About DSP

The DMP-A8 integrates a DSP module, performing digital processing on the signal between the digital signal and DAC.

When using the DSP function, please ensure that you are familiar with acoustics and DSP-related knowledge, and make sure you areaware of the consequences by adjusting the corresponding parameters. Otherwise, unexpected results might occur.

Each DSP model has certain limitations, such as sampling rates and memory size. Similarly, the DSP used in DMP-A8 has limitations regarding its sampling rate.

The supported PCM sampling rates for DMP-A8 are as follows:
44.1K / 48K / 88.2K / 96K / 176.4K / 192K

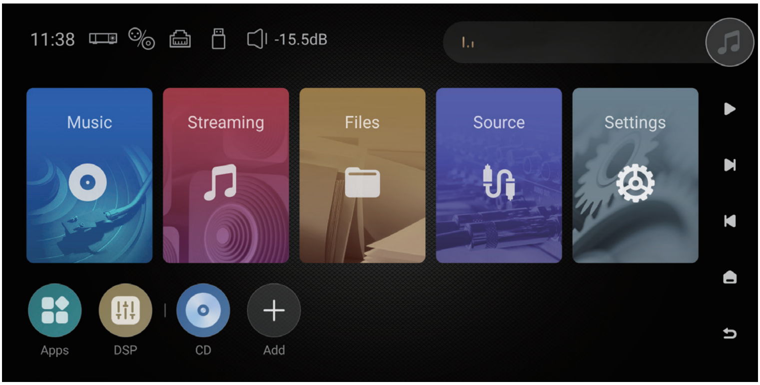
For PCM and DSD signals higher than 192K, DMP-A8 will pass them to the DAC in Bypass mode. The internal working sampling rate of the DSP is 48KHz. When the DSP function is enabled, all signals are asynchronously sampled (ASRC) to 48KHz before undergoing signal processing.

Output ports supported by DSP:
XLR balanced output / RCA analog output
Input ports supported by DSP:
Internal player / Bluetooth input / USB-B input / Optical input 1
Optical input 2 / Coaxial input 1 / Coaxial input 2 / ARC input

Using DSP

1.Enter DSP interface.

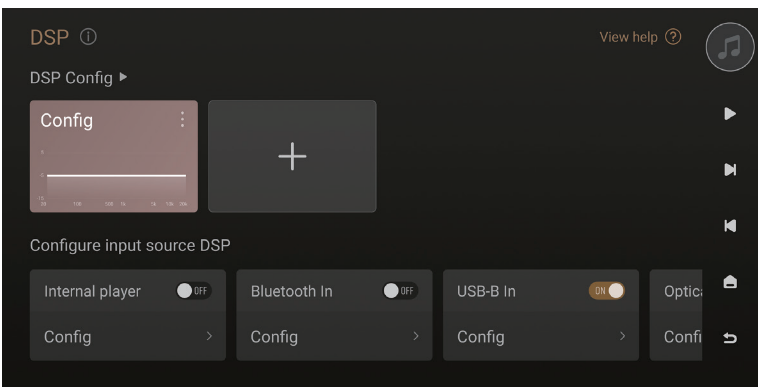
Locate the  on the Home page and click to enter.



2.Access the DSP Configuration Interface

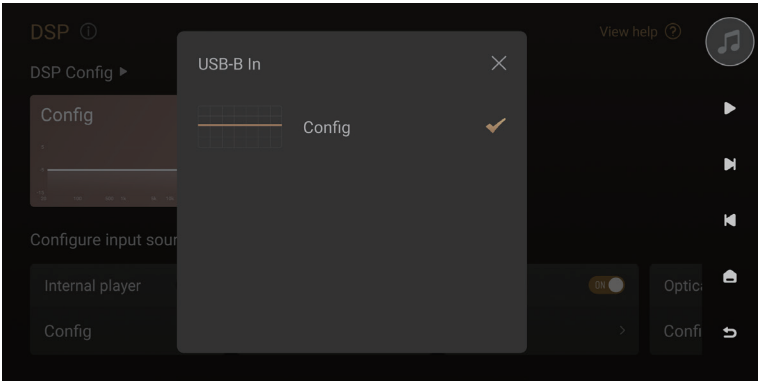
① DSP Configuration

This configuration pertains to the DSP parameters of DMP-A8. There is a default DSP configuration upon entering. Additionally, you can create new configurations and give them custom names.



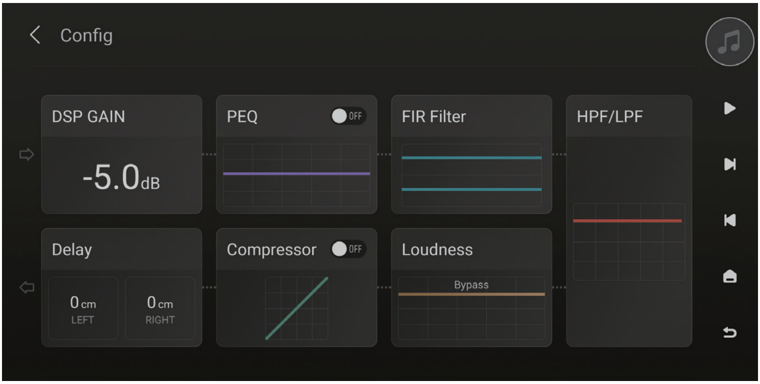
② Configure input sources for DSP

We can choose different DSP configurations for each input source, and at the same time, we can deactivate the DSP function for a specific input source."



3.Configure DSP

Choose a DSP configuration you want to design, click to enter:

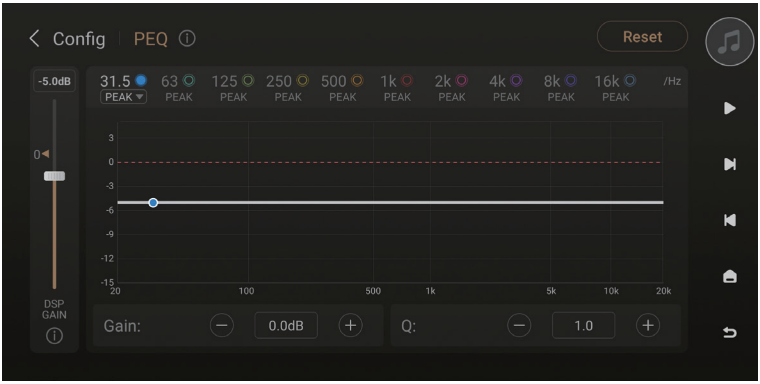


This diagram illustrates the internal signal processing flow of the DSP. The signal enters from 'In' and goes through each module. Finally, it exits from 'Out'.

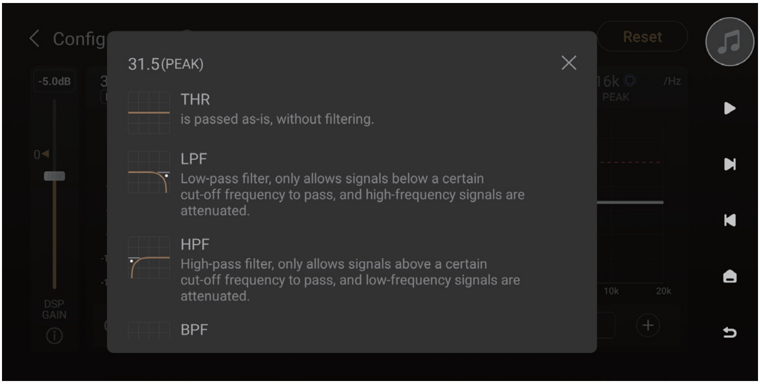
① DSP GAIN

To prevent signal overflow in backend modules, such as increasing positive gain in PEQ, it is generally necessary to reserve enough gain margin. It is recommended to set it to -5dB.

② PEQ (Parametric Equalizer) Parameters

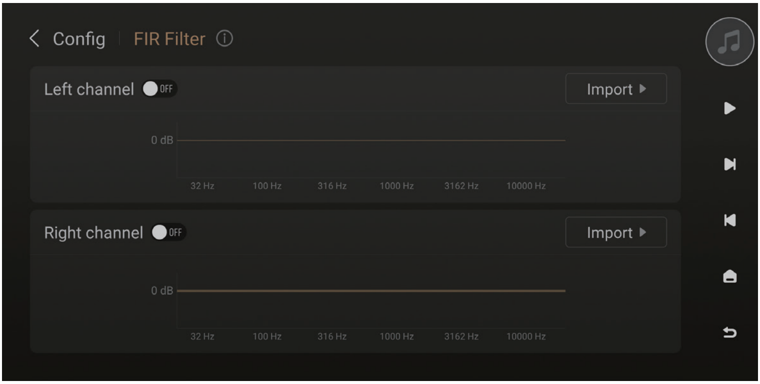


The PEQ can be customized according to your needs, and different filters can be set for each frequency band.



After completing the design, save and exit. To activate the PEQ, the switch outside needs to be enabled.

③ FIR Filter Finite Impulse Response Filter



The FIR Filter design only supports importing files. Different values can be set for left and right channels.The supported file types are bin and WAV formats.

Bin Format：

Supports 2046 taps, the file must be a filter coefficient file generated by the FIR filter design program, such as REW. The coefficient file must be in IEEE 754 single-precision binary floating-point format.

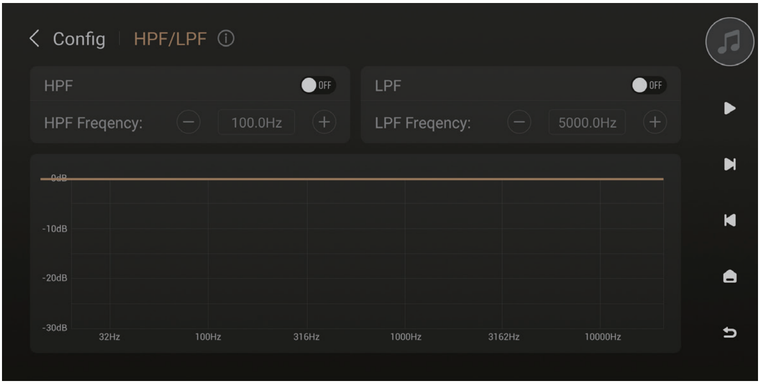
WAV Format：

Filter impulse response files in WAV format are recommended to use a 32-bit floating-point data format.

For example, filters generated using the REW tool in WAV format should adhere to this recommendation.

④ HPF/LPF

You can individually activate HPF or LPF, and you can also set the frequency.



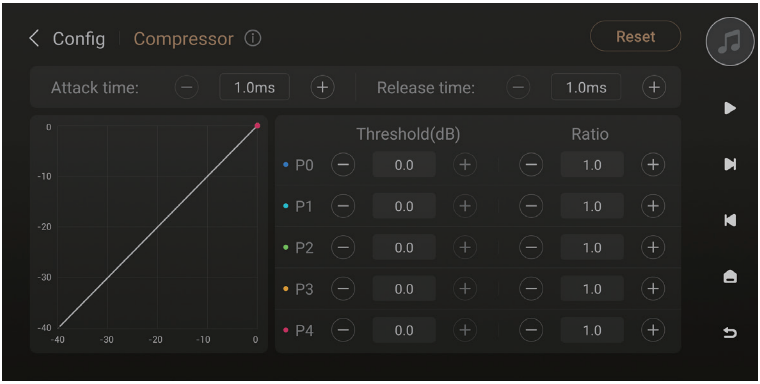
⑤ Loudness

The human ear is not very sensitive to high and low frequencies, so you can change the loudness by adjusting the gain of high and low frequencies.

 There are a total of 4 levels that you can choose from according to your needs, with "Bypass" indicating that this function is turned off.

⑥ Dynamic Compressor

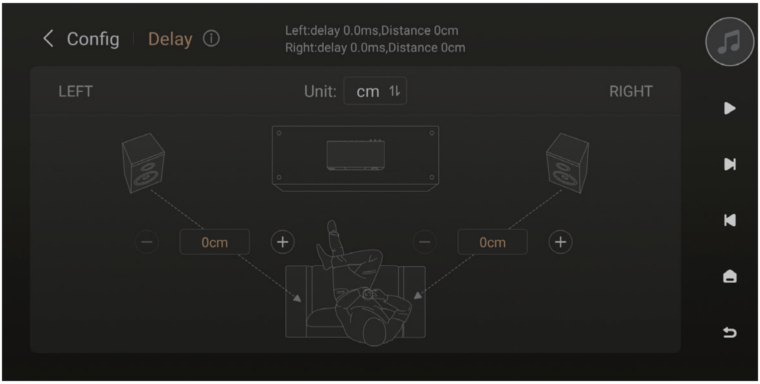
A compressor actually changes the ratio between input and output signals. For high-gain signals, if you want to reduce some of the gain, you can achieve this through the Compressor function. There are a total of 5 points available for setting. For example, in the diagram below, if you want to decrease the gain of signals above -10dB, you can set it up like this, and you can also adjust the compression ratio.



⑦ Delay

This function is for setting the delay of the left and right channels.

According to the speaker's position, set the distance correctly so that the sound from the left and right channels arrives at the ears simultaneously.



关于 DSP

DMP-A8 集成了 DSP 模块，对数字信号到 DAC 之间的信号进行数字处理。

使用 DSP 功能时，请确保你具备一定的声学与 DSP 相关知识，并且清楚的知道调整对应的参数会带来的结果。否则有可能会产生不可预期的结果。

每款 DSP 都会存在一定的限制，比如采样率、内存大小等。同样， DMP-A8 使用的 DSP 也存在着采样率的限制。

DMP A8 支持的 PCM 采样率如下：44.1K / 48K / 88.2K / 96K / 176.4K / 192K

对于大于 192K 的 PCM 和 DSD 信号， DMP-A8 会采用 Bypass 的模式传递给 DAC 。DSP 内部的工作采样率为 48Khz，打开 DSP 功能时， DSP 会将所有信号异步 (ASRC) 采样至 48Khz, 再进行信号处理。

DSP 支持的输出端口：
XLR 平衡输出 / RCA 模拟输出
DSP 支持的输入端口：
内置播放器 / 蓝牙输入 / USB B 输入 / 光纤输入 1 / 光纤输入 2 /
同轴输入 1 /同轴输入 2 / ARC 输入

DSP 的使用

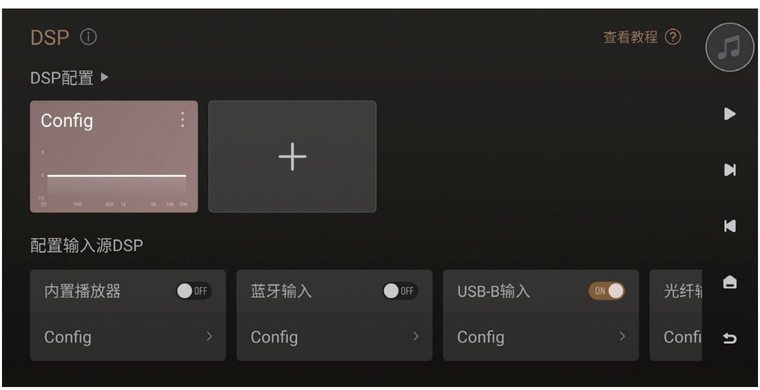
1. 打开 DSP

在主界面找到 ，点击进入。

2. 进入 DSP 配置界面

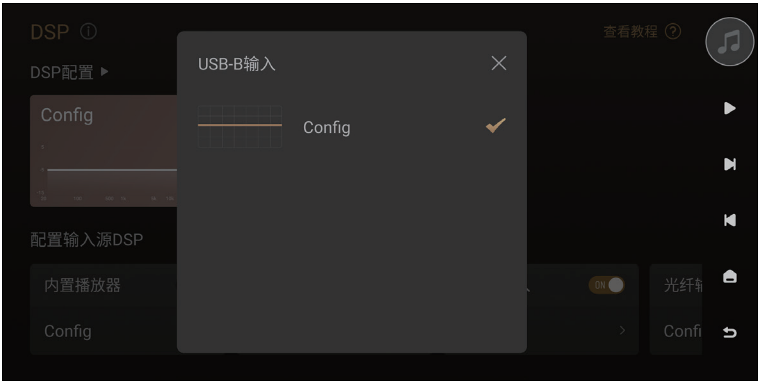
① DSP 配置

此配置是针对 DMP-A8 的 DSP 进行参数配置，进入有一组默认的 DSP 配置。同时我们也可以新增配置，并为我们新增的配置重新命名。



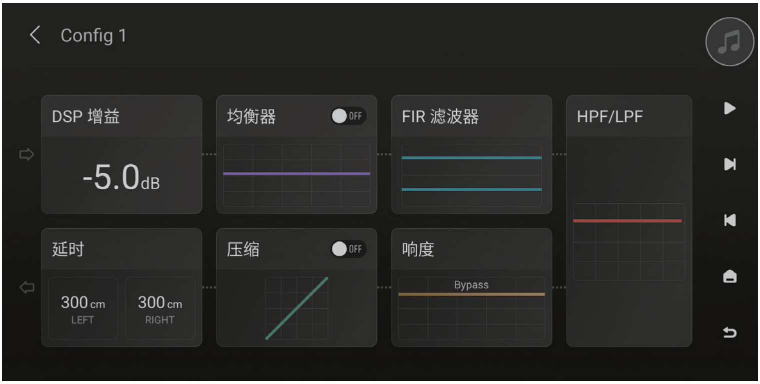
② 配置输入源 DSP

我们可以为每一个输入源选择不一样的 DSP 配置，同时我们也可以关闭指定输入源 DSP 功能。



3. 设计 DSP

选择一个自己想设计的 DSP 配置，点击进入：



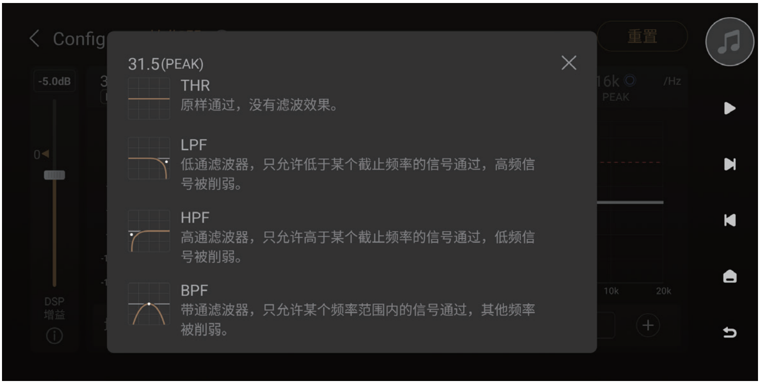
此图表示的 DSP 内部信号处理的流程，信号从 In 进入，通过每个模块处理后，最后从 Out 输出。

① DSP GAIN 增益

为了防止后面模块信号的溢出，比如 PEQ 增加正增益，一般需要预留足够的增益余量，建议设置成 -5dB。

② PEQ 参量均衡器

可以根据自己的需求设计 PEQ，同时，可以针对每个频段选择不同的滤波器：



设计完之后，保存退出。如果要使 PEQ 生效，需要在外面的开关打开。

③ FIR Filter 有限脉冲响应滤波器

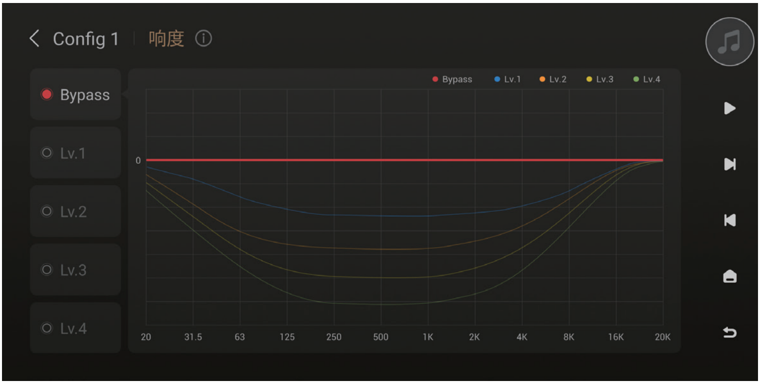
FIR Filter 只支持文件导入的方式。可以针对左右通道设计不同的值。
文件支持的类型是 bin 和 wav 格式。
Bin 格式：
支持 2046 个 tap，文件必须是 FIR 滤波器设计程序生成的滤波器系数文件，系数文件必须使用 IEEE 754 单精度二进制浮点格式。
Wav 格式：
wav 格式的滤波器脉冲响应文件，数据格式建议使用 32 位浮点。
比如使用 REW 工具 生成的 wav 格式的滤波器脉冲响应文件。

④ HPF/LPF 高 / 低通滤波器

可以单独打开 HPF 或者 LPF，同时也可以设置频率。

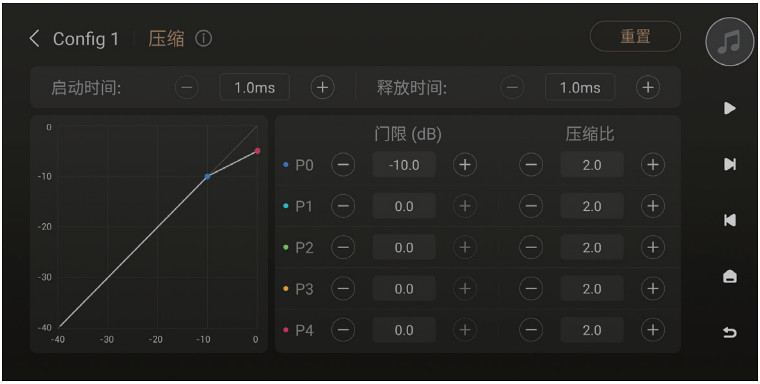
⑤ Loudness 响度

人耳对于高频和低频的听觉不是很灵敏，所以可以通过调整高频和低频的增益来改变响度。总共设置了 4 个等级，可以根据自己的需要来选择，Bypass 表示关闭此功能。



⑥ Compressor 动态压缩器

压缩器实际上改变的是输入与输出信号的比例，对于高增益的信号，如果想降低一些增益，可以通过动态压缩器功能来实现。总共提供了 5 个设置点。比如下图，如果想要 -10dB 以上的信号增益减小一些，可以这样设置，同时可以设置压缩斜率。



⑦ Delay 延迟

此项功能是设置左右声道的延时。

针对喇叭的位置，设置好距离，让左右声道的声音同时到达耳朵。