



# MIR480I/MIR880F

Audio system processor

POWERFUL  
MARANI DSP  
iFIR & AEQ



# Quick Start Manual



2021-11 V1.0

MARANI®



# INTRODUCTION

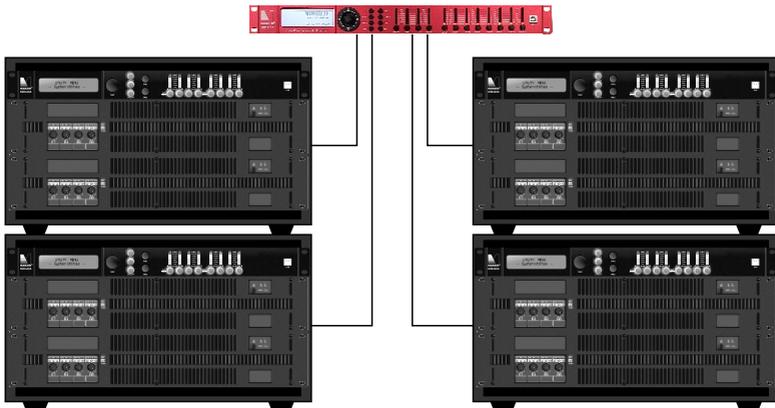
## MIR Series– MIR880F/MIR480I

The **MIR-F series** of digital processors are the flagship system processors newly launched by Marani, including two models: **MIR480I** and **MIR880F**. The hardware adopts the top AD/DA chip, the background noise is as low as -96dBu, and it has analog, digital AES/EBU, Dante three signal access capabilities, and has the automatic switching function. Any signal can be set to the highest priority. As the core processor of the system, it also has backup features, suitable for government projects, high-end clubs, large-scale mobile performances, theaters, radio and television and other places that have higher requirements for signal processing.

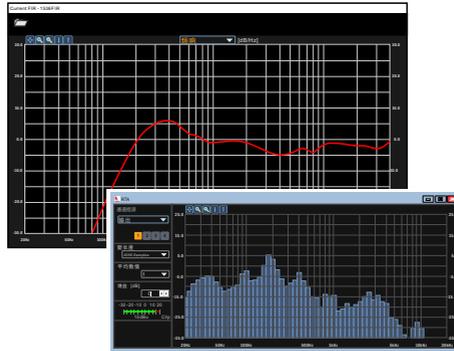
In the signal processing part, the powerful **MARANI DSP**, DSP and AD/DA converters are all running at **96KHz sampling rate**, and the complete processing functions provide a complete speaker x-over/protect solution.



From the input gain/delay/noise gate /PEQ/compressor FIR to the output gain/delay/polarity/crossover/FIR/PEQ/compressor/peak limit, the parametric equalizer (PEQ) has up to 13 filter types to choose from. The output crossover filter has the classic Linkwitz-Riley/Bessel/Butterworth, The NXF (North X-over Filter)and FIR filter with slope up to 120dB per octave are also available. The new MIR linear phase crossover filter makes it easier to engage the phase of the crossover point and produces lower latency. Everything we offer is for better sound.



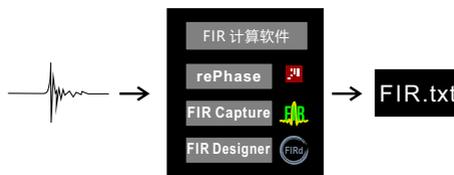
- Newly added DSP plug-in, providing additional practical plug-ins such as high-order FIR filter and feedback suppressor, including 96kHz FIR filter/dynamic EQ/ high order signal generator/RTA( real-time spectrum analyzer), etc.



- The newly designed Hard Limiter allows constant rate limiting of signals exceeding the threshold at any threshold value, which better protects the driver.



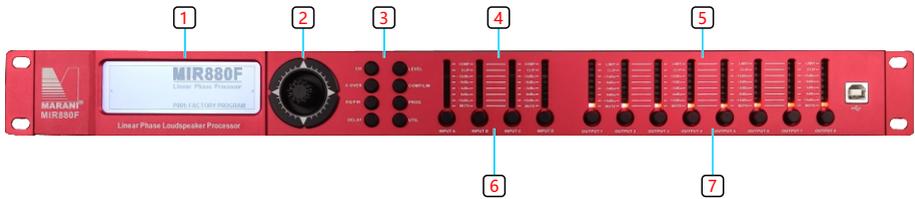
- Each input and output channel provides a 512-tap FIR filter, which can be customized by third-party software to generate the FIR convolution you need. It can be used for speaker presets to improve the phase response and control the directivity according to requirements



- The newly added MIR linear phase crossover filter has the filter shape of the traditional IIR filter (4th/8th order Linkwitz-Riley), but does not produce any phase shift, and the resulting delay is about 50% of the FIR filter.



# Front panel overview

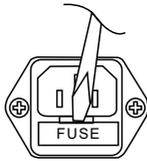


- 1— 320\*96 high score LCD screen
- 2— Rotary Encoder (can be rotated / up, down, left, and right)
- 3— Function selection
- 4— Input level meter: Display the pre-fader signal, the mute does not affect the level display. When the Mute light is on, it means that the current channel is muted. When the SIG light is on, it means that the input signal reaches -40dBu; -12dBu, 0dBu, +6dBu, and +12dBu represent the actual RMS value of the signal. When the Clip/Over light is on, the signal is close to the maximum value before analog to digital conversion. The Limit light lights up when the channel compressor/peak limiter/hard limiter is activated.
- 5— The output level meter displays the post-fader signal, when the output channel mute meter does not display any value. Mute light on means the current channel is muted, SIG light on means that the input signal reaches -40dBu; -12dBu, 0dBu, +6 dBu, +12 dBu represent the actual RMS value of the signal, Over light on means the signal reaches the Hard limiter threshold, The Limit light lights up when the channel compressor/peak limiter/hard limiter is activated
- 6— Input channel selection/mute: press this key to edit the processing of the current input channel, including channel name, gain /polarity/delay/parametric equalization/compressor. Hold for three seconds to mute the current channel.
- 7— Output channel selection/mute: press this key to edit the processing of the current output channel; including input channel matrix routing, high pass and low pass, slope, filter type; also include gain/polarity/delay/parameters Equalization/RMS compressor/peak limiter/hard limiter and other parameters. Hold for three seconds to mute the current channel.

# Rear panel overview



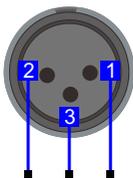
- 1\_\_\_ AC power input, standard C13 interface, please ensure that the power grounding pin is well grounded, otherwise electric shock may occur.
- 2\_\_\_ Ship type power switch.
- 3\_\_\_ The fuse box contains a spare fuse, which can be replaced in case of emergencies. The fuse specification is 220V0.8A



- 4\_\_\_ Ethernet control interface, support TCP/UDP protocol, the default IP address is DHCP , The port below can be changed to an RS485 loopout port.
- 5\_\_\_ Rs485 protocol interface, which can be used for connection software, and can also be used for central control protocol transmission,**The wiring is defined as:**

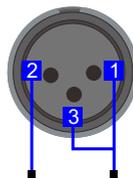


- 6\_\_\_ Analog signal input interface, which can be switched to digital AES/EBU signal input, processor audio signal input, maximum input level +20dBu, input impedance 20KΩ.
- 7\_\_\_ Analog signal output interface, processor audio signal output, maximum output level +20dBu, minimum load 1000.



XLR Balanced :

- 1 Ground,shielding
- 2 Signal(in phase+ )
- 3 Signal(out of phase- )



XLR Unbalanced :

- 1-3 Ground,shielding
- 2 Signal

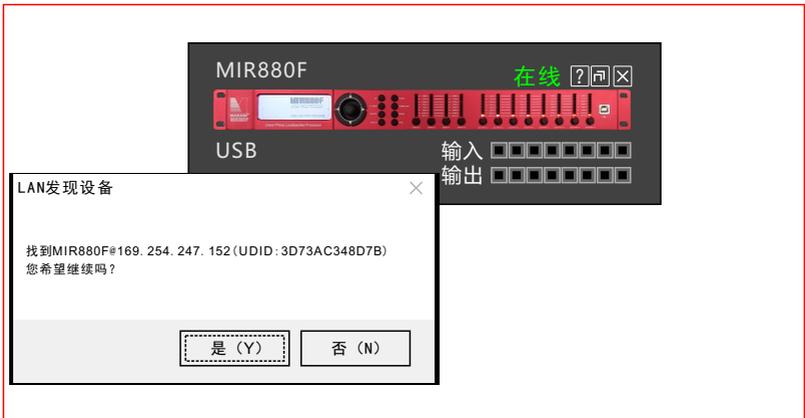
# HOW TO QUICKLY CONNECT TO THE SOFTWARE

The MIR-F/I series provides 3 kinds of control interfaces, namely USB/RS485/ETHERNET, the connection switching of the three modes does not need to be set.

- 1\_\_ When using a network interface to connect, use a Cat 5/6 cable to connect the ETHERNET port of the processor to the network adapter interface of the PC, and 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.



## Pop-up diagram



## 2\_\_ To use RS485 often requires a converter

- ① Traditional PC will provide DB9 serial port, which needs to use DB9 to RJ45 conversion line to adapt to the high-level and low-level signals of RS485 protocol usually at pin 2 and pin 3, and pin 5 is grounded.

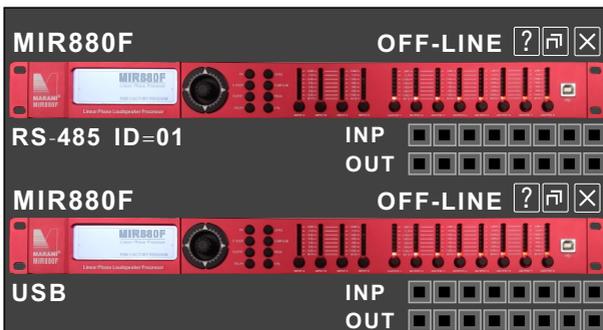
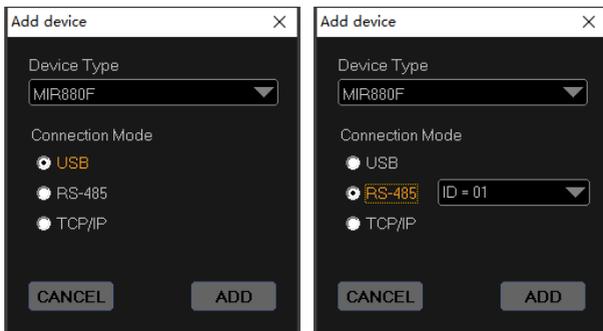


- ② Usually now common PC models do not provide serial ports, so a USB-to-serial converter is required. Marani provides a USB-to-RJ45 serial converter (USB-485-RJ)



③ The interface conversion diagram of USB to XLR converter is as follows

After the hardware is successfully connected, the interface prompt will automatically pop up when using the network method, RS485 or USB need to manually add the corresponding device, and then choose to read or write



# SOFTWARE MAIN INTERFACE

The screenshot displays the software main interface with the following components and callouts:

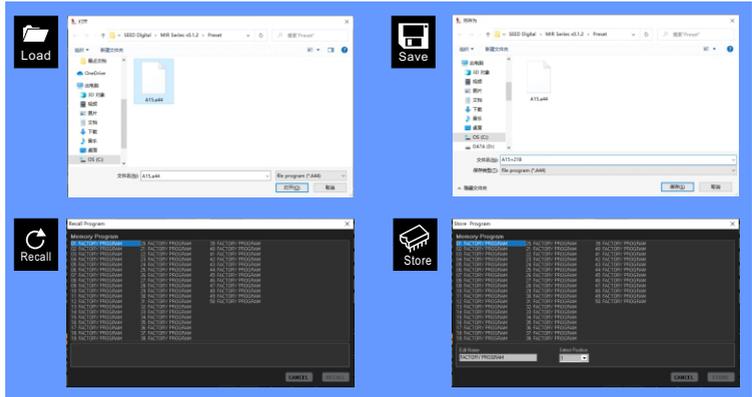
- 1:** Top toolbar containing icons for file operations (Load, Save, Refresh, Stop, Copy) and system settings (gear icon).
- 2:** The settings gear icon in the top toolbar.
- 3:** The 'Routing' tab selected in the top navigation bar.
- 4:** The 'Load' button in the top toolbar.
- 5:** The 'Backup' button in the top toolbar.
- 6:** The 'Refresh' button in the top toolbar.
- 7:** The 'Input' section in the right sidebar, showing 'Input Level' and 'Input' buttons.
- 8:** The 'OFF-LINE' button in the bottom left corner.

The main interface is divided into several functional areas:

- SOURCE:** A list of input sources including ANALOG, DANTE, and DANTE2.
- ROUTING:** A central table with columns for routing paths (e.g., IN X -> CH X, CH X -> PLUG-IN Y, PLUG-IN Y -> CH Y, CH Y -> OUT Y) and a grid of routing parameters (GATE, GAIN, DELAY, DLF, EQ, COMP).
- OUTPUT:** A list of output destinations including DANTE1, DANTE2, DANTE3, DANTE4, and DANTE5.
- WAVEFORMS:** A section at the bottom showing signal waveforms for various channels (e.g., IN A, IN B, IN C, IN D, IN E, IN F, IN G, IN H).
- LATENCY:** A counter showing latency in samples (e.g., 0, 0, 0, 0, 0, 0, 0, 0, 0, 0).
- RIGHT SIDEBAR:** Contains system controls such as 'Input Level', 'Input', 'Output', and 'Version'.

# 1

Use Save to save the current preset to the PC, and use Import to import the preset that has been saved in the PC to the processor. Use Recall to recall the preset stored in the processor, and click Save to store the preset in the processor's preset library.



# 2

The channel name and processor name can be changed arbitrarily; After entering the administrator password, you have the highest authority of the machine, and you can freely lock some or all of the parameters of the output channels.



3

Input processing part, including gain, delay, polarity, noise gate, maximum 27-band parametric equalization, or use FIR filter and 13-band PEQ, dynamic processor part includes compressor and dynamic loudness booster, the input part is All open and unlocked by default



4

The administrator has the highest authority of the machine. After the administrator account is logged in, some or all of the output processing modules can be locked. The locked part can be hidden or not hidden. The output part is completely invisible after being hidden.



5

Overall preset saving and restoration, suitable for machine migration or OEM customers writing data in multiple machines



6

Provides many plug-ins, currently including 48kHz sampling rate FIR filter and 96kHz extended bandwidth FIR filter, the default is 48kHz sampling rate, 1024Taps per channel; when a high number of taps is required, multiple channels can be aggregated into one channel to achieve the highest 4096Taps FIR filter, enough to cover the whole frequency band processing



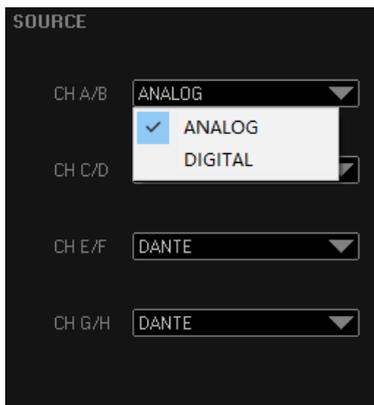
7

The output part, including gain, polarity, delay, IIR high and low pass filters, can be switched to MIR linear phase filter, or FIR filter can be used. Contains three dynamic processing modules: 8-band PEQ, RMS compressor, peak limiter, and hard limiter.





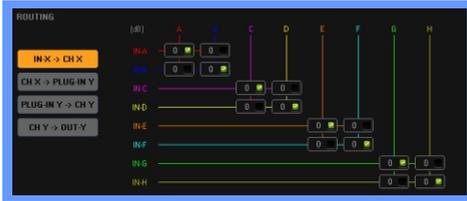
Signal source selection, AB/CD channel can be selected as analog signal or AES digital signal input in two groups



# 9

## Matrix routing design, completely free signal transmission design

In the input part, the adjacent physical input channels can be combined and sent to one input processing channel;



From the input processing channel to the plug-in channel, 8×8 matrix transmission can be performed;



The plug-in channel to the output processing channel can also be sent in 8×8 matrix



The output processing channel to the physical output channel can be superimposed on adjacent channels.



**Marani Proaudio Srl**

Via Vittorio Veneto N.11  
I-42022 BORETTO RE  
[www.marani-proaudio.com](http://www.marani-proaudio.com)  
[info@seed-digital.com.cn](mailto:info@seed-digital.com.cn)



**China Facotry**

SEED Guangzhou Electronic LTD.  
4F/5F, Building B, NO.885, Shenzhou Road  
Scientific City, Guangzhou, China  
[www.marani-proaudio.com](http://www.marani-proaudio.com)  
[info@seed-digital.com.cn](mailto:info@seed-digital.com.cn)  
Tel: 18665089038

