DSP36-II
DSP48-II
User Manual
software-v0.3.3

2 IMPORTANT SAFETY INSTRUCTIONS

The Lightning flash with arrowhead symbol within an equilateral triangle is intended to alert the user to the presence of uninsulated “dangerous voltage” within the product’s enclosure,that may be of sufficient magnitude to constiute a risk of electric shock to persons.

The exclamation point within an equilateral triangle is intended to alert the user or the presence of important operating and maintenance(servicing)instructions in the literature accompanying the appliance.

1. Read these instructions.
2. Keep these instructions.
3. Heed all warnings.
4. Follow all instructions.
5. Do not use this apparatus near water.

6. Clean only with dry cloth.
7. Do not block any ventilation openings. Install in accordance with the manufacturer's instructions.
8. Do not install near any heat sources such as radiators, heat registers, stoves, or other apparatus (including amplifiers) that produce heat.
9. Do not defeat the safety purpose of the polarized or groundingtype plug. A polarized plug has two blades with one wider than the other. A grounding type plug has two blades and a third grounding prong. The wide blade or the third prong are provided for your safety. If the provided plug does not fit into your outlet, consult an electrician for replacement of the obsolete outlet.
10. Protect the power cord from being walked on or pinched particularly at plugs, convenience receptacles, and the point wherethey exit from the apparatus.
11. Only use attachments/accessories
12. Use only with the cart, stand, tripod, bracket, or table specified by the manufacturer, or sold with the apparatus. When a cart is used, use caution when moving the cart/apparatuscombination to avoid injury from tip-over.

13. Unplug this apparatus during lightning stormsor when unused for long periods of time.
14. Refer all servicing to qualified servicepersonnel. Servicing is required when the apparatus has been damaged in any way, such as power-supply cord or plug is damaged, liquid has been spilled or objects have fallen into the apparatus, the apparatus has been exposed to rain ormoisture, does not operate normally, or has been dropped.
15. This apparatus shall not be exposed to dripping or splashing. And no object filled with liquid such as vases, shall be place on the apparatus.
16. Do not overload wall outlets and extension cords as this can result in a risk of fire or electric shock.
17. This apparatus has been designed with Class-I construction and must be connected to a mains socket outlet with a protective earthing connection (the third grounding prong).
18. This apparatus has been equipped with a rocker-style AC mains power switch. This switch is located on the rear panel and should remain readily accessible to the user.
19. The mains plug or an appliance coupler is used as the disconnect device, so the disconnect device shall remain readily operable.

WARNING-To reduce the rish of fire or electric shck,do not expose this appliance to rain or moisture.

CAUTION-Internal lithium battery.Danger of explosion if battery is incorrectly replaced. Replace only with the same or equivalent type.

WARNING-This equipment has been designed be installedby qualified professionals only! There are many factors to beconsidered when installing professional sound reinforcement systems,including mechanical and electrical considerations,as well as acoustic coverage and performance.MARANI Commercial strongly recommends that this equipment be installed only by a professional sound installer or contractor

Please think of our environment and don’t bin any materials, including this manual. When the producthas reached the end of its useful life, please dispose of it responsibly through a recycling centre.

CAUTION

Risk of electric shock-do not open risque de choc electrique ne pas ouvrir

Caution:to reduce the risk of electric shock do not remove cover(or back) no user-serviceable parts inside refer servicing to qualified personnel attention:pour eviter les risques de choc electrique,ne pas enlever le couvercle aucun entretien de pieces interieures par l’usager. confier l’entretien au personnel qualifie avis:pour eviter les risques d’incendie oud’electrocution,

4-INSTALL

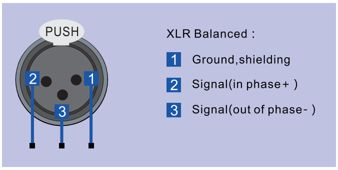
4.1-AC power input

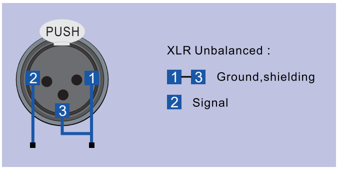
The AC voltage provided by the audio processor with built-in global voltage operation must be within ±10% of the specified line voltage (100~240V). The third pin (grounding pin) on the power cord that comes with the package is A necessary safety component, please do not try to disable the ground connection by using an adapter or other methods.

4.2-Signal input and output

4.2.1 -Analog signal input

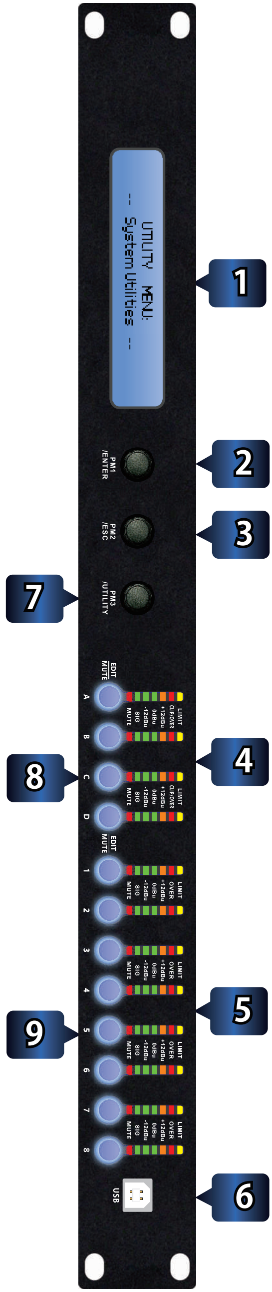
Like all digital signal processing equipment, the signal level supplied to the unit must be appropriate to avoid working in a low signal-to-noise ratio or distortion. The MIR-E series processors can accept up to +20dBu analog signal level, which is greater than the linear output level of most mixers, so the impact of this type of problem is reduced. When performing equalization processing, pay attention to the boosted gain value not to be too large, so as to avoid the gain of the unit from causing digital clipping. (Of course, you can use the built-in anti-clipping function to prevent this event from happening). It must be noted that the maximum input level in the specification is a clipping level, not a safe practical level. It must be ensured that the clipping level is not lower than the next device in the signal chain, and a certain margin must be left during use.





4.3.Overview of the processor front panel

1. 2*24 LCD display.
2. The PM1 knob is responsible for the main function switching, menu up and down.ENTER, click to next step
3. The PM2 knob is responsible for turning on/off some functions and coarse adjustment of some values.ESC, click to escape
4. Input signal level meter: display the pre-fader signal, the mute does not affect display. When the Mute light is on, it indicates that the current channel is muted, and when the SIG light is on, it indicates that the input signal reaches -40dBu; -12dBu, 0dBu, +6 dBu, and +12 dBu represent the actual RMS value of the signal. When the Clip/Over light is on, it means that the signal has been distorted before the analog-to-digital conversion. The Limit light will be on when the input channel compressor is activated.
5. Output signal level meter: display the post-fader signal, When the Mute light is on, it means that the current channel is muted, level meter does not display any value after mute. When the SIG light is on, it means that the input signal reaches -40dBu; -12dBu, 0dBu, +6 dBu, and +12 dBu represent the actual RMS value of the signal. When the Over light is on, it means the signal reaches the Hard limiter threshold value, the Limit light will light up when the output channel RMS compressor and peak limiter are activated.
6. USB Type B interface, for PC connection.
7. The PM3 knob is responsible for turning on/off some functions and fine adjustment of some values.UTILITY, click to escape
8. Input channel selection key: Click this key to edit the processing of the current input channel, including channel name, gain/polarity/delay/PEQ/compressor and other parameters. Hold for three seconds to mute the current channel.
9. Output channel selection button: Click this button to edit the processing of the current input channel; include input channel matrix routing, high and low pass filter frequency, slope, filter type; also include gain/polarity/delay/parametric equalization /RMS compressor/peak limiter/hard limiter and other parameters. Hold for three seconds to mute the current channel.



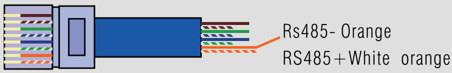
4.4.Overview of the processor back panel

1 AC power input, standard IEC interface, please ensure that the power grounding pin is well grounded, otherwise electric shock may occur.

2 Power switch.

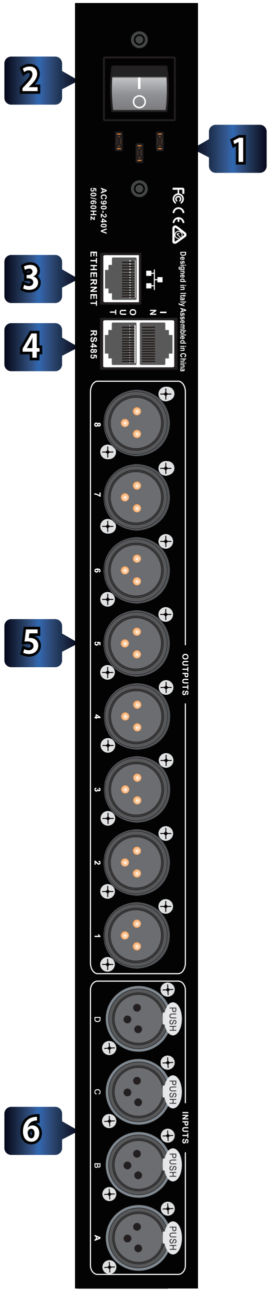
3 LAN control interface, supports TCP/UDP protocol, IP address defaults to DHCP.

4 Rs485 protocol interface, providing 1 input and 1 output dual interface, which can be used to connect to software, and can also be used for 3rd device protocol transmission.



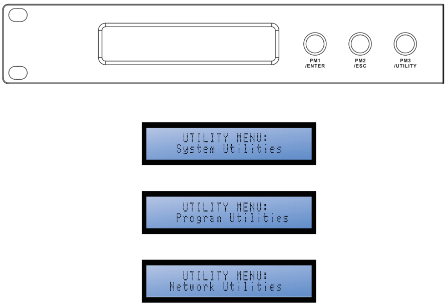
5 Analog signal output interface, maximum output level +18dBu, minimum load 100O.

6 Analog signal input interface, maximum input level +20dBu, input impedance 20KO.



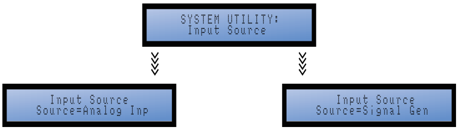
5.1 System Settings

Utility general menu contains 3 sub-menu System settings,Under Utility Menu, turn the PM1 navigation key to select System Utilities



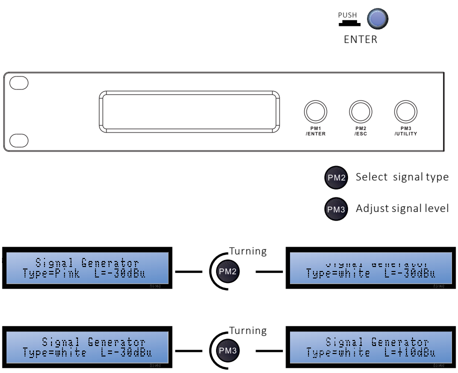
5.1.1 Input Source Selection

Press PM1 to enter the secondary sub-menu, where you can select Analog input or built-in Signal generator.



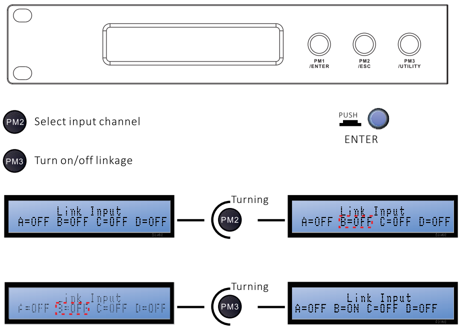
5.1.2 Signal Generator Settings

Use the PM2 knob to select pink noise or white noise, and use the PM3 knob to adjust the signal level from -30dBu to +10dBu, with a step of 1dB.



5.1.3 Link Input

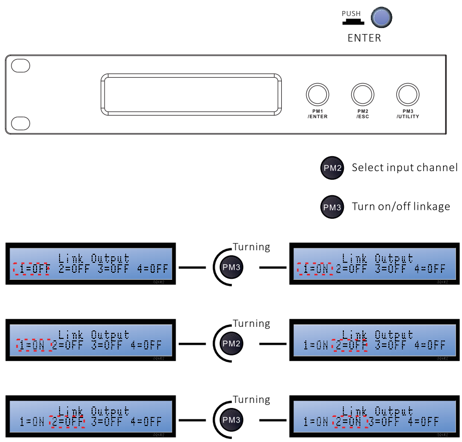
For the linked input channel, modify the settings of any one of the linked channels, and the other linked channels will synchronously modify the same value.



Turn the left and right knobs PM2 to select the ABCD channel, turn PM3 to the right to turn on the selected channel linkage, turn PM3 to the left to turn off the selected channel linkage.

5.1.4 Linkage output

For the input channel after linkage, modify the settings of any linkage channel, and the other linkage channels will simultaneously modify the same value.



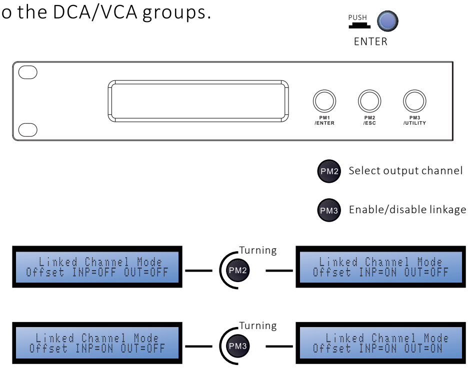
Knob PM2 selects the 1234 channel, turn PM3 to the right to turn on, and turn PM3 to the left to turn off the linkage.

For example, if you need to link output channels 3 and 4, the operation sequence is:

- Click the PM3 button
- Use the PM1 knob to select system
- Click PM1 to enter the submenu
- Rotate PM2 to the right to select 3 channels
- Turn right PM3 to turn on
- Continue to rotate PM2 to the right to select 4 channels
- Turn right PM3 to turn on

5.1.5 Keep Linked channels level offset (linked by DCA mode)

Allows the fader to be link in equal proportions in different positions. When the proportion of the input or output channels needs to be adjusted as a whole during use, the default linkage function will make the gain of each channel change along with the first changed channel. If you do not want to use it If the gain ratio changes, you can use this function, similar to the DCA/VCA groups.

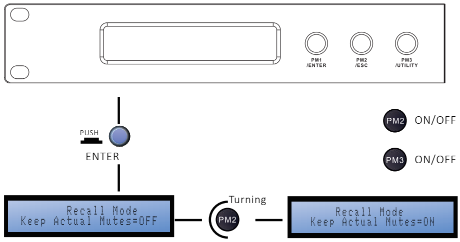


When the channel linkage is not turned on, this function is not av

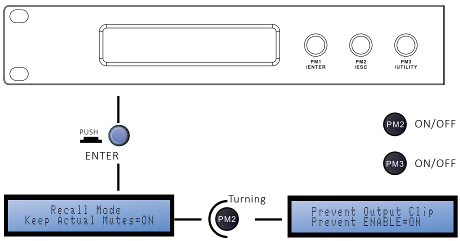
5.1.6 Recall mode (mute safety)

When this mode is turned on, the current channel mute state will be maintained regardless of reading any preset, and only the muted state of the preset will be read when this mode is turned off.

This function can prevent the loud sound pressure from the loudspeaker from frightening the staff when switching presets.



Protect the output from distortion Keep the signal after the processor A/D will not be distorted, and ensure that no distortion occurs in the processor regardless of any improvement.



5.1.7 View firmware version

You can check the firmware version of the current processor, you can go to the our website to download the latest version of the firmware to get function updates and bug fixes.

Among them, FP is the front panel firmware and MB is the main board firmware.

Note:

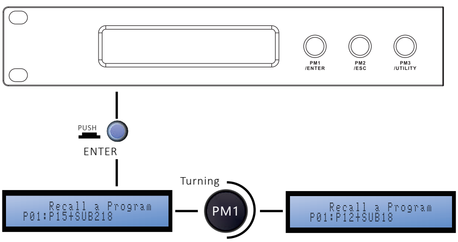
1. The firmware version can be upgraded online.
2. Back up all presets before the firmware upgrade.
3. During the upgrade, keep the power supply intact. If the upgrade fails midway, irreversible data loss may occur.

5.1.8 Program (preset) settings

Under UTILITY MENU, turn the PM1 knob to select program Utilities.

5.1.8.1 Read preset

Can read previously saved presets from this unit (Before pressing read, please save the current preset, otherwise the current modification will be lost) Add a box.



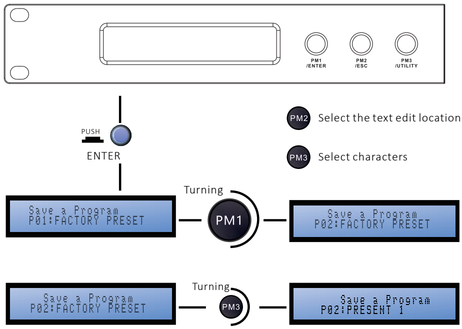
For example, read the present 3 from the machine

Press utility

- Turn the PM1 knob to the right
- Go to Program Utilities and press Pm1
- Press PM1 to Recall a Program
- Turn the PM1 knob to the right
- Choose P03
- Press PM1 to wait for the machine to finish reading

5.1.8.2 Save preset

Save the current input / output / routing, and the entire state of the filter to the machine.



For example, if some changes have been made to the channel and need to be saved in the processor, follow the steps below to save:

- Click the PM3 button
- Use the PM1 knob to select program
- Press PM1 to enter the submenu
- Rotate PM1 to the right to select save a program
- Press PM1 to enter the next level
  Rotate PM1 to the right to select the preset position (such as P01)
- Press PM1 to confirm

- Enter the name, PM1 select the character position, PM2 select the character from the gallery below, the maximum allowable 16 characters

	!	”	#	$	%	&	’	(	)	*	+	,	-	.	/	0	1	2	3	4	5	6	7
8	9	:	;	<	=	>	?	@	A	B	C	D	E	F	G	H	I	J	K	L	M	N	O
P	Q	R	S	T	U	V	W	X	Y	Z	[	¥	]	^	_	`	a	b	c	d	e	f	g
h	i	j	k	l	m	n	o	p	q	r	s	t	u	v	w	x	y	z	{	|	}	→	←

- After the modification is completed, click PM1 to save

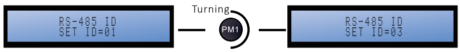
5.1.9 Network options

Under the settings, turn the PM1 knob, the last option is the network setting, the network setting is used to connect to the processor management software, and provides 3 different interfaces, namely USB, RS485, LAN,Among them, the USB connection is the easiest, and the installation driver can be directly connected to the PC;RS485 needs to set the serial port number, ID, baud rate, etc., and need to be consistent with the PC settings;LAN is suitable for multiple processors or long-distance connections. Through the local area network, stable connection quality and high connection speed can be obtained.
The MIR series processors provide three connection methods: USB/RS485/TCPIP, and you can choose any one of them to connect without choosing.
The RS485 serial port can be set in the network option menu, and the network settings of the machine can be viewed.

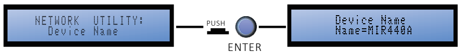
5.1.9.1 Set RS485 ID number

When multiple devices are connected via RS485 protocol, they need to be set to different ID numbers and cannot be repeated. After clicking PM1 , rotate PM2 to select ID numbers from 1-32.
For example, the steps to set 485ID to 3 are:

- Click the PM3 button
- Use the PM1 navigation knob to select Network
- Click PM1 to confirm
- Rotate PM1 to the right to select RS-485 ID
- Click PM1 to confirm
- Rotate PM2 to the right to select 03
- Click PM1 to confirm



5.1.9.2 Device name



The model of the machine is displayed by default, if you need to modify it, please follow the following order:

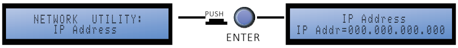
- Click the PM3 button
- Use the PM1 knob to select Network
- Click PM1 to confirm
- Rotate PM1 to Device Name
- Click PM1 to confirm
- Rotate PM2 to the right to select the edit character position
- Rotate PM3 to the right to select characters
- Rotate PM2 to modify the character position, PM3 to select the character, the maximum allowable 16 characters

	!	”	#	$	%	&	’	(	)	*	+	,	-	.	/	0	1	2	3	4	5	6	7
8	9	:	;	<	=	>	?	@	A	B	C	D	E	F	G	H	I	J	K	L	M	N	O
P	Q	R	S	T	U	V	W	X	Y	Z	[	¥	]	^	_	`	a	b	c	d	e	f	g
h	i	j	k	l	m	n	o	p	q	r	s	t	u	v	w	x	y	z	{	|	}	→	←

- Enter to confirm the modification and return

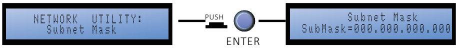
5.1.9.3 IP IP address

Only the IP address is displayed here, and the default is automatic acquisition, and the real address will be displayed after the connection is successful. It cannot be modified on the panel.



5.1.9.4 Subnet mask

The display is 255.255.255.0, which is only for display and cannot be modified on the panel.



5.1.9.5 Gateway

Only for display, if you need to modify, you need to use the software to modify online.

5.1.9.6 LAN Mode

The LAN mode defaults to DHCP, and the panel is only used as a display. It can be modified to DHCP or a fixed IP address through software. When the number of devices exceeds a certain number, that is, there are too many DHCP servers in the LAN, network congestion may occur, so it is recommended to >5 pcs processors When connecting at the same time, use the fixed IP method to connect.

5.2 Setting of input channel

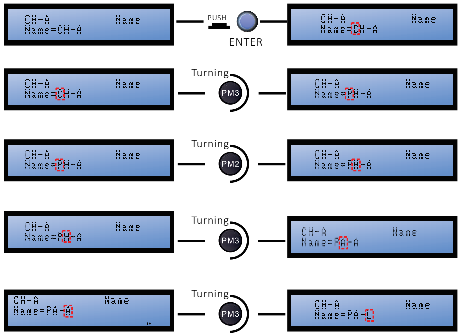
Edit input channel

Click the EDIT button below the processor level meter to edit the processing module of the input channel, long press to mute the current channel.

The level meter of the input channel shows the pre-fader level, that is, it is not affected by mute; the level meter of the output channel shows the post-fader level, and the level is not displayed after the mute.

5.2.1 Channel name editing

Click the edit button and the first item displayed is the channel name edit.



The default display, if you need to modify, please follow the following order:

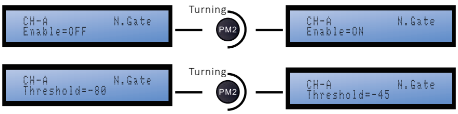
- ●Click the Edit button to edit the channel
- ●Click PM1 to confirm
- ●Rotate PM2 to the right to select the edit character position
- ●Rotate PM3 to the right to select characters
- ●Rotate PM2 to modify the character position, PM3 to select the character, the maximum allowable 6 characters

	!	”	#	$	%	&	’	(	)	*	+	,	-	.	/	0	1	2	3	4	5	6	7
8	9	:	;	<	=	>	?	@	A	B	C	D	E	F	G	H	I	J	K	L	M	N	O
P	Q	R	S	T	U	V	W	X	Y	Z	[	¥	]	^	_	`	a	b	c	d	e	f	g
h	i	j	k	l	m	n	o	p	q	r	s	t	u	v	w	x	y	z	{	|	}	→	←

- ●Click PM1 to save after modification

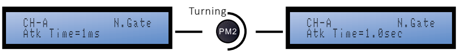
5.2.2 Noise gate

The noise gate can appropriately suppress the obvious background noise caused by the accumulation of front-end equipment or improper system settings. On the current page, click PM1 to edit the overall state of the noise gate, and rotate PM2/PM3 to open or close the noise gate.



The threshold can be adjusted from -80 dBu to -45 dBu through PM2.

The startup time can be adjusted from 1ms to 1000ms by rotating PM2.



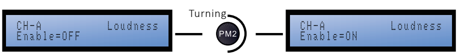
The release time can be adjusted from 1ms to 1000ms by rotating PM2.

5.2.3 Dynamic loudness booster

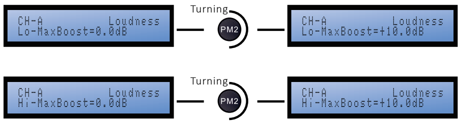
In the input channel, the working principle is based on the equal loudness curve of the human ear to compensate and enhance the ultra-low frequency and ultra-high frequency.

When the signal strength is low, the boost ratio is relatively large; when the signal strength is high, the boost ratio is relatively small or even zero.

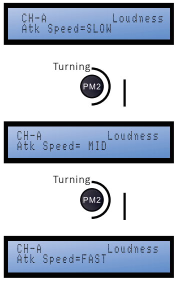
The maximum allowable boost gain value can be set separately for UHF and UHF, and the setting of starting speed is also provided.



When adjusting the maximum allowable gain, the adjustment step of the knob PM2 is 1dB, and the adjustment step of PM3 is 0.1dB.

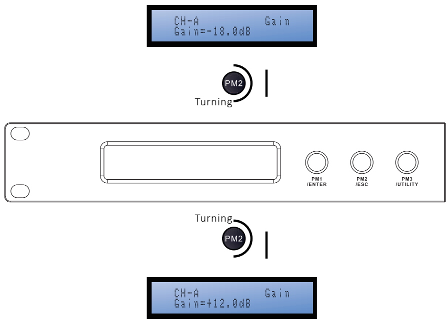


There are three levels of start-up time adjustable, namely fast, medium and slow.



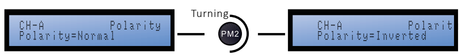
5.2.4 Gain

Controllable gain range from -18 to + 12dB, 0.1dB step.



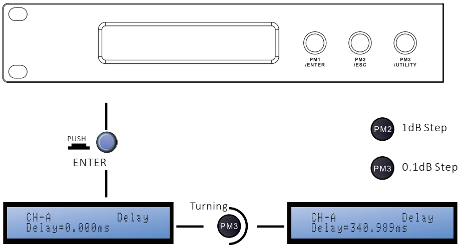
5.2.5Polarity

The overall polarity can be selected as positive polarity or -180° reverse polarity, which can be used to match the phase of the loudspeaker or correct the overall inversion caused by the wrong connection of the signal cable.



5.2.6 Delay

The input part can provide a maximum delay of 340 milliseconds, which can be used to delay the alignment of the tower speaker and the main amplifier speaker, with a maximum distance of 115 meters.



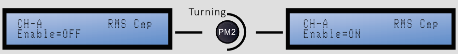
5.2.7 Compressor

RMS compressor, reduce the start threshold to use the compressor to control the dynamics of the input signal.

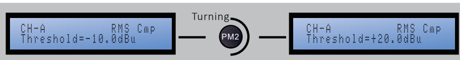
Using a lower start time (such as 5ms) and a higher release time (such as 1000ms) can firmly control the input signal within a certain range, of course, this will bring an obvious sense of compression (flatness).

Setting different start/release times in combination with music types will bring different listening feelings, and make up can also be used to compensate for the loss of loudness. And with a variable knee setting, the default 0 is the hard knee, and the threshold is the compression starting point; 100% is the soft knee, when the compression starting point is lower than the set threshold, and gradually exceeds the threshold to achieve full effect.

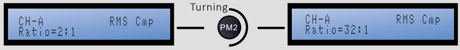
Under high compression ratios, it is easy to hear unnatural and abrupt sounds when using hard knees. On the contrary, using soft knees can make the compression transition smoothly and bring a natural sense of hearing.



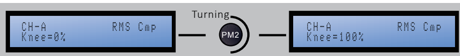
Compressor switch



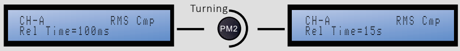
The start threshold can be adjusted from -10dBu to +20dBu.



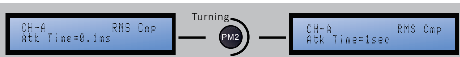
The compression ratio can be adjusted from 2:1 to 32:1.



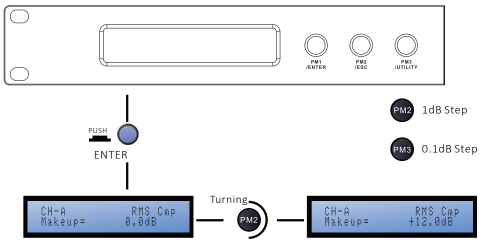
The soft and hard inflection point can be adjusted, among which 0% is the hard inflection point and 100% is the soft inflection point.



The adjustable range of the release time is: 100ms to 15000ms.



The start-up time can be adjusted from 0.1ms to 1000ms.



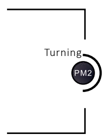
Provides automatic compressor settings, and automatically sets the compression start threshold

5.2.8 Equalizer Mode

Provide 4 combinations,

When multiple PEQs are needed, 31 bands of PEQ can be provided at most;

You can also use the Target EQ+PEQ that can be saved separately;



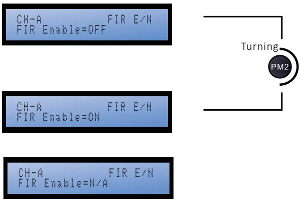
5.2.9 FIR filter

The input channel allows the user to use a 512taps FIR filter, its operating sampling rate is 48kHz.

But do not do any processing for the signal after 24000Hz, keep it as it is.

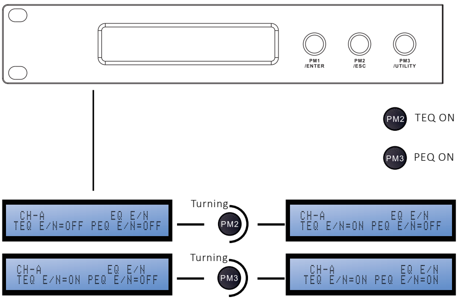
You can connect the software to edit the FIR filter, import or generate FIR high/ low / band pass filters according to the guide.

When PEQ mode is selected as with FIR filter, this option can be turned on.



5.2.10 Equalizer state

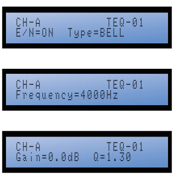
PEQ overall on/off, TEQ overall on/off.



Note: When PEQ mode is selected as full PEQ, TEQ is not available, that is, N/A is displayed.

5.2.11 TEQ (Target EQ)

Types include: BELL, HiShvQ, LoShvQ, LP Q, HP Q.

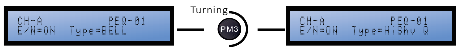


5.2.12 PEQ1-31

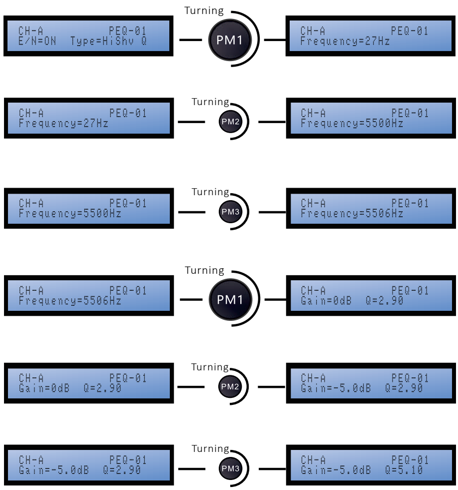
Specific turning of each band of parametric equalization, after Enter, you can rotate PM1 to select the filter type, you can select high and low Shelf /Bell/High and low pass/all pass/notch, etc. rotate the PM1 button to enter the specific parameter adjustment, PM2= gain, PM3= Q value , Frequency and so on.

To add a high-shelf Q with a frequency of 5506Hz, a bandwidth of 5.1, and a gain of -5dB to input channel B, follow the steps below.

- Click the Edit button of input B
- Use the PM1 knob to select PEQ1
- Click PM1 to enter the edit
- Rotate PM3 to the right to select HiShvQ
- Continue to rotate PM1 right to select frequency adjustment
- Turn PM2 to the right to adjust the frequency to 5500Hz, use PM3 to fine-tune to 5506Hz
- Click PM2 to return
- Right-turn PM1 to adjust gain and Q value
- Use PM2 to adjust gain and PM3 to adjust Q value
- Complete and return



Types include: BELL, HiShv1, HiShv2, HiShvQ, LoShv1, LoShv2, LoShvQ, LP 1st, LP 2nd, LP Q, HP 1st, HP 2nd, HP Q, Notch, AllPs1, AllPs2.



5.3 Edit output channel

EDIT button click into the lower level processor can edit the table processing module output channels.

Long press the Edit button to mute the current channel, the level meter of the output channel displays the post-fader level, and does not display the level after mute.

5.3.1 Channel name

1 Channel name edit

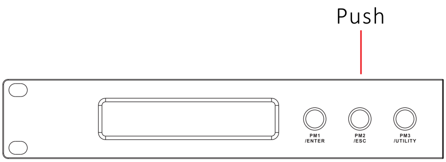
Click the Edit button and the first item displayed is the channel name edit

The default display, if you need to modify, please follow the following order:

- Click the channel edit button
- Click PM1 to confirm
- Rotate PM2 to the right to select the edit character position
- Rotate PM3 to the right to select characters
- Rotate PM2 to modify the character position, PM3 to select the character, the maximum allowable 6 characters

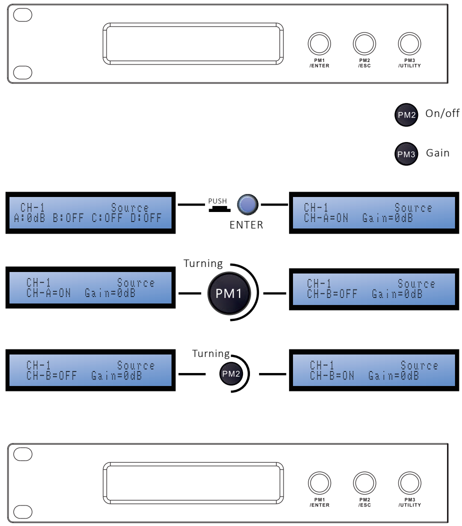
	!	”	#	$	%	&	’	(	)	*	+	,	-	.	/	0	1	2	3	4	5	6	7		
8	9	:	;	<	=	>	?	@	A	B	C	D	E	F	G	H	I	J	K	L	M	N	O		
P	Q	R	S	T	U	V	W	X	Y	Z	[	¥	]	^	_	`	a	b	c	d	e	f	g		
h	i	j	k	l	m	n	o	p	q	r	s	t	u	v	w	x	y	z	{	|	}	→	←		

- Click PM1 to save after modification



5.3.2 Input matrix

The input matrix allows the user to mix 4 input signals to any output channel in any ratio. The knob PM1 selects the ABCD input source, and the PM3 adjusts the gain amount, which can be adjusted from -30 to 0dB.

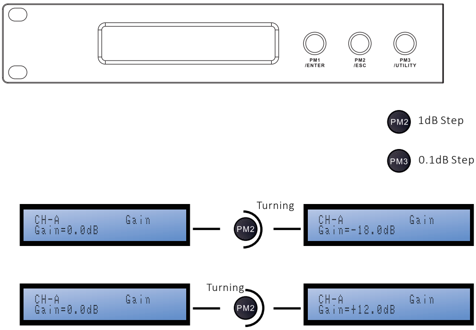


The default route of processor is channel ABCD pass-through output 12345678. In actual use, it may be necessary to assign one input signal to multiple output channels, such as routing input channel A to output 1/2 (since output channel 1 has sent 0dB from input A by default, so directly start from channel 2).

- Press the select button for output 2
- Select the PM1 navigation button to go to the Source screen
- Press Pm1
- To rotate PM1, select B
- Rotate PM3 to adjust the gain of input B to minus infinity
- To rotate PM1, select A
- Rotate Pm3 to adjust the gain of input A to 0dB
- press PM2 to complete

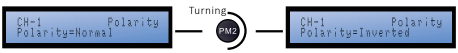
5.3.3Gain

The controllable gain range is from -18 to +12dB in 0.1dB steps.



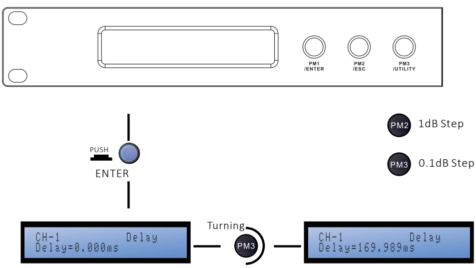
5.3.4 Polarity

The overall polarity can be selected as normal polarity or -180° reverse polarity, which can be used to match the phase of the loudspeaker or correct the overall inversion caused by the wrong connection of the signal cable.



5.3.5 Delay

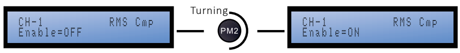
The output part provides a maximum delay of 170 milliseconds, which can be used to align the time between the multi-way units, in a step of 10.4 microseconds/0.01 milliseconds.



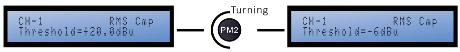
5.3.6 RMS compressor

It is mainly used to limit the RMS power of the unit. It needs to cooperate with the AES power provided by the unit manufacturer and the amplification factor of the power amplifier to calculate the threshold; the attack time is often determined by the period corresponding to the frequency of the high-pass filter, and the release time is always set to 16 times the startup time。

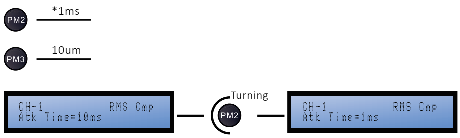
For example, the AES power (2h) of a HF driver is 100 watts, the impedance is 16 ohms, the crossover point is 1000 Hz, and the amplification factor of the power amplifier is 40 dB, then according to P=U2/R, the maximum input voltage of the unit is 40 volts, divided by The power amplifier magnification is 100 times, the voltage limit should be activated when the signal level is 0.4v, 0.4v is converted to 20log (0.4/0.775) to get -5.84 about -6dBu, that is, the threshold is -6 dBu; the cycle corresponding to the crossover point 1000 If it is 1 ms, then the start time can be set to 1 ms, and the release time is 100 ms.



Compressor switch.



The attack time can be adjusted from 0.1ms to 1000ms.



The start threshold can be adjusted from -10dBu to +20dBu.

Release Time can be adjust from 100ms～15000ms.

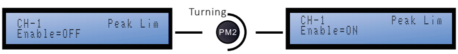
The compression ratio can be adjusted from 2:1 to 32:1.

The soft and hard inflection point can be adjusted, among which 0% is the hard inflection point and 100% is the soft inflection point.

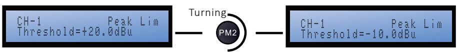
Compression compensation can be set range: 0~12dB.

5.3.7 Peak limit

It is mainly used to limit the peak signal and protect the woofer from mechanical damage caused by the voice coil movement exceeding the linear stroke. If you need to accurately set the stroke/frequency and voltage curve of the unit, you can consult the manufacturer of the unit Or measure and obtain data by yourself.
For example, using the ** model woofer, it is found that the stroke of the unit reaches the limit of 12mm at 103 volts. At this time, the amplifier magnification is 38dB, 103/80=1.2875v, 20log (1.2875/0.775)=4.4, that is, the peak compressor's The startup threshold is 4.4dBu.

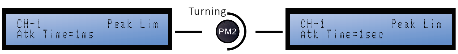


Peak limiter switch.



The threshold can be adjusted from -10dBu to +20dBu.

The adjustable range of the release time is: 100ms to 15000ms.



The attack time can be adjusted from 0.1ms to 1000ms.

5.3.9 X-over mode

The MIR-E series has built-in three kinds of crossover filters.

① Traditional IIR filter, including three types of Linkwiz-Raylei/Bessel/Butterworth, with a maximum slope of 48dB/oct.

② MIR linear phase IIR filter, using Marani's exclusive algorithm, retains the shape of the Linkwitz-Riley /Bessel/Butterworth filter types without phase distortion, maintaining the linear phase, and controlling the time delay at the same time In a reasonable range, it can directly replace the traditional IIR filter for speaker frequency division, and only need to align the delay.

③FIR finite impulse response filter, allowing users to customize the window function in the software to generate high-slope and steep filter shapes, and also accept FIR convolution generated by external third-party software based on measurement, which can realize complex filter shapes and multiple sections The parameters are equalized and the phase is corrected at the same time.

Only three modes are displayed here. The specific parameters are adjusted in the software high and low pass filter options, which cannot be adjusted on the front panel.

IIR mode

MIR mode

FIR mode

IIR+FIR mode (only 4 band of PEQ can be used in this mode)

The software contains 4 combinations of filter forms, including IIR/MIR/FIR/IIR+FIR four combinations are optional. When a single IIR/MIR/FIR mode is selected, a total of 8 band PEQ can be used, while when IIR+FIR filter is selected, only 4 band PEQ can be used.

5.3.10 FIR filter

The output channel allows the user to use a 512-taps FIR filter, and its operating sampling rate is 48kHz, but it does not do any processing for the signal beyond 24000Hz, , keep it as it is. (For the FIR filter status switch in the previous X-over mode)

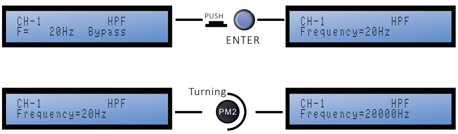
Select the mode with FIR filter to display ON/OFF.

Otherwise, it shows unavailable N/A.

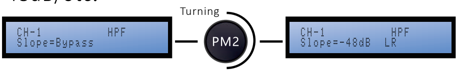
5.3.11 High pass filter

In IIR mode, the type, slope, frequency and other specific parameters of the high-pass filter can be adjusted.

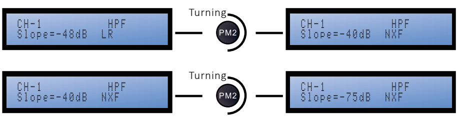
Click PM1 to edit.



Traditional filter categories include: Bessel, Butterworth, Linkwitz-Riley, with a maximum slope of 48dB/otc.



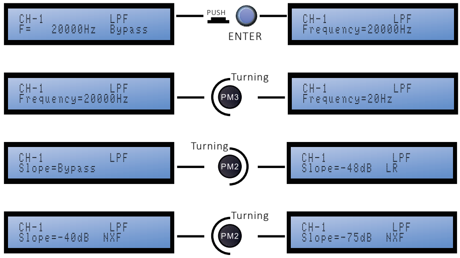
It additionally contains NXF （Notched X-over Filter）filter, the slope range is -40~-75dB/otc.



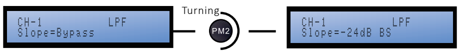
When the combination of IIR and FIR filter is selected, the highest slope of IIR filter is -24dB/Oct, and only 4 band PEQ are available.

5.3.12 Low-pass filter

In IIR mode, the type, slope, frequency and other specific parameters of the low-pass filter can be adjusted.



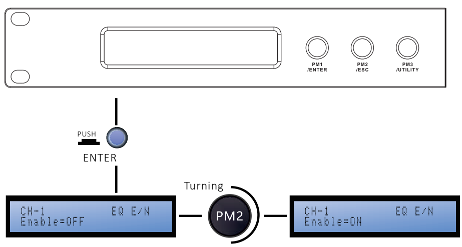
When the combination of IIR and FIR filter is selected, the highest slope of IIR filter is -24dB/Oct, and only 4 band PEQ are available.



When the MIR or FIR filter type is selected, the high/low pass filter cannot be adjusted on the panel, only by software.

5.3.13 Equalizer on state

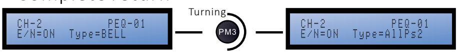
PEQ overall on/off.



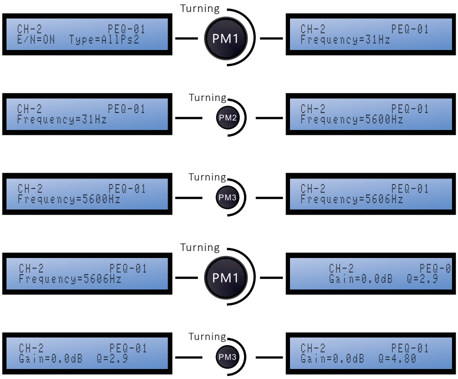
5.3.14 PEQ1-8

Specific turning of each stage of parameter equalization, after enter, you can rotate PM1 to select filter type, you can rotate high and low Chevron/Bell/high and low pass/all pass/notch, etc., rotate PM1 navigation key to enter specific parameter adjustment, PM2= gain, PM3= Q value, frequency and so on.

- Pressthe Edit buton of Output 2
- Rotate the PM1 to select PEQ1
- Press Pm1
- Rotate PM3 to the right to select AllPs2 filter
- Press Pm2
- Continue to rotate the PM1 selection frequency adjustment to the right
- Rotate PM2 to adjust the frequency to 5600Hz, use PM3 to fine tune to 5606Hz·
- Press PM2 to return
- Rotate PM1 right to adjusts the Q value
- Complete return



Types include: BELL, HiShv1, HiShv2, HiShvQ, LoShv1, LoShv2, LoShvQ, LP1st, LP 2nd, LP Q, HP 1st, HP 2nd, HP Q, Notch, AllPs1, AllPs2.

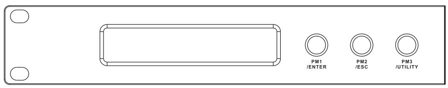


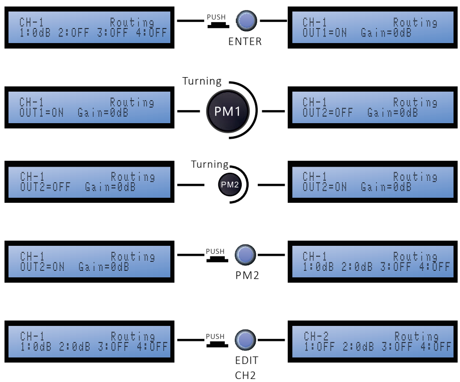
5.3.15 Output matrix

The output matrix allows the user to mix output channel signal to any physical output channel in any ratio. The knob PM1 selects the source, and the PM3 adjusts the gain amount, which can be adjusted from -30 to 0dB.

For example, to mix output channel 1.2 to physical output 2, you need to do the following：

- Click the EDIT button below output 1
- Rotate PM1 key to the end of CH1 routing
- Rotate PM1 to select output channel 2
- Rotate PM3 to adjust the gain to 0dB
- Click PM2 to return
- Click the EDIT button below output 2, and it will automatically appear on the Ch2 routing interface
- Rotate PM1 to select output channel 1
- Rotate PM3 to adjust the gain to 0dB Finish





7. Use of MIR-E series software

Minimum system requirements for running MIR-E software:

Operating system	Microsoft Windows
CPU	single core 2.0GHz
Running memory	2GB
Storage space	1GB

Need Microsoft Net Framework 4 or higher and Microsoft Visual C++2015-2022

The MIR-E series provides 3 kinds of control interfaces, namely USB/RS485/TCP IP, and there is no need to set up the connection switching of the three ways.

Connecting to a device

1 Using a network cable to connect it is the most direct and safe way of connecting. Use a network cable to connect the PC's network port directly to the processor's Ethernet port, and make sure that the computer's IP is set to automatically get connection (DHCP).

in order to get things working follow this actions sequence: Settings---Network and Internet---Ethernet---Change adapter options---Select the currently used adapter---Properties-Internet Protocol Version 4 (TCP/IPv4)---Select Automatic Get IP address.

After the network cable connection is confirmed, double-click the MIR-E series software, the device will be automatically discovered and a pop-up window will prompt.

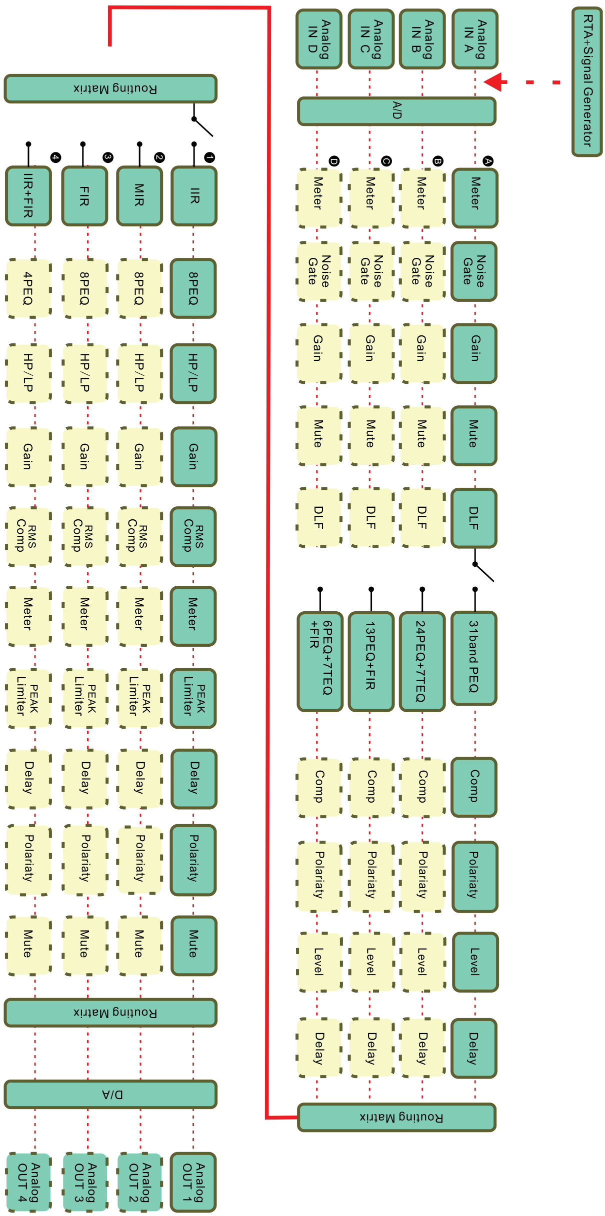
2 Use RS485 protocol connection, through USB to RS485 device, you can get the serial port of RJ45 port and connect directly to RS485-IN/OUT on the rear panel. The RS485 input and output on the rear panel are designed with loop-out design. When only one device is connected, Can be plugged into RS485-IN/OUT at will.

When using the USB interface to connect, you need to pay attention to the following points.

* When your PC operating system is Windows XP or Windows7, you need to install the STM32 virtual COM driver so that the processor can be correctly identified and connected.

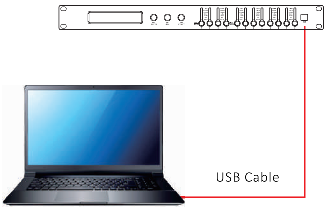
* When your PC operating system is Windows 8 or above, there is usually no need to install a driver, and the system will correctly identify the processor. (If the driver is not installed automatically, install the STM32 virtual COM driver manually).

P6 Overview of signal processing



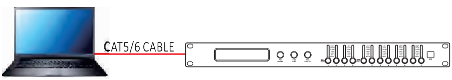
7.1 Connect with USB

When using the USB interface, directly connect to the USB port of the PC with a B-type cable, add USB to the software, and select the corresponding model.



7.2 Use network cable to connect

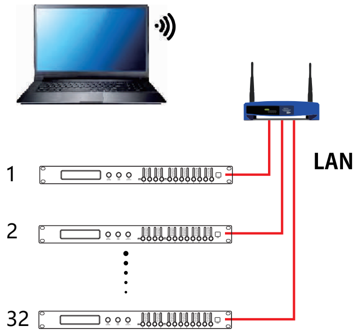
When using the network interface to connect, use a Category 5/6 network cable to connect the Ethernet port of the processor to the network adapter interface of the PC, adjust the IPV4 address option of the corresponding network adapter in the Windows network settings to automatically obtain DHCP, and then open the software Then the current processor can be found automatically. As shown：

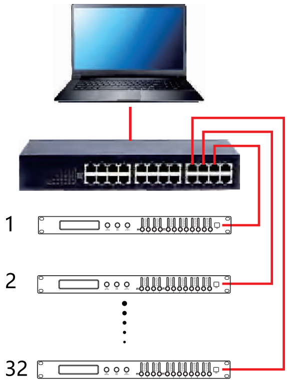


When using only one processor, you can use DHCP to connect to the processor. The specific methods are as follows:

The PC does not need to change the settings by default, directly connect the Ethernet port of the PC and the processor with a network cable, and open the software in the CD to automatically connect.

When using multiple processors, you can use a fixed IP to reduce the error rate. You need to set the pc to the same network segment as the processor, but different IP addresses.





7.3 Use RS485 to connect

The use of RS485 often requires a patch cord

① Traditional PC will provide DB9 serial port, which needs to use DB9 to RJ45 conversion cable to adapt.

② Generally, current household models do not provide serial ports, so USB to serial converters are needed, and MARANI provides USB to serial ports。

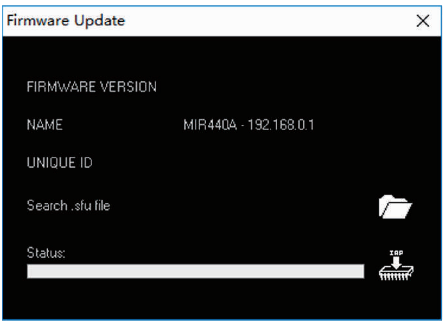
RJ45 type converter (USB-485-RJ).

The interface conversion diagram of USB to XLR converter is as follows.

7.4 Firmware upgrade

When Marani releases a new version of the firmware, users can download the latest firmware from the official website.

You can use TCP/IP to upgrade the firmware to ensure the stability of the power supply during the upgrade process. In case of a power outage, you may need to return to the factory for repair.

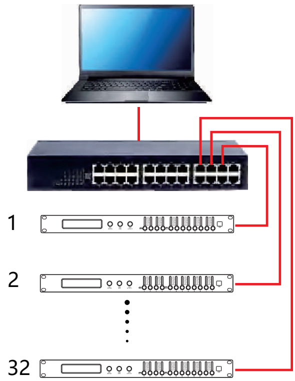


P8. Advanced features

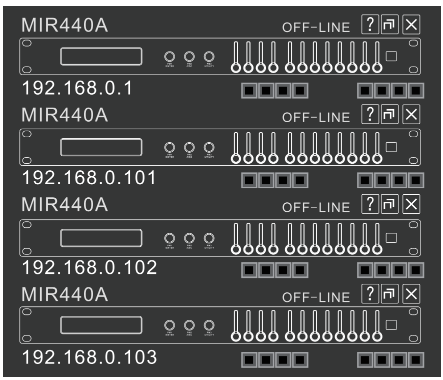
8.1 Group control Overall marshalling control

The simultaneous marshalling control of multiple processors simplifies a lot of work processes for on-site debugging, can quickly respond to emergencies, and reduces the amount of repetitive work.

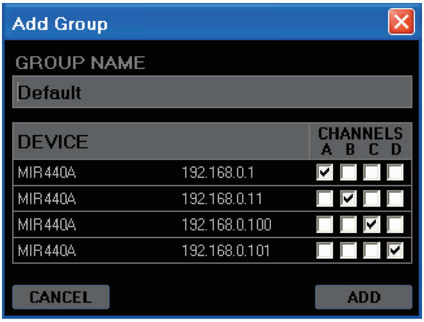
7.3.1 To use the marshalling function, it is recommended to use a switch to connect multiple processors with a network cable, and then modify the IP address of each to a fixed IP and use a different IP address.



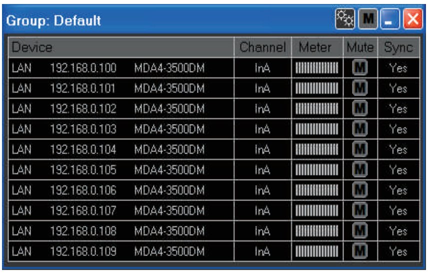
7.3.2 Connect the processors that need to be grouped online in the software, and choose to add grouping.

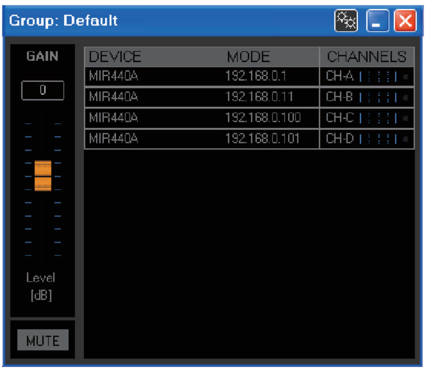


Tick the input channels that need grouping control, you can freely group the input channels, you can create a new group to control the remaining ungrouped channels, click Add after the selection is complete.



At this time, there is a small window for group control, you can control the overall gain, mute, monitor level, etc.





1--Software

1 Home page overview

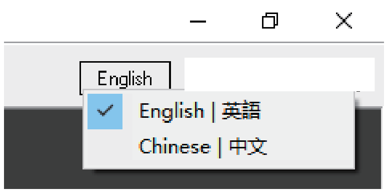
Enter the main page of the software, the upper left corner respectively shows: New/Import/Save project.

Add device: For the case of a known machine with a fixed IP address/USB/RS485, use to add the corresponding device for online operation.

Find device: For the situation where the IP address of the device is unknown and the port number of 485 is unknown, you can select the search function to quickly find the corresponding device.

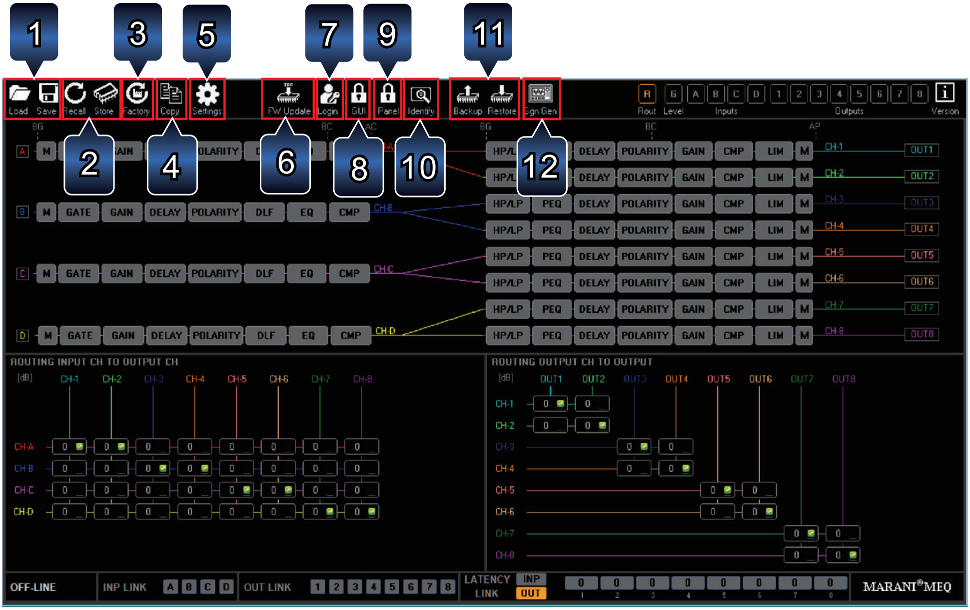
Grouping: You can group input channels of the same model using any connection medium, including but not limited to gain, polarity, delay, and PEQ.

In the upper right corner of the software, Chinese and English bilingual switching is provided, click the red box to switch.



2 Software main interface

On the main interface of the processor, the overall status of the audio paths routing and the order of the signal processing modules are displayed. Note: The signal processing modules here cannot be directly clicked to access, need to be modified separately in the corresponding input or output channels.



1 

Load& Save: You can save a single preset as a PC file or load a preset from a PC file.

2 

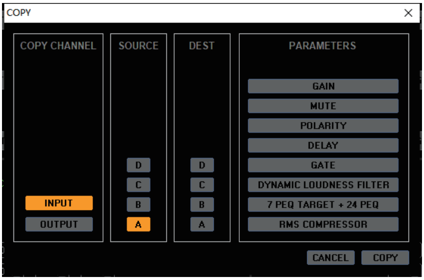
Recall& Store: Read the preset from the device , Store the current preset to the device.

3 

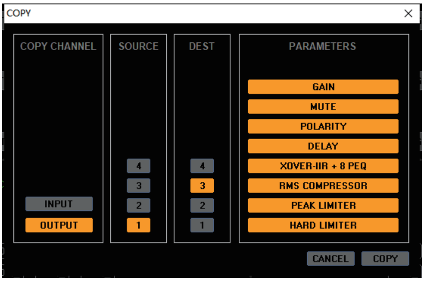
Use the ***Factory (reset)*** button to clear the current preset to the factory default state without affecting other presets stored in the machine.

4 Copy

Channel Functions Copy: select (highlight) the channel to be copied, and select the function to be copied at the same time, then the required function can be copied to the corresponding channel.



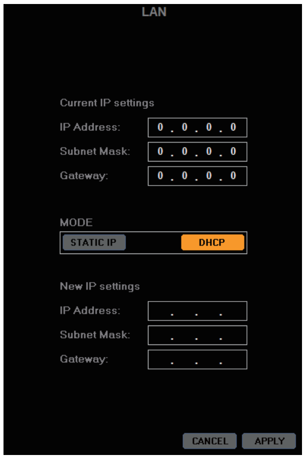
For example, copy all the contents of output 1 channel to output 3, just follow the diagram.



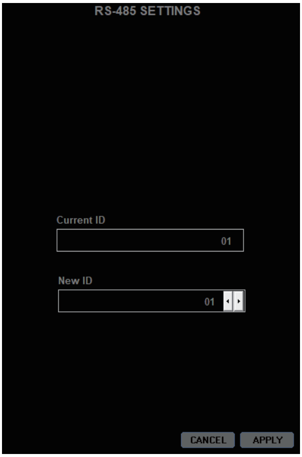
5 Settings

Settings: LAN, RS485, channel label, group setting, recall mode, user authority, administrator password, output anti-overflow and other settings.

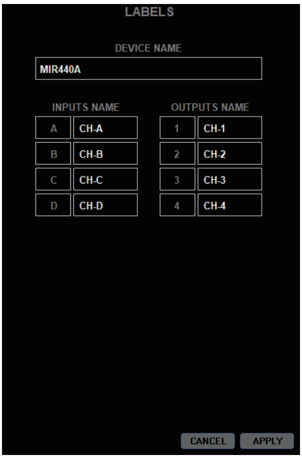
LAN settings: MIR series IP address defaults to DHCP, if you need to adjust it to a fixed IP, you need to modify it in



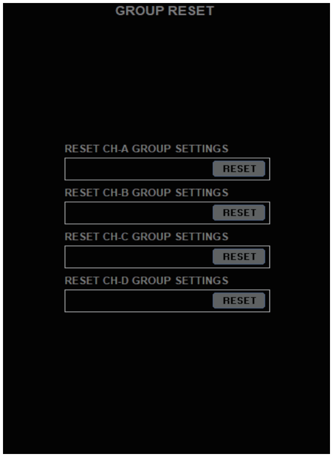
Rs485: Set the COM ID of RS485, the default is 01, the maximum is 32.



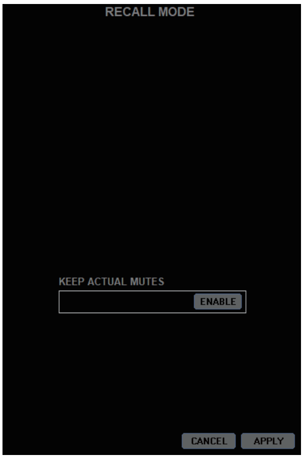
Channel label: you can set different labels for the input and output channels (involving the front panel display, only English characters and Arabic numerals are supported).



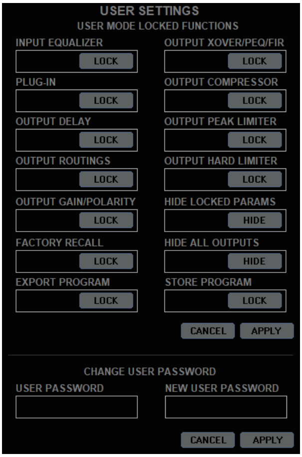
Group Reset: after using the group, the last saved group information is still in the input channel. If you need to remove it, you need to reset the required channels in the group reset.



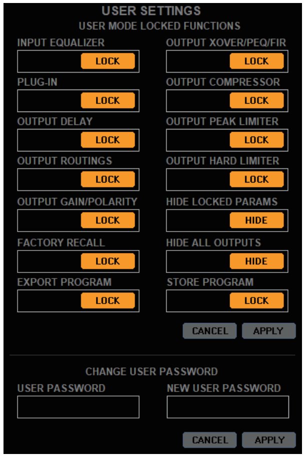
Recall Mode: When recall presets from the device, if you need to keep the mute state by current setting, you can turn on this function.



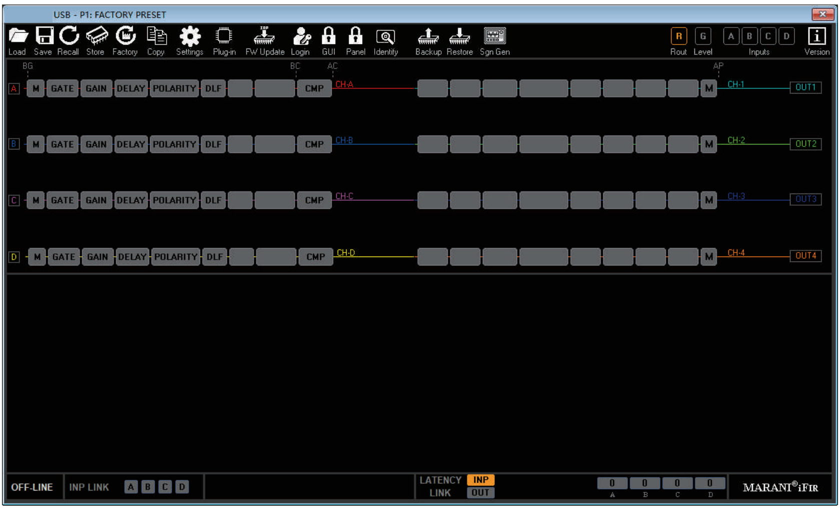
User Authority Setting: When logging in as an administrator, part or all of the output channel functions of the machine can be locked to ensure the integrity of the preset. The administrator password is 111111 by default. Click Lock and log out of the administrator account. The locked items will be grayed out and cannot be modified in the software or the front panel. When the parameters are locked and hidden, the hidden area will not be visible in the software or the front panel.



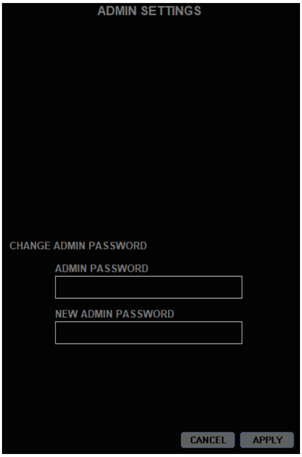
For example: now the administrator has logged in, select all locked items and hide them.



After logging out of the administrator account, the output part/routing part becomes empty.



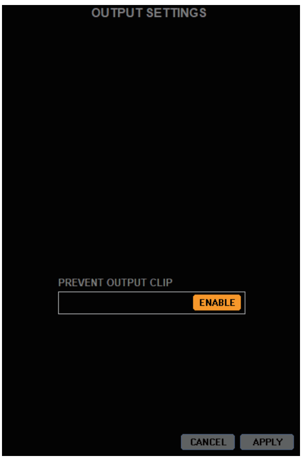
Administrator Settings: modify the administrator password, the default



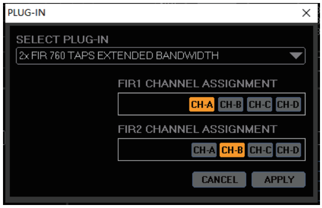
output setting:

The output setting includes a hidden Anti-Clip limiter: Max input Level is +20dBu and Max output level is +18dBu, this means that when input is going over +18dBu, if no Limit process on output, the output will be clipped up to +2dB in excess.

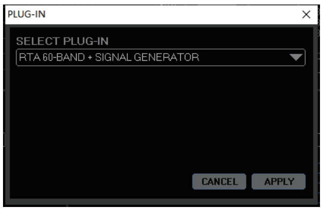
The Anti-Clip Limiter is Limiting up to +2dBu the output when input is exceeding



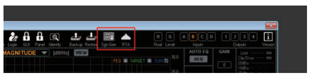
The same applies to dual-channel, that is, two channels can be applied with plug-ins, and the corresponding FIR with a sampling rate of 96KHz needs to be used, and the FIR generated by a third party can be imported here (Note: the FIR filter of the input and output channels runs at 48kHz sampling rate, the audio stream is processed by FIR through down-sampling, and then up-sampled back to the DSP processing chain. Similarly, the FIR convolution generated by AEQ is also running at a sampling rate of 48kHz, so here 96kFIR can only be imported from the file).



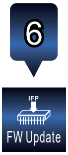
When selecting RTA and advanced signal generator.



The signal generator and RTA icons will appear.



This is the main interface when the advanced generator and RTA are not selected.

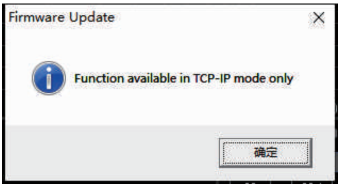


The MIR series provide firmware upgrade function, users can experience the latest version without leaving home, including new plug-ins, bug fixes, new functions, etc. At present, this function needs to be connected via a network cable, that is, the firmware cannot be updated via

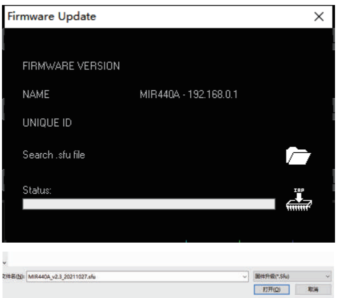
To get the latest firmware, you can visit Marani official website:

Marani Pro Audio (marani-

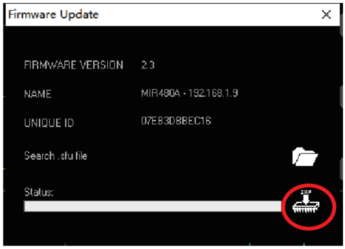
When using the USB|RS485 connection, clicking on the firmware upgrade will pop up such a prompt.

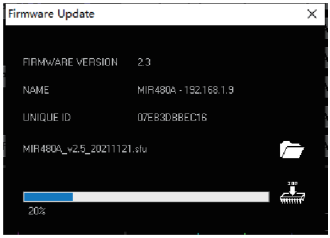


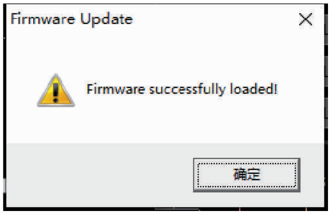
When obtaining the latest firmware IAP firmware package, click Firmware Upgrade. After the serial number and firmware version of the processor are correctly identified, select the correct firmware file "****.sfu".



Then Click.







When the progress bar is completed, the machine will automatically restart and the upgrade is complete.

Tips: The upgrade process takes about 1 minute. If the progress bar freezes or fails to enter the system after the upgrade, please retry the above steps to ensure that the machine's power supply and pc software are running normally during the upgrade process. If the upgrade fails, you need to contact your MARANI sales staff.

7 Login

Login: Administrator login, the administrator has the highest management authority of this

8 GUI

To lock the **software interface**, you need to enter a user password, the default user password is

9 Panel

The **device panel** lock, need to enter the user password, the front panel is unavailable after being locked, and it can be restored after restarting the processor. When the MIR processor is online, the front panel and software can be adjusted at the same time instead of locking the panel. When you have to leave your position in front of the Pc sw

10 Identify

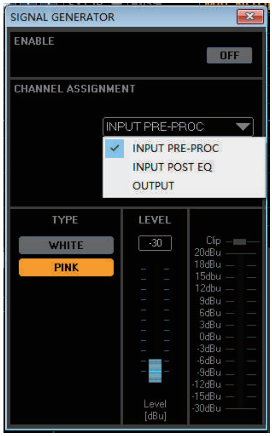
After pressing the identification button, the level lights on the front panel of the corresponding device will all light up and flash for 10 seconds, which is used to quickly find the currently edited one among multiple devices. And every time the pc software is operated, the three white

11 Backup Restore

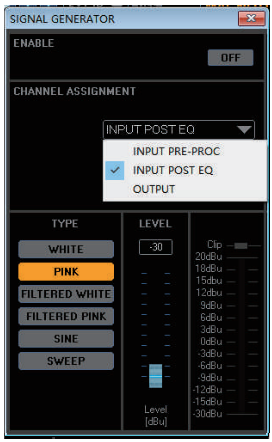
For the copy of all presets and functions of the whole machine, all 32 presets and network settings can be quickly imported into the new machine, which is very convenient for the migration or backup of the whole machine for engineering companies and OEM customers.

12 Sgn Gen

Advanced generator Provides a choice of insert position, which can be selected before the input channel processing (after A/D), after the input channel processing, or output (before D/A), when the input channel is selected after processing, or output (before D/ A) , when select INPUT PRE-PROC ,there are two noises: pink noise/white noise.

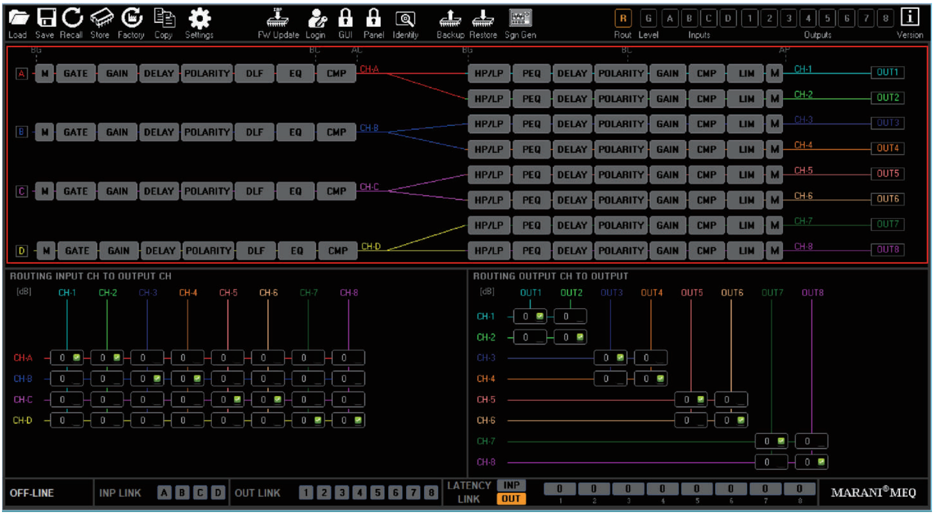


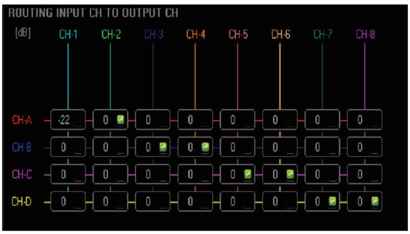
When selecting input post EQ or output, different filter types can be selected, including filtered white noise, filtered pink noise, sine wave, sweep sine wave, etc., which can only be sent to 1 channel of the input/output, The



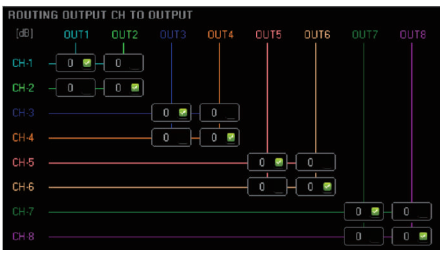
3 Signal routing part

Signal flow chart: Indicate the processing path after analog signal input A/D conversion, including routing status, plug-in status, etc. It is only for display and cannot be clicked to enter the modification. If you need to make changes, you can make detailed modifications in the input and output interface.



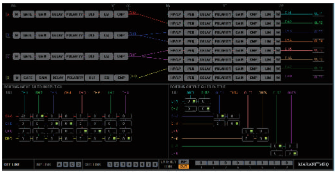


In the input matrix, you can arbitrarily route the signal from the input processing channel to the output processing channel. The default sending volume is 0dB, which is equivalent to group output. When choosing different sending volume, it is equivalent to real Mixing Matrix sending, and the sending volume selection range is -30~0 dB. You can



Similarly, the output matrix is responsible for routing the output channel to the physical output channel. The clever use of input and output routing can achieve powerful functions that cannot be achieved by conventional processors, such as Multi Band compression.

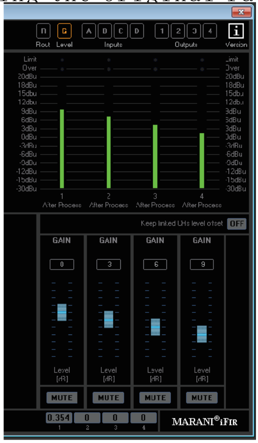
In example, use the processor A channel input, the output channel 1 is connected to a 2 way passive speaker, and the route is as shown in the above figure. The output channel uses a MIR linear filter to perform a virtual X-over on the speaker (it is better to know the crossover Point), so that different compressor activation thresholds can be set for the high and low drivers, and at the same time, this setting will not destroy the original amplitude and phase of the speaker, can add some color or an extra protection as required.

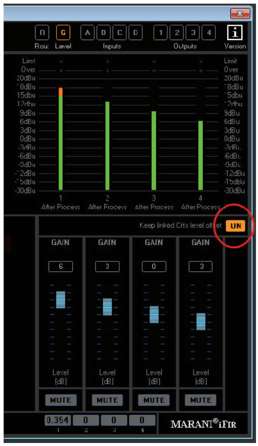


The channel level page displays the input post-A/D and output pre-D/A signal levels of all channels, corresponding to the physical level meter on the front panel, where you can intuitively adjust the channel level and mute.

A new function called ″ Keep Linked CHs Level Offset ″ has been added. The conventional linkage channel will link the level faders together. For some active speakers, the output level ratio should remain the same. When this happens, select ″Keep Linked CHs Level Offset ″

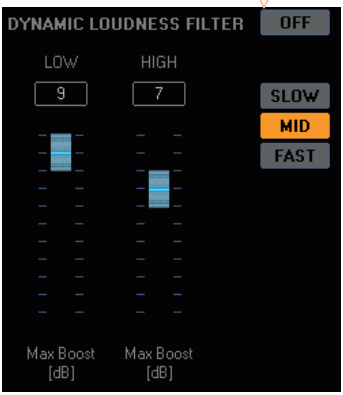
.The specific method is to assume that the output channels need to be linked. When the output channel linkage is turned on, turn on the ″ Keep Linked CHs Level Offset ″ at the red circle, so that the output channel can be linked while maintaining the original ratio.





4 Input part

Input channel processing includes noise gate, DLF, equalizer, compressor, delay,



Process 1 Dynamic Loudness filter.

Working principle: According to the human ear equal loudness curve, when the sound pressure is low, the ultra-low and ultra-high frequencies are increased; as the sound pressure level increases, the increase ratio approaches 0, so that a more average sound pressure level can be obtained Sense of hearing.

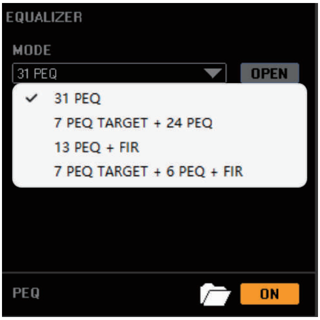
How to set: First set the maximum allowable boost level, the default is 9dB for low frequency, 7dB for high frequency, and the maximum boost is 10dB.

At the same time, there are three start-up speeds to choose from to adapt to different styles of music.

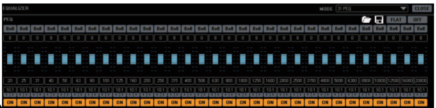
(Don't worry too much about the maximum boost, because the maximum boost will only

Process 2 Variable Equalizer Configuration

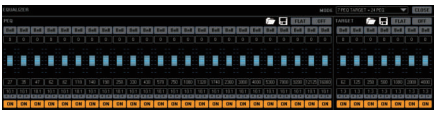
Contains four categories.



1 **31 bands of PEQ,** PEQ with enough bands, the default Q value is 10.5, which is equivalent to 31 bands of graphic equalization (GEQ), in fact, the frequency Q value of each band of equalization and even the filter type can be changed.

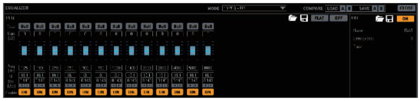


2 **7-bands target EQ and 24-bands PEQ.** The design purpose of target EQ is that users can create a small preset that is different from the overall presentation, which can be used for room acoustic correction and stored and recalled independently, which is very convenient.

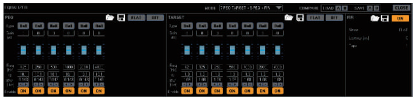


3 **13-bands PEQ+512-taps FIR filter,** when you need to use FIR filter on the input channel, you can select this option to provide users with FIR correction at the input, which is suitable for AEQ room

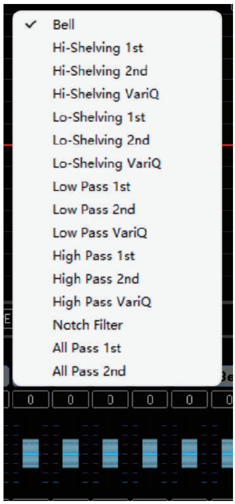
Note: This FIR filter is 48kHz sampling rate 512Taps. Since the overall sampling rate is 96kHz, the sampling rate will down to 48kHz before FIR, and after processing, the sampling rate will be re-extended back to 96kHz.



4 **6-bands PEQ+7-bands target EQ+512Taps FIR filter,** the most complete type, can do almost all the EQ cofigurations you need.

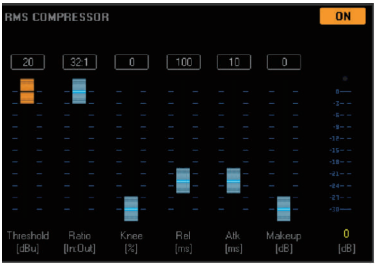


There are as many as 16 types of filters. In addition to the most conventional Bell filter, it also includes high-shev/low-shev/high-low-pass/notch/all-pass, etc., which can be selected as well.

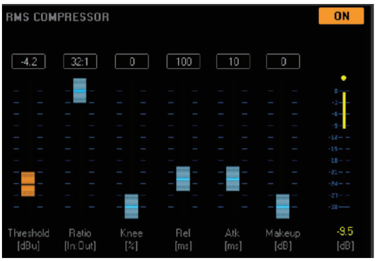


Process 3 Compressor

Conventional compressors will provide: threshold, attack time, ratio, release time, gain compensation, soft and hard knee.



The compressor is a very important part of audio processing. It can change the timbre of our output audio or make the sound stable.

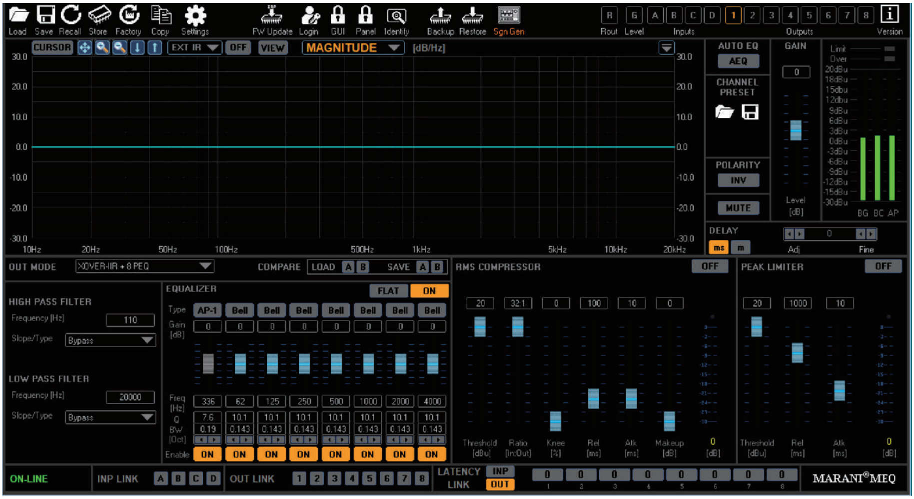


The delay alignment option is provided in the lower right corner of the software, which can be used to align the delays generated by FIR filters with different latency applied to the different channels.

On

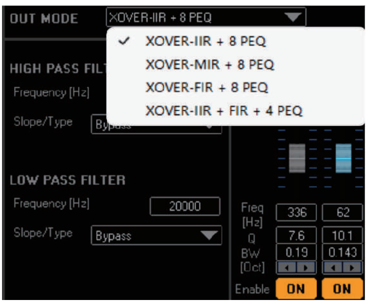
5 Output section

The overall overview of the output section, including crossover, polarity, delay, gain, equalizer, RMS compressor, peak limiter, hard limiter, etc.

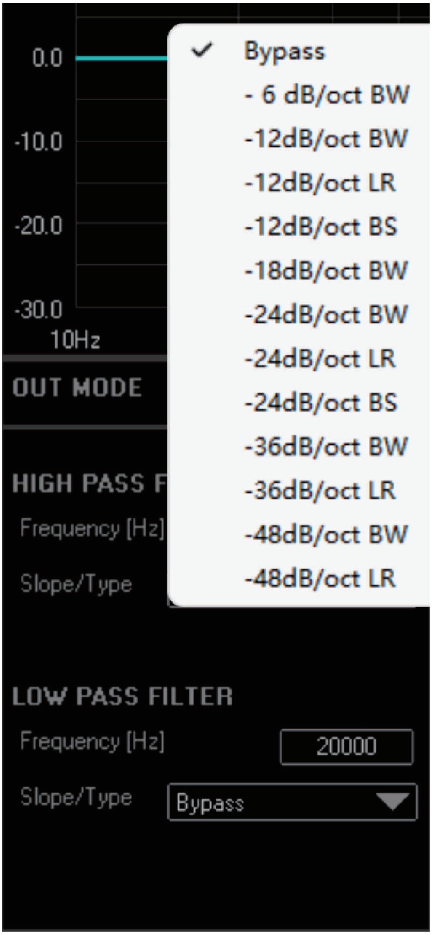


Process 1. MIR linear phase filter/FIR filter/FIR+IIR

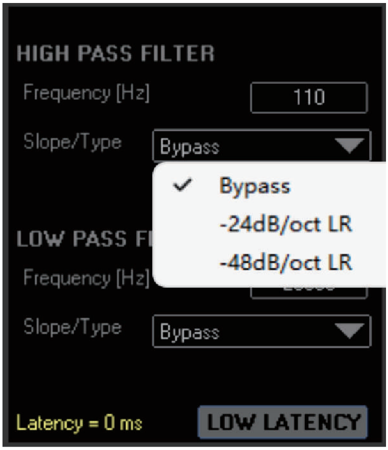
The conventional IIR filter maintains the slope of the analog filter, but will produce a phase deviation. The higher the order, the more the phase shift, which in turn causes magnitude problems.



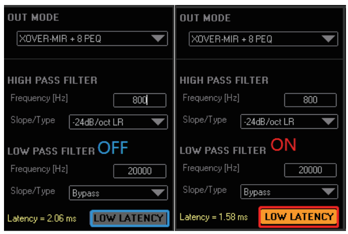
In the high and low pass filters of IIR, we provide three types of traditional Butterworth/Linquez-Rayleigh/Bessel slopes ranging from -6dB/oct to -48dB/oct.



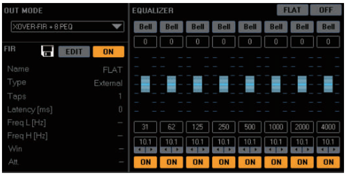
The MIR linear phase filters replicate exactly the slope of the analog filter (LR24/LR48), thereby keeping the phase Linear and easily coupling the phase at the crossover point.



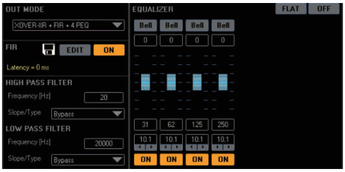
After the low latency mode is turned on, the delay caused by the latency coming with MIR linear phase filters can be further reduced, but the disadvantage is that the in band ripple close to the knee will increase from the default +-0/01% to ±0.5dB will be generated at the crossover point. However, this is not enough to affect the hearing.



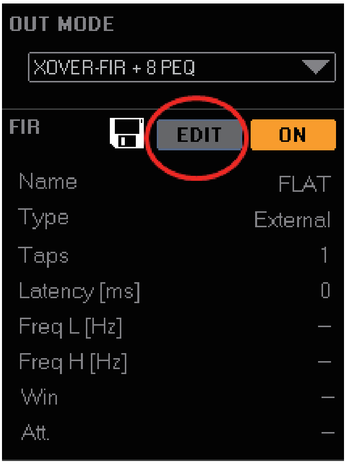
For the FIR filter, two modes can be selected, FIR+8-segment PEQ, FIR+ IIR high and low-pass with a slope of up to -24dB + 4-segment PEQ.



You can use a simple FIR filter + 8-segment PEQ, or you can choose a FIR+IIR mixed mode + 4-segment PEQ. There are always a variety of solutions that are suitable for your use scenario.

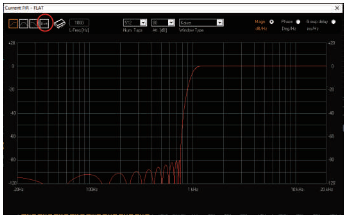


Click Edit to enter the FIR editing options,



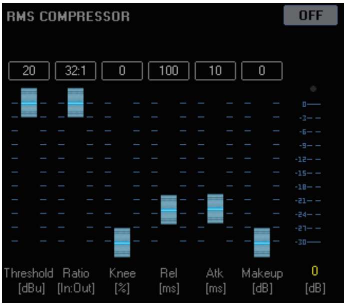
Provides three guides of high-pass/low-pass/band-pass, you can enter the frequency that needs to be band-passed according to your needs, choose more or less taps, slopes from -20dB to -120dB/oct, and multiple window function types , Click save at the red circle (here we use band pass filter as an example).

If you need to use external FIR coefficient import, you can select "EXT" in the red circle here, click enter to import FIR coefficient file.

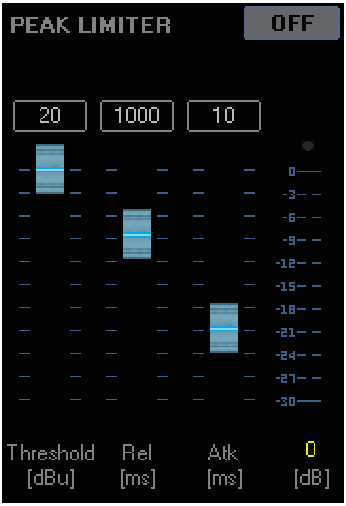


External FIR supports three formats of FIR coefficients, namely ".csv, .txt, .saf". The first two file formats can be generated by mainstream FIR convolution software, and the .saf file can be saved and generated from AEQ in the Marani processor. It should be noted that the FIR filter in the input and output part of the MIR-E series works at a sampling rate of 48kHz and a maximum of 512 taps. The FIR filter in the plug-in works at a sampling rate of 96kHz and a maximum of 1536 taps.
Compatible FIR convolution software has been tested: rePhase, FIR Designer, FIR Capture, Filter hose.

The output part provides three levels of compression limiter settings, namely RMS compressor, which can be used to set the compression option of the average level over a period of time. When the compression ratio is high, it is the limiter.



The peak limiter has a very low attack time and quickly suppresses the burst signal. It can be used to limit the maximum displacement of the speaker voice coil in conjunction with the X-max-voltage data given by the speaker manufacturer.



P8. Specificlations and Data

DSPprocessing parameter

Signal generator---------------	white noise/pink noise, level range: -30dBu~+10dBu
nput & output gain-------------	-18 dB ~ +12 dB, step accuracy is 0.1dB
Noise gate-----------------------	Threshold range: -80dBu~-45dBu Start-up time: 1ms~1000ms; Release time: 1ms~1000ms
Dynamic loudness filter ------	Gain range: 0dB-10dB Start-up speed: fast/medium/slow
Parametric equalizer----------	Input channels up to 31 optional types of PEQ, output channels up to 8 optional types of PEQ
Optional types include --------	Bell filter, 1st order high Shelf filter, 2nd order high Shelf filter Variable Q high Shelf filter, 1st order low Shelf filter, 2nd order low Shelf filter Variable Q low Shelf filter, 1st-order low-pass filter, 2nd-order low-pass filter Variable Q low pass filter, 1st order high pass filter, 2nd order high pass filter Variable Q high pass filter, notch filter ,1st order all-pass filter, 2nd order all-pass filter with variable Q value
The center frequency---------	adjustable within the frequency range of 20Hz~20kHz with a step accuracy of 1Hz
Q value/bandwidth-------------	The Q value range of Bell filter is 0.4~128, the step is 0.01 The range of the Q value of the Chevron/high-pass/low-pass filter is: 0.1~5.1, and the step is 0.01 The value range of bandpass/notch filter Q is: 4~104, step is 1
Equalizer gain range----------	-15dB ~ +15dB
IIR crossover filter-------------	Butterworth slope: 6/12/18/24/36/48dB per octave Bessel slope: 12/24dB per octave Linkwitz-Riley slope: 12/24/36/48dB per octave NXF horn filter slope is 40/45/50/50/55/60/65/70/75dB per octave
MIR linear phase filter--------	Linkwitz-Riley: 24/48dB per octave,NXF-40
FIR crossover filter------------	type; high pass/low pass/band pass/external import Taps range: 256 ~ 512, slope range 21 ~ 120dB per octave Time window type: Rect / Sinc / Keiser / Hanning / Hamming / Blackman /Blackman-Harris/ Blackman-Nuttal / Nuttal/ Keiser -Bessel/Sine
RMS compressor--------------	Starting threshold range: -10dBu~ +20dBu; Compression ratio range: 2~32: 1; Soft and hard inflection point: 0~100% start time: 0.1ms~1000ms; Release time: 10ms~15000ms Gain compensation: Maximum 12dB
Peak limiter ---------------------	Threshold range: -10dBu~ +20dBu Start-up time: 1ms~1000ms; Release time: 10ms~3000ms
Delay----------------------------	The adjustable delay time of each input channel + output channel is 452ms, Step accuracy 0.0104ms (10.4us)

Input impedance---------------------	20KΩ
Output impedance-----------------	100 Ω
A/D dynamic range-----------------	118dB
D/A dynamic range----------------	118dB
Maximum input level---------------	+20dBu
Maximum output level-------------	+18dBu
Total harmonic distortion-----------	≦ 0.003% (+4dBu 1KHz)
Frequency response---------------	20Hz ~ 20kHz ±0.3dB
Crosstalk----------------------------	≦-95dB
Signal-to-noise ratio----------------	>111dB(A weighting)
Noise floor --------------------------	≦-91dB (A weighting)
Common Mode Rejection Ratio------	60dB
Number of analog input channels----	2/4
Number of analog output channels--	4/6/8
Rs485 control port-----------------	2
Network control port---------------	1
size--	482x44x207mm 1RU
Net/Gross weight---	3.0 Kg / 3.5 Kg