MIR-A

Speaker Management
Systems

MIR-A Series User Manual

software-v0.3.3

MARANI®

REVISION-A1 DEC-2021

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 specified by the manufacturer.
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, n'exposez pas cet article a la pluie ou a l'humidite

3 INTRODUCTION

The MIR-A series processor is a digital audio processor newly designed by Marani. It has extremely low noise floor, high dynamic range and powerful internal DSP to cover most of the usage scenarios. Whether in meeting rooms, small theaters, touring performances, or even large-scale tours, you can use the A series processors. Advanced circuit design and original DSP algorithm are the core of pure and high-quality sound.There are 4 different models of MIR-A series, namely MIR260A/360A/440A/
480A, which means that they have 4 different channel configurations of 2 in6 out, 3 in 6 out, 4 in 4 out, and 4 in 8 out. . In the signal processing part, they all use powerful MARANI DSP, DSP and AD/DA are running at 96KHz sampling rate, complete processing functions provide a complete speaker X-over solution.
From input gain/delay/noise gate/EQ/compression/FIR， to output gain/delay/polarity/X-over/FIR/EQ/RMS comp/Peak limiter, there are up to 13 types of parametric equalizer (PEQ) can be selected. The output crossover filter includes the classic Linkwitz-Riley/Bessel/Butterworth, even NXF (Norched X-over Filter), and the FIR filter with a slope of up to 120dB per octave. Optionally, the newly added MIR linear phase filter can make the phase of the crossover point easier to join and produce a lower delay. Everything we provide is for better sound.
Newly added DSP plug-in, providing high-precision 96kHz sampling rate FIR filter/high-order signal generator/RTA real-time spectrum analyzer and other practical plug-ins.
The newly designed Hard Limiter allows a constant rate limit on signals exceeding the threshold at any threshold to better protect the speaker unit.

Each input and output provides a maximum 512-taps FIR filter, which can be customized by third-party software to generate the FIR convolution you need. Used for speaker presets, it can improve the phase response and control the directivity according to requirements.

The newly added MIR linear phase filter has the shape of the traditional IIR filter (Linkwitz-Riley 24/oct), but does not produce any phase distortion, and the resulting delay is about 50% of the FIR filter.

4-INSTALL

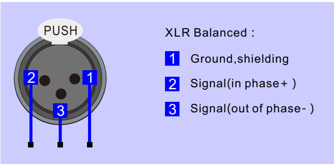
4.1-AC power input

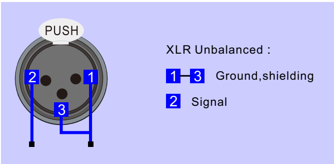
The AC voltage provided by the audio processor with built-in global voltage operation must be within ±10% of the specified line voltage (100~240V). The third pin (grounding pin) on the power cord that comes with the package is A necessary safety component, please do not try to disable the ground connection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

4.2-Signal input and output

4.2.1-Analog signal input

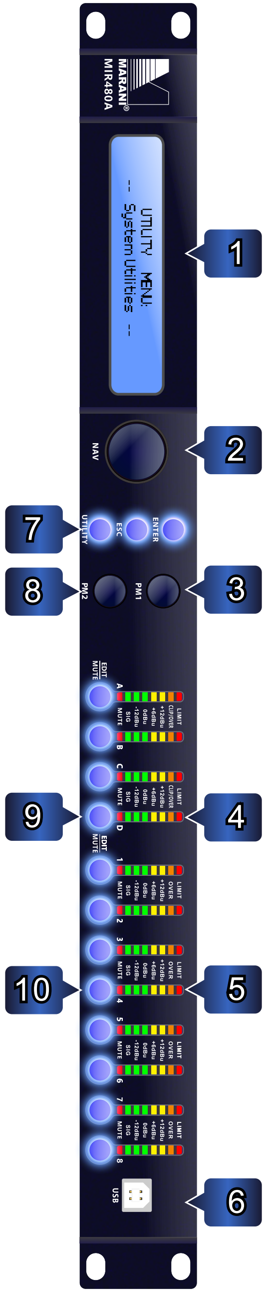
Like all digital signal processing equipment, the signal level supplied to the unit must be appropriate to avoid working in a low signal-to-noise ratio or distortion. The MIR-A series processors can accept up to +20dBu analog signal level, which is greater than the linear output level of most mixers, 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*20 LCD display.
2. Navigation knob, responsible for the main function switching, menu up and down.
3. The PM1 knob is responsible for turning on/off some functions and coarse adjustment of some values.
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. UTILITY, click to open the main menu. ESC, click to escape. ENTER, click to next step.
8. The PM2 knob is responsible for turning on/off some functions and fine adjustment of some values.
9. 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.
10. 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.



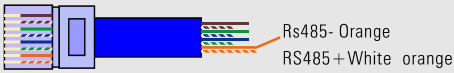
4.4.Overview of the processor back panel

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.



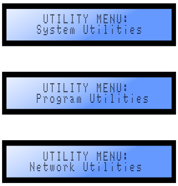
5. Analog signal output interface, maximum output level +18dBu, minimum load 100Ω.

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

5.FRONT PANEL OPERATION

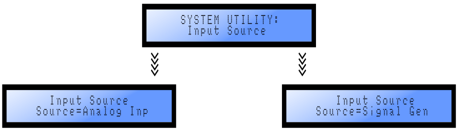
5.1 System Settings

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



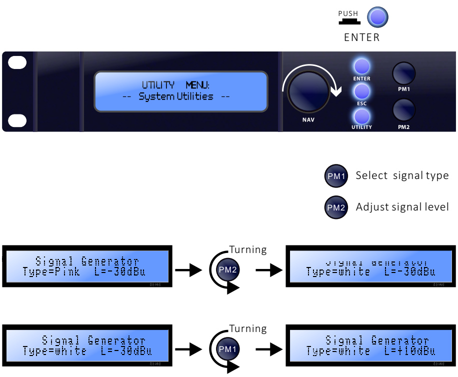
5.1.1 Input Source Selection

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



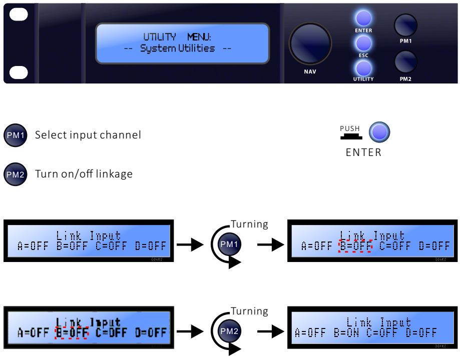
5.1.2 Signal Generator Settings

Use the PM1 knob to select pink noise or white noise, and use the PM2 knob to adjust the signal level from-30dBu to +10dBu, with a step of 1dB.



5.1.3 Link Input

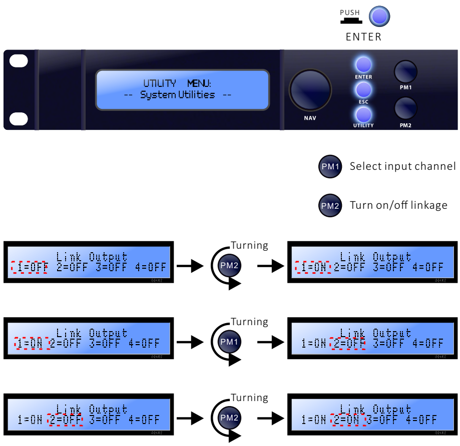
For the linked input channel, modify the settings of any one of the linked channels, and the other linked channels will synchronously modify the same value.



Turn the left and right knobs PM1 to select the ABCD channel, turn PM2 to the right to turn on the selected channel linkage, turn PM2 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.



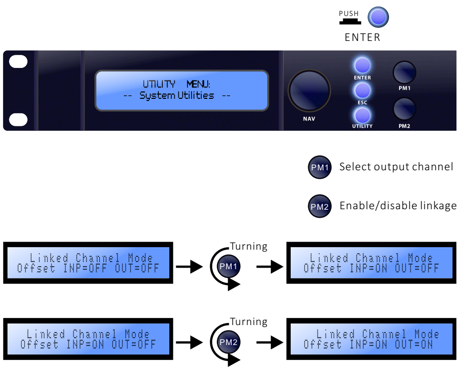
Knob PM1 selects the 1234 channel, turn PM2 to the right to turn on, and turn PM2 to the left to turn off the linkage.

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

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

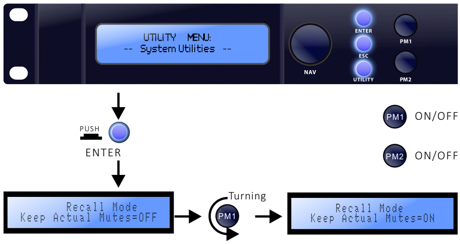


When the channel linkage is not turned on, this function is not available.

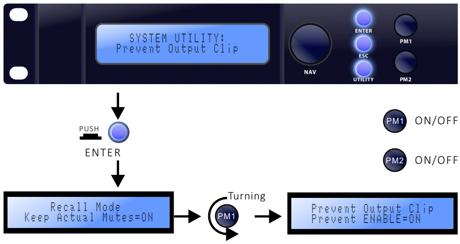
5.1.6 Recall mode (mute safety)

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:

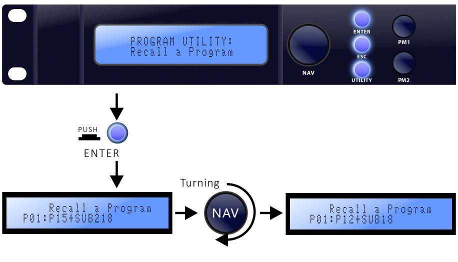
1. The firmware version can be upgraded online.
2. Back up all presets before the firmware upgrade.
3. During the upgrade, keep the power supply intact. If the upgrade fails midway, irreversible data loss may occur.

5.1.8 Program (preset) settings

Under UTILITY MENU, turn the RAV 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.



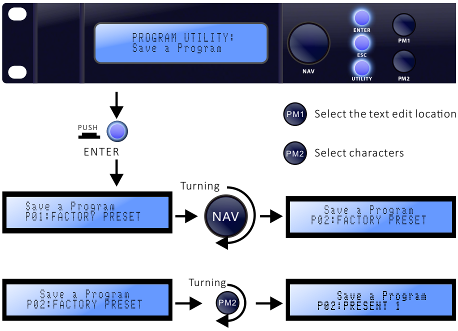
For example, read the present 3 from the machine

Press utility

- Turn the NAV knob to the right
- Go to Program Utilities and press Enter
- Press Enter to Recall a Program
- Turn the NAV knob to the right
- Choose P03
- Press Enter 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 utility button
- Use the NAV knob to select program
- Press Enter to enter the submenu
- Rotate NAV to the right to select save a program
- Press Enter to enter the next level
- Rotate PM1 to the right to select the preset position (such as P01)
- Press Enter 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 Enter to save

5.1.9 Network options

Under the settings, turn the RAV 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.

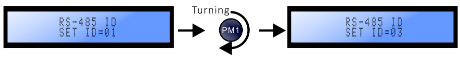
The RS485 serial port can be set in the network option menu, and the network settings of the machine can be viewed.

5.1.9.1 Set RS485 ID number

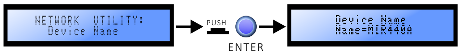
When multiple devices are connected via RS485 protocol, they need to be set to different ID numbers and cannot be repeated. After clicking Enter , rotate PM1 to select ID numbers from 1-32.

For example, the steps to set 485ID to 3 are:

- Click the Utility button
- Use the NAV navigation knob to select Network
- Click Enter to confirm
- Click Enter to the next level
- Rotate NAV to the right to select RS-485 ID
- Click Enter to confirm
- Rotate PM1 to the right to select 03
- Click Enter 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:

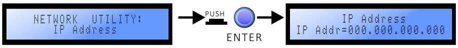
- Click the Utility button
- Use the NAV knob to select Network
- Click Enter to confirm
- Rotate NAV to Device Name
- Click Enter to confirm
- Rotate PM1 to the right to select the edit character position
- Rotate PM2 to the right to select characters
- Rotate PM1 to modify the character position, PM2 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

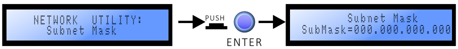
5.1.9.3 IP IP address

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

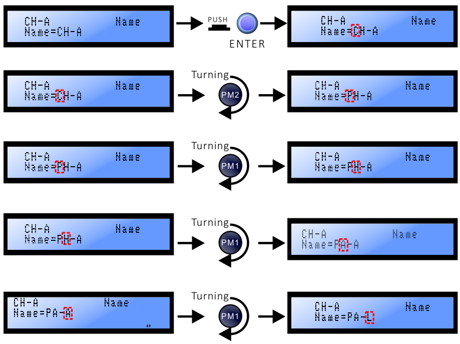
Edit input channel

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:

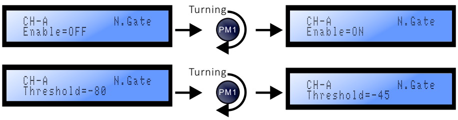
- ●Click the Edit button to edit the channel
- ●Click enter to confirm
- ●Rotate PM1 to the right to select the edit character position
- ●Rotate PM2 to the right to select characters
- ●Rotate PM1 to modify the character position, PM2 to select the character, the maximum allowable 6 characters

	!	"	#	$	%	&	'	(	)	*	+	,	-	.	/	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	{	|	}	→	←

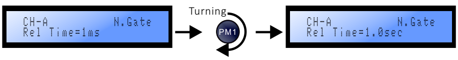
- ●Click Enter to save after modification

5.2.2 Noise gate

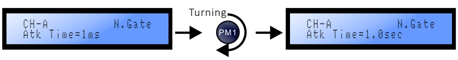
The noise gate can appropriately suppress the obvious background noise caused by the accumulation of front-end equipment or improper system settings. On the current page, click Enter to edit the overall state of the noise gate, and rotate PM1/PM2 to open or close the noise gate.



The threshold can be adjusted from -80 dBu to -45 dBu through Pm1.



The startup time can be adjusted from 1ms to 1000ms by rotating Pm1.



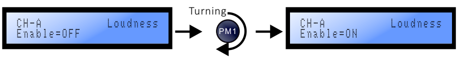
The release time can be adjusted from 1ms to 1000ms by rotating Pm1.

5.2.3 Dynamic loudness booster

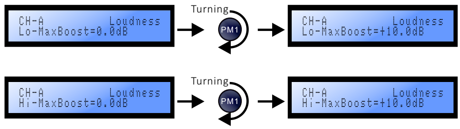
In the input channel, the working principle is based on the equal loudness curve of the human ear to compensate and enhance the ultra-low frequency and ultra-high frequency.

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.

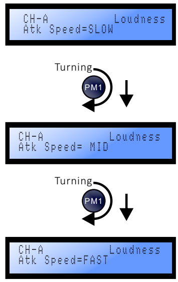
The maximum allowable boost gain value can be set separately for UHF and UHF, and the setting of starting speed is also provided.



When adjusting the maximum allowable gain, the adjustment step of the knob PM1 is 1dB, and the adjustment step of PM2 is 0.1dB.

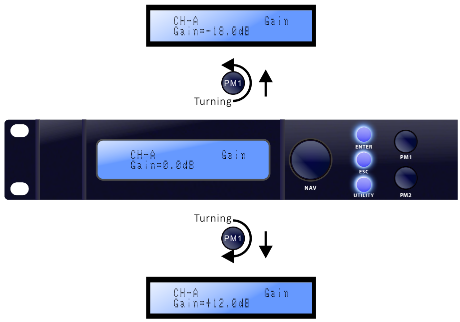


There are three levels of start-up time adjustable, namely fast, medium and slow.



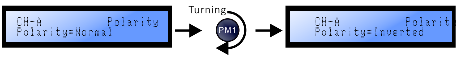
5.2.4 Gain

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



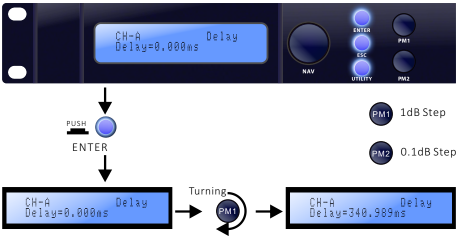
5.2.5Polarity

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.



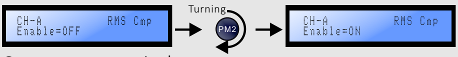
5.2.7 Compressor

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

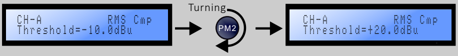
Using a lower start time (such as 5ms) and a higher release time (such as 1000ms) can firmly control the input signal within a certain range, of course, this will bring an obvious sense of compression (flatness).

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.

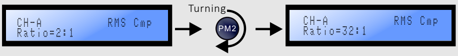
Under high compression ratios, it is easy to hear unnatural and abrupt sounds when using hard knees. On the contrary, using soft knees can make the compression transition smoothly and bring a natural sense of hearing.



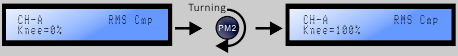
Compressor switch



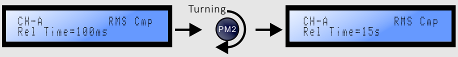
The start threshold can be adjusted from -10dBu to +20dBu.



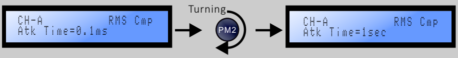
The compression ratio can be adjusted from 2:1 to 32:1.



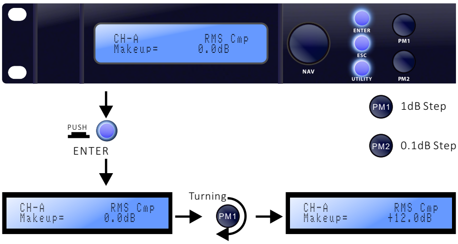
The soft and hard inflection point can be adjusted, among which 0% is the hard inflection point and 100% is the soft inflection point.



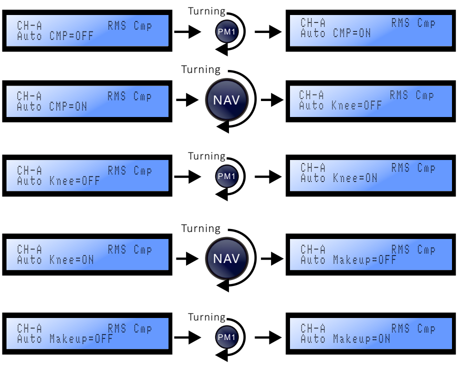
The adjustable range of the release time is: 100ms to 15000ms.



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



Provides automatic compressor settings, and automatically sets the compression start threshold and soft and hard inflection points according to the size of the music signal.



5.2.8 Equalizer Mode

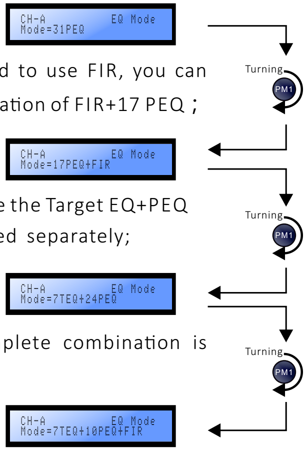
Provide 4 combinations,

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

When you need to use FIR, you can use the combination of FIR+17 PEQ；

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

The most complete combination is PEQ+TEQ+FIR.



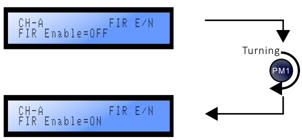
5.2.9 FIR filter

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.

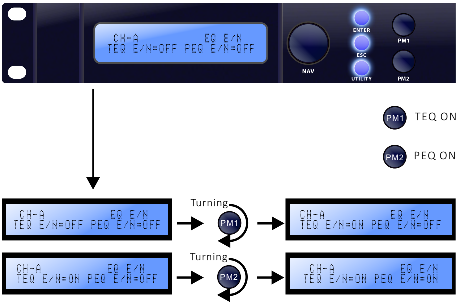
When PEQ mode is selected as with FIR filter, this option can be turned on.



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

5.2.10 Equalizer state

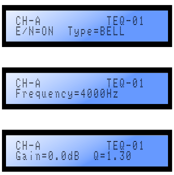
PEQ overall on/off, TEQ overall on/off.



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.

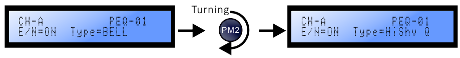


5.2.12 PEQ1-31

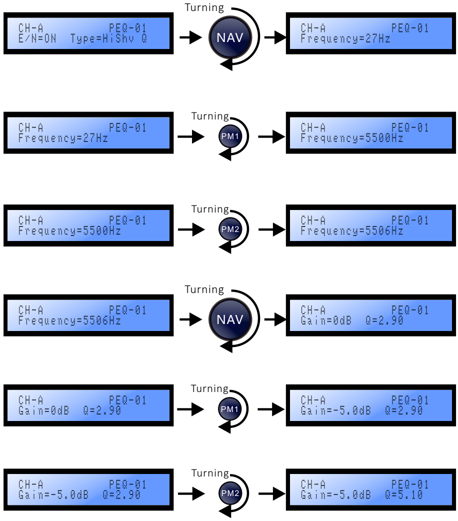
Specific turning of each band of parametric equalization, after Enter, you can rotate PM1 to select the filter type, you can select high and low Shelf /Bell/High and low pass/all pass/notch, etc. rotate the RAV button to enter the specific parameter adjustment, PM1= gain, PM2= Q value , Frequency and 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 NAV knob to select PEQ1
- Click Enter to enter the edit
- Rotate PM2 to the right to select HiShvQ
- Click Esc to return
- Continue to rotate NAV right to select frequency adjustment
- Turn PM1 to the right to adjust the frequency to 5500Hz, use PM2 to fine-tune to 5506Hz
- Click Esc to return
- Right-turn NAV to adjust gain and Q value
- Use PM1 to adjust gain and PM2 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.



5.3Edit 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

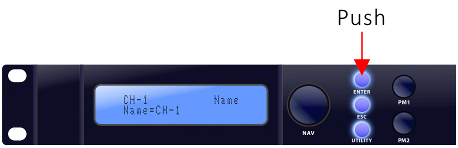
Click the Edit button and the first item displayed is the channel name edit

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

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

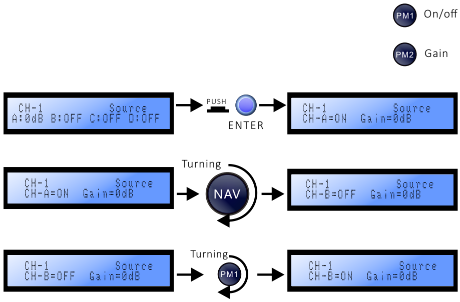
	!	"	#	$	%	&	'	(	)	*	+	,	-	.	/	0	1	2	3	4	5	6	7
8	9	:	;	<	=	>	?	@	A	B	C	D	E	F	G	H	I	J	K	L	M	N	O
P	Q	R	S	T	U	V	W	X	Y	Z	[	¥	]	^	_	`	a	b	c	d	e	f	g
h	i	j	k	l	m	n	o	p	q	r	s	t	u	v	w	x	y	z	{	|	}	→	←

- Click Enter 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 PM2 adjusts the gain amount, which can be adjusted from -30 to 0dB.

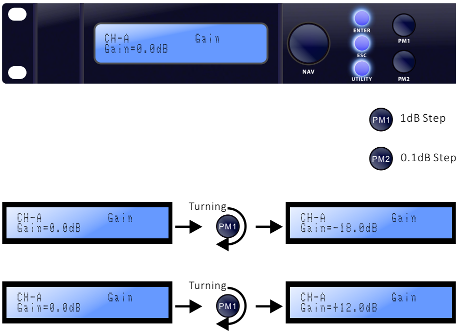


The default route of MIR440A is channel ABCD pass-through output 1234. In actual use, it may be necessary to assign one input signal to multiple output channels, such as routing input channel A to output 12 (since output channel 1 has sent 0dB from input A by default, so directly start from channel 2).

- Press the select key for output 2
- Select the NAV navigation button to go to the Source screen
- Press Enter
- To rotate NAV, select B
- Rotate PM2 to adjust the gain of input B to minus infinity
- To rotate NAV, select A
- Rotate PM2 to adjust the gain of input A to 0dB
- press Esc to complete

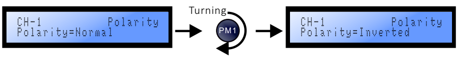
5.3.3Gain

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



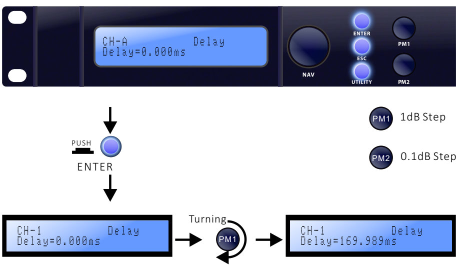
5.3.4 Polarity

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

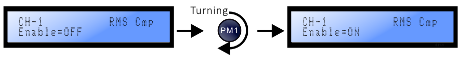
The output part provides a maximum delay of 170 milliseconds, which can be used to align the time between the multi-way units, in a step of 10.4 microseconds/0.01 milliseconds.



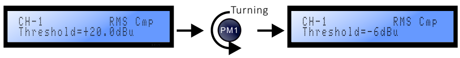
5.3.6 RMS compressor

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。

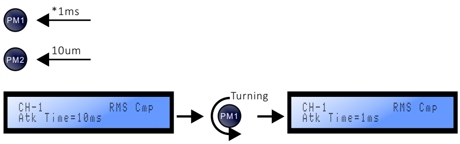
For example, the AES power (2h) of a HF driver is 100 watts, the impedance is 16 ohms, the crossover point is 1000 Hz, and the amplification factor of the power amplifier is 40 dB, then according to P=U2/R, the maximum input voltage of the unit is 40 volts, divided by The power amplifier magnification 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.

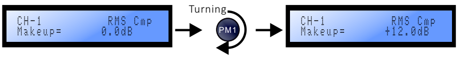


The start threshold can be adjusted from -10dBu to +20dBu.

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.

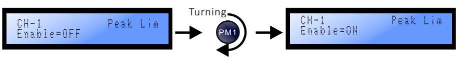


Compression compensation can be set range: 0~12dB.

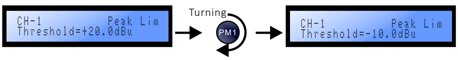
5.3.7 Peak limit

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.

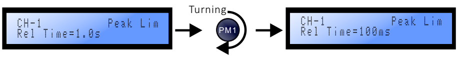
For example, using the ** model woofer, it is found that the stroke of the unit reaches the limit of 12mm at 103 volts. At this time, the amplifier magnification is 38dB, 103/80=1.2875v, 20log (1.2875/0.775)=4.4, that is, the peak compressor's The startup threshold is 4.4dBu.



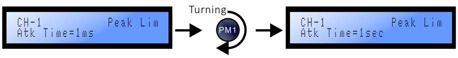
Peak limiter switch.



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



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

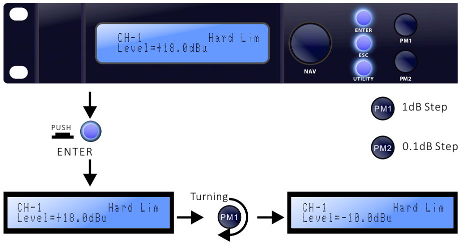


The attack time can be adjusted from 0.1ms to 1000ms.

5.3.8 Hard Limit

There is no setting of start time and release time in the hard limit. By directly setting the limit threshold, the internal signal flow is prevented from being distorted.

It can be used to match the gain structure of the front and rear systems. The simple ones are often the most intuitive and effective.



5.3.9 X-over mode

The MIR-A 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

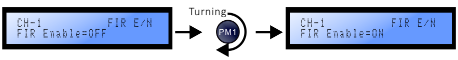
IIR+FIR mode (only 4 band of PEQ can be used in this mode)

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.

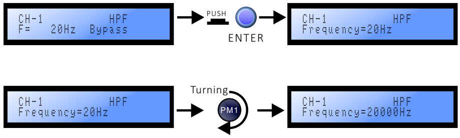


Otherwise, it shows unavailable N/A.

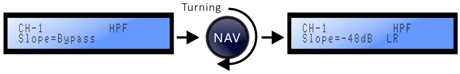
5.3.11 High pass filter

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

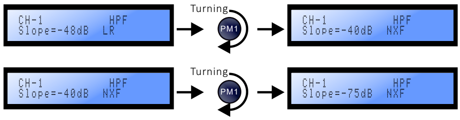
Click Enter to edit.



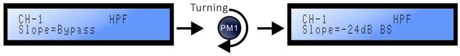
Traditional filter categories include: Bessel, Butterworth, Linkwitz-Riley, with a maximum slope of 48dB/otc.



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

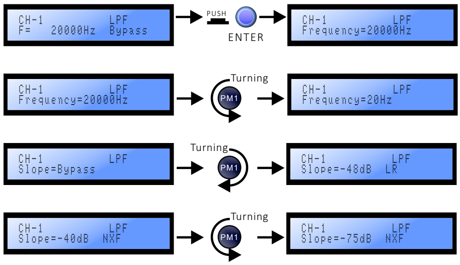


When the combination of IIR and FIR filter is selected, the highest slope of IIR filter is -24dB/Oct, and only 4 band PEQ are available.

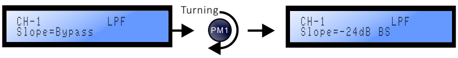


5.3.12 Low-pass filter

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



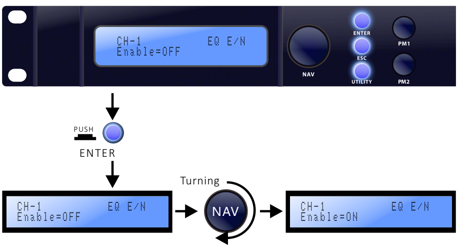
When the combination of IIR and FIR filter is selected, the highest slope of IIR filter is -24dB/Oct, and only 4 band PEQ are available.



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

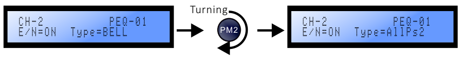
PEQ overall on/off.



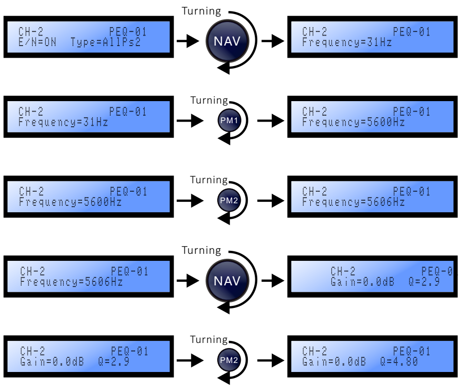
5.3.14 PEQ1-8

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 RAV navigation key to enter specific parameter adjustment, PM1= gain, PM2= Q value, frequency and so on.

- Press the Edit key of Output 2
- Rotate the NAV to select PEQ1
- Press Enter
- Rotate PM2 to the right to select AllPs2 filter
- Press ESC
- Continue to rotate the NAV selection frequency adjustment to the right
- Rotate PM1 to adjust the frequency to 5600Hz, use PM2 to fine tune to 5606Hz·
- Press ESC to return
- Rotate NAV 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.

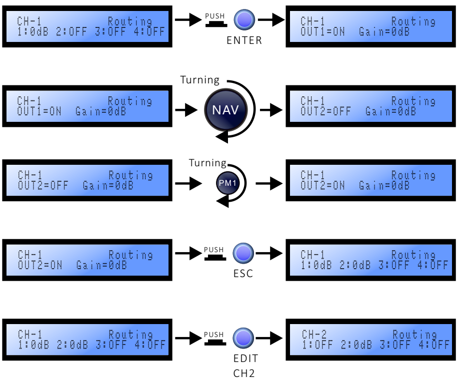


5.3.15 Output matrix

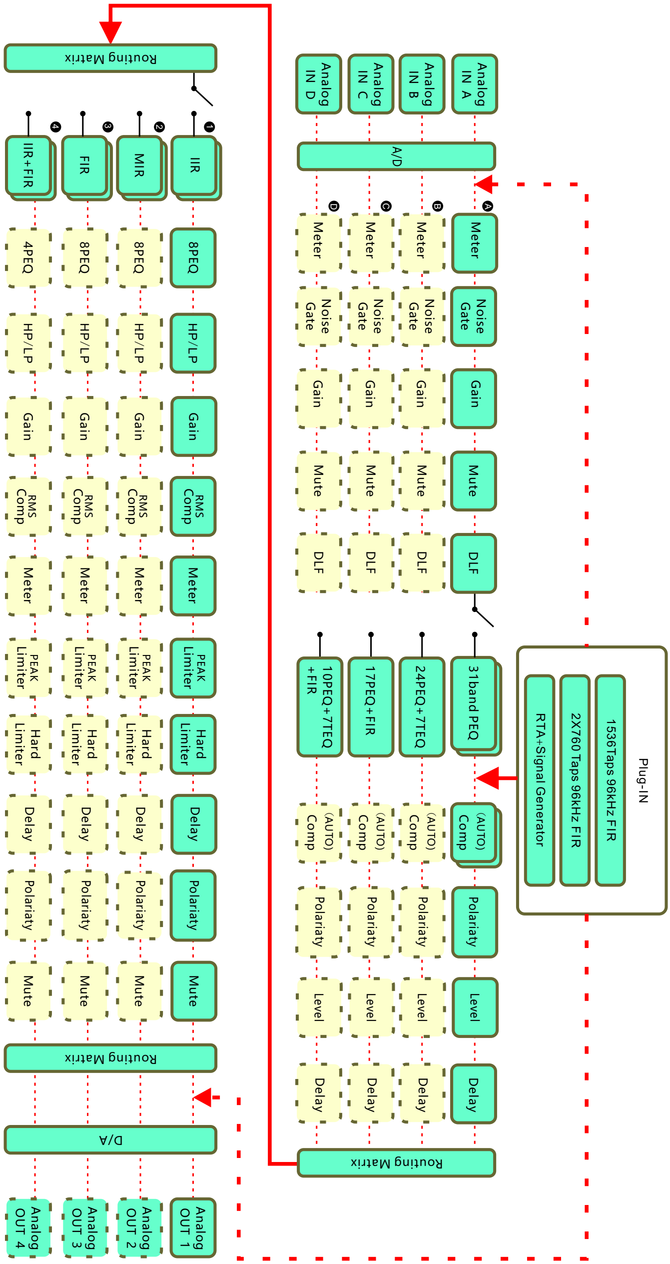
The output matrix allows the user to mix output channel signal to any physical output channel in any ratio. The knob PM1 selects the source, and the PM2 adjusts the gain amount, which can be adjusted from -30 to 0dB.

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 NAV key to the end of CH1 routing
- Rotate PM1 to select output channel 2
- Rotate PM2 to adjust the gain to 0dB
- Click Esc 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 PM2 to adjust the gain to 0dB Finish



P6 Overview of signal processing



7. **Use of MIR-A series software**

Minimum system requirements for running MIR-A 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-A 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-A 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).

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.

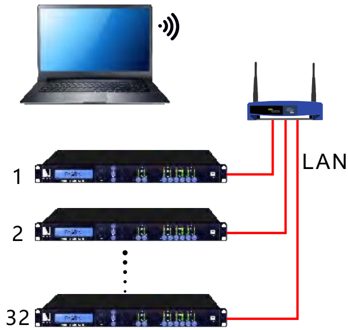
7.2 Use network cable to connect

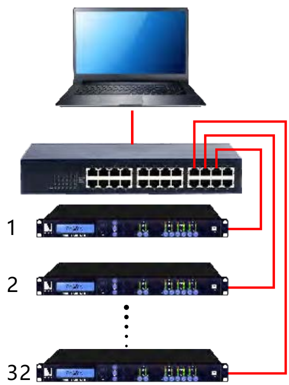
When using the network interface to connect, use a Category 5/6 network cable to connect the Ethernet port of the processor to the network adapter interface of the PC, adjust the IPV4 address option of the corresponding network adapter in the Windows network settings to automatically obtain DHCP, and 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.

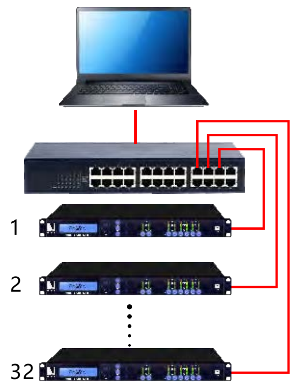
You can use USB or TCP/IP to upgrade the firmware to ensure the stability of the power supply during the upgrade process. In case of a power outage, you may need to return to the factory for repair.

P8. Advanced features

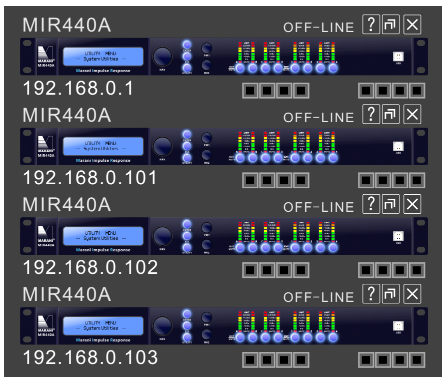
8.1 Group control Overall marshalling control

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.



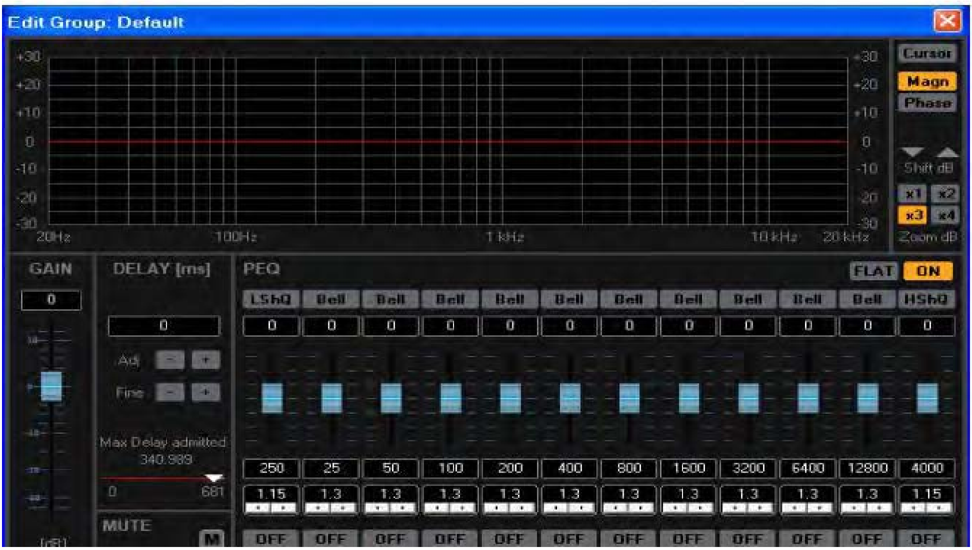
7.3.2 Connect the processors that need to be grouped online in the software, and choose to add grouping.



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.

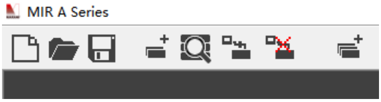
At this time, there is a small window for group control, you can control the overall gain, mute, monitor level, etc.

Double click the border to modify the detailed content：



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.

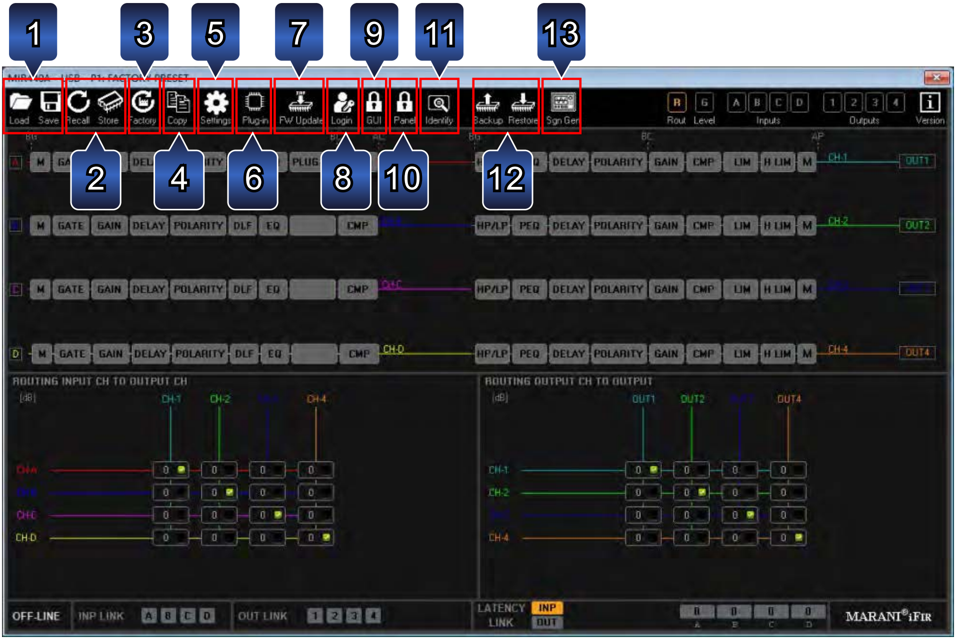
Find device: For the situation where the IP address of the device is unknown and the port number of 485 is unknown, you can select the search function to quickly find the corresponding device.

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.

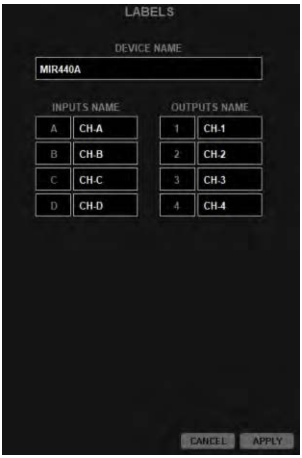
5 Settings

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 the software after you are online.

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).



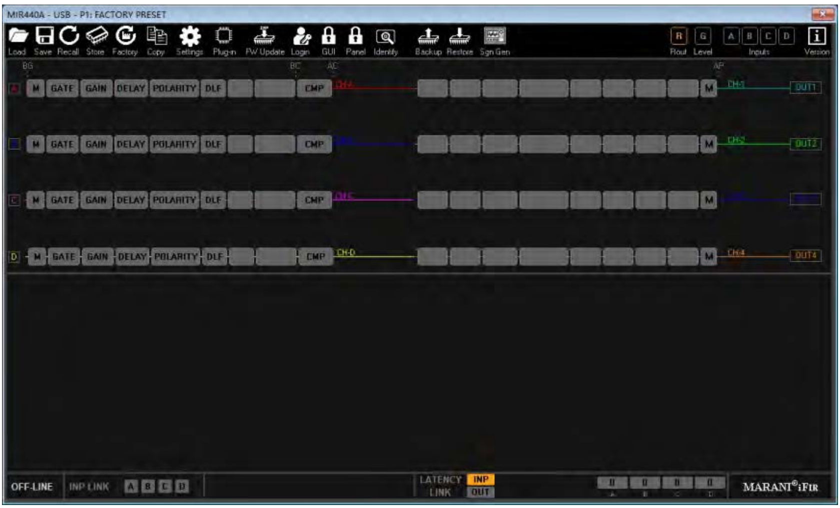
Group Reset: after using the group, the last saved group information is still in the input channel. If you need to remove it, you need to reset the required channels in the group reset.

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.

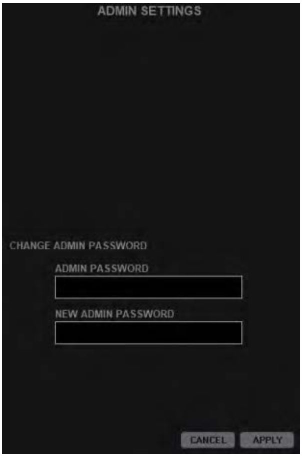
User Authority Setting: When logging in as an administrator, part or all of the output channel functions of the machine can be locked to ensure the integrity of the preset. The administrator password is 111111 by default. Click Lock and log out of the administrator account. The locked items will be grayed 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.



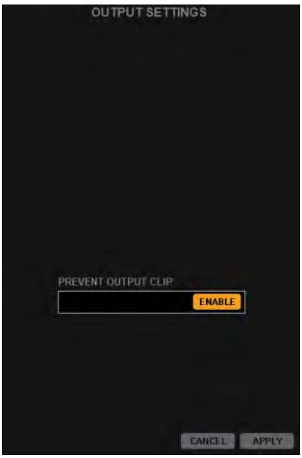
Administrator Settings: modify the administrator password, the default password is "111111".



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 +18dBu, e nd all processes are in Flat. It is turned on by default.



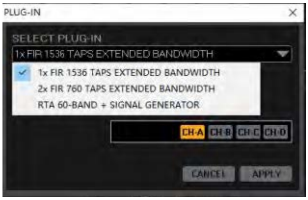
6 Plug-in

Plug-in function: Extra processes can be added in plug in shape on Input path, in addition to the fixed processes of the MIR controller. Plug-ins will be made available in time. The first available Plug ins are including.

1.Single channel 96kHz sampling rate 1536 taps FIR filter.

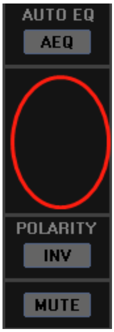
2.Dual channel 96kHz sampling rate 760 taps FIR filter.

3.60-band RTA spectrum analysis and advanced signal generator.

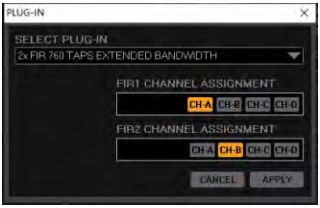


Plug-in 1 is applied to channel A by default.

Channels without plugins are empty here



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).



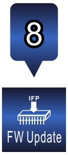
When selecting RTA and advanced signal generator.

The signal generator and RTA icons will appear.

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

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 level range is-30dBu to +10dBu.

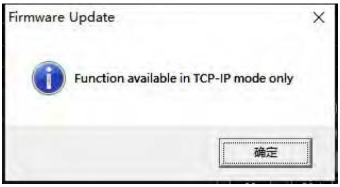


The MIR series provide firmware upgrade function, users can experience the latest version without leaving home, including new plug-ins, bug fixes, new functions, etc. At present, this function needs to be connected via a network cable, that is, the firmware cannot be updated via USB/RS485 connection.

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

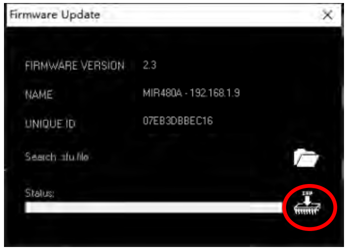
[Marani Pro Audio (marani-proaudio.com)](marani-proaudio.com)

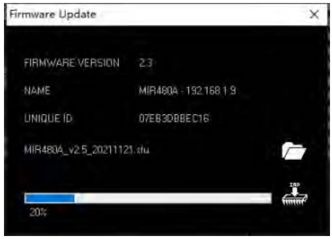
When using the USB|RS485 connection, clicking on the firmware upgrade will pop up such a prompt.



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.

9

Login

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

10

GUI

To lock the **software interface**, you need to enter a user password, the default user password is "000000".

11

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 controls, the panel can be locked to prevent the panel from being mis-operated by others.

12

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 buttons of the device will flash the blue status light once.

13

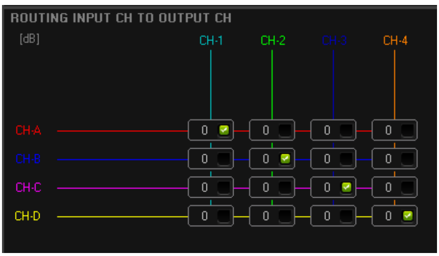
Backup

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.

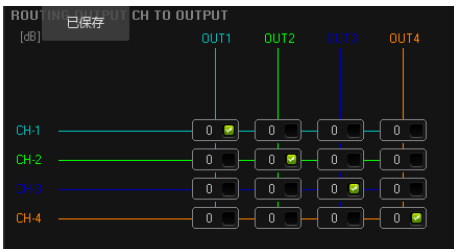
3 Signal routing part

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.

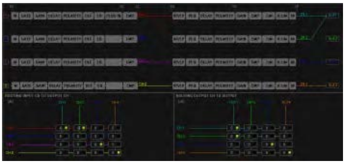


In the input matrix, you can arbitrarily route the signal from the input processing channel to the output processing channel. The default sending volume is 0dB, which is equivalent to group output. When choosing different sending volume, it is equivalent to real Mixing Matrix sending, and the sending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

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.



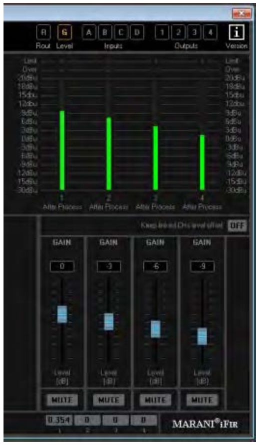
In example, use the processor A channel input, the output channel 1 is connected to a 2 way passive speaker, and the route is as shown in the above figure. The output channel uses a MIR linear filter to perform a virtual X-over on the speaker (it is better to know the crossover Point), so that different 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.

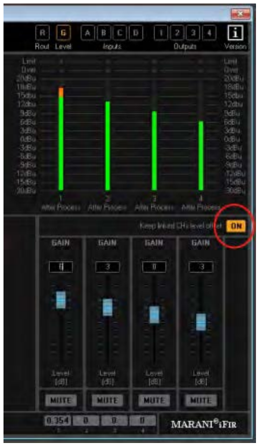


The channel level page displays the input post-A/D and output pre-D/A signal levels of all channels, corresponding to the physical level meter on the front panel, where you can intuitively adjust the channel level and mute.

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 "

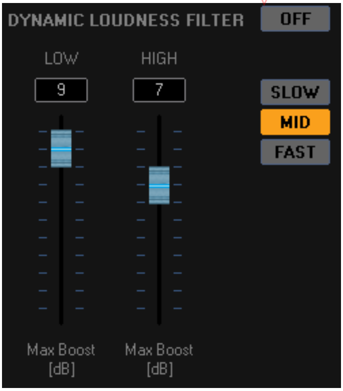
.The specific method is to assume that the output channels need to be linked. When the output channel linkage is turned on, turn on the " Keep Linked CHs Level Offset " at the red circle, so that the output channel can be linked while maintaining the original ratio.





4 Input part

Input channel processing includes noise gate, DLF, equalizer, compressor, delay, polarity, etc.



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.

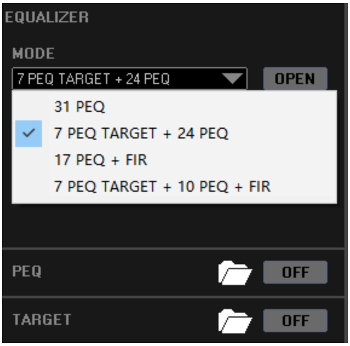
How to set: First set the maximum allowable boost level, the default is 9dB for low frequency, 7dB for high frequency, and the maximum boost is 10dB.

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 be boosted when the level is very low).

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.

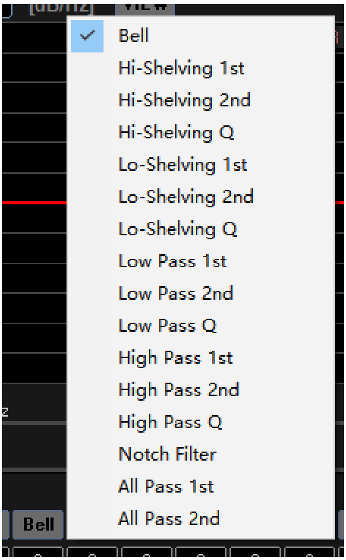
2 **7-bands target EQ and 24-bands PEQ**. The design purpose of target EQ is that users can create a small preset that is different from the overall presentation, which can be used for room acoustic correction and stored and recalled independently, which is very convenient.

3 **17-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 acoustic correction.

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 **10-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 Automatic Compressor

Conventional compressors will provide: threshold, attack time, ratio, release time, gain compensation, soft and hard knee. For some non-professional users, too many settings can cause headaches.

At this time, the new automatic compressor can help users solve the problem, as long as the threshold and ratio are set, and the automatic compression switch is turned on, the system will automatically calculate the best attack and release time according to the comparison of the Signal Crest Factor and RMS value within a period of time. Makes the compressed signal sound more natural and smooth, rather than the abrupt feeling caused by Constant Attack and Release times.

Automatic make up can also be used to Improve the match between input to Comp and output levels of the processed signa of the music signal.

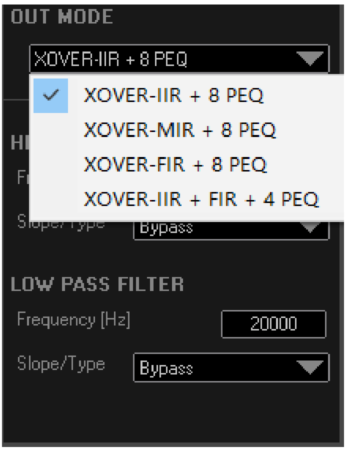
The delay alignment option is provided in the lower right corner of the software, which can be used to align the delays generated by FIR filters with different latency applied to the different channels.

On

5 Output section

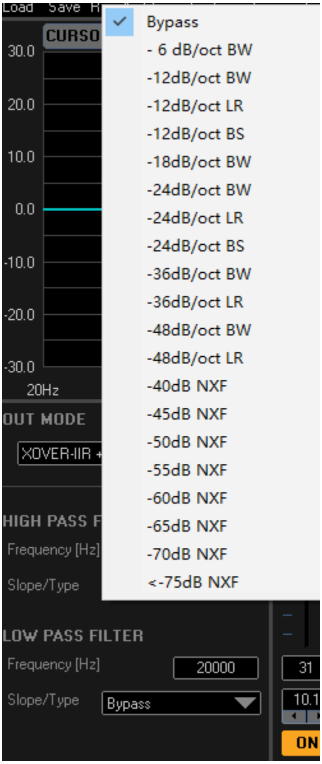
The overall overview of the output section, including crossover, polarity, delay, gain, equalizer, RMS compressor, peak limiter, hard limiter, etc.

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

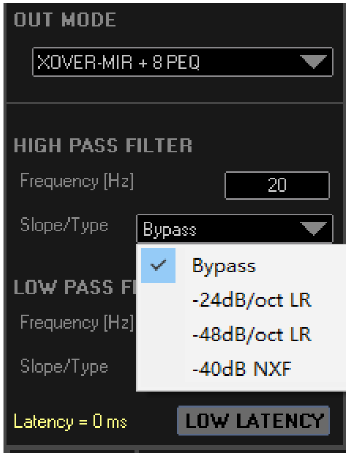


The conventional IIR filter maintains the slope of the analog filter, but will produce a phase deviation. The higher the order, the more the phase shift, which in turn causes magnitude problems.

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, and innovatively provide NXF (Notched X-over Filter) to create a deeper slope—from-40dB/oct to -75dB/oct, a deeper slope can bring a more thorough cut, make the sound clearer, and better protect tweeter driver.



The MIR linear phase filters replicate exactly the slope of the analog filter (LR24/LR48), and adds NXF-40dB without phase shifting, thereby keeping the phase Linear and easily coupling the phase at the crossover point.

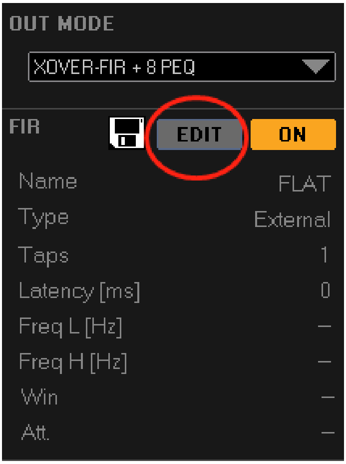


After the low latency mode is turned on, the delay caused by the latency coming with MIR linear phase filters can be further reduced, but the disadvantage is that the in band ripple close to the knee will increase from the default +-0/01% to ±0.5dB will be generated at the crossover point. However, this 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,



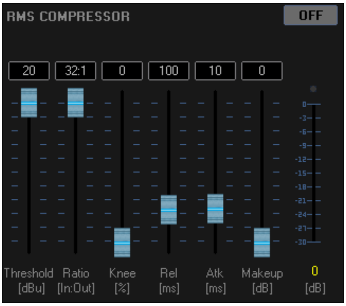
Provides three guides of high-pass/low-pass/band-pass, you can enter the frequency that needs to be band-passed according to your needs, choose more or less taps, slopes from -20dB to -120dB/oct, and multiple window function types , Click save at the red circle (here we use band pass filter as an example).

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.

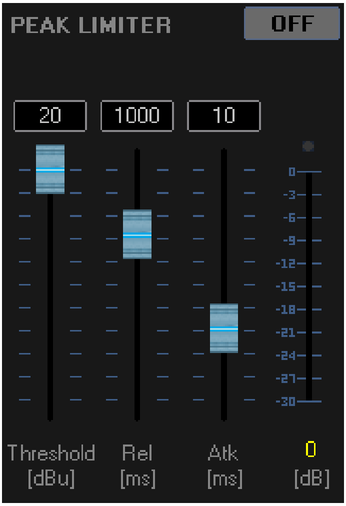
External FIR supports three formats of FIR coefficients, namely ".csv, .txt, .saf". The first two file formats can be generated by mainstream FIR convolution software, and the .saf file can be saved and generated from AEQ in the Marani processor. It should be noted that the FIR filter in the input and output part of the MIR-A 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.

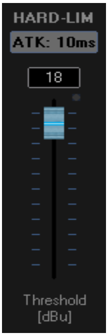
The output part provides three levels of compression limiter settings, namely RMS compressor, which can be used to set the compression option of the average level over a period of time. When the compression ratio is high, it is the limiter.



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 sp



Hard limiter Three-speed attack time can be adjusted, simple and direct to quickly match the overall gain structure of the system, the minimum attack speed is 0ms, almost zero delay processing, at the same time the distortion is extremely low, ensuring high system reliability.



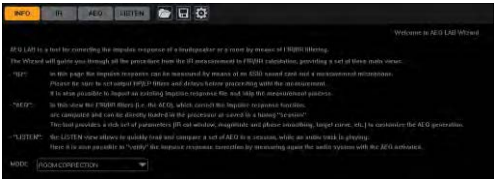
6 AEQ automatic equalization

The brand-new AEQ function can generate real-time IIR and FIR coefficients for on-site measurement, and directly store them in the processor, eliminating the need for third-party software to measure-export-convolution-import complicated steps, simply and directly apply FIR to live sound reinforcement or the speaker is preset.

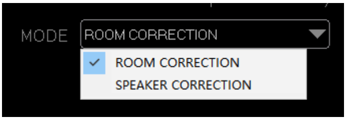
Note: AEQ License is sold independently, and the default model does not contain AEQ. You can click the software version to check AEQ authorization status after the processor is online. The device in the picture is not activated. If you need to activate, please contact your sales.

For the Licensed device, you can click "AEQ" in the upper right corner of the input or output to open AEQ.

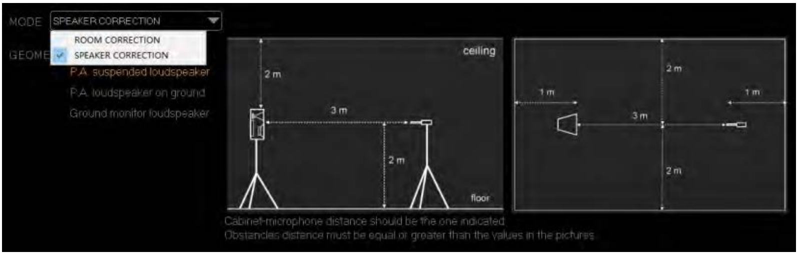
After entering AEQ, the homepage shows the introduction and mode selection of AEQ.



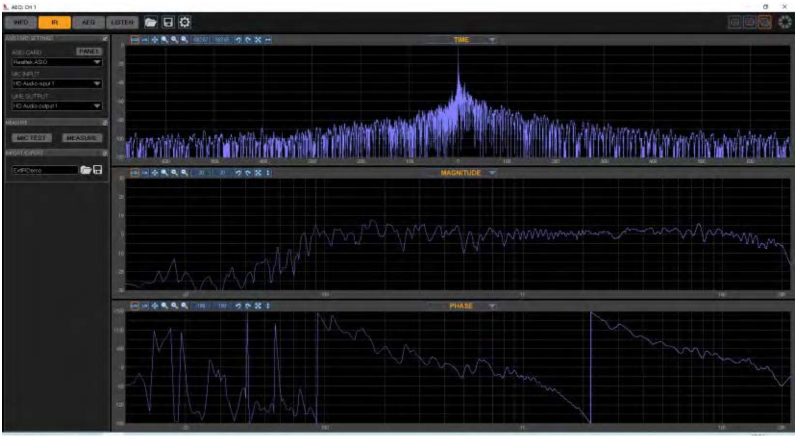
The conventional room calibration mode simulates the position of the measuring microphone at the listening point of the human ear, considering the superposition or cancellation of different frequency bands caused by the direct sound of the speaker and the reflected sound in the room, and finally the calculation of processing these factors that are not conducive to the sense of hearing model.



The speaker calibration mode is designed to reduce the impact of reflected sound in the environment. It simulates the amplitude and phase response of the speaker in the anechoic room environment. It provides several different placement methods, but the choice of different placement methods will not affect the calculation results. Reminder before measurement.

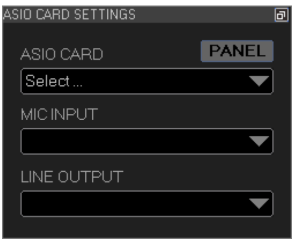


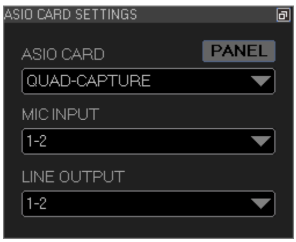
After selecting the mode, go to the second page of IR (impulse response) acquisition.



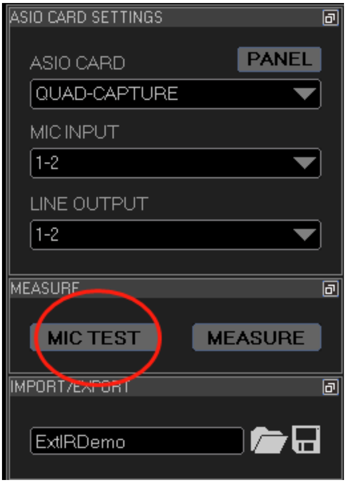
The main problem here is to correctly select the measurement microphone channel and the output channel of the test signal.

Remarks: Only sound cards compatible with ASIO drivers can be used. Wave drivers are not compatible, so first confirm that your sound card is compatible with ASIO drivers. (Usually the onboard sound card that comes with the motherboard cannot drive the measurement microphone, so it is usually impossible to use the realtek that comes with windows to complete the measurement).

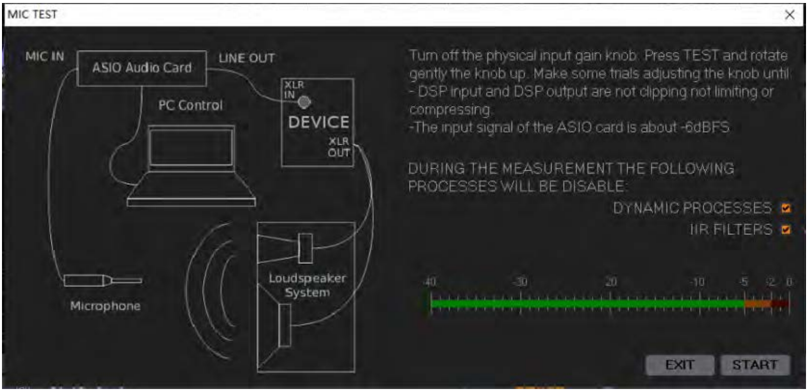




After setting up the sound card and measurement/output channel, you can proceed to the next step, that is, the calibration of the test soundpressure level.

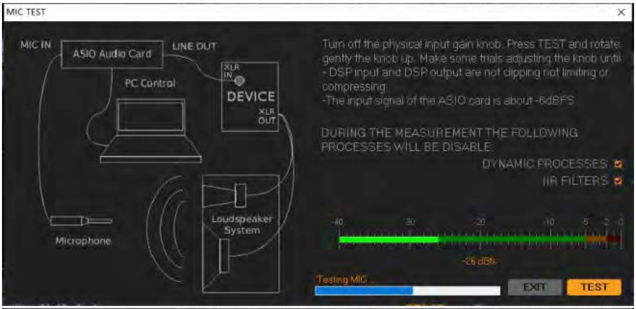


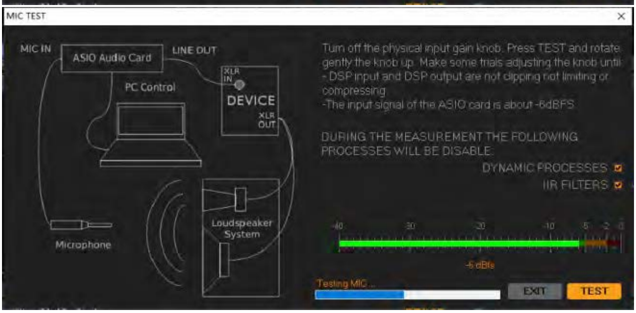
Click MIC TEST to enter.



Connect the sound reinforcement system and the measurement system according to the instructions on the figure. Note: No NEED "reference channel" of other measurement software always required Click start.

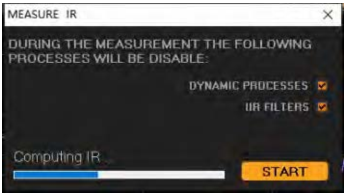
Adjust the output knob and input knob of the ASIO sound card to make the sound pressure level sufficient during the whole measurement process, and the measured signal level indicated by the level meter does not exceed-6dBFs at the highest and-30dBFs at the lowest to ensure sufficient signal-to-noise ratio.



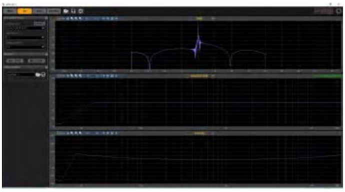


After confirming, you can close the test page and click measure .

The calculation will start automatically after the measurement is over.

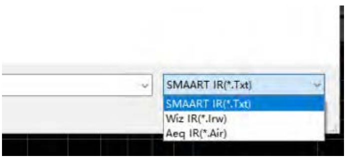


Obtain the measurement results; what is demonstrated here is to measure the impulse response of the ASIO sound card itself, the actual measurement is far less silky.



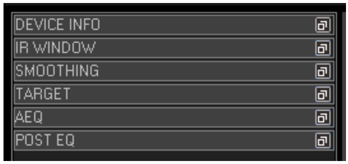
The measurement can be done by the software itself, and it also supports importing measurement results from other measurement software. For example, if you need multi-channel averaging, you can use Smaart V8 software to perform multi-channel measurement, and then export the ASCII file in txt format, and then import it here.

Click to open the file, you can choose to import.



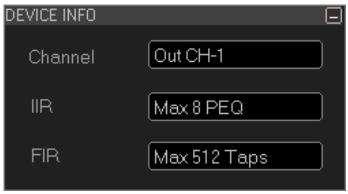
In the AEQ page, it is the adjustment of the FIR calculation process.

Expand here, you will get detailed information for each step.

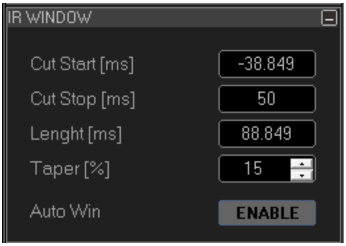


Those are: device information (these info are related to the Filtering Resoruces present on the signal path where from the AEQ has been launched)

Output channel 1, maximum 8 segments of PEQ are used, FIR has 512 taps.



The **time window** of the impulse response can avoid the reflected sound pulse visible to the naked eye and improve the accuracy.



smoothing

Different smoothing can be set for amplitude and phase, or it can be smoothed separately for a certain segment.

For example, divide the amplitude into 20-2000 Hz to smooth in 1 octave, and 5000 to 20000 Hz to smooth in 1/3 oct .

Target curve setting

You can increase or decrease the overall level

Obtained, similarly, the phase can also be segmented and smoothed with different smoothness.

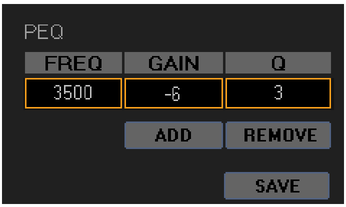
There are two presets, FLAT and Follow

The target curve can be boosted or attenuated in sections

For example, to compensate for High frequency.

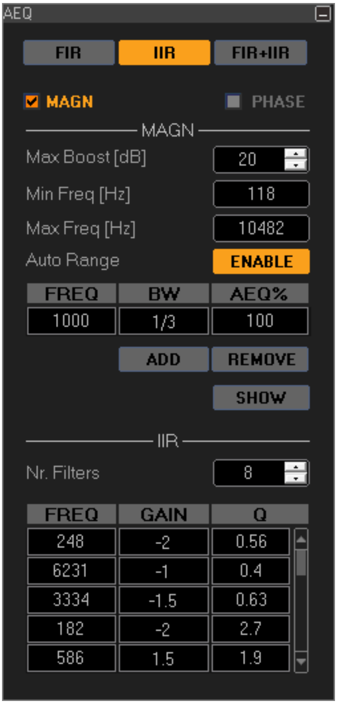
Get the green curve.

You can also add PEQ manually in order to get, starting from the flat or Auto curve, a real TARGET RESPONSE where the Curve can be set y the user on the base of the PEQ's filters set.



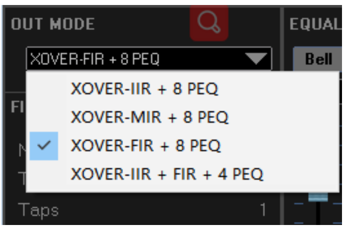
Next are the most important filter settings

The default state is IIR filter. If you need to use FIR or FIR+IIR hybrid filter, you need to manually select.

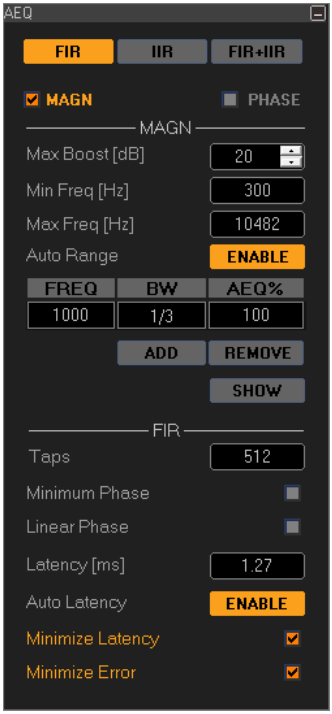


(The filter type here depends on the filter type selected at the input or output part).

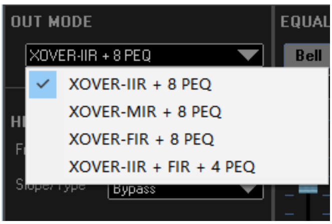
For example, we select FIR+8PEQ on output channel 1,



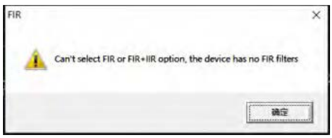
FIR filter can be used at this time.



If selected as IIR+8PEQ.



It prompts that FIR filter is not available.

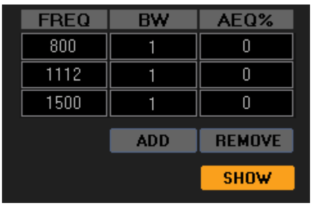


When the pure IIR filter is selected, only the amplitude can be corrected.

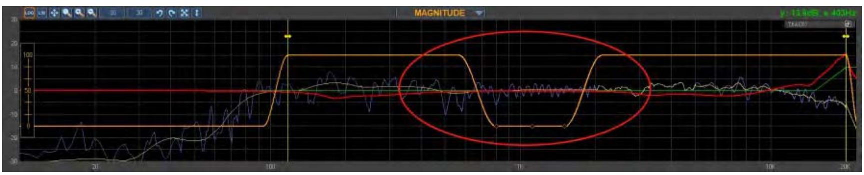
You can choose the maximum boosted gain amount, frequency lower limit and frequency upper limit.

AEQ coverage can choose the range of application of AEQ, similar to segment smoothing, the entire frequency band can be divided into several segments for processing.

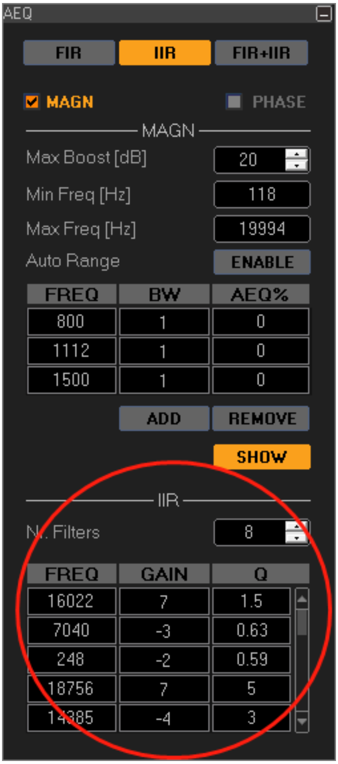
For example, it is necessary to avoid the 800-2000Hz segment and process 20-800Hz and 2000-20000Hz,



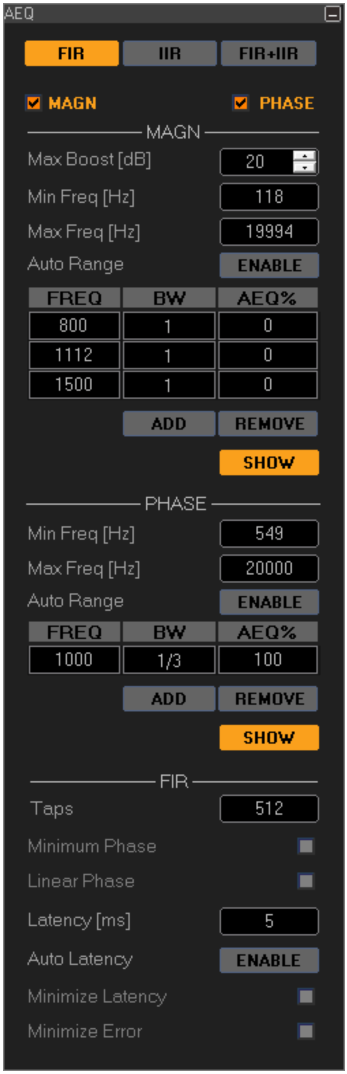
It can be obtained that the amplitude of 800-2000Hz remains as it is, and only the positions lower than 800Hz and higher than 2000Hz are processed.



The number, frequency, gain, and Q value of the IIR filter used will be displayed here.



If you choose FIR mode, you can tick PHASE, that is, process the phase at the same time.



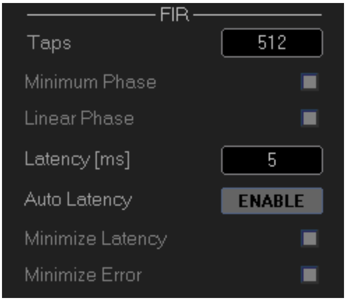
In addition to the range and avoidance processing similar to the amplitude, it can also be set separately for the FIR filter

For example; the number of taps used by the filter

You can also use automatic delay to reduce the amount of delay in FIR calculations

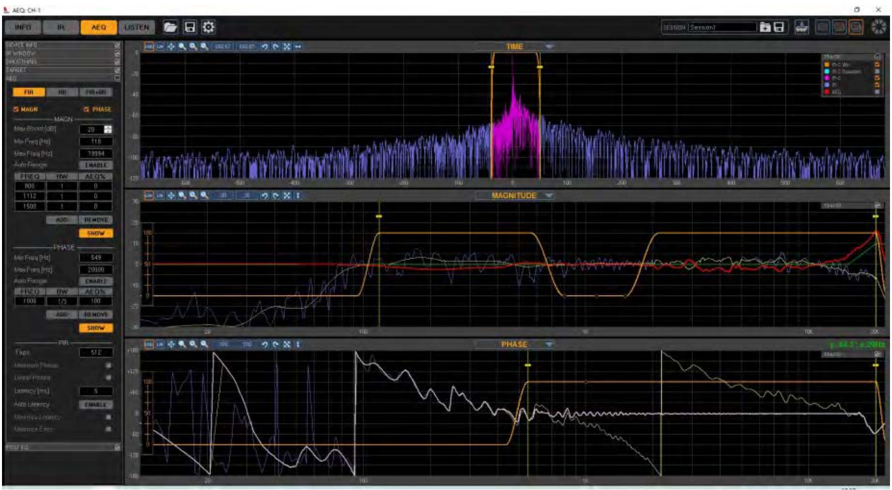
Optional minimum phase/linear phase

The minimum delay and minimum error can be selected in the delay part.



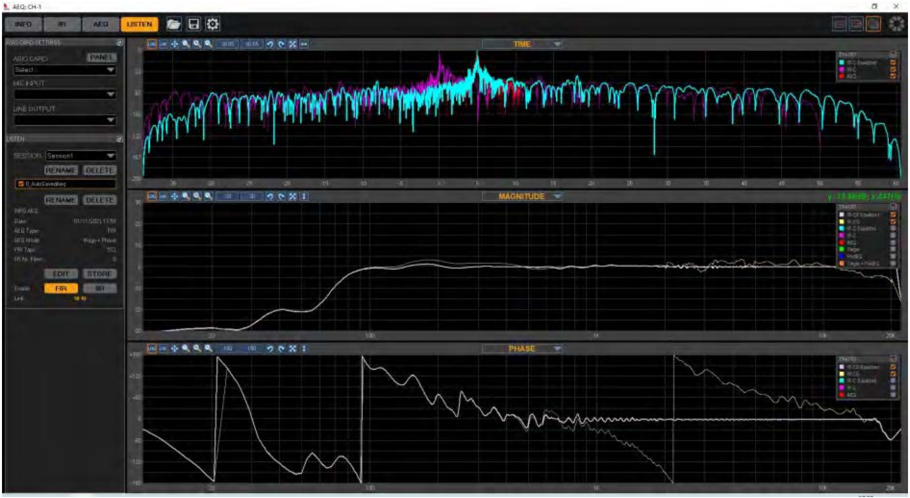
Post EQ can be added to PEQ after FIR processing.

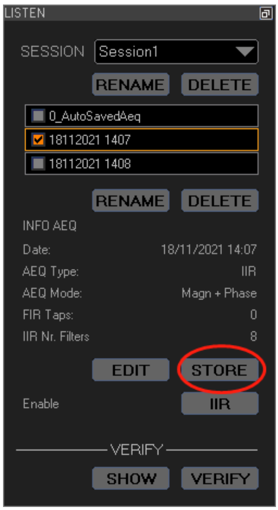
Can store presets for contrast switching



Save and listen

You can save and upload AEQ results on this page.





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