MDM-D8 / MDM-D16

DSP Matrix Audio Processor 8x8 / 16x16

16x16 Dante channels

User manual

Contents

Chapter 1 Introduction

MDM-D8 / D16 is a 8/16 inputs and 8/16 outputs DSP matrix audio processor for conference systems and can be used as interface for the MusiCall Music & Message systems. It provides with useful voice algorithm AFC, AEC, ANC, and DSP functions including auto mix, matrix mixer, noise gate, crossover, parameter EQ, delay, compressor, limiter etc. Supports Dante network audio 16x16 channels.

Applications

- Meeting room
- Multi-functional hall
- Auditoriums
- Sports stadium

Features

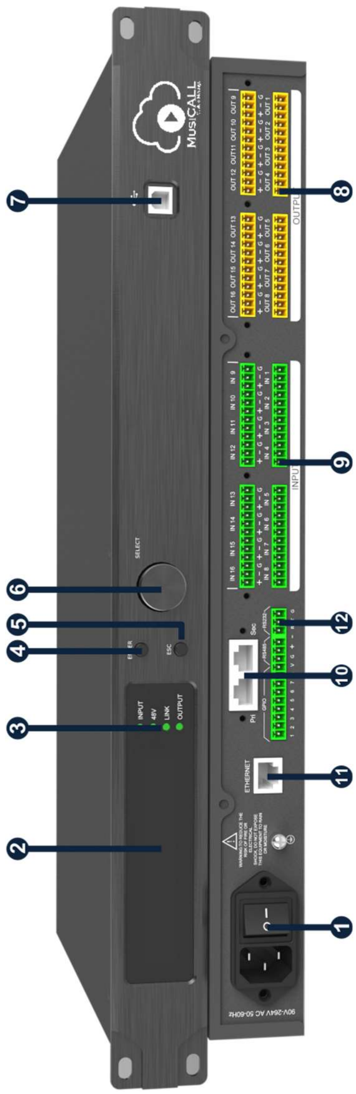
- 8 analog inputs and 8 analog outputs, supports to select Line level and Mic level
- Dante network audio 16×16.
- Supports 48V phantom for each Mic level input, 40 level sensitivity (1dB in step).
- Built-in AFC(feedback control) , 2 level to select.
- Built-in AEC(echo control) for remote video-conference system.
- Built-in ANC(noise control) for optimizing local meeting system.
- Built-in AGC(automatic gain control) for optimizing microphone signals in complex scenarios
- Input with 8 PEQ and output with 8 PEQ. Support LSLV, HSLV, ALL-PASS, PHASE, ELLIPTIC, LOW PASS AND HIGH PASS filters. Support HPF and LPF with Butterworth / Bessel / Linkwitz-Riley.
- Supports auto mix and matrix mix.
- Supports camera tracking with most of camera control.
- Supports presets archiving and locking, help project to hide parameters of setting.
- Control connections: USB or TCP/IP. Configured with RS232 and RS485 central control connection. Configured with GPIO external control connection.
- Nice GUI windows7/8/10/11 software MDM-DSP.
- Optional touch screen wall control panel (RS485 wired control).

Chapter 2 Technical parameters

MDM-D8	
1. DSP Processor	
Process:	32bit float point DSP 400MHz
System delay:	<3ms
AD/DA:	24-bit 48KHz
2. Analog Audio Inputs and Outputs	
Input:	8 channels balanced. Line/Mic level to switch / 16 Dante channels
Input interface:	3.81mm phoenix, 12-pin
Input impedance:	16KΩ
Max input level:	17dBu/Line; -3dBu/Mic@20dB sensitivity
Phantom supply:	+48V DC, 5.5mA in each input channel
Output:	8 channels balanced. Line level / 16 Dante channels
Output interface:	3.81mm phoenix, 12-pin
Output impedance:	150Ω
3. Audio Performance Specifications	
Frequency response:	20Hz-20kHz(+-0.5dB)/Line, input 0dBu; 20Hz-20kHz(+-1.5dB)/Mic, 20dB gain sensitivity, input -10dBu
THD+N:	-90dB(@17dBu, 1kHz, A-wt)/Line -90dB(@-6dBu, 1kHz, A-wt)/Mic, 20dB gain sensitivity
SNR:	110dB(@17dBu, 1kHz, A-wt)/Line 100dB(@-6dBu, 1kHz, A-wt)/Mic, 20dB gain sensitivity
4. Connect Ports and Indicators	
USB:	Type A-B, free driver
RS232:	Serial port communication
TCP/IP interface:	RJ-45
Indicator light:	Input signal, +48V, Link, Output signal
5. Electrical and Physical	
Supply:	AC 90V ~ 264V 50/60 Hz
Products Dimensions	483mmx265mmx44.5mm
Packaged Dimensions	540mmx390mmx80mm
Net Weight	3.3kg
Packaged Weight	4.4kg
Operating temperature:	-20℃ ~ 80℃

MDM-D16	
1. DSP Processor	
Process:	32bit float point DSP 400MHz
System delay:	<3ms
AD/DA:	24-bit 48KHz
2. Analog Audio Inputs and Outputs	
Input:	16 channels balanced. Line/Mic level to switch / 16 Dante channels
Input interface:	3.81mm phoenix, 12-pin
Input impedance:	16KΩ
Max input level:	17dBu/Line; -3dBu/Mic@20dB sensitivity
Phantom supply:	+48V DC, 5.5mA in each input channel
Output:	16 channels balanced. Line level / 16 Dante channels
Output interface:	3.81mm phoenix, 12-pin
Output impedance:	150Ω
3. Audio Performance Specifications	
Frequency response:	20Hz-20kHz(+-0.5dB)/Line, input 0dBu; 20Hz-20kHz(+-1.5dB)/Mic, 20dB gain sensitivity, input -10dBu
THD+N:	-90dB(@17dBu, 1kHz, A-wt)/Line -90dB(@-6dBu, 1kHz, A-wt)/Mic, 20dB gain sensitivity
SNR:	110dB(@17dBu, 1kHz, A-wt)/Line 100dB(@-6dBu, 1kHz, A-wt)/Mic, 20dB gain sensitivity
4. Connect Ports and Indicators	
USB:	Type A-B, free driver
RS232:	Serial port communication
TCP/IP interface:	RJ-45
Indicator light:	Input signal, +48V, Link, Output signal
5. Electrical and Physical	
Supply:	AC 90V ~ 264V 50/60 Hz
Products Dimensions	483mmx265mmx44.5mm
Packaged Dimensions	540mmx390mmx80mm
Net Weight	3.3kg
Packaged Weight	4.4kg
Operating temperature:	-20℃ ~ 80℃

Chapter 3 Functions structure and panel operation



1. Power connector
2. Display
3. Indicator (input, +48V, link, output)
4. Enter (display menu)
5. ESC (display menu)
6. Select (display menu)

7. USB control and audio

11. Ethernet (TCP/IP) control

    support external control system

12. -GPIO control

    (8 channel)

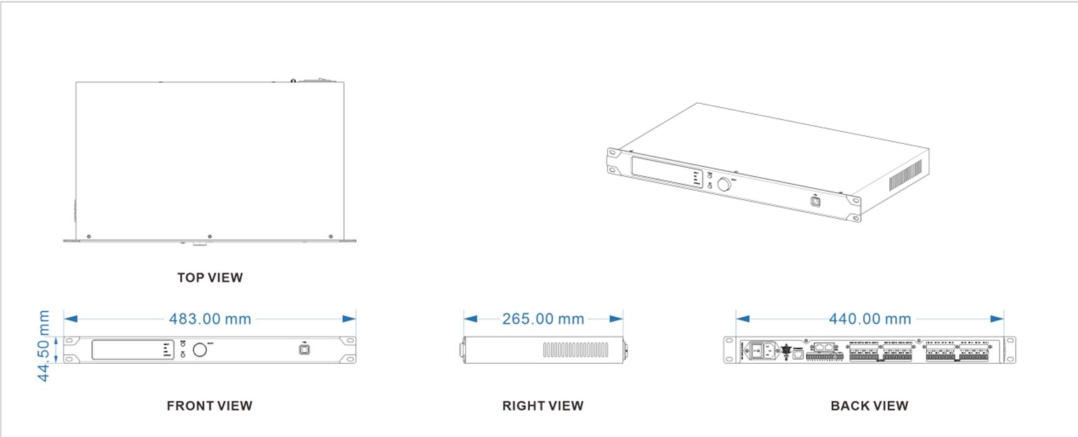
    - RS485 control

    - RS232 control

    support external control system

8. Output channels (Line level)
9. Input channels (Line/Mic level)
10. Dante network audio

Dimensions (mm)



Operating front panel

①Press **ENTER** or **SELECT** 5 seconds to unlock device, it will show menu list, turn the **SELECT** knob clockwise or counterclockwise to select functions: INPUT VOLUME, OUTPUT VOLUME, PRESETS, INPUT SOURCE, IP SET, RENAME, SECURITY. Press **ENTER** or **ESC** button to set function.

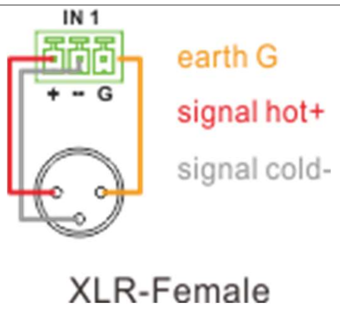
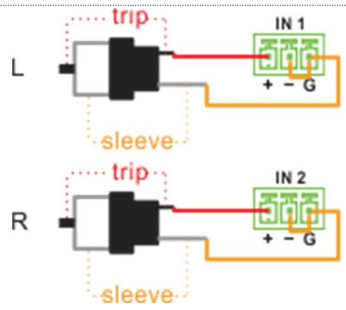
Functions in panel	Menu list	Remark
1.INPUT VOLUME	---- 1 IN VOLUME ---- 10 >20 30 40	Value from mute, -60 to 15dB 8 or 16 channels
2.OUTPUT VOLUME	---- 2 OUT VOLUME ---- 10 >20 30 40	Value from mute, -60 to 15dB 8 or 16 channels
3.PRESET	---- 3 PRESETS ---- >*01.Default Preset	LOAD THIS PRESET ? YES NO
4.INPUT SOURCE	---- 4 INPUT SOURCE ---- >1LINE 2LINE	DANTE, LINE, MIC (DB), PHAN (DB)
5.IP SET	---- 5 IP SET ---- IP : 192.168.0.115 ->	---- 5 IP SET ---- GATE: 0. 0. 0. 0 OK
6.RENAME	---- 6 RENAME ---- [MATRIX DE_] OK	
7.SECURITY	SCREEN AUTO LOCK ? *YES NO	

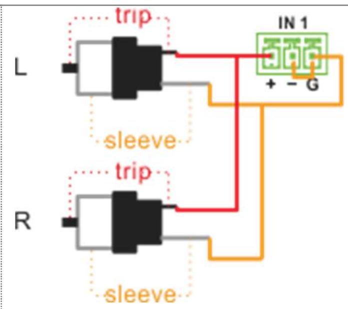
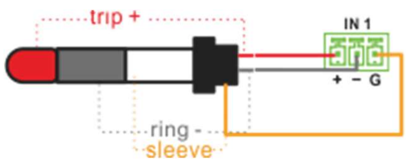
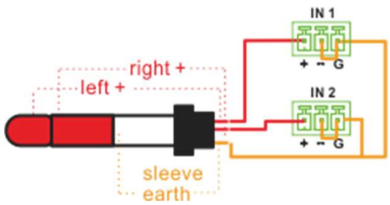
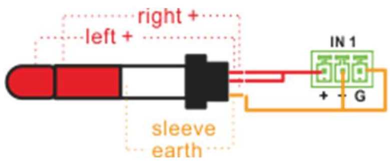
②Turn the **SELECT** knob clockwise or counterclockwise to set volume of device.

Chapter 4 Connection of balanced and unbalanced signal

One of the most powerful ways to prevent noise from entering audio signals is to make balanced connections between devices. Some cables are capable of balanced connections, others aren't.

Some common connection methods are listed as below:

Source	Type	Direction	Receiver	Connection diagram
XLR (Canon)	balanced	to	Phoenix	
RCA (Stereo)	unbalanced	to	Phoenix 2 channels	

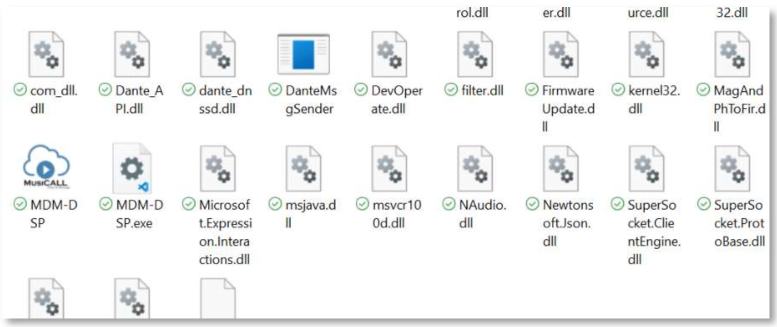
RCA (Stereo)	unbalanced	to	Phoenix 1 channel	
Standard Jack (6.3mm)	balanced	to	Phoenix	
Mini Jack stereo (TRS 3.5mm)	unbalanced	to	Phoenix 2 channel	
Mini Jack stereo (TRS 3.5mm)	unbalanced	to	Phoenix 1 channels	

Chapter 5 Operation of control software – MDM-DSP

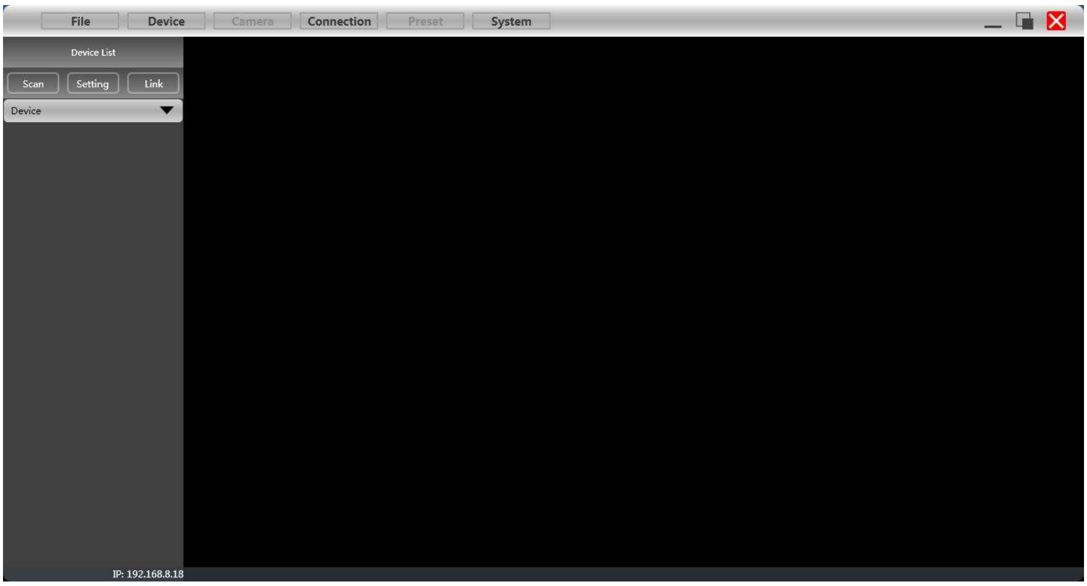
MDM-DSP provides users a fast tool to control one or more devices through multiple methods: TCP/IP, USB, common serial port (RS232). Easily set DSP functions of device, and check central control codes.The configuration can be stored in presets, convenient for various applications.

5.1 Operating condition

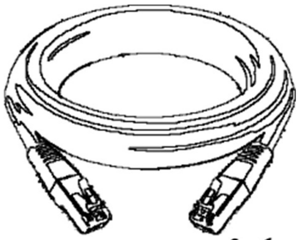
MDM-DSP is suitable for Win7/8/10/11 x86/x64 PC systems with Microsoft .NET Framework 4.0 installed. Double click the file with the MusiCall logo:



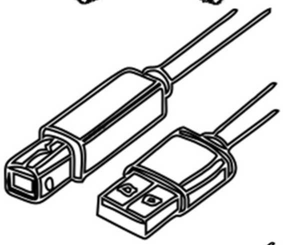
the main interface will pop up:



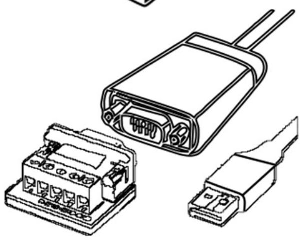
5.2 Connect to PC

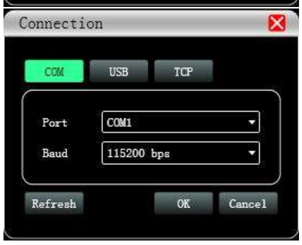


If you connect the device by using network cable, click Setting in Device List, choose TCP in Connection window.



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

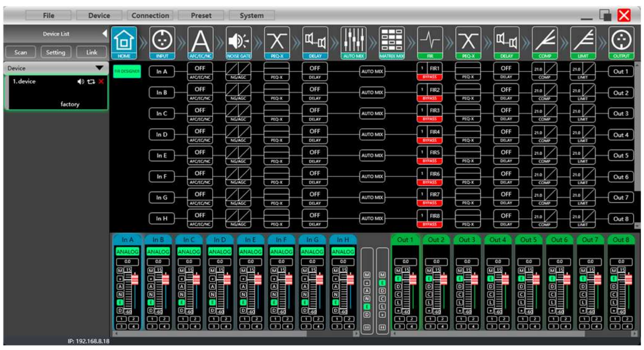


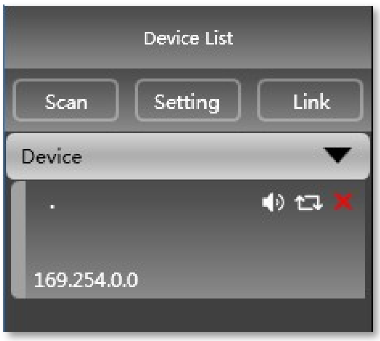


If you connect the device by using serial cable, click Setting in Device List, choose COM in Connection windows. Please check port and baud rate carefully for 232 before connecting.

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

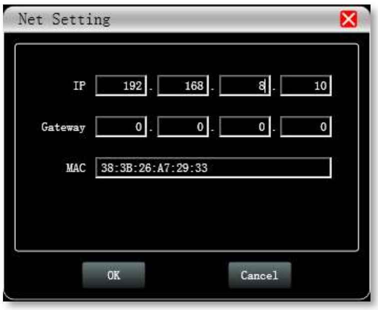
User can mute devices, refresh connections, or delete devices in this window. Single click the device to load the function interface.





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

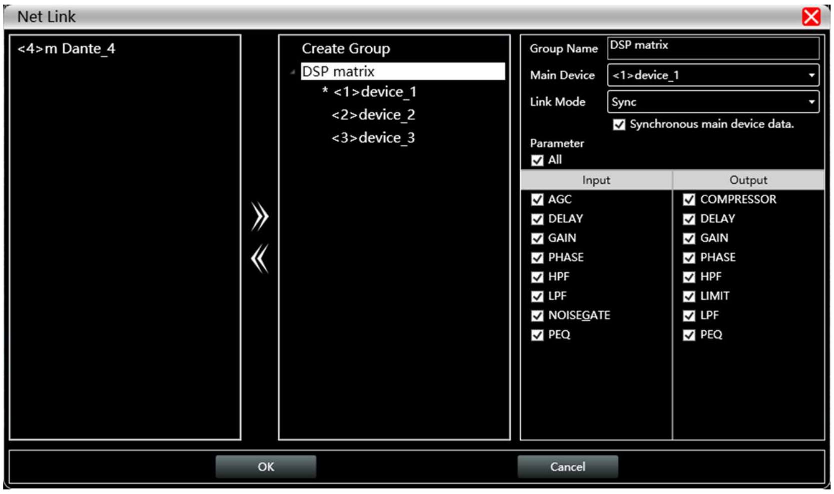
Right-click the device enclosure, a Net Setting window will show.



Set IP address of device refer to IP showed in the bottom of the software. Set the first three paragraphs same with the PC IP.

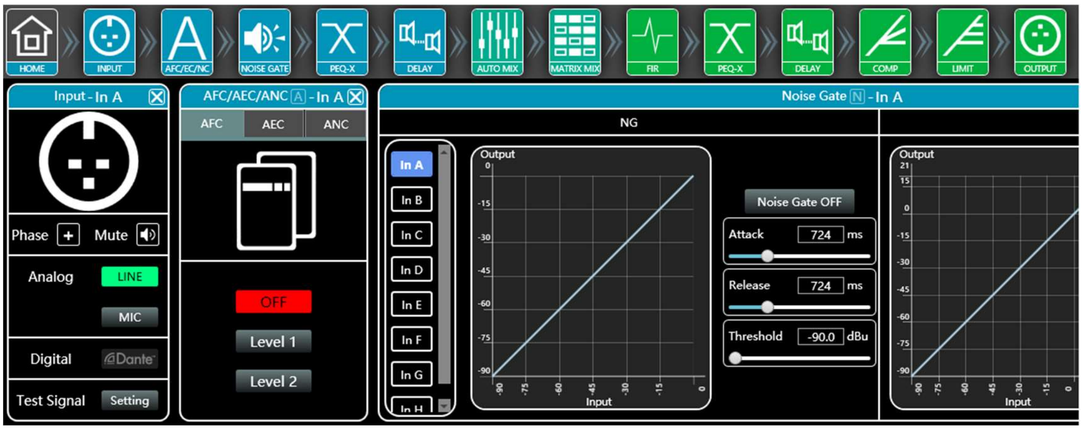
Successfully scanned and connected. Then click the device to load all parameters from the processor.

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



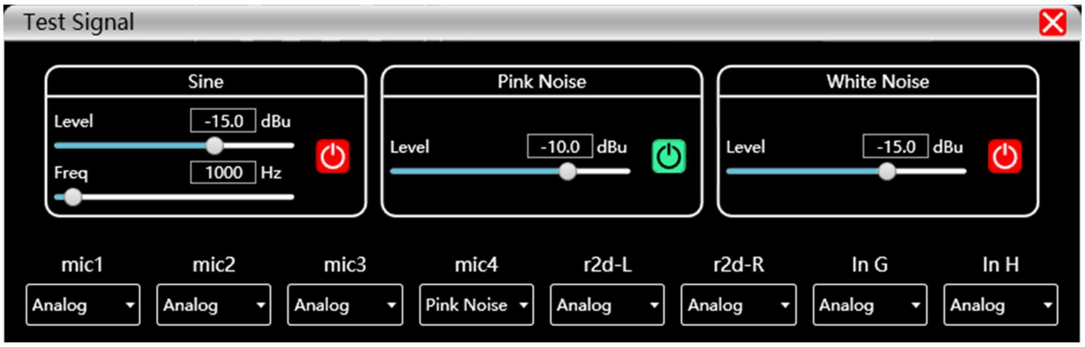
5.3 DSP functions

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.

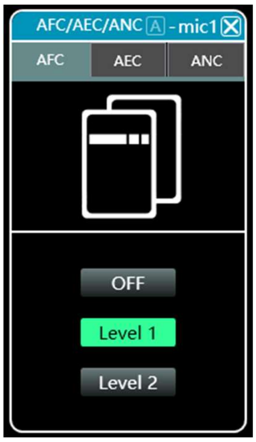


5.3.1 DSP functions - INPUT(Line, Mic, Dante, USB audio and Test signal)

- Set Phase of input;
- Set Mute of input;
- User can select Line/Mic input;
- When Dante in option, user can select Dante network audio;
- When matrix channels in 16x16, user should select USB audio in In A and In B;
- When choosing test signal, choose from Sine/Pink Noise/White Noise for each input channel.



5.3.2 DSP functions - AFC



Matrix processor provides AFC (acoustic feedback control) function for microphone. With two level to select, user can control howl round easily when setting each microphone;

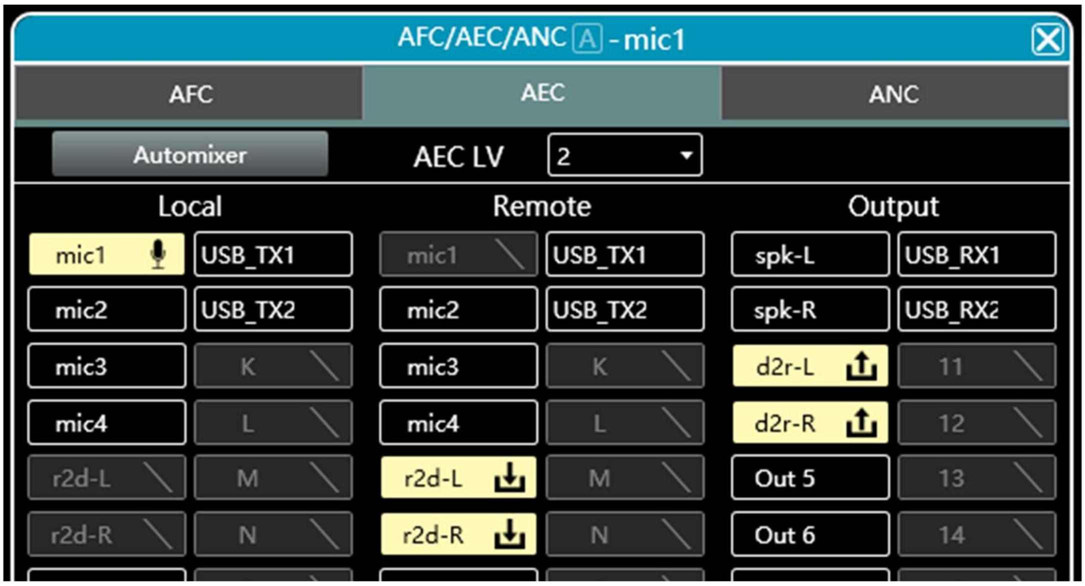
- Level 1, low degree process;
- Level 2, high degree process;
- The window of AFC will show input channel name, please select corresponding input channel to set.

5.3.3 DSP functions - ANC

Matrix processor supports to control a certain degrees of noise, which comes from microphone, ground noise of input source, or surroundings. This function is mainly used for voice process.

- User can route input channels and output channels in this window.
- ANS Level: 0 to 4, from low to high degree to process.

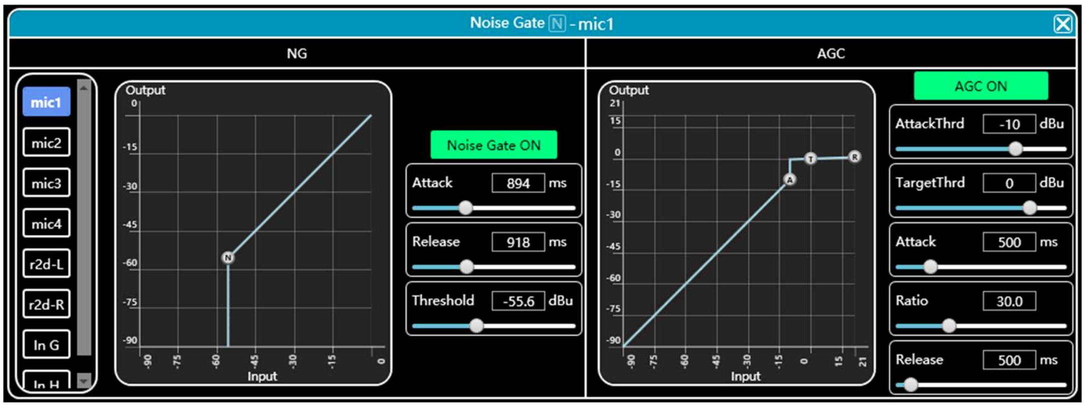
5.3.4 DSP functions - AEC



Matrix processor supports to control a certain degrees of echo from local room to remote room, which usually happened in remote video conference via network software Skype, Zoom, or via remote conference system controller(terminal). This function is mainly used for voice process.

- User can route input channels and output channels in this window.
- AEC Level: 0 to 5, from low to high degree to process.

5.3.5 DSP functions - NOISE GATE and AGC



NG (noise gate)

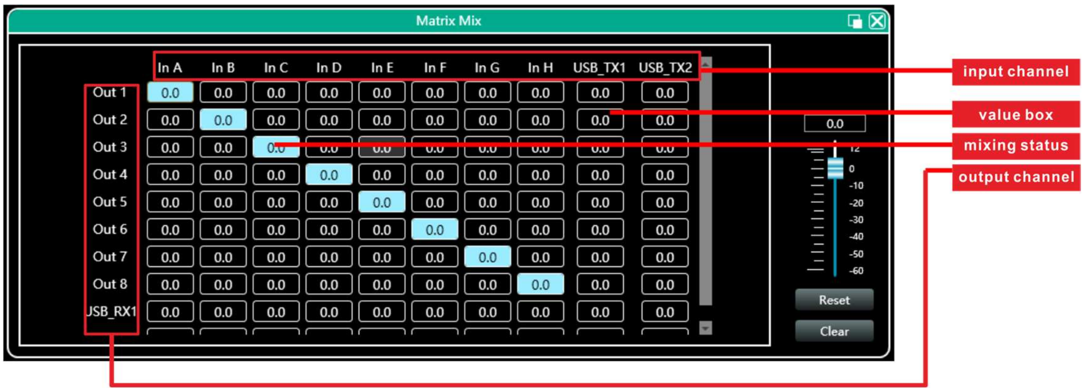
- Attack: 1 to 2895ms;
- Release: 1 to 2895ms;
- Threshold: -90.0 to 0.0dBu;
- Click Noise Gate ON to enable this function.

AGC (automatic gain control)

- AttackThrd: -90 to 21dBu;
- TargetThrd: -90 to 21dBu;
- Attack: 1 to 2895ms;
- Ratio: 1.0 to 100.0;
- Release: 1 to 10000ms;

Click AGC ON to enable this function.

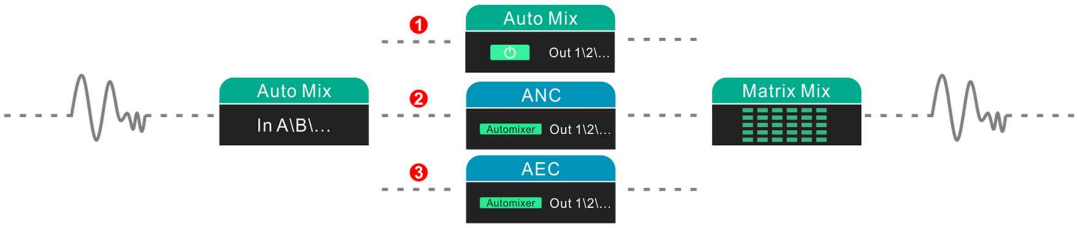
5.3.6 DSP functions - MIXING PROCESS (Matrix Mix, AMX, AEC, ANC)



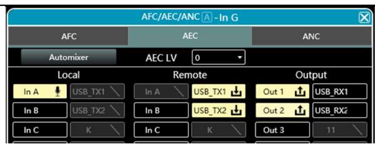
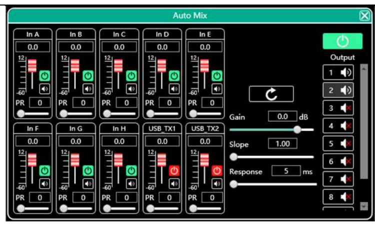
Input channel (on top side) corresponds to output channel. The box with a value is mixing key of channels. When the mixing key is cyan (double-click the value box to switch), the input channel and output channel signal realizes the mixing function.

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.

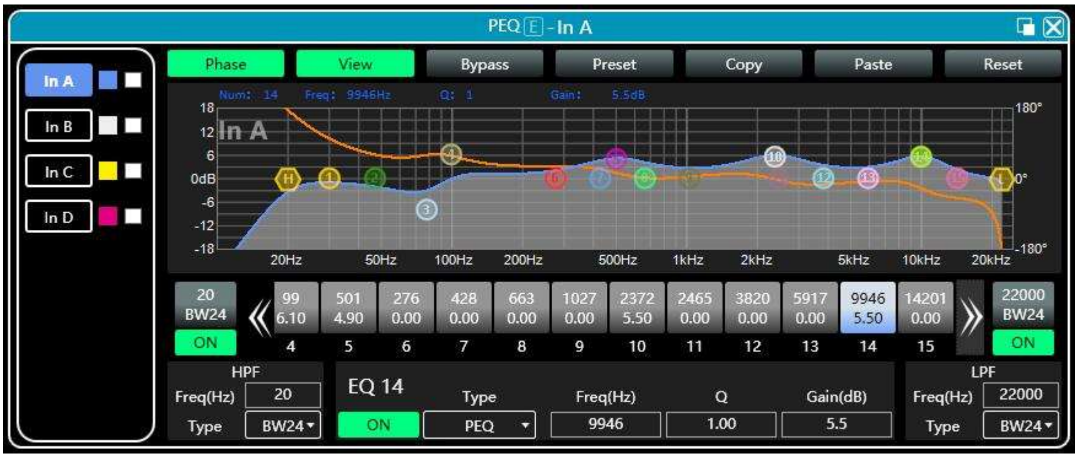
User can find there is a Automixer button in ANC and AEC windows, which provides user a signal routing from Auto Mix to ANC or AEC process.



some case setting for reference:

Applications	Auto Mix Input	Auto Mix Output	ANC Input	ANC Output	AEC Input	AEC Remote	AEC Output	Schematic diagram
single Mic with ANC			•	•				
single Mic with AEC					•	•	•	
multiple Mic with AMX	•	•						
multiple Mic with AMX ANC	•			•				
multiple Mic with AMX AEC	•					•	•	

5.3.7 DSP functions - 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 for input channels

Type: PEQ/LSLV/HSLV/ALLPASS-1/ALLPASS-2/PHASE;

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 for output channels

Type: PEQ/LSLV/HSLV/ALLPASS-1/ALLPASS-2/PHASE;

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.

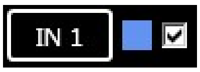
Bypass: turn on or off all equalizer EQ of the current channel at the same time

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).

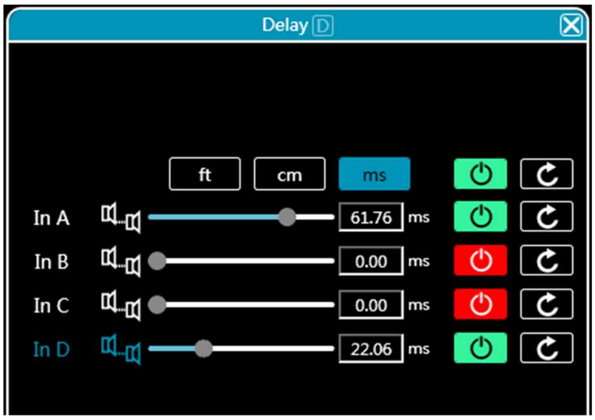
Paste: used in combination with the copy button to paste the last copied equalizer parameter value to the current channel.

Reset: reset the equalizer parameter to the default parameter values.

In the PEQ-X window, the left side 

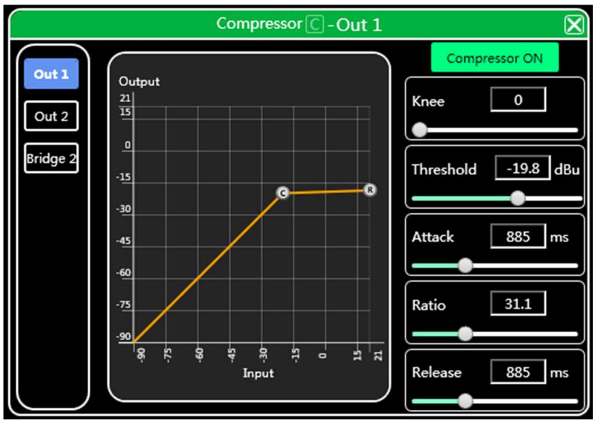
is the interface switching button for each channel. Click to switch the EQ channel, and the color is the currently selected channel.  is the curve color of the EQ channel.  For each channel's EQ curve display switch, check it to enable it to display the curves of other channels in the current channel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

5.3.8 DSP functions - DELAY (input and output)



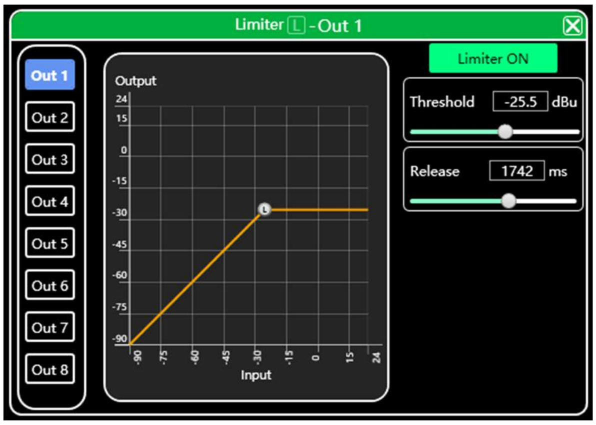
- Max 2000ms for input channel;
- Max 2000ms for output channel;
- Click  to enable this function;
- Click  to reset each channel;
- User can switch ft/cm/ms measurement for delay.

5.3.9 DSP functions - COMPRESSOR



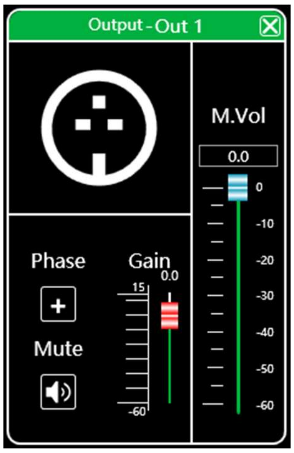
- Soft knee: 0 to 30;
- Threshold: -90.0 to 21.0 dB;
- Attack: 1 to 2895 ms;
- Ratio: 1.0 to 100.0;
- Release: 1 to 2895 ms;
- Click  to enable this function;

5.3.10 DSP functions - LIMITER



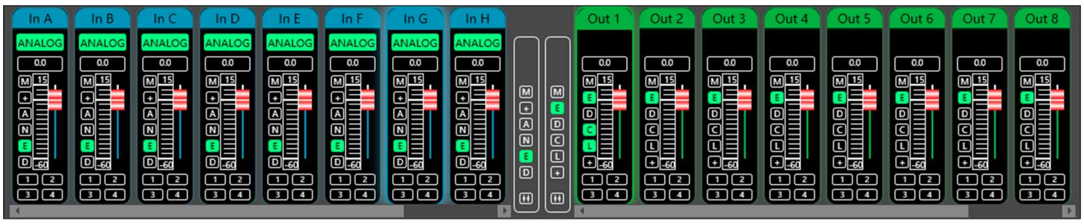
- Threshold: -90.0 to 21.0dBu;
- Release: 1 to 2895 ms;
- Click  to enable this function;

5.3.11 DSP functions - OUTPUT



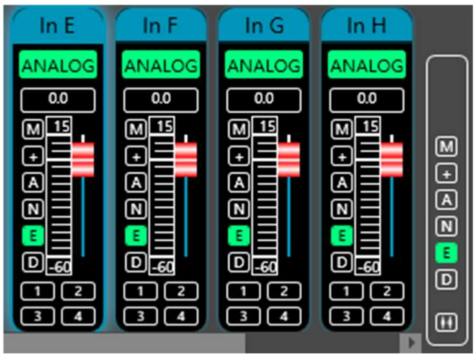
- Set phase of output signal;
- Set mute of output channel;
- Set gain of output channel.

5.4 Monitoring and setting of channels



User can monitor gains level of input and output channels.

5.4.1 Channel gain level

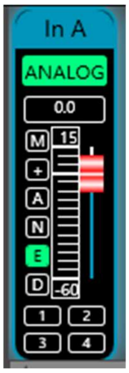


There are 3 kinds of input signal in device: ANALOG, Mic and testing signal. It will show a label for user.

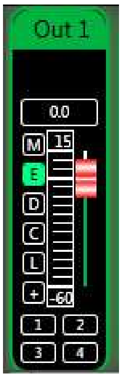
Some devices support to switch USB audio in In A and In B, all analog channels switch to Dante audio.

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

5.4.2 Quick buttons of DSP in channels

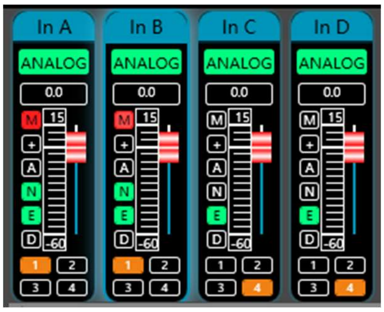


M Mute
+ Phase
A AFC
N Noise Gate
E PEQ
D Delay



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

5.4.3 Group and channels link



User can quickly set channels in groups for opening or closing mute, phase, noise gate, PEQ and delay function. AFC should be set independently, which can't be linked.

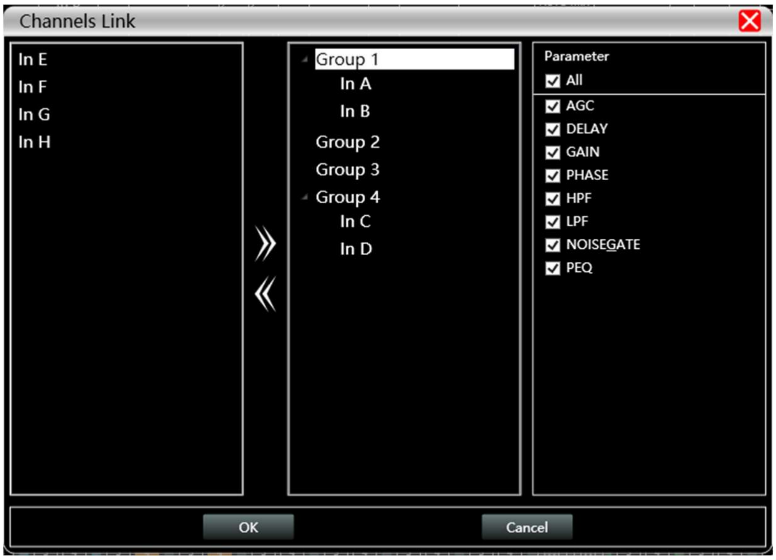
M Mute
+ Phase
N Noise Gate
E PEQ
D Delay

Channels link for input

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

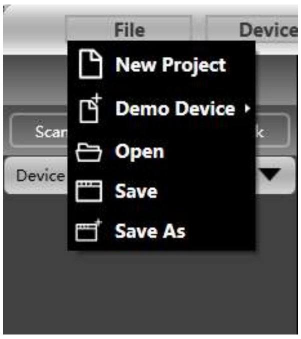
Channels link for output

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.

5.5 Menu - File



New project: the project is restored to the initial open state.

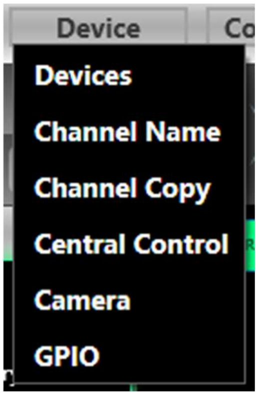
Demo Device: user can view all the functions of the device without affecting the specific device connected.

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.

5.6 Menu - Device (Central control, Camera tracking, GPIO)



Devices: view or modify the software version information, device name and device IP address of the upper and lower computer of the device.

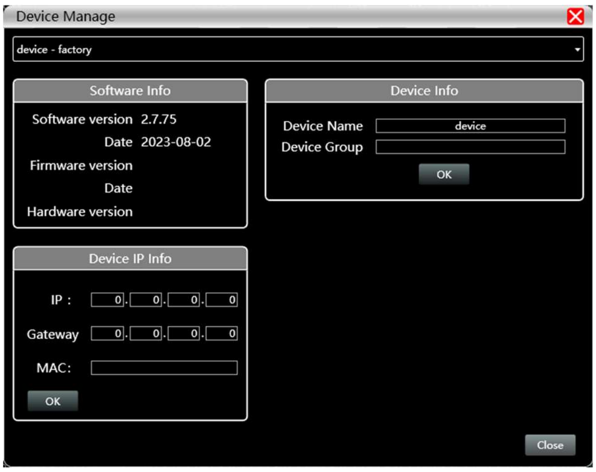
Channel name: set the name of each input and output channel.

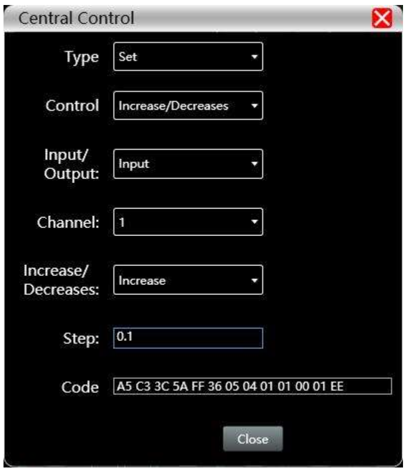
Channel copy: copy device input and output channel parameter, can realize cross-device copy parameter (Note: the same type of device is required).

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.

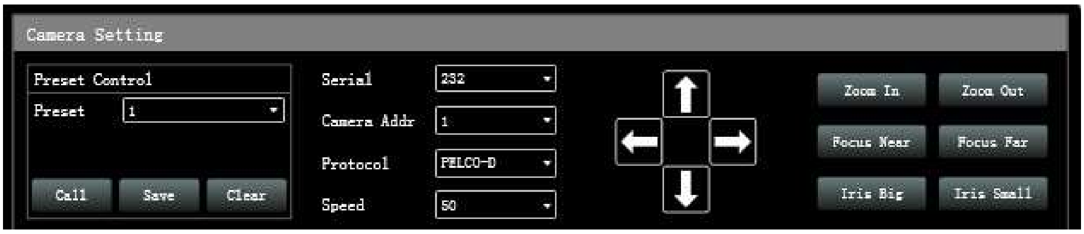
Camera: provides user with camera tracking function.

GPIO: provides user a quickly way to inquiry direction of GPIO setting.





Camera setting

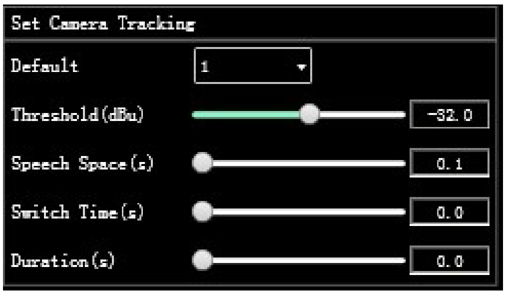


Generally, the camera position should be debugged before the tracking starts, and finally the parameter of this part are saved on the camera.

1. Set the serial ports via RS232 or RS485.
2. Set the camera address and protocol type refer to the protocol depends on the camera model.
3. The preset No. is defined by the user for the camera, and then adjust the up, down, left, right, focal length, aperture and other parameter.
4. Click "Save" to save the parameter to the camera. "Clear" is to delete the information of the current preset, and "Call" is used to view the camera position saved by the current preset NO.

Note: A camera address can contain multiple preset No., but one preset No. corresponds to only one camera address. Camera Settings and Mic Settings have preset NO., serial port numbers, camera addresses, and protocols, which need to be considered in actual situations.

Set Camera Tracking



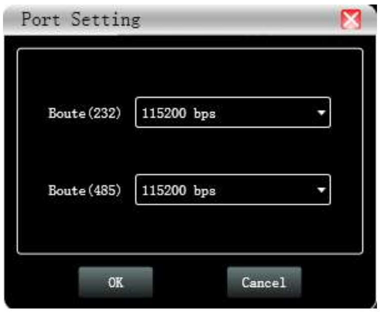
Default mic: when all mics have no input, turn the camera to the default MIC setting or send the associated command defined by the default MIC.
Tracing threshold: Indicates that the detected input signal must be greater than or equal to the tracing threshold. The system automatically enables tracing parameter.

Speech gap: the maximum discontinuous time of a valid signal. If the microphone is used to speak, the reaction time is set to 3 seconds. The signal considered to be continuously valid within 3S of the pause during speech, and invalid if it exceeds 3S.
Rotation time: the minimum speaking time required for the camera to switch to a valid position. If the microphone is used to speak for longer than the "rotation time", the channel signal is regarded as valid, and then the camera will automatically switch to the set position. Usually the "rotation time" is greater than the "rotation period".
Rotation interval: indicates the interval for sending the camera switching command or user-defined command. If the interval is 0, no camera switching command is sent.

Set Mic Tracking

Mic No.: corresponds to the input channel of device. (parameter need to be set separately for each channel)
Priority: Higher number for priority. If the priorities are the same, the processing is performed in the sequence of triggering priorities. If two mics speak at the same time, the camera automatically rotates to the preset position corresponding to the mic with a higher priority or sends the command corresponding to the mic with a higher priority. However, if the two mics have the same priority, the signal detected first prevails.
Active: Enables camera tracking for this channel.
Apply: Saves the current microphone camera tracking parameter to the device. (After camera tracking is enabled, the parameter must be applied to take effect)
The preset point, serial port number, camera address, and protocol are related to the camera and must correspond to the actual camera connection.

5.7 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.

5.8 Menu - Preset

Save: select the saved gear, save all the parameter of the current automatic gear of the machine to the device preset (2~59 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.

5.9 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.

Chapter 6 FIR filter and function (available in MAP08E)

6.1 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:

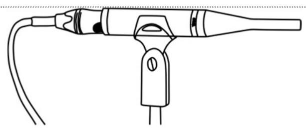
Delay = (1/Sampling Hz)*Taps/2

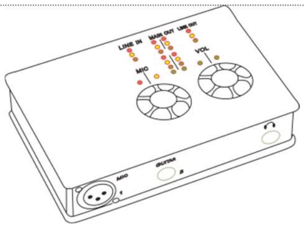
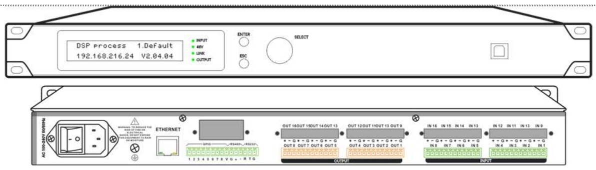
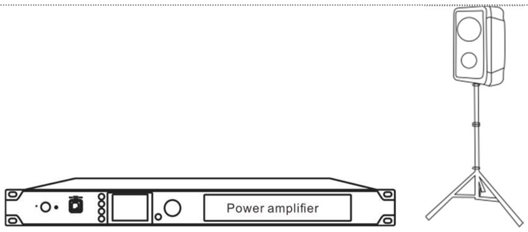
Taps / Sampling	48kHz	96kHz
256	2.67ms, LF 563Hz	1.33ms, LF 1125Hz
512	5.33ms, LF 279Hz	2.67ms, LF 558Hz
768	7.99ms, LF 188Hz	4.00ms, LF 375Hz
1024	10.67ms, LF 141Hz	5.33ms, LF 281Hz
2048	21.33ms, LF 70Hz	10.67ms, LF 141Hz

Applications:

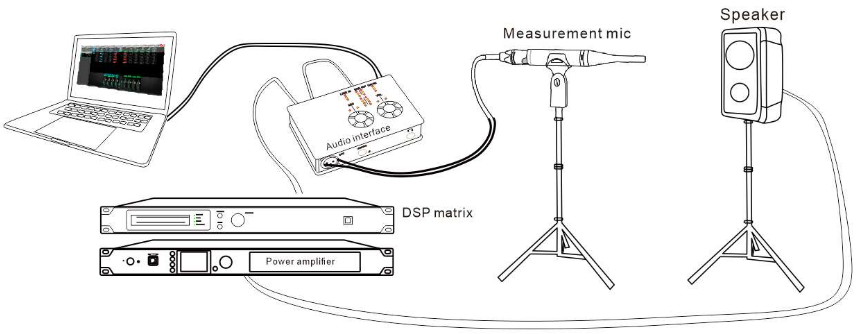
- Linear of the phase curve of the speaker;
- Match the phase and magnitude of different speaker models within the same product line, as well as different speaker models in the installation project to make it easier to debug speaker groups and arrays;
- Dealing with linear array systems (for audience 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/REW, MDM-DSP)	×1	
FIR matrix processor	×1	
Power amplifier and speaker	×1	

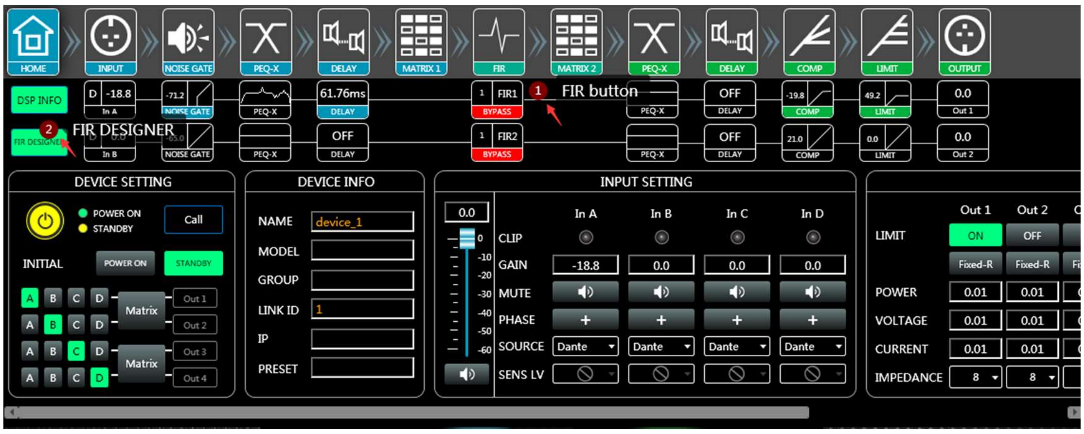
Connection schematic diagram:

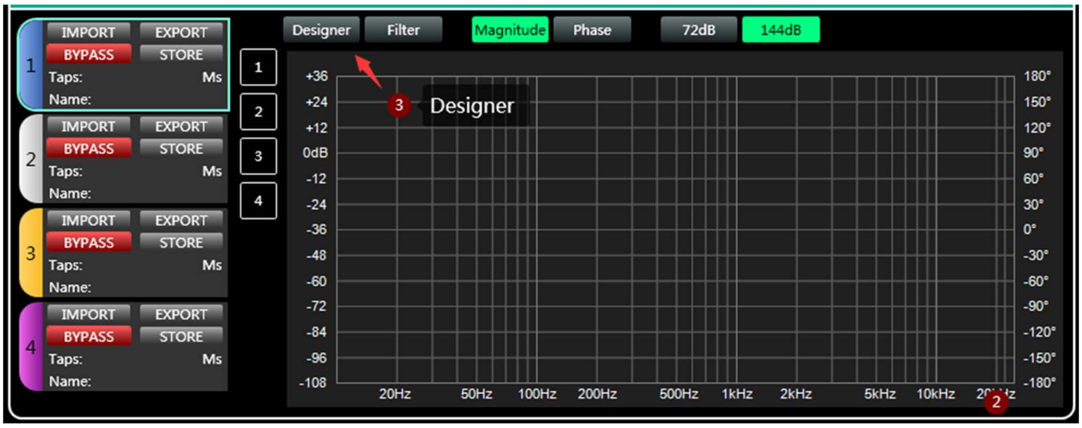


6.2 FIR DESIGNER in MDM-DSP to set FIR magnitude and phase

Beside using third party software, MDM-DSP provides user a more convenient way to set FIR magnitude and phase of each channels.

There are two ways to open FIR DESIGNER interface:



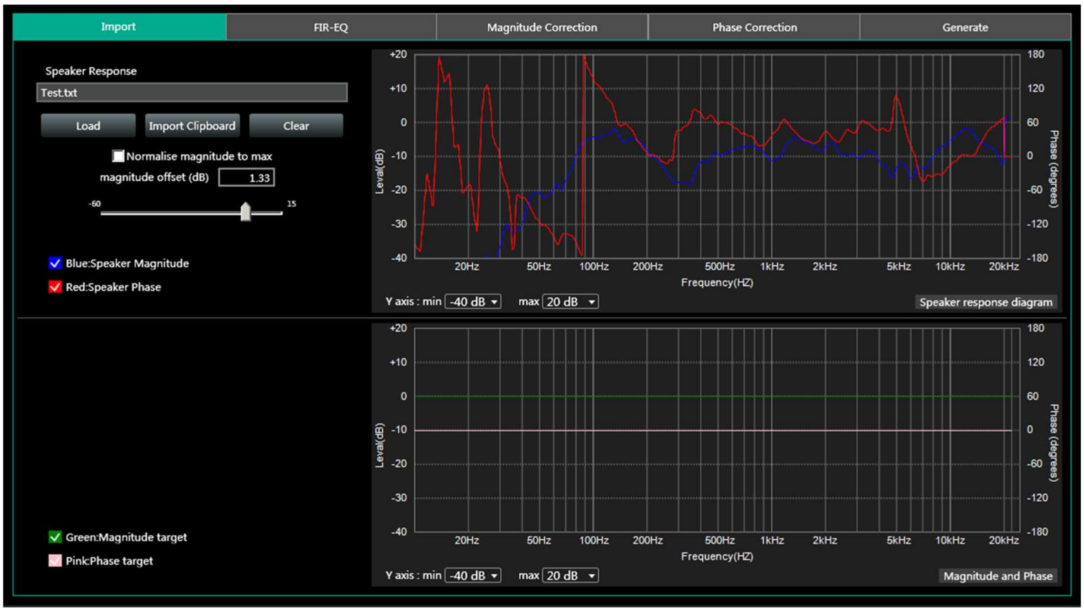


① Click “FIR” - “Designer” button to enter FIR automatic linear magnitude and phase function interface.

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.

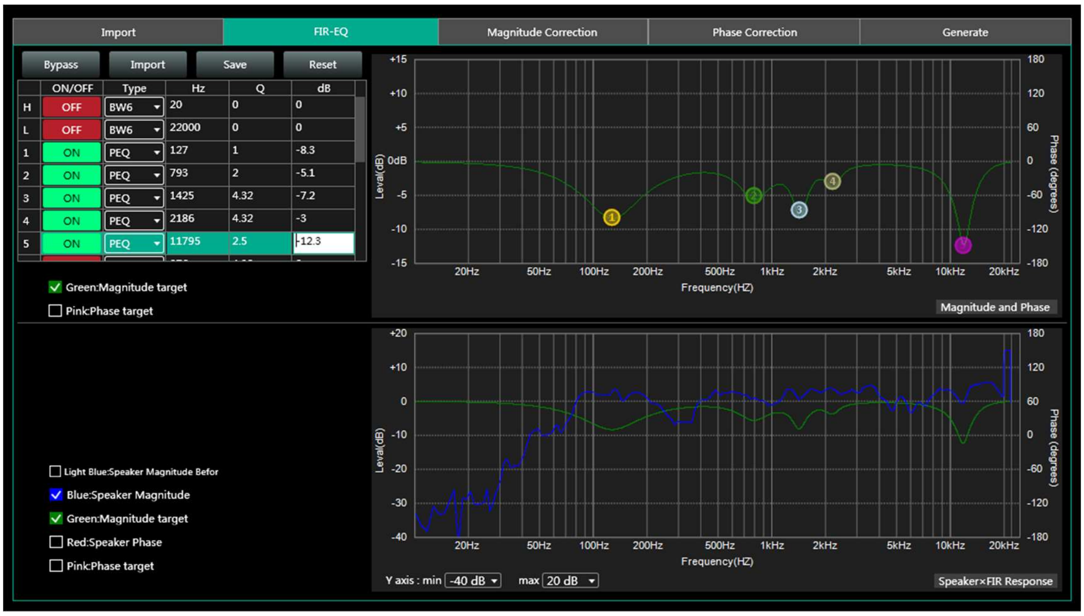
Let’s begin to set:

6.2.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.

6.2.b FIR DESIGNER - FIR-EQ

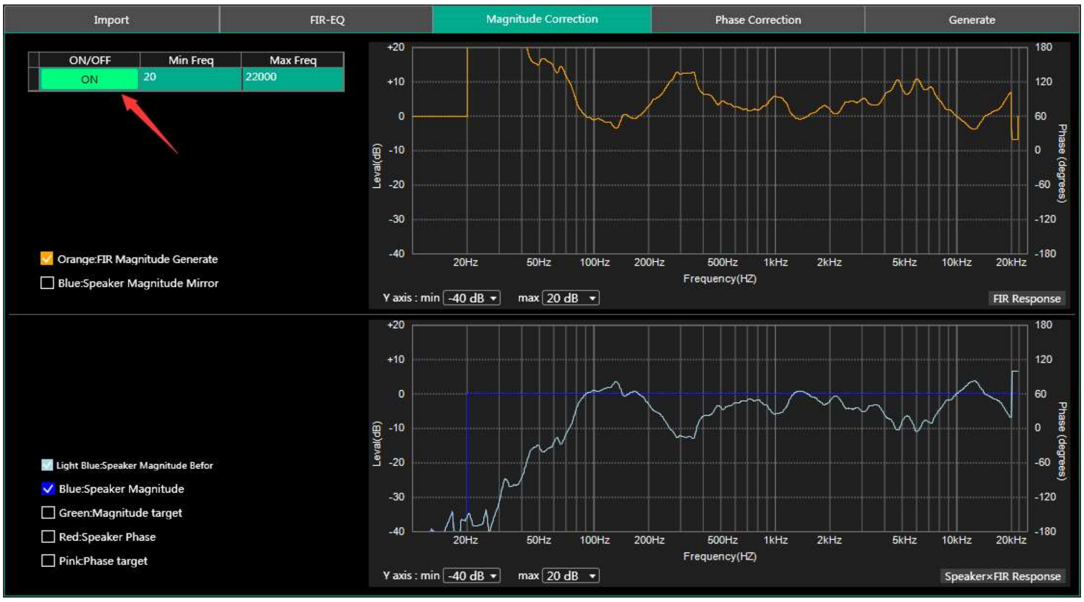


There are High pass filter and low pass filter for setting frequency divider, and 15 bands of PEQ \ LSLV \ HSLV to adjust magnitude. Try to set a linear magnitude of target speaker.

Mark: changing FIR magnitude doesn't effect its phase.

6.2.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.

6.2.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.


Import
FIR-EQ
Magnitude Correction
Phase Correction
Generate
Num.Taps
Ms
Design sample rate
512
5.33
48 kHz
FIR Name:
Default 1
Channels
FIR1 - 2048
STORE
EXPORT
Orange:FIR Magnitude Generate
Yellow:FIR Phase Generate
White:FIR Generate
Successful execution.
OK
FIR Response
Blue:Speaker Magnitude
Red:Speaker Phase
Green:Magnitude target
Pink:Phase target
Leval(dB)
Phase (degrees)
Frequency(HZ)
Y axis : min -40 dB
max 20 dB
Speaker×FIR Response


IMPORT
EXPORT
BYPASS
STORE
Taps:512
5.33
Ms
Name:Default 1
Taps:
Name:
Designer
Filter
Magnitude
Phase
72dB
144dB
20Hz
50Hz
100Hz
200Hz
500Hz
1kHz
2kHz
5kHz
10kHz
20kHz

DSP MATRIX PROCESSOR