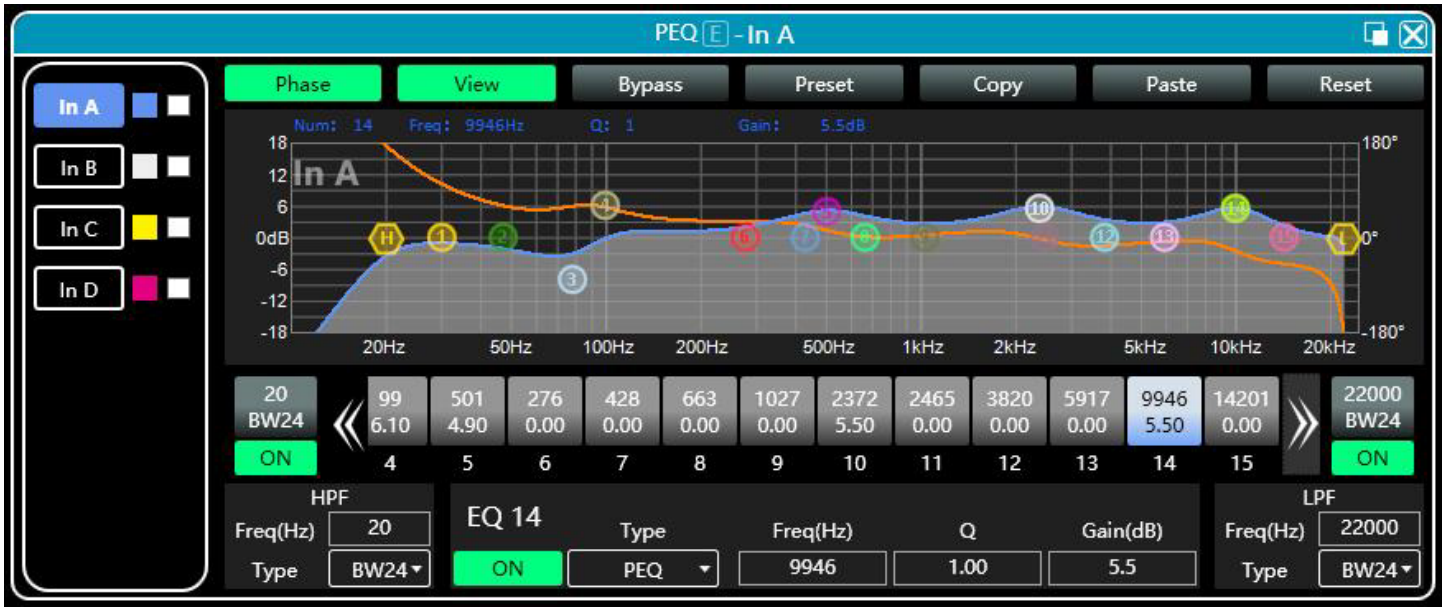


DSP Function Configuration – NOISE GATE



- Attack: 1 to 2895 ms;
- Release: 1 to 2895 ms;
- Threshold: –90 to 0 dBu;
- Click “Noise Gate ON” to enable this function.

DSP Function Configuration – PEQ-X (Inputs and Outputs)



High-Pass Filter (HPF)

Enter the frequency value and select the filter type; press ON to activate this function:
Butterworth 6/12/18/24/36/48, Bessel 12/24/36/48, Linkwitz-Riley 12/24/36/48.



Low-Pass Filter (LPF)

Enter the frequency value and select the filter type; press ON to activate this function:
Butterworth 6/12/18/24/36/48, Bessel 12/24/36/48, Linkwitz-Riley 12/24/36/48.

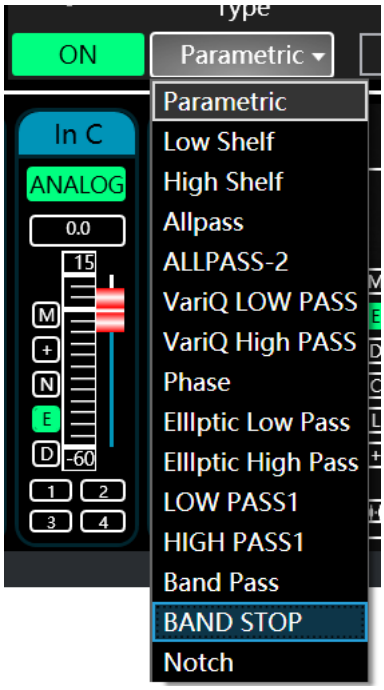


PEQ 15 bands for input channels and 10 bands for output channels.

Type: PEQ / LSLV / HSLV / ALLPASS-1 / ALLPASS-2 / three types of high-pass and low-pass filters / Phase / Band-pass / Band-stop / Notch filter;

Freq (Hz), Q, Gain (dB): enter values manually or use the mouse wheel to adjust;

Users can also drag the frequency point on the curve to make adjustments.



• **Phase Curve:** displays the phase curve of the current channel.

• **View:** shows or hides all control points of the equalization curve.

• **Bypass:** enables or disables all EQs of the current channel simultaneously.

• **Preset:** saves all equalization parameters of the current channel to the computer and allows recalling them, which can be applied to different channels and devices.

• **Copy:** copies the equalization parameters of the current channel, which can be pasted into other channels of the same type (e.g., input parameters can only be copied to other input channels).

• **Paste:** used together with the Copy button to paste the last copied equalization parameters into the current channel.

• **Reset:** restores the equalization parameters to the factory default values.



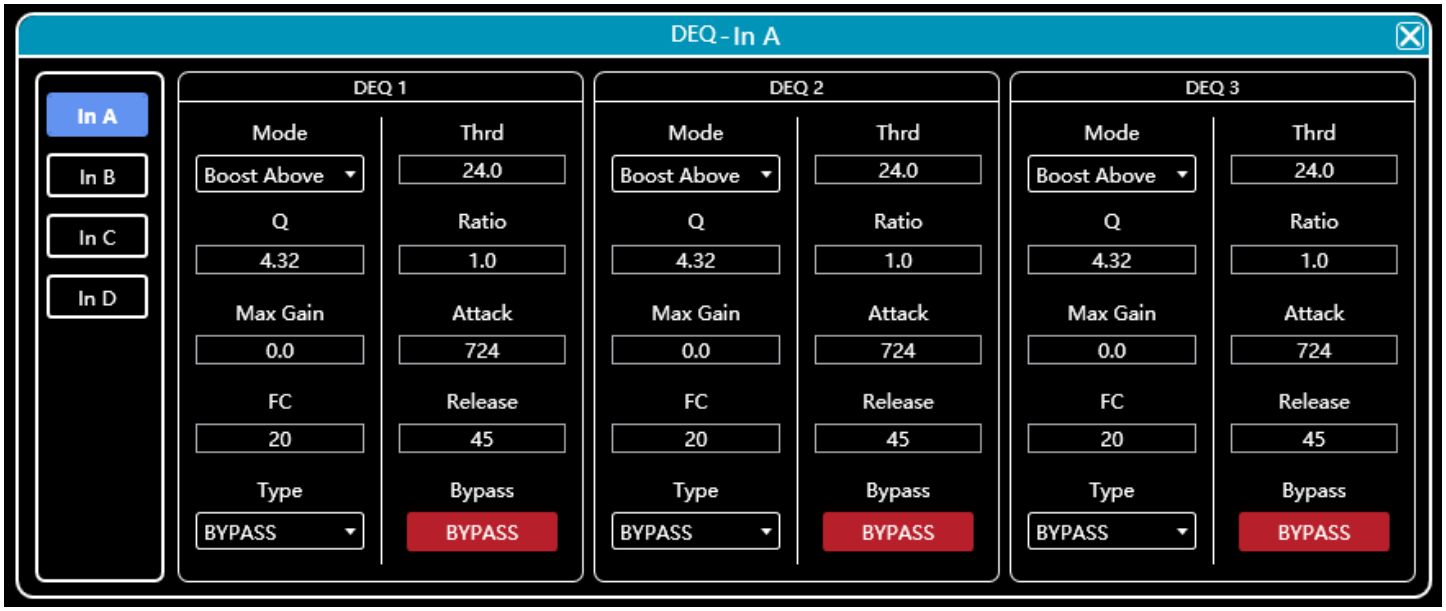
As shown in the figure above, on the left side are the interface selection buttons for each channel. Click to switch the EQ channel; the color indicates the currently selected channel.



The color of the curve represents the corresponding EQ channel.

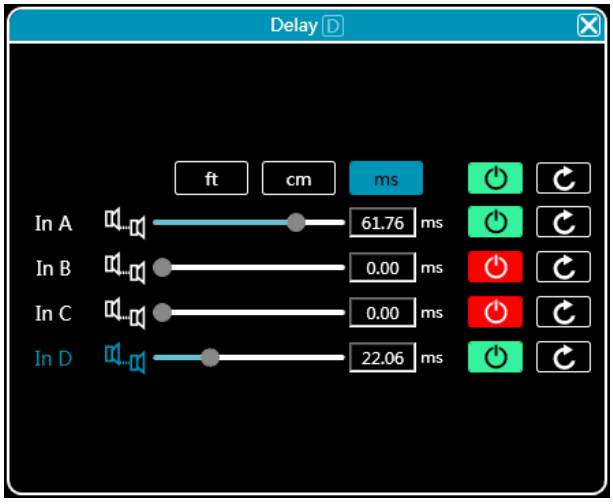
For each channel, the EQ curve display control allows enabling the visualization of curves from other channels simultaneously within the current channel interface.

DSP Function Configuration – DYNAMIC EQ



- Mode: Boost Above / Boost Below / Cut Above / Cut Below
- Threshold: -90 to 24.0 dBu
- Q: 0.27 to 15
- Ratio: 1.0 to 100.0
- Max Gain: 0.0 to 12.0 dB
- Attack: 1 to 2895 ms
- Frequency: 20 to 22,000 Hz
- Release: 1 to 2895 ms
- Type: BYPASS / PEQ
- Bypass button to enable or disable the function

DSP Function Configuration – DELAY (Input and Output)



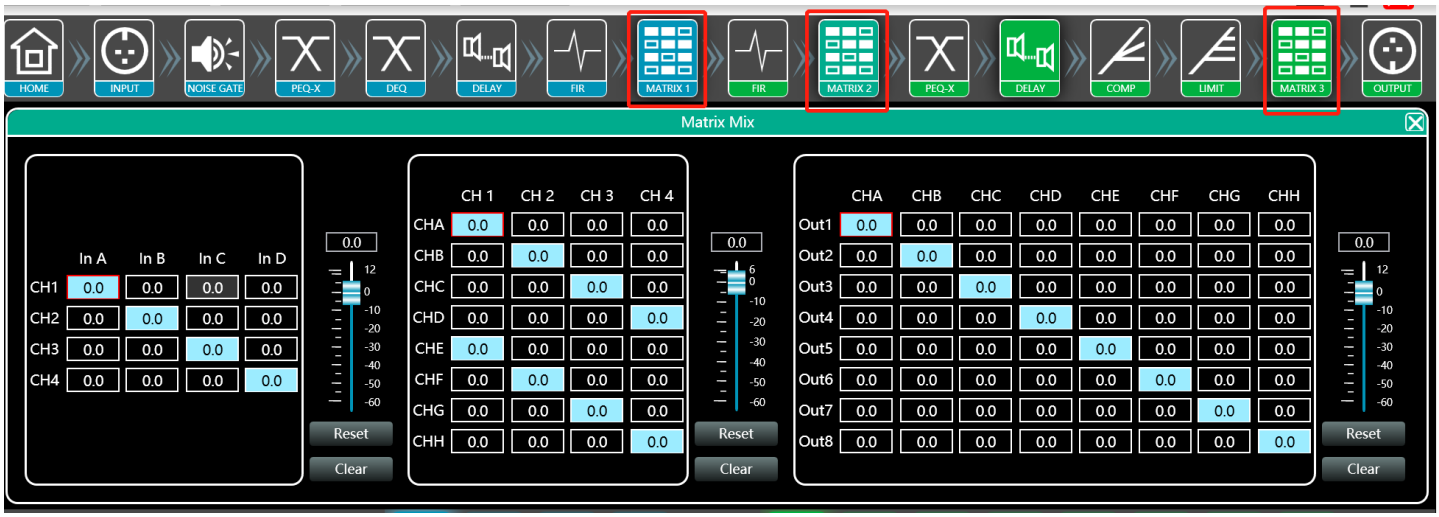
- Maximum of 2000 ms for input channels;
- Maximum of 2000 ms for output channels;
- Click to enable this function;



- Click to reset each channel;

• The user can switch the delay measurement unit between feet (ft), centimeters (cm), or milliseconds (ms).

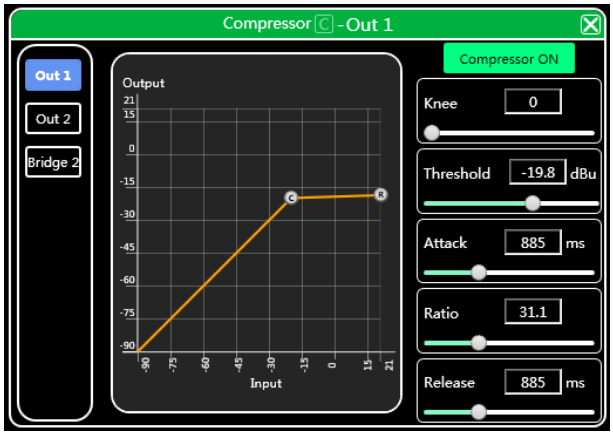
DSP Function Configuration – MATRIX MIX



In the figure above, the input channels (top) correspond to the output channels (left side). The value box with a number is the channel mixing switch. When the mixing switch is green (double-click the value box to toggle the state), the input and output channels perform the mixing function.

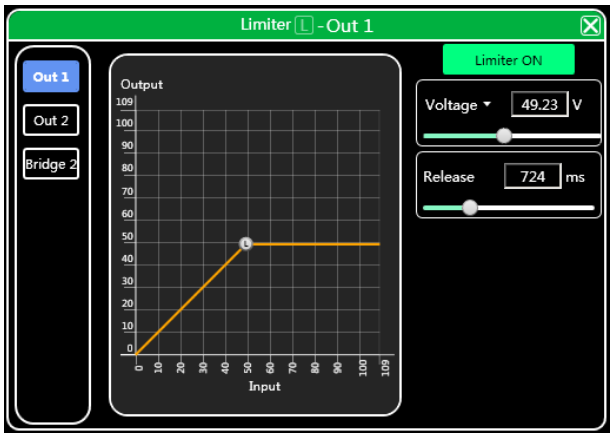
On the right side of the figure are the gain control, reset button, and clear button for the Matrix Mix. Click the value box on the left, then drag the Matrix Mix gain slider or enter a value in the box to adjust the matrix block. Click the reset button to restore the matrix mixing function to the initial one-to-one state; click the clear button to remove all matrix mixing functions, leaving no routing between the device's inputs and outputs.

DSP Function Configuration – COMPRESSOR



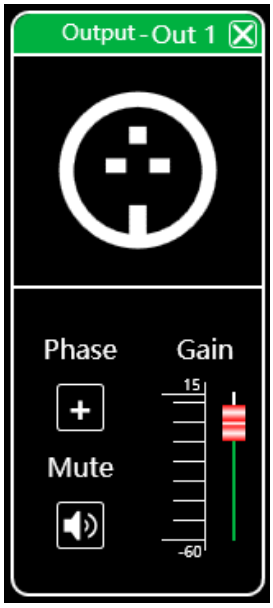
Soft knee: 0 to 30;
• Threshold: -90.0 to 24.0 dB;
• Attack: 1 to 2895 ms;
• Ratio: 1.0 to 100.0;
• Release: 1 to 2895 ms;
• Click "Compressor ON" to enable this function;

DSP Function Configuration – LIMITER



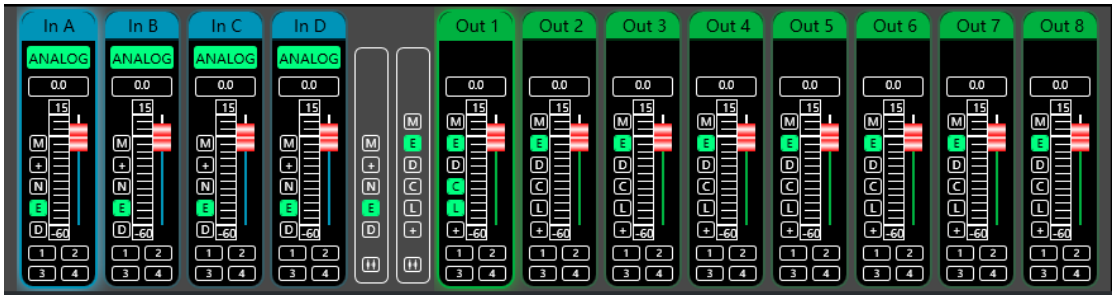
• Threshold: 0.0 to 24.0 dBu;
• Release: 1 to 2895 ms;
• Click "Limiter ON" to enable this function;

DSP Function Configuration – OUTPUT



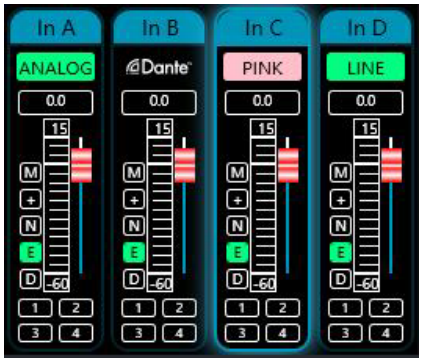
• Adjust the signal polarity;
• Adjust the output channel mute;
• Adjust the output channel gain.

Channel Monitoring and Configuration



The user can monitor the gain levels of input and output channels.

Channel Gain Level



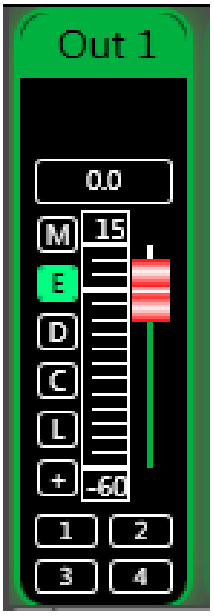
Some products feature 3 types of input signals: ANALOG, AES digital audio, and test signal. A label will be displayed for the user.

Input Value: drag the **gain fader** or use the **mouse wheel** to set the gain value.

DSP Quick Buttons on Channels

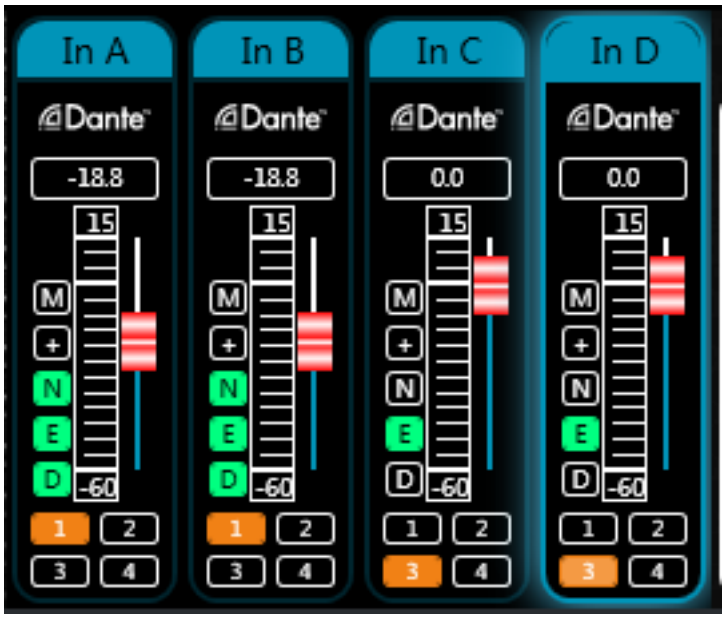


M Mute
+ Polarity
N Noise Gate
E PEQ
D Delay

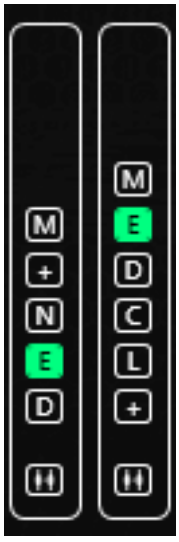


M Mute
E PEQ
D Delay
C Compressor
L Limiter
+ Polarity

Group and Channel Linking



The user can quickly configure channels into groups to enable or disable functions such as mute, polarity, noise gate, PEQ, and delay.



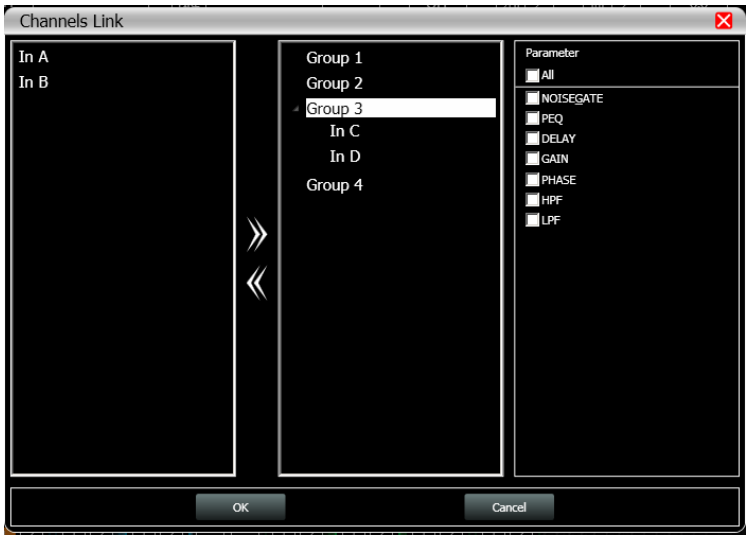
M Mute
+ Polarity
N Noise Gate
E PEQ
D Delay

Channel Linking for Inputs

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

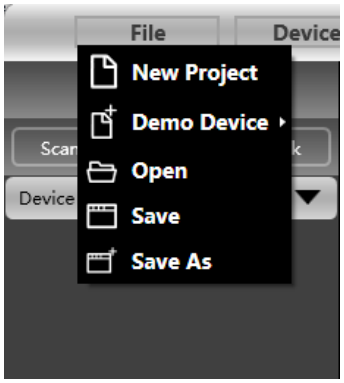
Channel Linking for Outputs

When clicking the Link button, the Channels Link window will be displayed as shown below:



Select the corresponding channels to link; they will be grouped so the user can adjust their parameters.

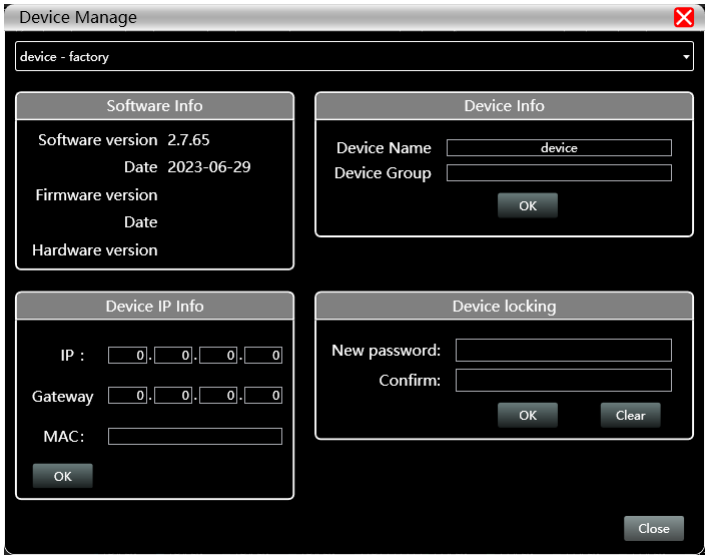
Menu - File



New project: the project is restored to its initial state when opened.
Demo Device: the user can view all device functions without affecting the connected device.
Open: open an existing device management project from the computer disk.
Save: save the current equipment management project to the computer disk.
Save As: save the current equipment management project to the computer disk with a new name. This saved project can be loaded virtually later. To load a saved project, the processor must be disconnected.

Menu – Device (including Device lock)



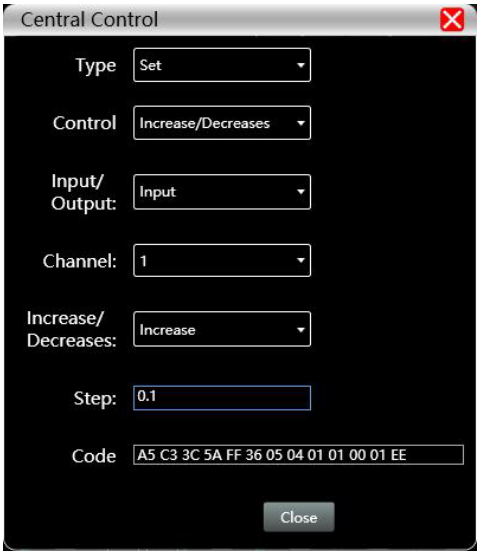


Devices: view software version information, modify the device name and IP address, and set the device password.

Channel name: define the name of each input and output channel, with memory function.

Channel copy: copy parameters of the device's input and output channels; allows parameter copying between different devices (Note: devices must be of the same type).

Central control: provides a quick way for the user to check the Central Control configuration code.



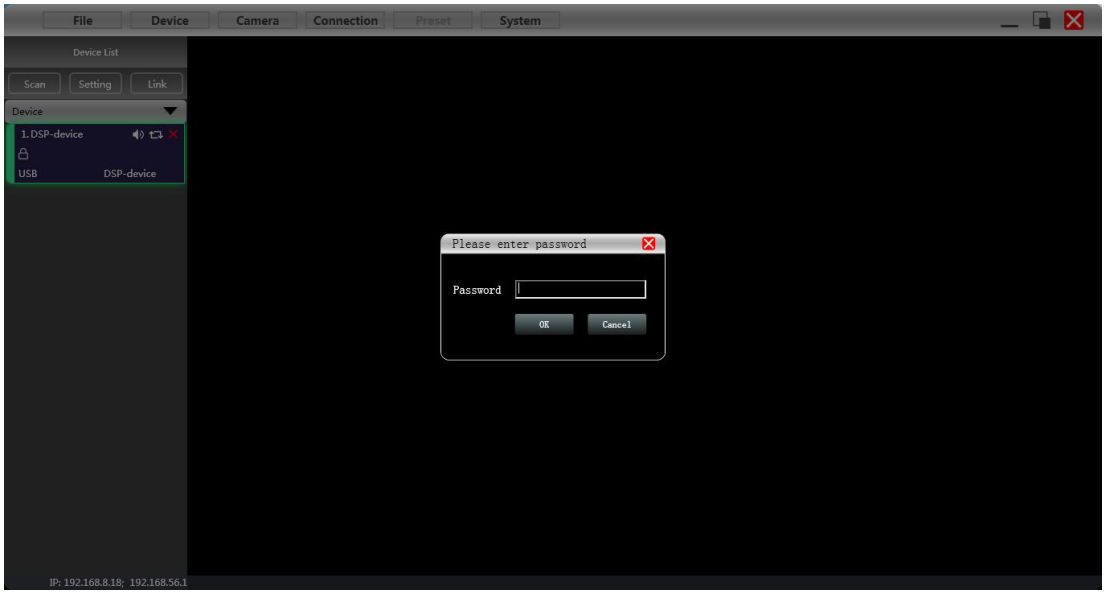
Device Locking: the user can define a preset for this device to protect the audio project after configuring the parameters. After unlocking the device in the software, the user can clear or reset the password.

IMPORTANT: Once a password is set, no processor presets can be modified without it. If the user forgets the password, there will be five attempts to enter it; after that, the processor will be permanently locked, and only a factory reset— which deletes all saved presets — will unlock the device.

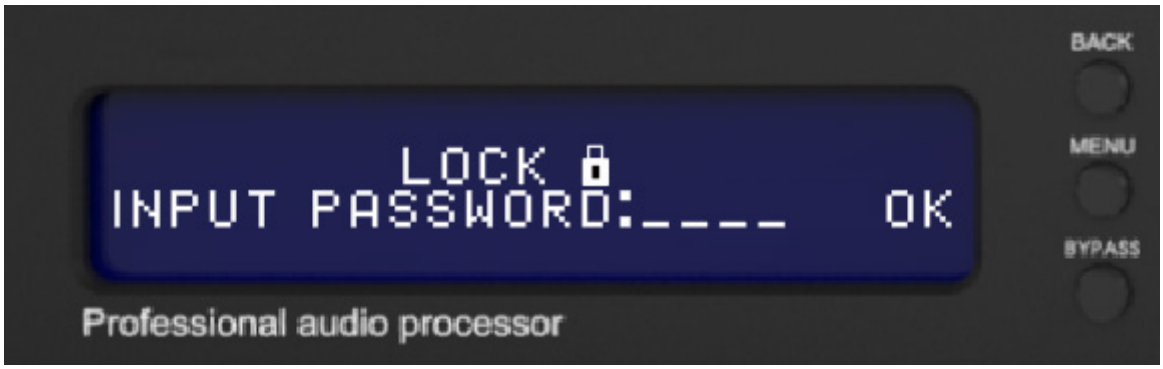
The password can have four digits (0,1,2...9), allowing the user to unlock the device using the control software or the device's front panel. If the device is locked, an icon will be displayed in the software and on the LCD screen, as shown in the image below:





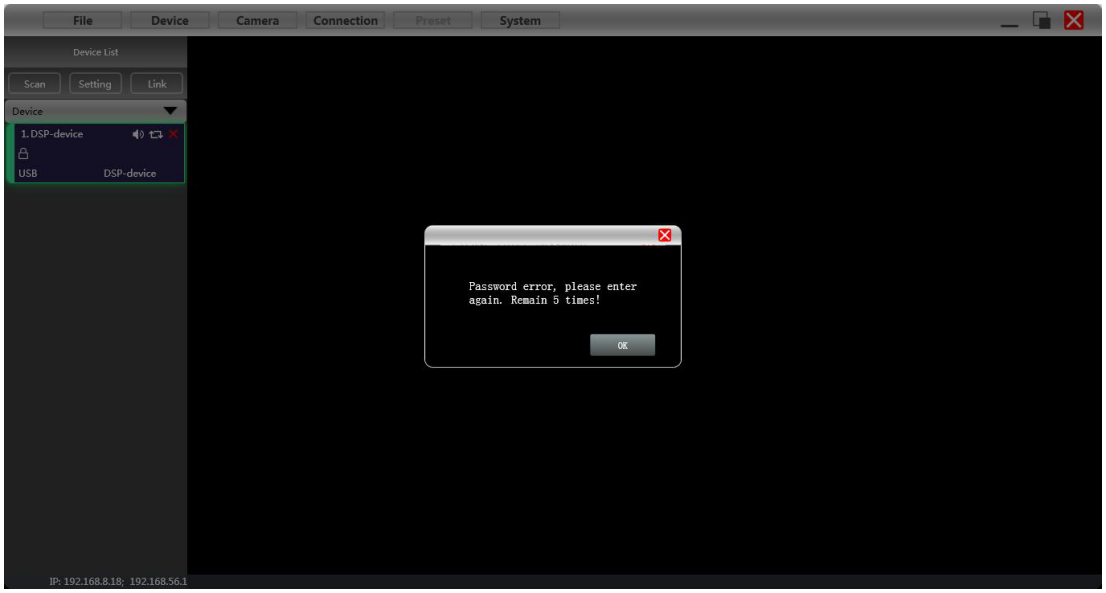


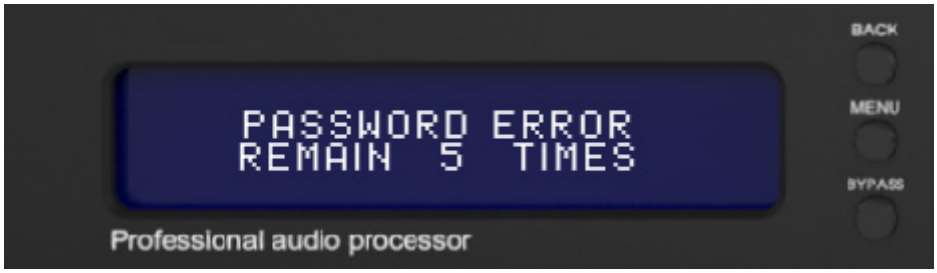
Enter the password in the software to unlock the device.



Enter the password on the LCD display to unlock the device.

When the device is locked, the user can press the MUTE button of each channel on the front panel. Press any of the BACK, NEXT, MENU, ENTER, BYPASS, or QUIT buttons, and the LCD will display the password input interface; press GAIN to select the digit and press BACK or NEXT to choose the digit position. Then select “OK” and press ENTER to unlock the device.





If the password is entered incorrectly, the software will warn that there are only 5 attempts to enter the correct password. After more than 5 attempts, the device can no longer be unlocked, and the user must perform a factory reset, which will delete all saved presets in the processor.





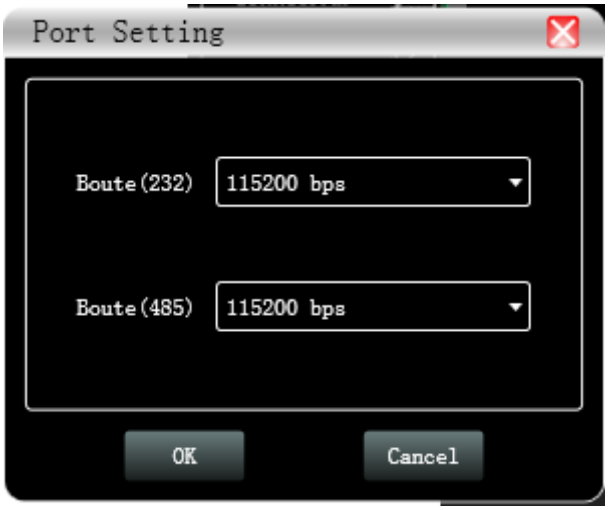
Password entered successfully and access to the main interface granted.

Factory Reset

To reset the processor and restore factory settings, press and hold the BYPASS and ENTER buttons simultaneously for at least 10 seconds.

This function is only available when the processor is locked.

Menu - Connection



• **Port:** set the connection mode, port number, and baud rate; confirm the connection mode, then select the corresponding port.

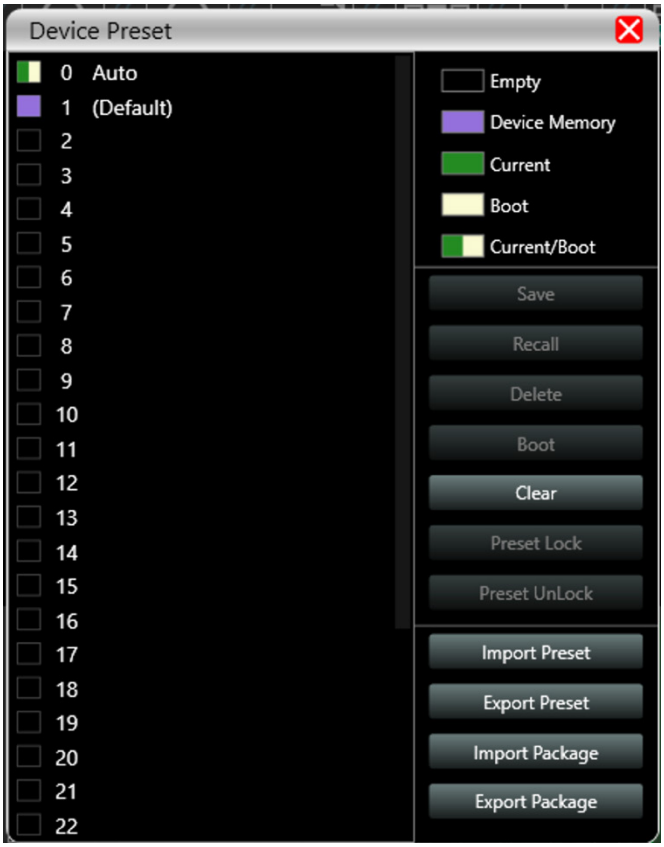
• **Connect:** connect and download parameters from the device.

• **Disconnect:** disconnect the connected device.

• **Connect all:** connect and download parameters from all devices in the device list.

• **Disconnect all:** disconnect all connected devices in the device list.

Menu - Preset



• **Save:** select the preset to be saved and store all current machine configuration parameters into the device preset (2 to 30 presets are possible).

• **Recall:** load a device preset into the current configuration position in the processor.

• **Delete:** delete an existing preset; the default file cannot be deleted, overwritten, or saved. Password: 1111.

• **Clear:** delete all presets from the device. Password: 1111.

• **Boot:** select a specific preset and set it as the boot file; each time the device is powered on, it will automatically load the saved parameters. If the last configured parameters need to be saved automatically, set the auto file as the boot file.

• **Preset Lock:** allows locking a preset so it cannot be edited.

• **Preset Unlock:** allows unlocking a preset so it can be edited.

• **Import preset:** import a single preset file from the computer.

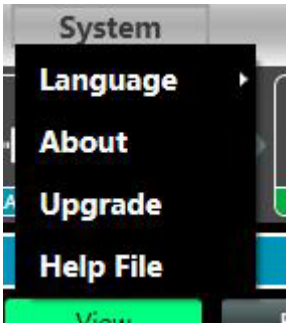
• **Export preset:** export all current parameters to the computer and generate a single preset file.

• **Import preset package:** import a preset package file containing multiple presets from the computer.

• **Export preset package:** group multiple device presets into a preset package and export it to the computer.

Important: always save a copy of all presets on your computer. This can be very important in case you forget the access password, since without it the processor will need to be factory reset, and all user-defined preset configurations will be lost.

Menu – System



Language: switch between multiple languages; supports Simplified Chinese, Traditional Chinese, and English.
About: information about the current version of the control software and the device firmware.
Upgrade: firmware can be updated using this function; a .bin upgrade file must be provided by the manufacturer.

FIR Filter and Functions

FIR Filter and Applications

When the user uses **PEQ** to adjust the audio signal and sets a linear magnitude, they may notice that the signal phase is altered due to the IIR filter. By using an FIR filter, it is possible to adjust the audio signal with linear phase.



Some Calculations:
Frequency resolution = Sampling rate / coefficients
Minimum available frequency ≈ Frequency resolution × 3

This means that when adjusting the audio signal at 48 kHz with 1024 coefficients, the FIR filter will affect frequencies above 141 Hz of the audio signal. The higher the number of coefficients, the steeper the FIR filter curve.

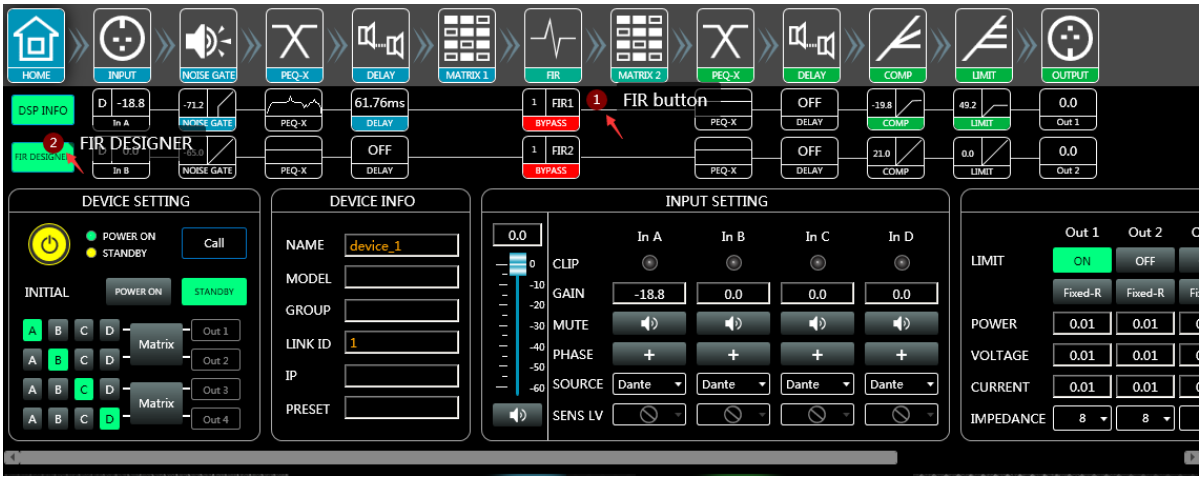
Processing the audio signal with an FIR filter introduces a certain delay:
Delay = (1 / sampling rate Hz) × coefficients / 2

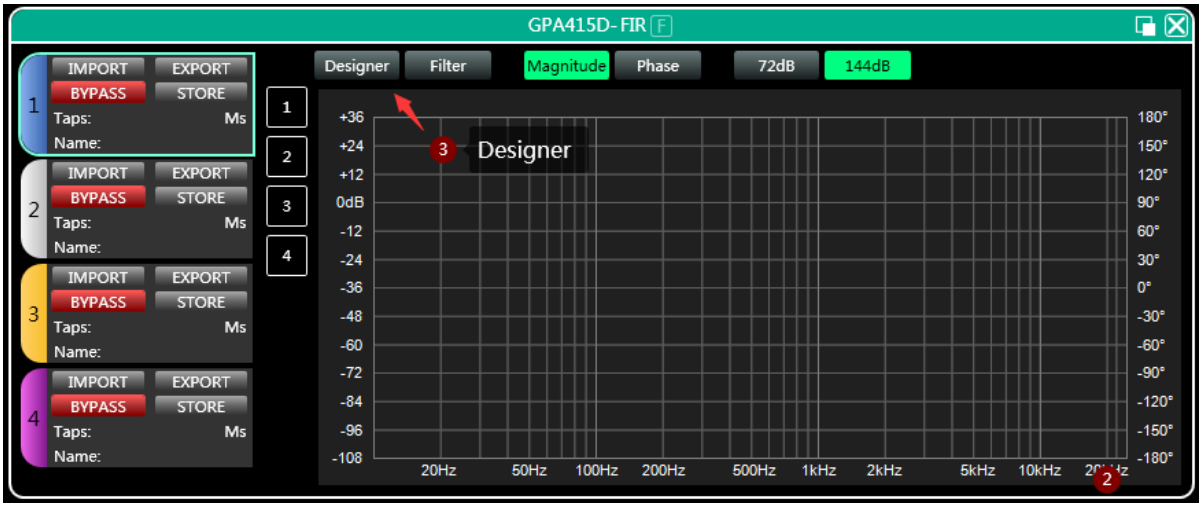
Coefficients	**48 kHz Sampling**	**96 kHz Sampling**
256	2,67 ms, LF 563 Hz	1,33 ms, LF 1125 Hz
512	5,33 ms, LF 279 Hz	2,67 ms, LF 558 Hz
768	7,99 ms, LF 188 Hz	4,00 ms, LF 375 Hz
1024	10,67 ms, LF 141 Hz	5,33 ms, LF 281 Hz
2048	21,33 ms, LF 70 Hz	10,67 ms, LF 141 Hz

Applications:
• Linearization of the loudspeaker phase response;
• Adjust phase and magnitude of different loudspeaker models within the same product line, as well as across different models in installation projects, facilitating the configuration of speaker groups and arrays;
• Use in line array systems (to optimize audience coverage);
• Optimization of frequency crossover to improve consistency of frequency response in multi-way loudspeakers across their coverage angle.

Using the FIR DESIGNER in LMControl+ to Adjust FIR Magnitude and Phase

The LMControl+ software provides a more convenient way for users to adjust the FIR magnitude and phase of each channel.

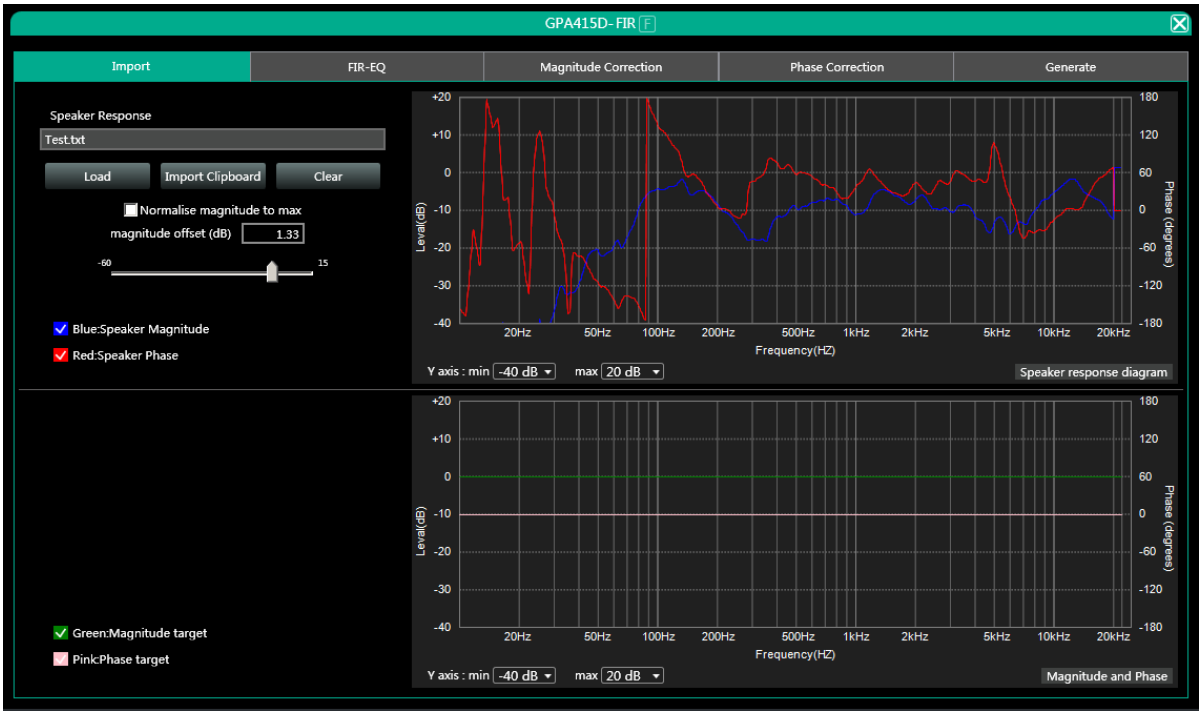




1- 3- Click the "FIR" – "Designer" button to enter the automatic FIR magnitude and linear phase function interface.

2- Click "FIR DESIGNER" on the main (Home) interface to enter the automatic FIR magnitude and linear phase function interface, allowing the user to quickly return to the last configured page.

Configuration:
FIR DESIGNER – Import



• **Load:** load the loudspeaker measurement file from Smaart (or another tool), usually a .txt file.
• **Import Clipboard:** load ASCII data directly from Smaart.
• **Clear:** clear the measurement data.
• **Normalise magnitude to max** or **Magnitude offset (dB):** helps the user adjust a specific magnitude value in dB while minimally altering the magnitude curve.

FIR DESIGNER – FIR-EQ



There are high-pass and low-pass filters to define the crossover, and 15 bands of PEQ / LSLV / HSLV to adjust the magnitude. Try to configure a linear magnitude for the target loudspeaker.

Note: changing the FIR magnitude does not affect its phase.

FIR DESIGNER – Magnitude Correction and Phase Correction

Of course, if there are many loudspeakers to adjust, the user would need to spend a lot of time manually adjusting their magnitudes. In this case, magnitude correction becomes more useful. Simply enable the ON button for the frequency.



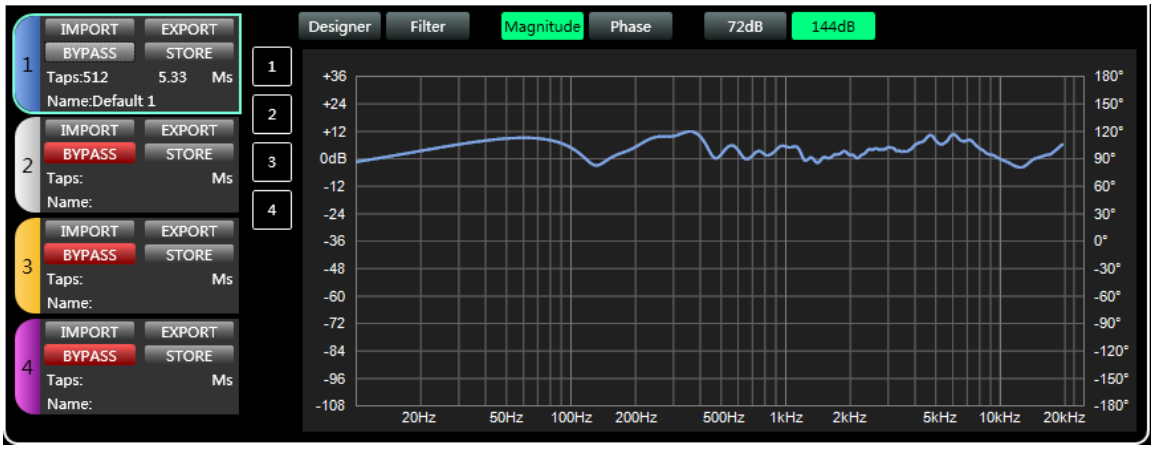
After adjusting the magnitude, configure the loudspeaker's linear phase.



FIR DESIGNER – Generate

Select the coefficients (e.g., 512) for this configuration and store it in an FIR channel. The user can also name this FIR setting and export it to a .KF file. After completing all configurations, return to the FIR interface and disable the **BYPASS** button to activate it.





ATTACK

AUDIO SYSTEM

HEARTHEDIFFERENCE

In order to improve our products, the specifications contained in this manual may be changed without prior notice.

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