Speaker Management Systems

MIR Series - MIR480A

MIR480A

MIR480A 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.
In the signal processing part of MIR480A, inside powerful MARANI DSP, DSP and AD/DA converters are running at 96KHz sampling rate, complete processing functions provide a complete speaker crossover solution.
From input gain/delay/noise gate/EQ/compression/FIR , to output gain/delay/polarity/frequency division/FIR/EQ/compression/compression, there are up to 13 types of parametric equalizer (PEQ) The filter type can be selected. The output crossover filter has the classic Linkwitz-Riley /Bessel/Butterworth, and there are also NXF horn-type filters and FIR filters with a slope of up to 120dB per octave. The newly added MIR linear phase frequency divider filter can make the phase of the crossover point easier to join and produce a lower delay. Everything we provide is for better sound.
Added DSP plug-in, providing 96kHz FIR filter/high-order signal generator/RTA real-time spectrum analyzer, etc.
The newly designed Hard Limiter (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 channel provides a maximum 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 divider filter has the filter shape of the traditional IIR filter (Linkwitz-Riley 24/48oct), but does not produce any phase shift, and the resulting delay is about 50% of the FIR filter.

Features

1. The machine runs at 96KHz sampling rate, the frequency response remains flat at 20-40Khz, the background noise is as low as -94dBu, and the maximum input level reaches +20dBu.
2. Full matrix mixing, any input channel can be sent to the output channel, and even several non-adjacent output channels can be superimposed and mixed to the physical output.
3. Each input and output channel is equipped with an RMS compressor, which can control the signal dynamics on the input channel or use it to shape the sound intensity. The newly designed extremely low distortion peak limiter can prevent sudden large dynamic signals from damaging the speaker unit and effectively guarantee the safety of the system.
4. Each channel is equipped with FIR filter up to 1024 Taps.
5. New MIR linear phase crossover filter: MIR linear phase filter is a new X-over filter, which has the shape of a classic filter (LR24) without any phase distortion and keeps the phase curve flat straight.
6. The machine is equipped with a standard network port, which can be directly connected to a PC through a network cable. The default DHCP automatically obtains an IP address and completes all connections with one click.
7. The new marshalling setting can control 128 processors at the same time, and can control gain, mute, PEQ and polarity uniformly, which increases the convenience of multi-machine debugging. 8. The input channel is equipped with a dynamic loudness booster, which can effectively enhance the sense of hearing.
9. Add plug-in, including 96kHz FIR filter with extended bandwidth, high-order signal generator and RTA spectrum analysis.

General

Preset---	32
size--	482x44x207mm 1RU
Net/Gross weight---	3.0 Kg / 3.5 Kg

Speaker Management Systems

MIR Series - MIR480A

Audio

Input impedance--------------	20KΩ	Crosstalk---------------------------	≦-95dB
Output impedance-----------	100Ω	Signal-to-noise ratio----------------	≧113dB
A/D dynamic range-----------	118dB	Noise floor --------------------------	≦-94dB (A weighting)
D/A dynamic range----------	118dB	Common Mode Rejection Ratio------	60dB
Maximum input level---------	20dBu	Number of analog input channels----	4
Maximum output level-------	18dBu	Number of analog output channels--	8
Total harmonic distortion----	≦0.003%	Rs485 control port-----------------	2
Frequency response---------	20Hz~40kHz	Network control port----------------	1

DSP processing

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

MIR480A