HI-Player introduction

Preface

HI-Player is a high-quality music playback software for music lovers; based on Mac/Windows operating system of x86/arm, it has the ability of playing local music files, streaming, playing, and pushing to professional audio decoders; the ability to frequency up/ down any PCM sampling rate file (including various PCM compression formats such as flac, ape, mp3, aac, etc.) to a specified PCM sampling rate or DSD sampling rate; it also has the ability to resample any DSD format (DSD64~DSD1024) to the specified DSD format or PCM format (PCM768K/705.6K)playback;

For DSD512, we use conventional equipment to play back stably and smoothly, the CPU occupation is low, so we do not need to buy expensive computing machines, which save costs and protects the environment.

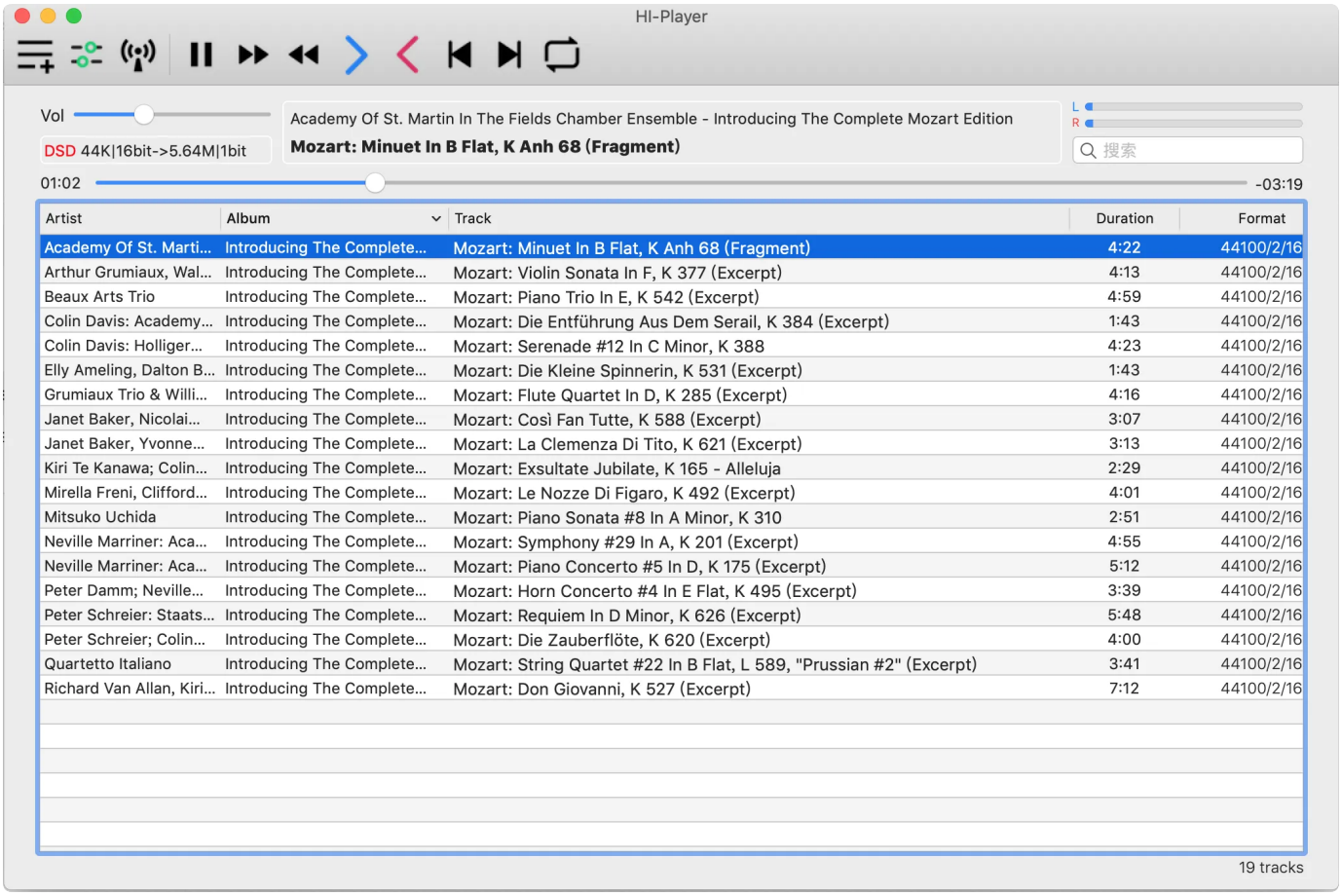
For streaming media support, HI-Player currently supports the mobile iPhone AIRPlay push-stream reception capabilities, deeply optimized AirPlay protocol stack, and supports smooth playback audio under weak networks; you can push the files on your phone to Mac/Windows devices for direct enjoyment or frequency up and forward to DAC for enjoyment;

The use of HI-Player is described below;

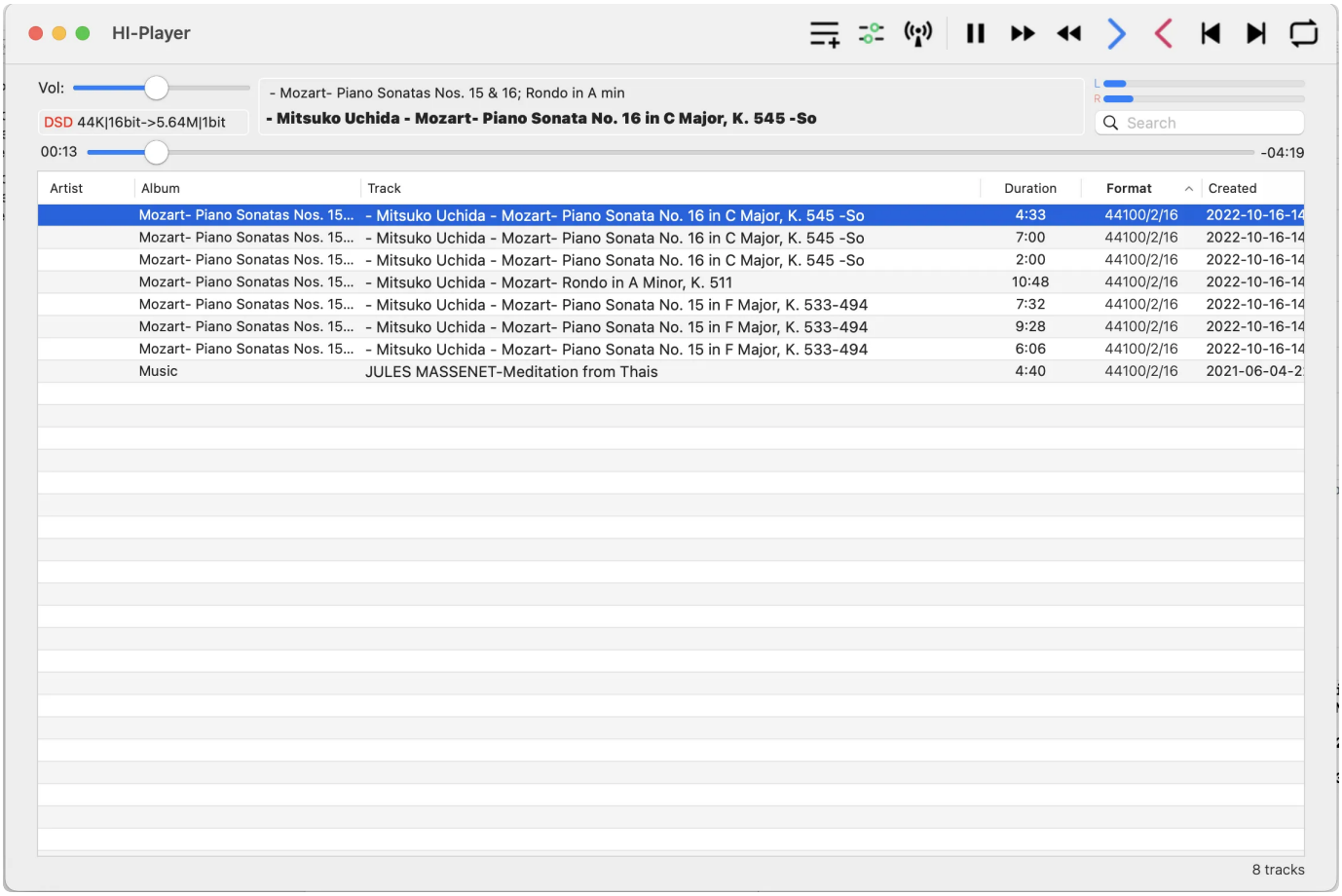
Use introduction

Main screen

mac BookPro



mac Apple M2 Pro

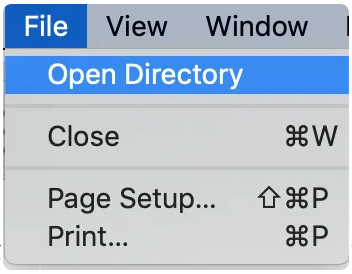


1, Add music files/ folders

1, You can directly drag music file to the file list component from the file manager, and it will be automatically added to the file list after dragging in;

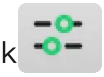
2, Click to open the file browser, select the music file directory to add, click OK, the player will automatically scan the file directory to record all files including music files in subdirectories, and will automatically add them to the file area later;

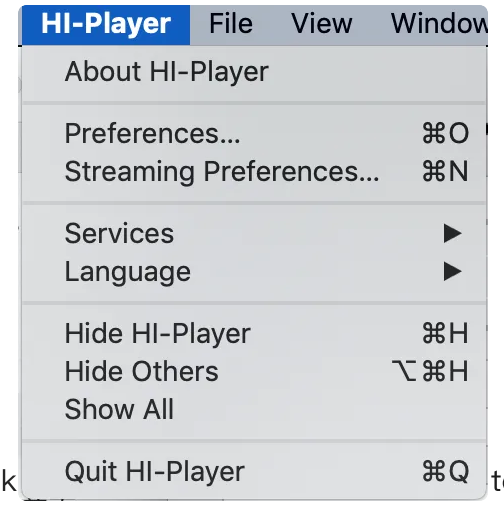
3, Select



to add music files, and click the same screen as option 2, after selecting the folder, click OK to scan the directory for adding music files;

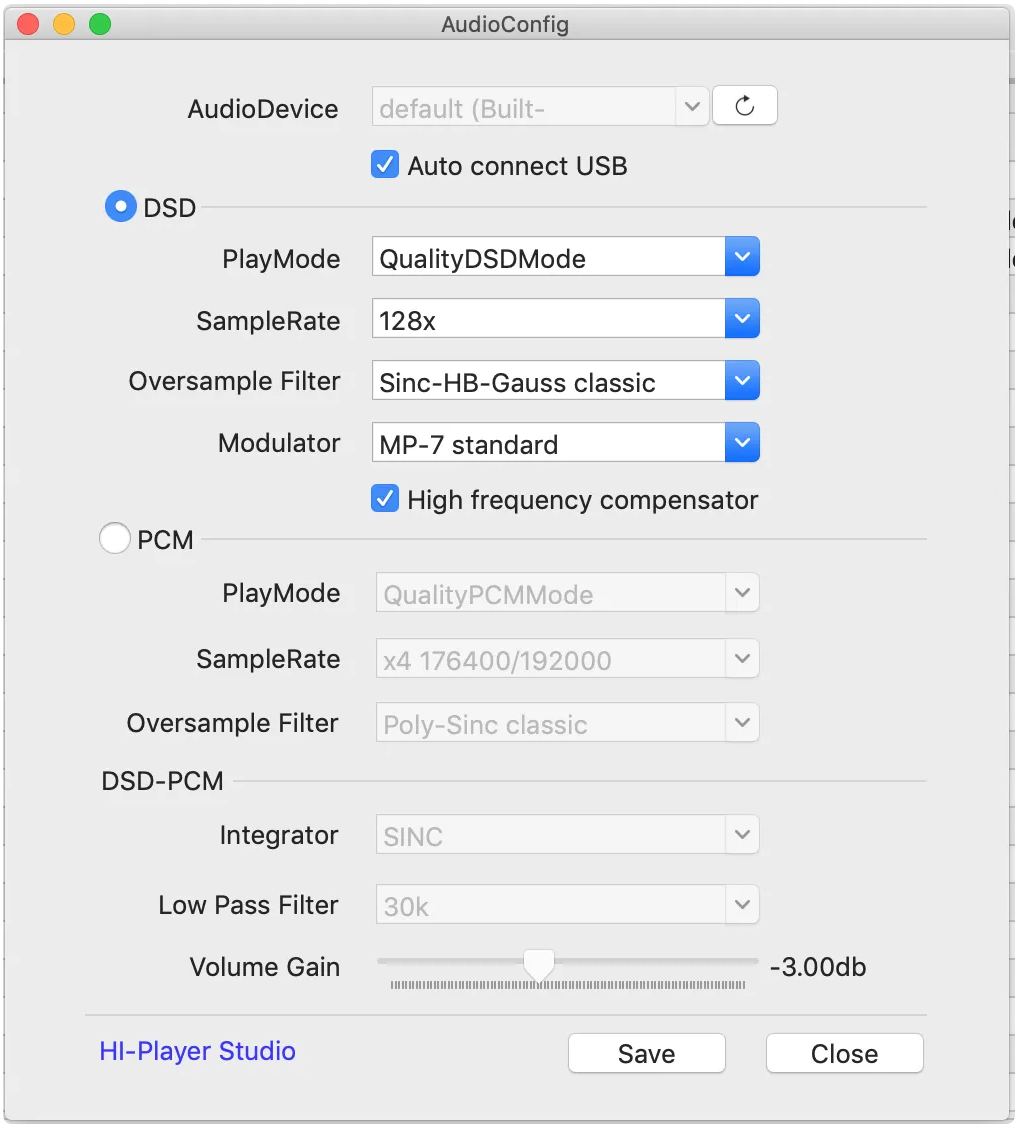
2, Set up audio device

1, Click  to open the audio settings page to set up, the following section will describe the functions of audio settings and the precautions for use.



2, Click to select the audio settings, that is, open the same screen as above 1 to set the audio;

3, Audio setting description



Automatically connect USB device

After selecting this item, HI–Player will automatically select an external USB device for playback every time the audio file is played, which avoids the problem of having to manually set up a DAC after switching between different USB DACs, it is recommended that users select this item, only turn off this item in the case of multiple external DACs. Select the DAC you want to use by selecting the combobox;

DSD mode

Write it in front of me

Mac currently **only supports dop mode,** so the DSD support range marked on the decoder must be reduced by one level to work normally on the Mac unless it is specifically stated to be DOP. For example, if a DAC says that it supports (DSD64 to DSD512), it will be (DSD64 to DSD256) in DOP mode. Requires nativeDSD mode to support DSD512,

DSD1024 must use AudioNetworkBridge mode in Mac;

Play mode

Name	Description	comment
FixedDSDMode	All audio files are output at the set sampling rate; 1. PCM file All files will be sampled to set the sampling rate and played back in DSD format. 2. DSD file Files higher the set sampling rate will be sampled to the set sampling rate output in DSD format output DAC; Files below the set sampling rate will be sampled to the set sampling rate and output in DSD format output DAC; Files equal to the the set sampling rate will be directly output to DAC;	Our DSD direct upsampling algorithm is deeply optimized for low power consumption and memory footprint; DSD up and down sampling theoretically has a minimal impact on sound quality, which is difficult to feel in daily listening, please choose AutoDSDMode if you mind;
AutoDSDMode	1. PCM file: all files will be sampled to the set sampling rate; 2. DSD file: output to DAC with DSD original sampling rate;	Due to the small loss of DSD up and down sampling rate, users with high sound quality requirements can select this option.

QualityDSDMode	1.Support 48K DSD mode, that is, the files whose original file sampling rate is 16K multiples, use 48K x mode for DSD conversion; 64x = 64x48K 128x = 128x48K 22050 multiples playback is the same as FixedDSDMode;	It mainly supports 48KDSD mode to avoid sound quality loss caused by non-integer upturn of 48K to 44100X sampling rate; 1. There are quite a few Dacs, especially some old DACs do not support 48K DSD mode, if you want to use this mode, please carefully test, if you encounter problems, please switch back to FixedDSDMode; 2. HI-Player will dynamically switch the sampling rate to support files with different sampling rates. Some DACs may switch the sampling rate slowly. In most cases, the switch can be successful within 1 second, but occasionally it takes 6 seconds or longer; if you prefer a smooth playback experience, it is recommended to use FixedDSDMode;

1，Some users' devices may only support DSD64/128; do not support high sampling rate DSD (DSD256/512 file) playback, we provide a scheme to resample any DSD sampling rate to a specified sampling rate, so that they can listen to the DSD files with high sampling rate not supported by the decoder in DSD format;

2，For those who want to try high sampling rate DSD playback, by locking the DSD sampling rate, you can placy DSD files with low sampling rate back with high sampling rate DSD to get a different HIFI experience;

Non-standard DSD sampling rate

Note

Since the standard DSD format such as DSD64 is 64 times of 44100, most of the HIRES files are 48K or higher integer multiple files. It is known that the sampling rate from 48K to 44100 integer

multiple files has a loss of sound quality, and HI-Player has made a lot of optimization, basically reduce the loss of non-integer multiples conversion to the minimum, but in order to pursue the HIFI spirit of uncompromising, we provide QualityDSDMode mode to solve this problem, for files with 16K multiples of sampling rate, we use 48K DSD mode, avoiding the sound quality loss caused by non-integer multiples upturn. Take sound quality to its limits;

Precautions

1, After our tests, quite a number of DACs do not support this sampling rate standard, especially the old DACs (such as AK4497EQ), and most of the new decoders support this sampling rate. Before using this mode, use 48K/96K/192K file to test whether your DAC has sound in QualityDSDMode, if not, please switch back to FixedDSDMode;

2, After our test, although some DDAC supports DSD48K standard, the sampling rate displayed is not accurate, and some DDAC is still displayed as DSD44100 standard, such as DSD48K*64 still displays DSD 2.8M, but the actual DSD is 3.072M/1bit. Please don't worry if you see such situation. The actual sampling rate is subject to the one displayed on the HI-Player UI, see the following figure for details;

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DSD 48K|16bit->6.14M|1bit
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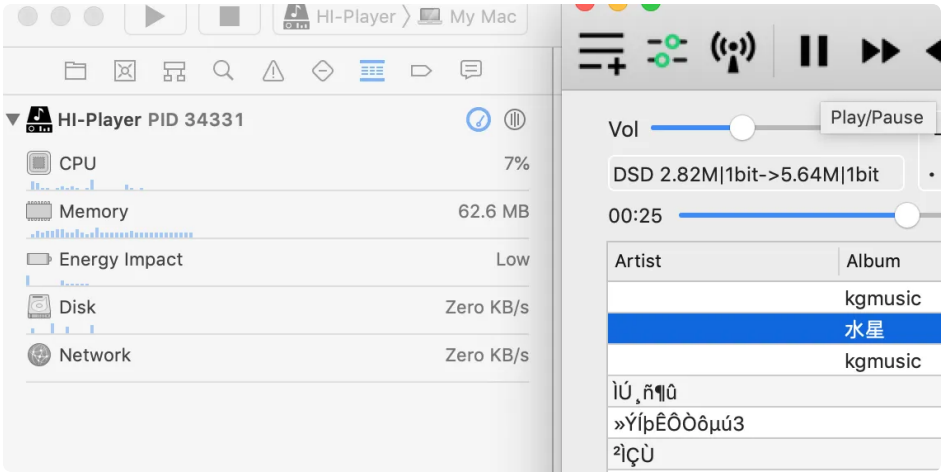
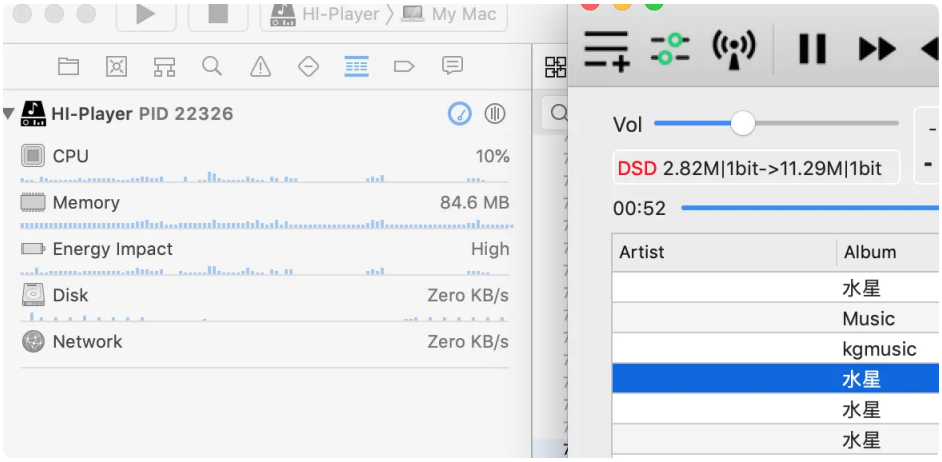
DSD frequency up and down Resample

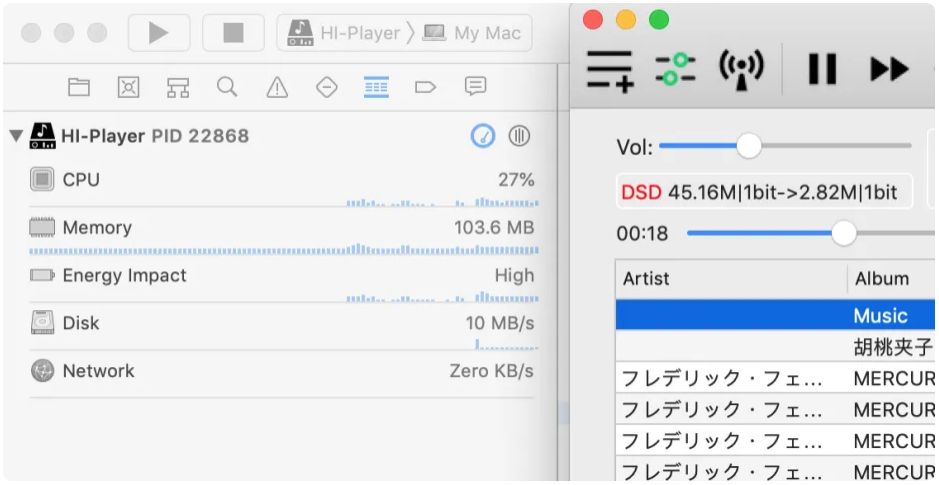
The latest version of HI-Player supports DSD direct frequency up and down playback, frequency up/ down adopts a new algorithm, with the advantages of low CPU occupation and low memory usage, DSD64 file frequency up to DSD128 only requires about 7% of the CPU, DSD64 frequency up to DSD256 only requires about 10% of the CPU (2015 MacBook Pro, see below for specific models), a similar software requires nearly 200% of the CPU on the same device, therefore HI-Player has extremely low power consumption, and there is no loss of sound quality, so you can use it with confidence;



DSD64 up sample rate to DSD128



DSD64 up sample rate to DSD256



DSD1024 down sample rate to DSD64

Interpolation filter

All filters have undergone multiple spectral comparison analyses and multiple auditions, including spectrum and hearing comparison with similar software, we recommend you to share it with your

friends, and widely collect their listening and suggestions;

The following collates the relevant listening conditions and analysis, the following filters unless otherwise specified, are 64bit filters, for a certain type of music, the details are not the more the better, it may be more atmospheric with less details, some expensive DAC decoder effective bit may only be 24bit or lower;

Name	Description	Remark
Sinc classic	Classic Sinc filter, moderate transition band, medium CPU usage, linear phase, 32bit filter;	Omnivorous, suitable for classical music;
Sinc-HB classic	Sinc-HB series are classics Sinc long filter, with steep transition band, moderate CPU usage, linear phase;	Medium to high resolution, moderate line sense, good high frequency, large sound field, suitable for classical music, fast;
Sinc-HB-Gauss classic	Sinc long filter based on Gaussian window, moderate CPU usage, linear phase, with steep transition band;	Medium resolution, slightly mushy style, strong air sense, suitable for music that needs a sense of atmosphere;
Sinc-HB super	Enhanced version of the Sinc-HB classic with steeper transition band, higher CPU usage, linear phase;	High resolution, sharper style, more detail, very good line, loud sound field, suitable for fields that require extremely high resolution;
Sinc-HB-Gauss super	Enhanced Sinc-HB-Gauss classic filter with steeper transition band, higher CPU usage, linear phase;	The resolution is medium and upper, the style is slightly mushy, the air sense is strong, suitable for music that needs a sense of atmosphere, it also has a good speed;

Poly–Sinc classic	Classic Poly Sinc filter with moderate transition band, medium and low CPU usage;	Moderate resolution, natural style, omnivorous;
Sinc–SP	Optimized overband Sinc filter, flatter transition band, medium and high CPU usage, 32–bit filter;	Moderate resolution, sweet style, suitable for women and other popular music;
Sinc–Fast	Optimized low power consumption Sinc filter, Sinc–Fast series has a relatively gentle transition band, slow roll–off, and low CPU usage, 32–bit filter;	Medium resolution, delicate and fresh style, high speed, omnivorous;
Sinc–Fast–Gauss	Gaussian version of Sinc–Fast Filter, with a smoother transition band, slower roll–off, lower CPU usage, 32–bit filter;	The resolution is medium to low, suitable for ambient music, with a good sense of speed;
Poly–Sinc sharp LP	Optimized Poly–Sinc filter, steep transition band, low CPU usage;	The resolution is medium, the style is fresh and delicate, the speed is fast;
Poly–Sinc–Gauss sharp LP	Gaussian version of Poly–Sinc sharp LP, slightly steeper transition band, comparatively low CPU usage;	Resolution is medium to low, with a nice air feel and atmosphere, with moderate sense of speed;
Poly–Sinc sharp MP	Optimized Poly Sinc filter, minimum phase filter, slightly steeper transition band, and a low latency fast roll–off filter, with lower CPU usage;	Medium resolution, natural style, good sense of speed, suitable for jazz, vocals and other slightly slow music, loose and natural;

Poly-Sinc-Gauss sharp MP	Optimized Poly Sinc filter, Gauss-based minimum phase filter, with a flatter transition band and low latency medium-speed roll-off filtering, with low CPU usage;	Low resolution, sweet style, loose and natural, a better sense of atmosphere and suitable for people sound, jazz, music that needs a sense of atmosphere;
Sinc-SP turbo	The optimized Sinc filter based on Sinc-SP has a relatively smooth transition band, medium and low speed filtering, and moderate CPU occupancy.	Moderate resolution, sweet style, large sound field,suitable for women and other popular music;
Sinc-SL MP	Optimized Sinc filter, minimum phase filter, slightly steeper transition band, and a low latency fast roll-off filter, with lower CPU usage;	Medium resolution, natural style, good sense of speed, suitable for jazz, vocals and other slightly slow music, loose and natural;
Sinc-SL MP super	Enhanced version of Sinc-SL MP, minimum phase filtering, medium to high speed chattering, medium to low CPU occupation;	High analytical power, natural style, moderate speed sense, suitable for jazz, vocals and other slow music, loose and natural;
Sinc-HB HIRES	Sinc-HB uses the new algorithm to design the Sinc-HB filter using HIRES standard, linear phase, steeper transition band, medium to high CPU occupation.	High analytical power, good high frequency, sound full atmosphere, sound field, quiet background, suitable for classical music and other music that needs high resolution;

Sinc–HB MP HIRES	The minimum phase filter based on Sinc–HB HIRES design has a high speed and steep drop, and the transition zone is steep.	High analytical power, good high frequency, full voice, suitable for pop or non–classical music;
IIR–EL classic	The nonlinear phase filter is designed based on the classical IIR structure, the structure is based on analog filter simulation, high–speed steep descent filtering;	High analytical power, a good IF, suitable for listening to pop music, voice analysis is better;
Sinc–SL MP HIRES	Minimum phase filter based on Sinc design, low slow drop, smooth transition band, low CPU occupation;	The analytical power is medium, has a better IF frequency, the voice is closer, suitable for the voice and small compilation of classical music;
Auto–Sinc classic	The most suitable Sinc Phase step/Poly filter mode is automatically selected according to the sampling rate conversion ratio, which has moderate CPU occupation and steep transition band.	The analytical power is medium and high, with good medium and high frequency, suitable for classical music and other music that needs high quality;

Auto–Sinc–Cos	The most appropriate Sinc Phase step/Poly filter mode is automatically selected according to the sampling rate conversion ratio. The optimized Cos window is used, which has moderate CPU occupation and steep transition band.	Medium analytical power, sound oily, full, suitable for dry playback equipment;
Poly–Sinc MP classic	The minimum phase version of Poly–Sinc classic,The very classic poly sinc filter;	High analytical power, sound moist, suitable for human voice, small compilation and other music;
Sinc–SL alive	Linear filter, slow slow drop, smoother transition band, less front and back ringing very low CPU usage;	The analytical power is medium, the sound is fresh, suitable for vocal pop, jazz and small band instruments;
Sinc–SL space	Linear filter, slow slow drop, smoother transition band, less front and back ringing very low CPU usage;	The analytical power is medium, the sound naturalness is high, there is a good sound field, suitable for small Musical Instruments, popular music, especially suitable for sharper playback equipment;
Sinc–SL bright	Linear filter, slow speed drop, smoother transition band, less front and back ringing very low CPU usage;	The analytical power is medium, the sound is bright and transparent, easy to listen, suitable for small Musical Instruments and popular music;

Sinc–SL nature	Linear filter, slow speed drop, smoother transition band, less front and back ringing very low CPU usage;	The analytical level is moderate, the sound is extremely natural and moist, suitable for Musical Instruments, popular music, similar to the listening sense of tape;
Sinc–ML bright HIRES	Linear filtering, medium speed drop, slightly steep transition zone, moderate front and rear ringing, medium CPU usage;	Good analytical power, sound field and high frequency extension, optimized for harmonics, suitable for HIRES audio playback, ordinary audio playback also has a good sound field;
Sinc–ML HIRES	Linear filtering, medium speed drop, slightly steep transition zone, moderate front and rear ringing, medium CPU usage;	Good analytical level, good high–frequency extension, optimized overtone, slightly warm, suitable for HIRES audio playback, ordinary audio playback also has a good sound field;
Sinc–ML–Gauss HIRES	Linear filtering, medium speed drop, slightly steep transition zone, moderate front and rear ringing, medium CPU usage;	Good analytical level, good high–frequency extension, depth optimization overtone, sound field is good; Suitable for HIRES audio playback, general audio playback also has a good sound field;

Sinc-ML space HIRES	Linear filtering, medium speed drop, slightly steep transition zone, moderate front and rear ringing, medium CPU usage;	Moderate analytical level, good air sense, optimized overtone, HIRES version of Sinc-SL space; Suitable for HIRES audio playback, general audio playback also has a good sound field;

Our recommendation

Those who pursue high detail and clear image contours can choose Sinc-HB super series interpolation filters;

Those who pursue atmosphere can choose filter with Gauss, which is more HUFI and with a sense of atmosphere;

Voice aspect: Poly-Sinc sharp MP/Poly-Sinc-Gauss sharp MP is preferred, with minimum phase filter, more natural; Sinc-SP/Sinc-Fast is also optimized for voices, sweet and natural, you can choose it;

Classical: Sinc-HB classic/super and Poly-Sinc classic series have good resolution and speed;

Those who pursue low power consumption and a light style can choose Poly-Sinc sharp series filters;

We generally choose Sinc-HB classic for our own use, which is relatively balanced, and has a good sense of image and atmosphere;

Looking to the future

We are also updating the interpolation filter, and will update synchronously to HI-Player if there are better filters in the future;

Modulator

The modulator is a noise shaping filter used for DSD conversion, HI-Player's modulators are all using 64bit modulation, the biggest problem of the modulator is the stability, HI-Player's

modulators have been tested with a large number of audio files, some of the modulator algorithms with poor stability are optimized to avoid sudden silent problems, if you encounter a sudden silent file of the modulator in use, we will continue to optimize;

Name	Description	Suggestion
LP-3 classic	For third-order modulator, LP series is a classic 1-bit modulator algorithm, with good stability and very low power consumption;	The resolution is medium and low, focus on low frequency, and witha good sense of atmosphere, slightly slow, suitable for nostalgic music;
LP-4 classic	Fourth-order modulator with medium stability and low power consumption;	Medium resolution, decent low and medium frequency, medium speed and moderate sense of atmosphere;
LP-5 classic	Fifth-order modulator with good stability and medium power consumption;	The resolution is above average, with good low and medium frequencies, moderate speed and wide adaptability;
MP-4 standard	Fourth-order modulator with moderate stability, medium power consumption, MP series is standard 1bi t modulator algorithm;	Medium resolution, good high and medium frequencies, moderate speed, suitable for fast vocal songs or medium speed music in classical music, softer;
MP-5 standard	Fifth-order modulator with medium stability and medium power consumption;	Medium resolution, good sense of speed, suitable for fast vocal songs or medium speed music in classical music, softer;
MP-7 standard	Seventh-order modulator with medium stability and high power consumption;	Medium and high resolution, with good technical indicators, suitable for music that requires high resolution, and there is high speed;

HP–4 modern	Fourth–order modulator with good stability, medium power consumption, and it is a more balanced modulator algorithm in all aspects;	Medium resolution, with good technical indicators, omnivorous, with good speed;
HP–5 modern	Fifth–order modulator with medium stability and medium power consumption;	Medium and high resolution, medium to high technical indicators, vocal pop have good results, moisturizing, it is the moduator we often use, we recommend it;
HP–7 modern	Seventh–order modulator with medium stability and medium to high power consumption;	Medium to high resolution, extremely high technical indicators, delicate and moisturizing;
FP–4 future	Fourth–order modulator with medium stability and medium and low power, it is classical low power modulator algorithm;	Medium resolution, with medium technical indicators, fresh;
FP–5 future	Fifth–order modulator with medium stability and medium power consumption;	Medium resolution, with good technical indicators, fresh;
FP–7 future	Seventh–order modulator with medium stability and medium power consumption;	Medium to high resolution, with good technical indicators, fresh;
SP–4	Fourth–order modulator with medium and low stability, medium and high power consumption, and the SP series has standard DSD modulator;	Medium resolution, medium technical indicators, good high frequency, SP series is suitable for medium and high speed sense of music;

SP–5	Fifth–order modulator with low to medium stability and medium to high power consumption;	High resolution, good technical indicators, with good high frequency;
SP–7	Seventh–order modulator with low to medium stability and medium to high power consumption;	Good resolution, with good technical indicators, focus on high frequency;
SP–8	Eighth–order modulator with low to medium stability and high power consumption;	Excellent resolution, with good technical indicators, and high frequency;
MAD–4	Fourth–order modulator with low to medium stability and high power consumption, the MAD series has medium–term DSD modulator;	With medium resolution, medium technical indicators, medium and high frequency, the MAD series is suitable for type of music with high speed sense;
MAD–5	Fifth–order modulator with low to medium stability and high power consumption;	High resolution, with good technical indicators, with good high frequency, suitable for electronic music and other music;
MAD–7	Seventh–order modulator with low to medium stability and higher power consumption;	With high resolution, good technical indicators, focus on high frequency, suitable for electronic music and other music;
MAD–8	Eighth–order modulator with low stability and high power consumption;	High resolution, with good technical indicators, with good high frequency, suitable for electronic music and other music;

Our recommendation

LP–3/4/5 is suitable for listening to slow vocals or music with a sense of atmosphere, with good low frequencies and moderate sound stage;

MP–4/5/7 series is soft, with wide adaptability, if you want to listen to music plainly you can choose it, for some decoders that are a little rushed, you will feel surprised when you choose it;

HP–4/5/7 series is suitable for listening to pop music or medium vocals, it has a good sense of moisture, a wide sound field, it is our common option;

FP–4/5/7 is suitable for listening to classical music, which is relatively silky and delicate while having a good sense of speed, moderate sound stage;

SP–4/5/7/8 和 MAD–4/5/7/8 have higher frequency and greater dynamics, suitable for electronic music, etc.; after our many optimizations, basically the stability problem of their 7/8th order filter is solved, but the 7/8th order filter may have stability problems, if you encounter problems with these two types of filters, you can contact us;

Order of the modulator

The results of our audition test show that the 5th–order modulator has moderate hearing, mid–high frequency equalization, the 7th–order modulator is relatively more delicate and soft, and the 4th–order modulation is relatively tough; users can choose the right modulator according to their preferences;

Looking to the future

We also have better modulators in the test, which will be updated to our HI–Player after test;

High frequency compensation

When we develop the simulation of interpolation filter, we find that some interpolation filters have a certain attenuation of high frequency, so we compensated the high frequency attenuated after the interpolation filter to ensure a relatively flat curve in the audible frequency band.

This option is turned on by default. If some users feel that the high frequency energy is more when using a specific DAC, they can turn off this switch to retain the attenuation of the high frequency by the filter. The high frequency will be slightly dark, but there will be a looser hearing feeling, and the cold, thin and stinging of the specific DAC, the user can switch this option to compare;

PCM mode

PCM mode is suitable for those without external DAC or external R2R DAC, and the unique sound quality of R2R is suitable for upscaling playback with PCM mode, which solves the high-frequency dark problem in NOS mode;

Interpolation filter

The interpolation filter used in PCM mode is the same as the DSD interpolation filter, and the interpolation filter in PCM mode is slightly optimized for PCM, but the difference is not large, I will not repeat it here;

Play mode

Name	Description	Suggrestion
FixedPCMMode	All audio files are output at the set sample rate; 1. PCM files: all files are up-sampled/down-sampled to the set sampling rate; 2. DSD files: all DSD files are down-sampled to the set sampling rate;	When selecting this mode, it only works under options of DSD-PCM; We offer two kinds of integrators, and we have a very small difference in auditions, usually we choose SINC;we also have no problem using FIR, it has good sound quality;
AutoPCMMode	1, PCM files: all files will be sampled to the set sampling rate; 2, DSD file: DSD original sampling rate output to DAC;	Due to the small loss of DSD to PCM, users with high sound quality requirements can select this option.

QualityPCMMode	RAW mode is the original mode, PCM plays according to the original sampling rate, DSD plays according to the original sampling rate; X1 ~ X16 play back the PCM file at the N rate of the specified 44100/48000 sampling rate; The DSD file is set to N times the sample rate of 44100 set output; The maximum sampling rate is N times of 44100/48000, N=(1,2,4,8,16) to avoid sound quality loss caused by non-proportional resampling;	1. The original format is 44100 multiples, output according to the set N value and 44100xN sampling rate; 3. The original format is 48000 multiples, output according to the set N value and 48000xN sampling rate; 3. Set N times for DSD according to 44100, then the output PCM is 44100xN sampling rate; 4. In RAW format, if less than 44100, the sampling rate system automatically selects an appropriate sampling rate for playback; other sampling rates play according to the original sampling rate;

Instructions for use

FixedPCMMode mode

Suitable for playing back audio at a fixed sampling rate, avoiding relay sound caused by frequent switching of DAC sampling rate, and the DAC remains stable throughout the process;

AutoPCMMode mode

Use a fixed sampling rate for PCM files, and DSD raw frequency playback for DSD files, so that DSD has better sound quality;

QualityPCMMode mode

RAW mode is suitable for original sampling rate playback enthusiasts, we play back audio files with original file sampling rate;

X1 is suitable for some external DACs that only support lower sampling rates, such as classic DACs of TDA1541/1543/1547;

X2 is suitable for Mac devices built-in DAC playback, it can play back any audio file with the best playback sampling rate of Mac machine;

DSD-PCM

This option is to set the filter settings for DSD to PCM, including integrators and low-pass filters;

This option only works when selecting FixedPCMMode or QualityPCMMode non-RAW mode for PCM playback mode;

Integrator

Name	Description	Remark
FIR	Linear phase integrator	The most commonly used integrator;
SINC	Linear phase integrator	Optimized integrator with lower power consumption;

The difference between the two integrators is extremely small, and the results we auditioned are more supple, the atmosphere is a little better, the FIR details are more, and the dynamics are large;

SINC is suitable for vocals, jazz and other music, FIR is suitable for symphony, classical music;

Low pass filter

Name	Description	Remark
30K	The most commonly used settings, with a nice texture;	The default settings are recommended;
40K	If you pursue more balanced sound quality;	Users who have requirements for high frequency can set it;
50K	If you pursue high frequency, you can set it;	Users who have requirements for very high frequencies can choose it;

General users choose 30K is enough, there is good sound quality, some users with high frequency requirements can choose 40K or 50K filter, the results of our audition are not much different for all types of filters;

Volume gain

For DSD, due to the modulator, the maximum volume after direct conversion PCM can only reach –6db, so some users of DSD to PCM will feel that the volume is low, so we provide an adjustment volume gain interface to adjust the volume, up to 0db;

Default setting – 3db;

Resample Frequency Status

PCM mode

`PCM 44K 16bit->176K 24bit`

DSD mode

`DSD 44K|16bit->5.64M|1bit`

Frequency up mode will be displayed on the interface with real–time update, convenient for users to view the current software working status;

Recommended settings

Device	Setting	Description
With early and medium external decoders or Portable player;	FixedDSDMode 64X/128X/256X	Since the DAC does not support the 48KDSD format, it is set to FixedDSDMode mode, you can set the maximum format supported by the device. DSD64/DSD128/DSD256 is recommended;

With new external decoders or Portable player;	QualityDSDMode 256X/512X	New decoders and small tails both support 48K DSD, so you can set it to QualityDSDMode without worry, no 48K/44100 non-integer conversion loss, the best sound quality can be achieved, if the device supports DSD512, you can get more delicate sound performance;
With decent external decoders or portable player, like DSD format playback experience, focus on the playback experience and fluency;	FixedDSDMode 128X/256X	Switching the DAC sampling rate may require waiting, which affects the playback experience. If you care about the playback flow, it is recommended to use the FixedDSDMode. Our filter is deeply optimized for non-integer ratio sample rate conversion. If you are not too concerned about the problems caused by non-integer ratio sample rate conversion, you can safely play music in this way;
With good R2R external decoders or portable player, focus on replay experience and fluency;	FixedPCMMode	R2R DAC has a special treatment for PCM, and has a softer hearing sense. switching the DAC sampling rate may require waiting, which affects the playback experience. If you care about the playback flow, it is recommended to use the FixedDSDMode. Our filter is deeply optimized for non-integer ratio sample rate conversion. If you are not too concerned about the problems caused by non-integer ratio sample rate conversion, you can safely play music in this way.

With good R2R external decoders or portable player, at the same time like DSD files have a good listening sense	AutoPCMMode	R2R DAC has a special treatment for PCM, with softer hearing sense. DSD uses DSD original format to play, to ensure the original sound quality;
With good R2R external decoders or portable player, like PCM style;	QualityPCMMode 4X/8X/16x	R2R DAC has been specially treated for PCM, which has a softer listening feeling; those who like PCM sound quality can u listen to music in this mode;
Mac BookPro/M1/M2 has no external DAC or portable player	QualityPCMMode 2X	Since the Mac Book Pro/M1/M2 built-in sound card supports up to 88200/96000 32bit, setting it to QualityPCMMode 2x can get the best sound quality;
Normal Mac Mini sound card	QualityPCMMode 1X	Since the normal Mac Mini may only support the sound card up to 44100/48000, setting it to QualityPCMMode 1X supports any file playback including DSD files, with the best compatibility;

Audio device sampling rate and bits width

About audio sampling rate

HI-Player will select the most suitable device sampling rate according to user settings and file sampling rate, it is not recommended to open other audio players during playback, to avoid mutual preemption of audio equipment caused by lag and sound quality degradation;

About audio sampling bit width

1. External decoder, we will prefer 24bit;

2. Mac built-in sound card, we will prefer 32bit;

Use Guide

Type of music	Interpolation filter	modulator	Description
General	Sinc-HB classic Sinc-HB-Gauss super Sinc-SP turbo Poly-Sinc MP classic Sinc-ML bright HIRES Sinc-ML HIRES	HP-5 modern HP-7 modern	**HB** classic filter the series is balanced, with a good texture and sense of space, suitable for most music categories; **HP** series modulators have a good texture, this combination can be suitable for most music;
pop	Sinc-SL alive Poly-Sinc sharp MP Poly-Sinc-Gauss sharp MP Sinc-SP Sinc-HB-Gauss classic Sinc-Fast Sinc-SP turbo Sinc-SL MP Sinc-SL super MP Sinc-SL nature IIR-EL classic	LP-3 classic LP-4 classic LP-5 classic FP-4 future MP-4 standard	**LP-3 classic** Smell of nostalgia; **Gauss** Slightly granular sense, highlighting the texture of the music, suitable for male voice; **MP** modulator is the minimum phase filter to listen to the human voice popular will have a more natural sense of hearing; **Sinc-SP** Suitable for female voice, slightly pointed, highlighting medium and high frequency;

Jazz/blues	Poly–Sinc sharp MP Sinc–HB–Gauss classic Sinc–Fast–Gauss Sinc–SL MP Sinc–SL super MP Sinc–HB MP HIRES IIR–EL classic Sinc–SL alive Sinc–SL MP HIRES Sinc–SL space Sinc–SL nature Sinc–ML space HIRES	MP–4 standard MP–5 standard MP–7 standard	**Gauss** filter is the minimum phase filter to listen to the human voice popular will have a more natural sense of hearing; **MP–4 standard** The low–order modulator has a granular sense, and the high–order modulator is soft. It can be used with most interpolation filters to listen to music that needs a sense of atmosphere, such as jazz.
Classical Music – Piano/violin	Sinc–HB classic Poly–Sinc classic Sinc classic Sinc–HB HIRES Sinc–SL space/alive/bright Sinc–ML HIRES Sinc–ML bright HIRES	HP–5 modern HP–7 modern MP–7 standard	**HP** modulator has good quality, good indicators, suitable for listening to small and medium–sized classical repertoire;
Classical music – Symphonies	Poly–Sinc classic Sinc–HB super Sinc–HB–Gauss super Poly–Sinc sharp LP Sinc–SP turbo Sinc–HB HIRES Sinc–ML–Gauss HIRES	HP–5 modern HP–7 modern FP–5 future FP–7 future	**HP/FP** Medium and high order modulators with super class interpolation filters are suitable for large symphony;

Electronic /ACG/New age	Sinc-HB super Sinc-HB-Gauss super Poly-Sinc-Gauss sharp MP Sinc-SL super MP Sinc-ML space HIRES	SP-7 SP-8 MAD-7 MAD-8	**super** Similar interpolation filter with **SP/MAD** high order modulator, has excellent high frequency, suitable for electroacoustic ACG and other needs in high frequency music;;

Guide to DAC filtering collocation

Most DAC can set the filter type in DAC. After a long time listening and testing, as well as spectrum analysis, the general filtering in DAC is divided into FIR and IIR and minimum phase filtering (MP) type;

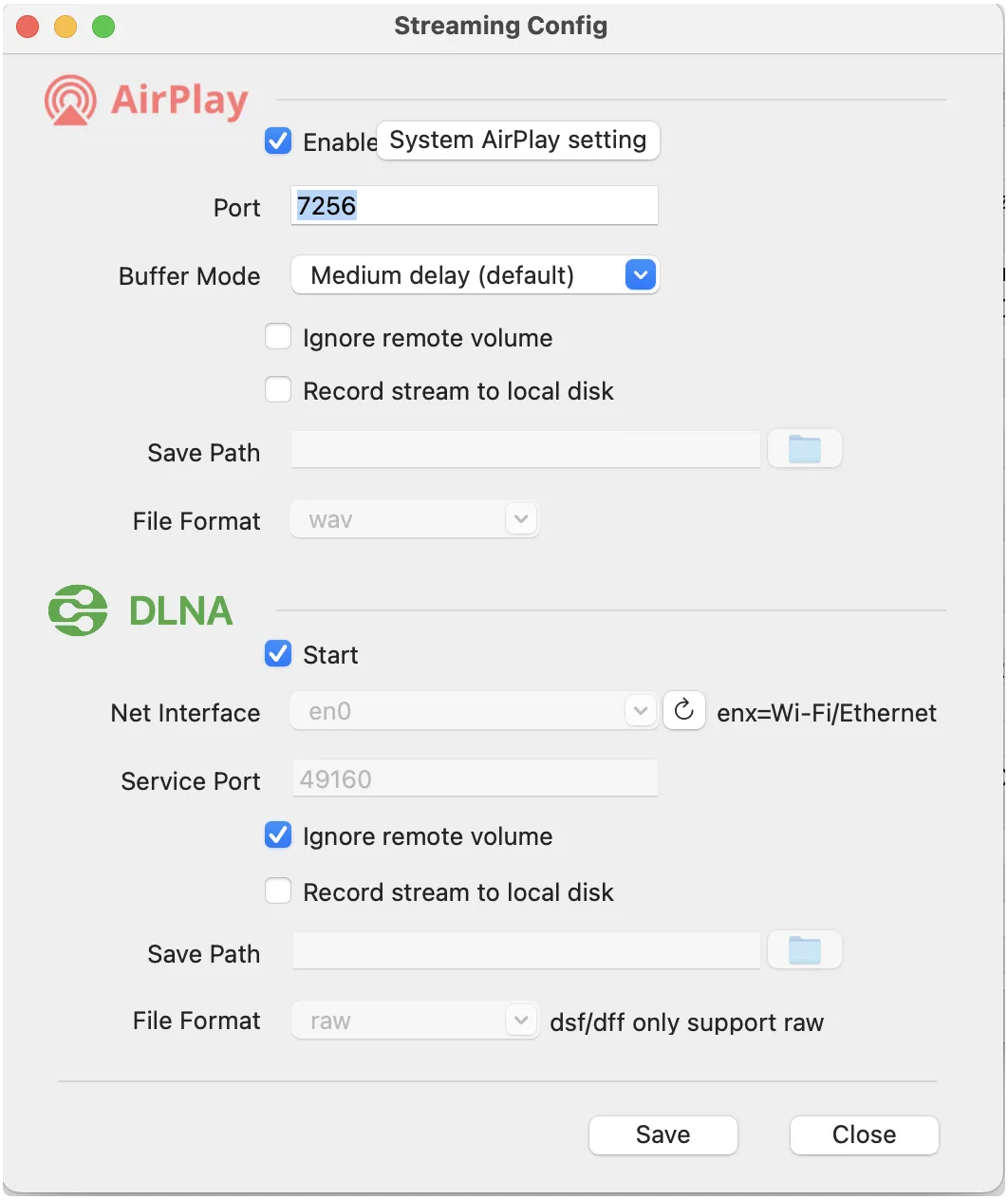
Principles of collocation:

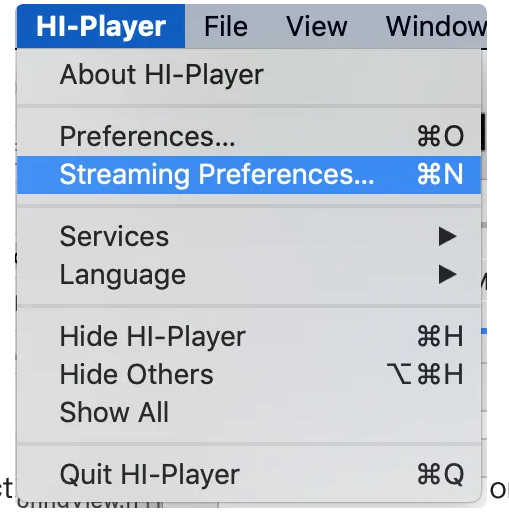
一, FIR type filtering because the linear phase can be matched with almost any DAC on-chip filtering;

二, MP/IIR type interpolation filtering is not recommended to match with DAC in-chip IIR/MP filtering. Because of multiple nonlinear phase filtering, the sound field positioning is not accurate and the hearing sense is cloudy. Our own collocation is not good, and multiple phase distortion will cause the hearing sense to decline.

三, Please consult the manufacturer's information and documentation to confirm the type of on-chip filter, so that you can choose the appropriate HI-Player interpolation filter;

4, Network settings





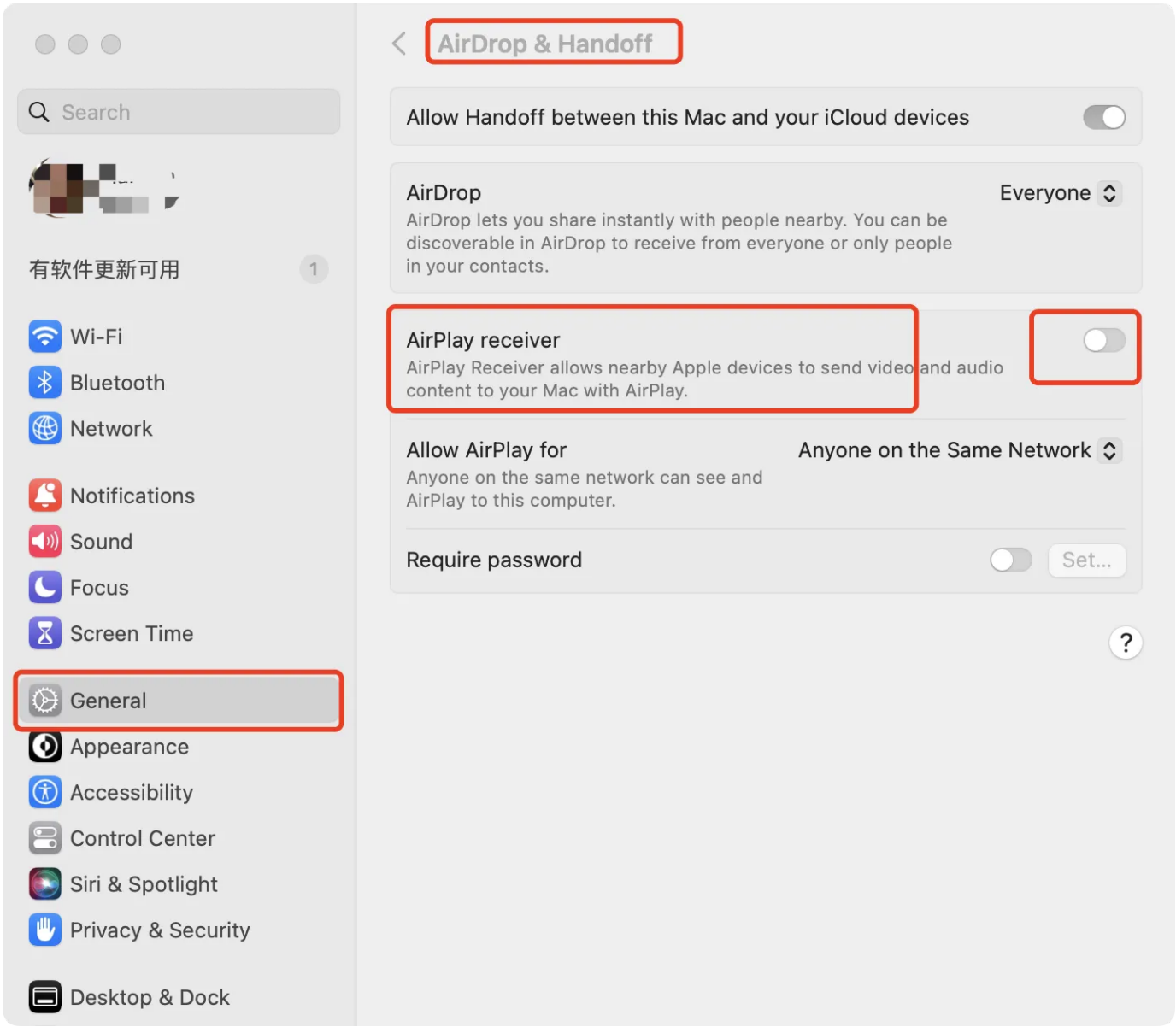
1,Select or click to open the above interface;

AirPlay

Function Description

1,Enable select to enable the AirPlay service; the port is opened for HI-Player to provide AirPlay services;

2,click`System AirPlay setting`After clicking, a pop-up window will pop up to help users with advanced Mac versions to enable AirPlay reception in HI-Player. (supported by v3.0);



3,The cache mode is the internal cache of AirPlay, if the user network situation is good, you can choose the Low delay mode, it is generally recommended to use the default setting Medium

delay mode; if the network situation is very bad, you can use the High delay mode, so that HI-Player will allocate larger cache for anti-packet loss processing to improve playback smoothness;

4,Ignore whether the remote device volume control is to synchronize the volume setting of the AirPlay push end, the user will adjust the volume setting on the AirPlay push end, if the remote volume setting is not ignored, the HI-Player will adjust the DAC volume in real time according to the user's volume setting on the AirPlay push end; HIFI audiophiles are recommended to use the volume adjustment function of the previous stage, so it is recommended to turn on this option;

4, record stream to local disk

This feature requires a subscription

一, Click the Record Audio Stream button

After selecting this button, after receiving the music pushed by AirPlay, HI-Player will save the received audio to the set path according to the set audio format, the file name is year, month, day, hour, minute and second.wav/flac format;

二, Audio stream save directory

Click on the right  You can choose to save the path, set the path, the future audio stream will be synchronized to save to the directory, at the same time has the year, month, day, hour, minute and second.

三, File Format

You can choose wav, flac format, flac format will attach tag information, AirPlay is a real-time streaming media protocol, so the sound of dragging the progress bar will not be recorded. Therefore, it is recommended not to drag the progress bar after opening the recording mode to avoid audio loss;

Notes

1, AirPlay for real-time streaming media, multiple tracks are continuous playback, so if the user is continuous listening, all the music recorded into a file, the additional log file will record the information of multiple tracks at the same time;

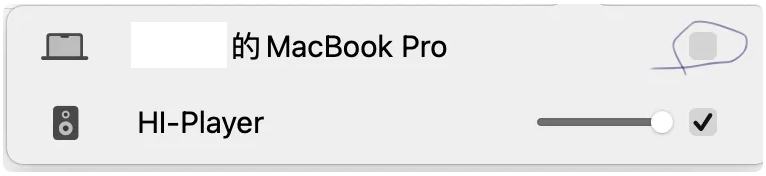
2, The latest iPhone13 or higher version of iOS devices in AirPlay streaming, if you do not touch the phone for a long time, it will enter the energy saving mode, turn off the network, resulting in serious streaming jam, it is recommended to operate the phone regularly to avoid network sleep, Mac high version also has a similar problem. If Apple music is not in the foreground or does not

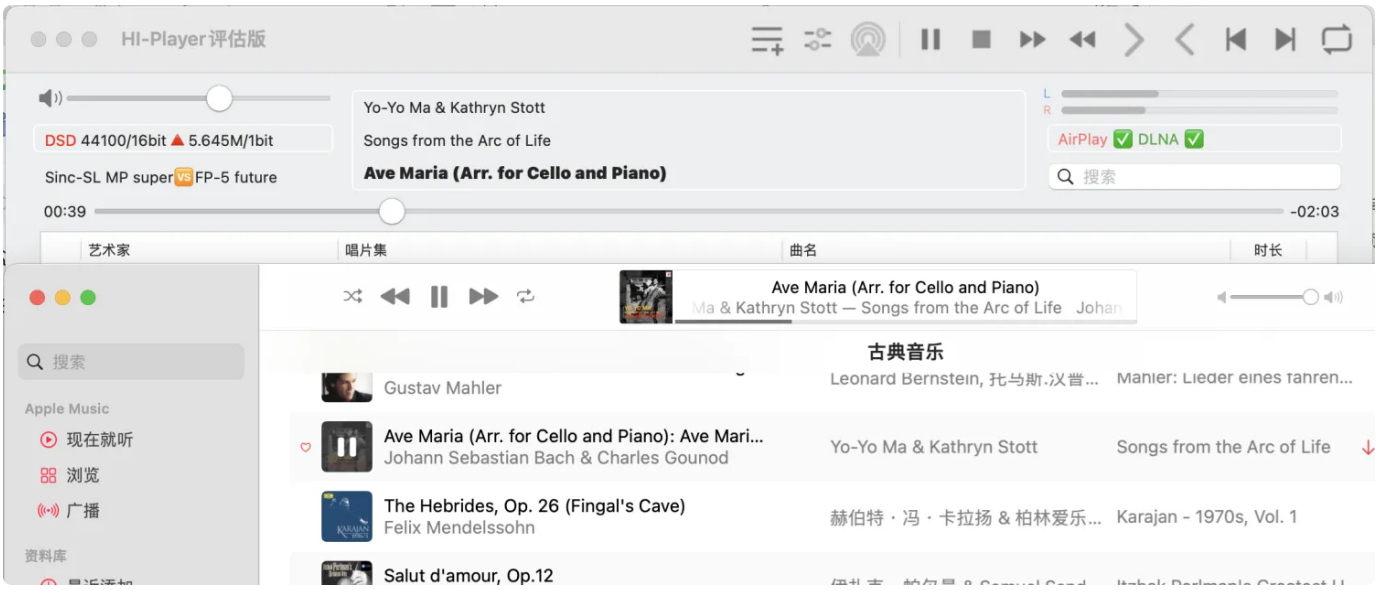
reveal the UI, it will be saved by the system. It is recommended to listen to Apple music push flow and use Apple Music to control play and music switch to avoid being saved.

3, At present, AirPlay only supports alac lossless format, I believe that everyone will enable the lossless mode of iPhone or Mac;

4, When using Mac to push stream, it is recommended to turn off apple music native playing on Mac to avoid 2 sounds and cancel the setting in the box;

5, Mac apple music push stream, the first link will be slow, there is a probability card or failure, retry can be connected, the second link is very fast, basically can ignore the link time-consuming;





Mac AirPlay push stream to HI-Player (v2.9 support);

DLNA

DLNA support file formats list

format	support	Notes
flac	support	16bit/24bit/32bit
wav	support	16bit/24bit/32bit(IEEE float / 32 bit signed int)

ape	support	16bit/24bit
mp3	support	16bit
m4a	support	16bit
dsf	support	1bit Format support, large file, need more time buffer;
dff	support	1bit/dst Format support, large file, need more time buffer;
alac	support	16bit support;

Function Description

一, after clicking the open button, you can use DLNA client to push music to HI-Player;

二, network interface, general Mac equipment, please choose en0, some users have multiple devices have multiple network cards, please choose their own network card as the network interface, after our test, some devices if you want to accept Mac native push flow, need to choose enX

[x= 0-9] to use the natively pushed DLNA stream;

三, listen to the port, use the default port, the recommended 49220 ~ 49300; It is mainly to avoid conflicts with other ports of the native machine. We found that ROON or third-party apps may occupy the same port, resulting in startup failure; We have internal error retry mechanisms;

四, The DLNA client can adjust the volume, so the interface is provided to facilitate the remote DLNA client to adjust the volume. If this option is selected, the volume control of the DLNA client will not work. On the contrary, HI-Player will adjust the volume.

五, Record stream to local disk

This feature requires subscription

1, When the button is selected, the audio file pushed by DLNA will be saved to the set directory synchronously in real time;

2, Audio stream save directory, click  You can choose to save to the directory, the recorded audio stream is saved in the directory, audio file name is year month day hour minute seconds.

3，File Format，support RAW、FLAC、WAV format；

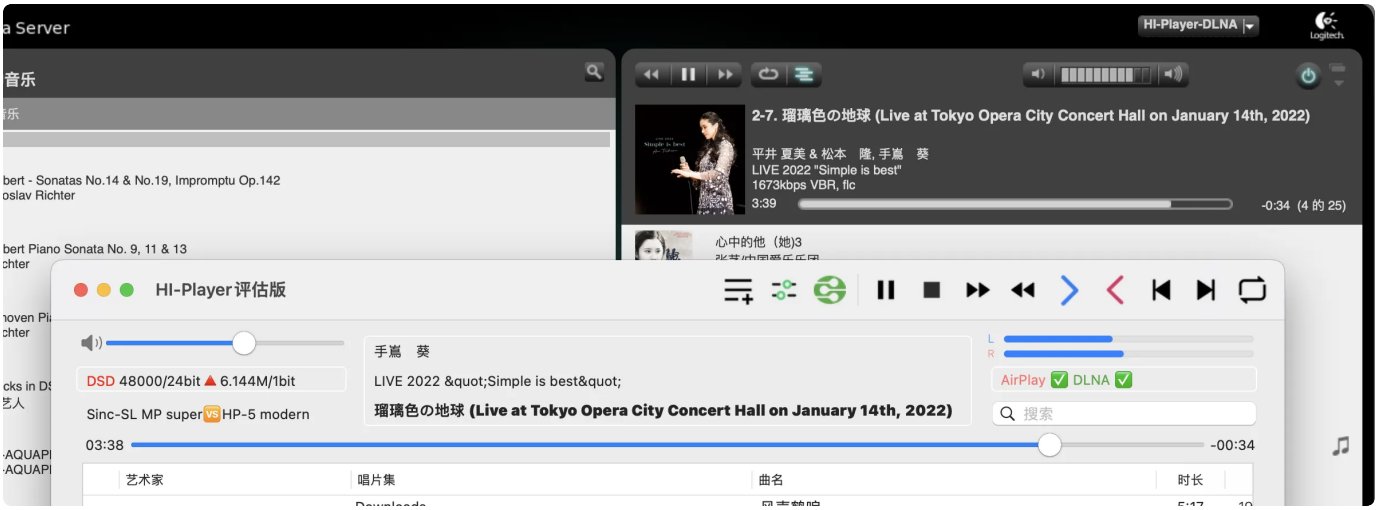
RAW format refers to the audio file format that is pushed by DLNA, without the process of decoding and recoding. It is recommended that users use it first. FLAC will use vorbis to record tag information,FLAC, WAV format will record in real time, so drag the progress bar on the UI will lose the audio that is not played, using RAW mode will not have this problem, if you record audio in other formats, it is not recommended to drag the progress bar;

Version compatibility report

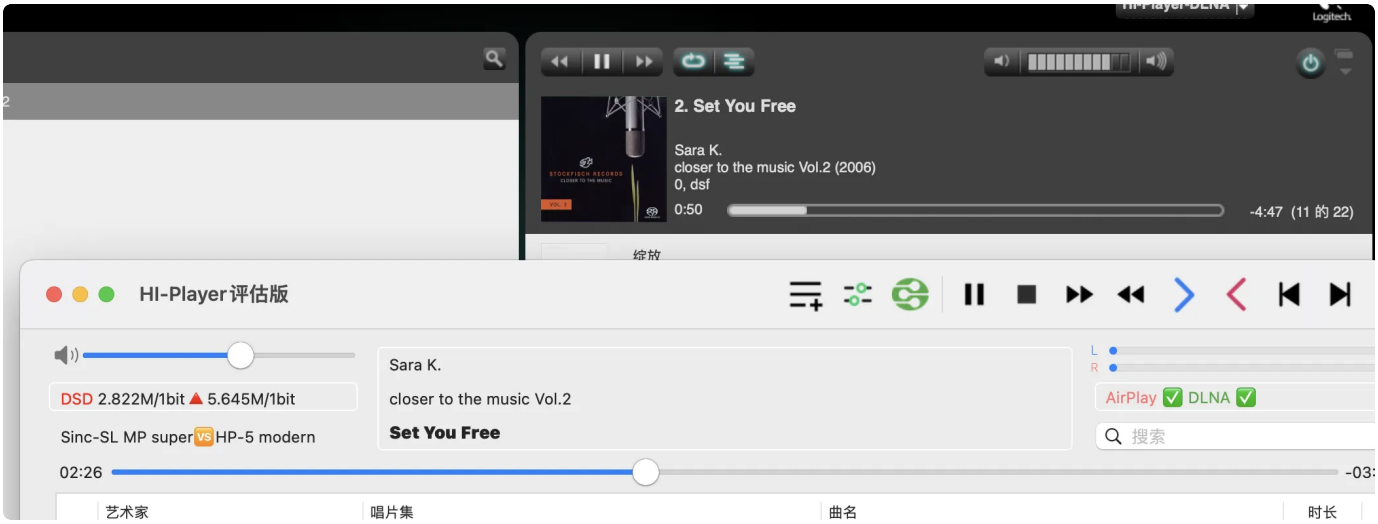
DLNA Client	support	note
BubbleUPnP	continuous playback support	support tag/DSD test pass!
Audirvana	continuous playback support	support tag/DSD test pass!
Logitech Media Server UPnP/DLNA bridge	continuous playback support	support tag/DSD test pass!
kugou	continuous playback support	support tag
Foobar2000	continuous playback support	support tag but show Foobar2000 streaming!
qq music	support，Continuous playback is not supported, you need to manually switch to the next song;	support tag
NetEase Cloud Music	support，Continuous playback is not supported, you need to manually switch to the next song;	Tag support, progress bar progress is out of sync, still in progress;
xiaomi music	support，Continuous playback is not supported, you need to manually switch to the next song;	support tag

LMS（Logitech Media Server ）Note

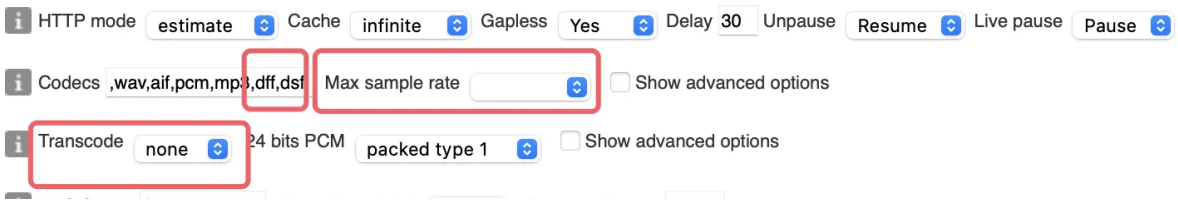
push pcm file



push dsd files



option note



push DSD file

1，need config Codecs add ,dff,dsf,

2，need set null to Max sample rate;

3，If you want the best sound quality, you can set Transcode为none；

Notes

1, AirPlay currently supports iPhone/iPad device push, Mac device push compatibility is still in the process;

2, our AirPlay2 protocol stack is deeply optimized for weak networks, which can resist about 20% of network packet loss and has high stability.

3, we tested the mainstream DLNA client software, BubbleUPnP Audirvana/kugou have good sound quality, running smoothly, recommended our users;

4, we support DSD audio streaming (dsf/dff), but dsf/dff audio files are large and need more time buffer, so when pushing dsd files, you need to wait for more time buffer, so there will be a higher sound quality, all the expectations are worth it;

5, our DLNA supports dragging the progress bar at any position during playback in HI-Player and DLNA client. For lossless format, there will be a delay, especially for DSD format files, we suggest you drag the progress bar carefully during playback;

Function description

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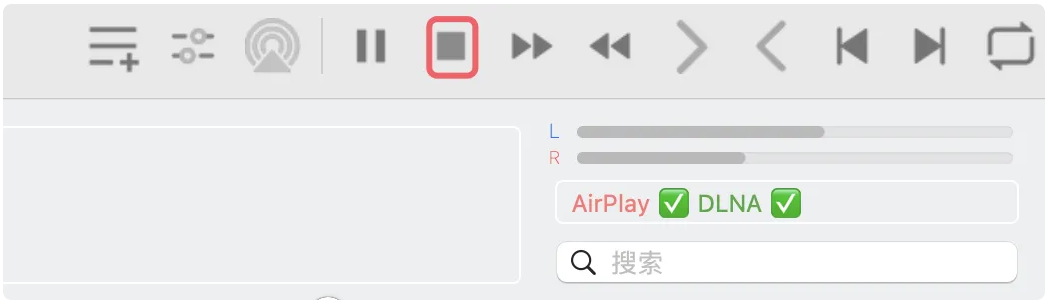
5, our DLNA supports dragging the progress bar at any position during playback in HI-Player and DLNA client. For lossless format, there will be a delay, especially for DSD format files, we suggest you drag the progress bar carefully during playback;

6，As for play/pause, HI-Player only does one-way state synchronization at present. Operations on the HI-Player interface can be stopped and played, but the push terminal is not notified in reverse. Operations on the operation terminal can be notified to the HI-Player terminal, and the

HI-Player interface will also be synchronized. This feature will be supported in the new version, it is recommended that you control play/pause on the push side;

7，Because DLNA/AirPlay status interaction is more complex, the status control of different versions of DLNA/AirPlay is different, HI-Player does not guarantee that it works normally under various circumstances, what problems do you encounter in use, can click



Stop playing, HI-Player will restart the audio system and AirPlay and DLNA service after the button is clicked; All state is restored.

5，Play

Supported file formats

Format type	Description	Support or not
flac	16/24/32bit PCM lossless compression format	Support vorbis tags support
ape	16/24/32bit PCM lossless compression format	Support ID3 tags support
dff	1bit DSD lossless format/dst encoded format (2/4/5/6 channels support)	Support ID3 tags support
dsf	1bit DSD uncompressed format	Support ID3 tags support
cue	Description file(flac/ape/wav support)	Support

ISO	1bit DSD dst encoded/uncompressed format(2/4/5/6 channels support)	only support DSD64
mp3	16bit PCM compression format	Support ID3 tags support
wav	16bit/24bit/32bit lossless format	Support ID3 tags support
aac/m4a	16bit compression format	Support
alac	16bit/24bit lossless compression format	Support
other format	For other formats, we will call the system API interface to obtain PCM data playback, if the default Mac system player can play the file, we can support it;	May support

CUE file

1. Due to the permission restriction of apple's sandbox, the selected file should be dragged into the UI interface operation, and the cue and the corresponding audio file should be dragged at the same time, so that the cue support can be used; Dragging the cue alone to the interface to open is invalid, because the app does not have access to the audio file at this time;

2, cue has the corresponding audio file type, but many of the cue files we collected only write wav, which corresponds to ape or flac files in fact, so we do the optimization, we do not make the audio format distinction, theoretically as long as there is a cue file, any format we can split track playback;

3. The file character encoding of cue file is complex, including GBK, UTF8, BIG5 and other formats. We have automatically identified the file, but there is still a possibility of garbled code. If you encounter this kind of problem, you can send the file to us, we will try to solve it;

DFF file

dff files are available in multiple formats. We support both uncompressed and dts compressed formats as well as (2/4/6/8 channels);

ISO file

1. We only support DSD64 ISO files, all the DSD64 iso files we can find on the market;

2, some ISO files are compressed format or some blocks are compressed format, we support both;

3, we support 2/4/6 channels or higher iso files for dual-channel playback;

4. If there are multiple channel format files in the ISO file, we will list the tracks at the same time for everyone to enjoy;

Supported sampling rate

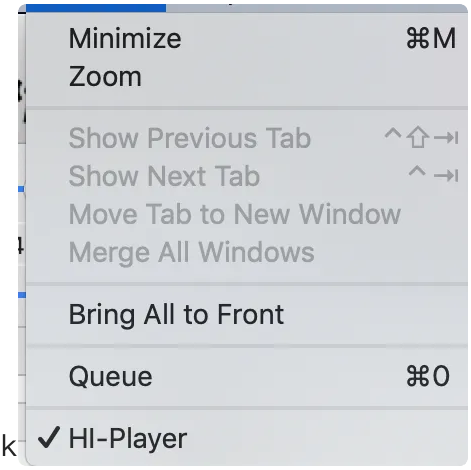
Formart type	Supported	Description
PCM	22050～705600 hz 16000～76800 hz	It supports up to 44100x16/48000x16, only supports 16000X/22050X times the sampling rate;
DSD	resample support: DSD64~DSD512 @ 44100 file format support: DSD64~DSD1024@44100 resample support: DSD64~DSD512 @ 48000	PCM frequency up supports DSD 64~512 44100X and 48000X standards; DSD file support DSD64~DSD1024 @44100; DSD files do not support 48000X standard for the time being;

Now the mainstream DSD files are only 44100X format, no 48000X format, so our DSD file playback only supports 44100X format;

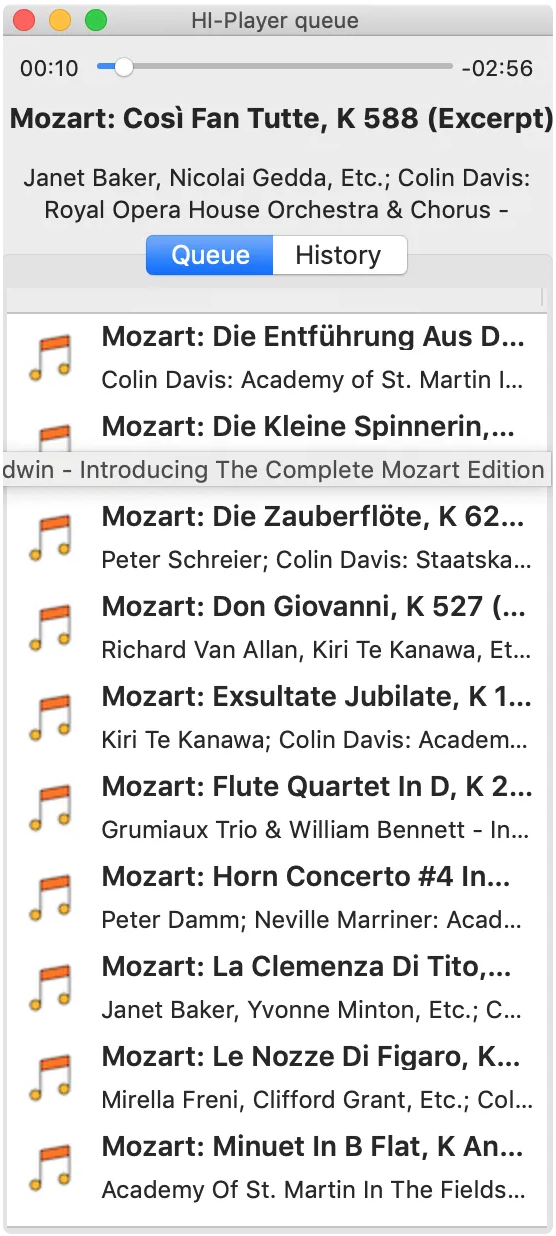
Supported sound track

HI–Player only supports 1/2 channel, and does not support audio files higher than 2 channels for the time being;

Play queue

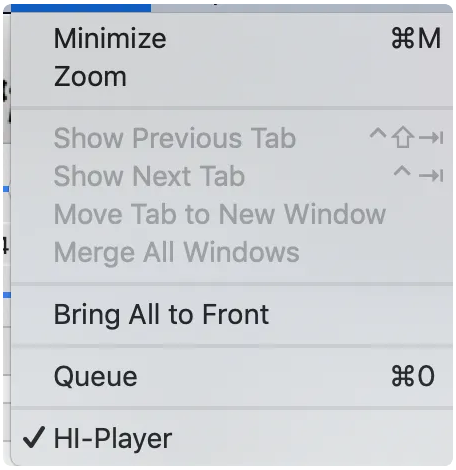
Click 

Queue See play queue window;



Play/ pause

1, Double-click/ or click  on the file in the Chinese list to play the file immediately, if a file has been played before, it will immediately stop playing the currently selected file;



2，in select the playback queue, in the pop-up window you can see the list of files in the current queue, Because it takes time to load the queue file, the playback queue is updated every time the file is selected, so it takes a few seconds to load after updating the playback file;

3，In the middle of playback, you can click at any time to pause playback, after pausing playback, the icon will be switched to , and you can click again to continue playback;

Stop

Click Stop playing, HI-Player will close the playing, stop the occupation of the audio device; In the case of other players open at the same time, it is recommended to click the button;

Notes

Because HI-Player has exclusive mode for audio equipment, it will detect the change of the sampling rate of the sound card, and if the sampling rate is not set, it will restart the sound card to avoid playing sound distortion. Therefore, opening multiple audio player software at the same time may affect each other in the playing state. It is recommended to click this button to stop HI-Player from occupying the device;

Fast forward

Click it can fast forward, about 10% of the overall progress of the file, because HI-Player in order to ensure a high sound quality, the cache time is slightly longer, about 1 second delay;

Rewind

Click  for fast forward, the fast forward ratio is about 10% of the overall progress of the file, because HI-Player in order to ensure a higher sound quality, the cache time is slightly longer, about 1 second delay;

Next

Click  to play the next song, which can be seen by viewing the playback queue;

Previous

Click  to play the next song, which can be seen by viewing the playback queue;

To the top

Click  to the head of the currently playing file, which is equivalent to starting playback from head;

To the end

click  to the end of the current playback file, which is equivalent to playing the next song;

Play mode

HI-Player supports two file playing modes, sequential playing and random playing.

 Is the state of the two playback modes and the toggle button;

一, Sequential playback mode

The default is sequential play mode, play to the last concert automatically back to the first song to play, click  Will switch to sequential playback mode.

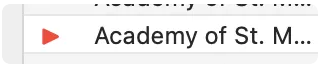
二，random play mode

Each time a song is played, the next song is randomly selected and clicked  Switch to random play mode

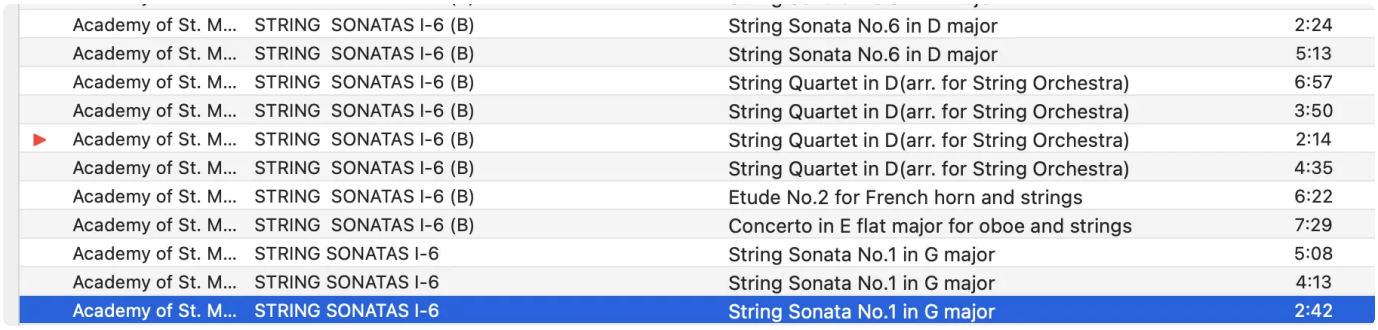
Notes:

Only when the music has finished playing in the file queue will it switch to random play mode.

Play status

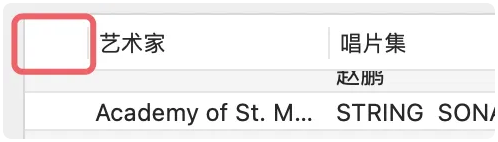
1，The currently playing music will be displayed in the file list 

The list of files is shown as follows:



2，When paused, the icon will also switch to the paused state

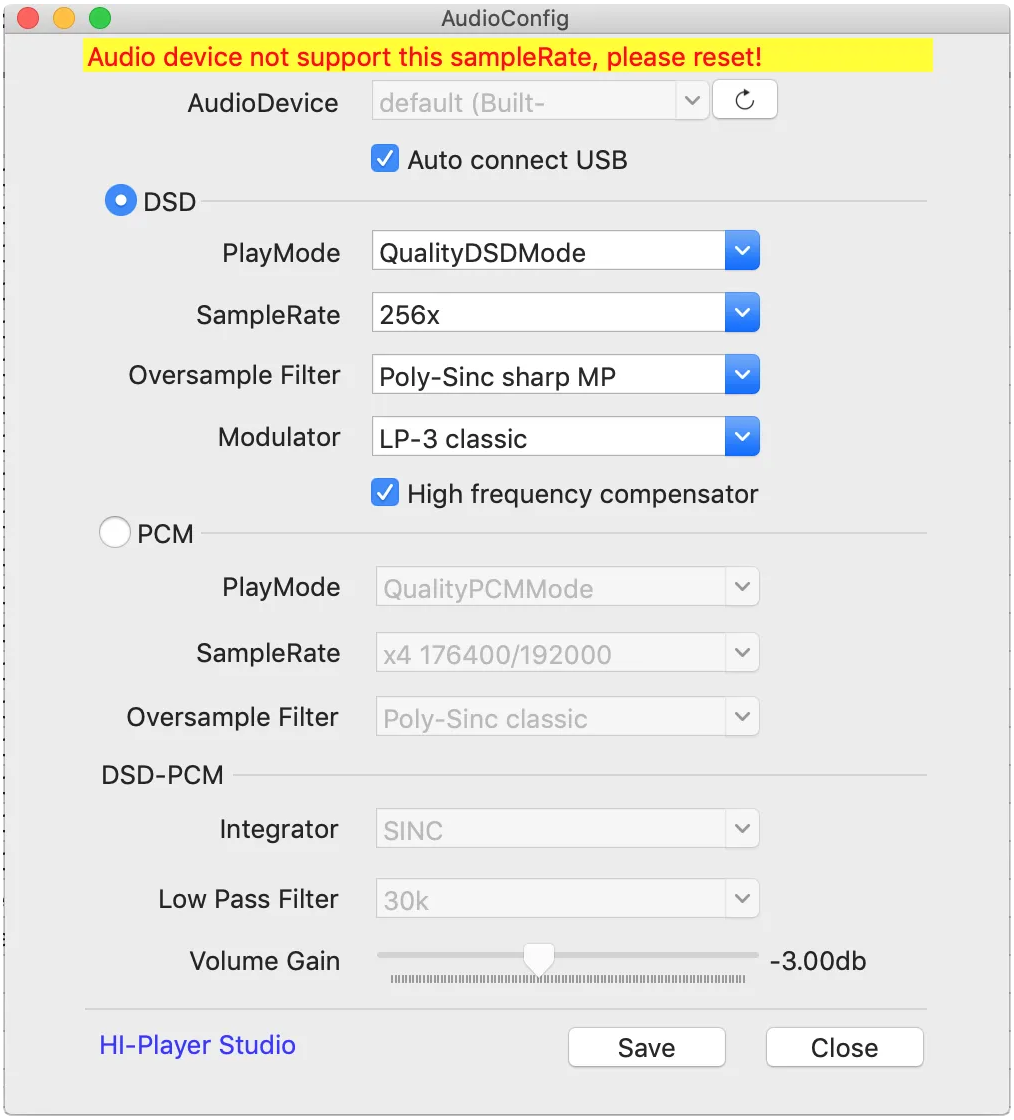
3，If you want to quickly find what music is currently playing, you can click



Middle red box, so HI-Player will automatically scroll to the currently playing music；

Anomalies

If the format of the audio setting does not support the current DAC, the setting interface will pop up, and the user shall reset the audio settings to ensure normal playback;



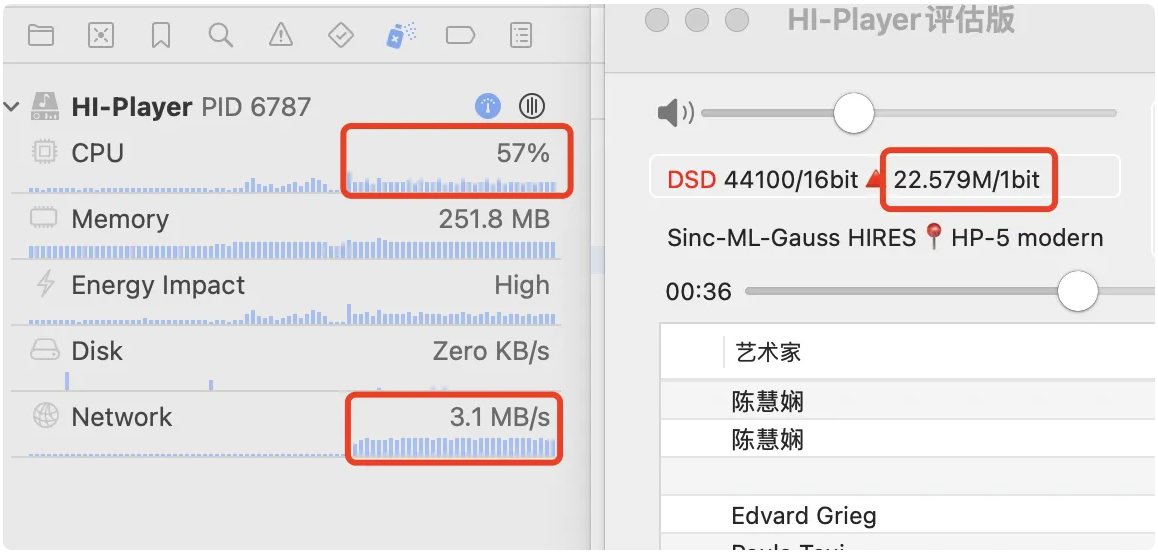
6, AudioNetworkBridge

The Denafrips Terminator plus 12st test was used for all tests

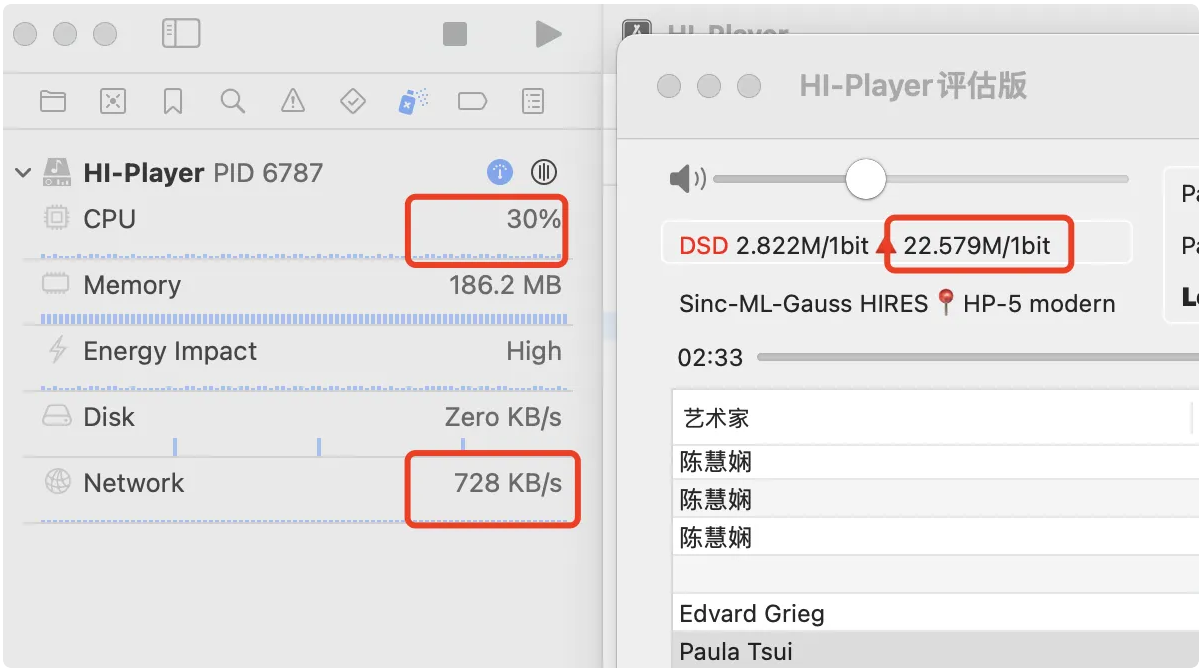
DSD64~DSD256, using conventional 100M network can run stably, here is not screenshot;

DSD512

PCM UpResample to DSD512

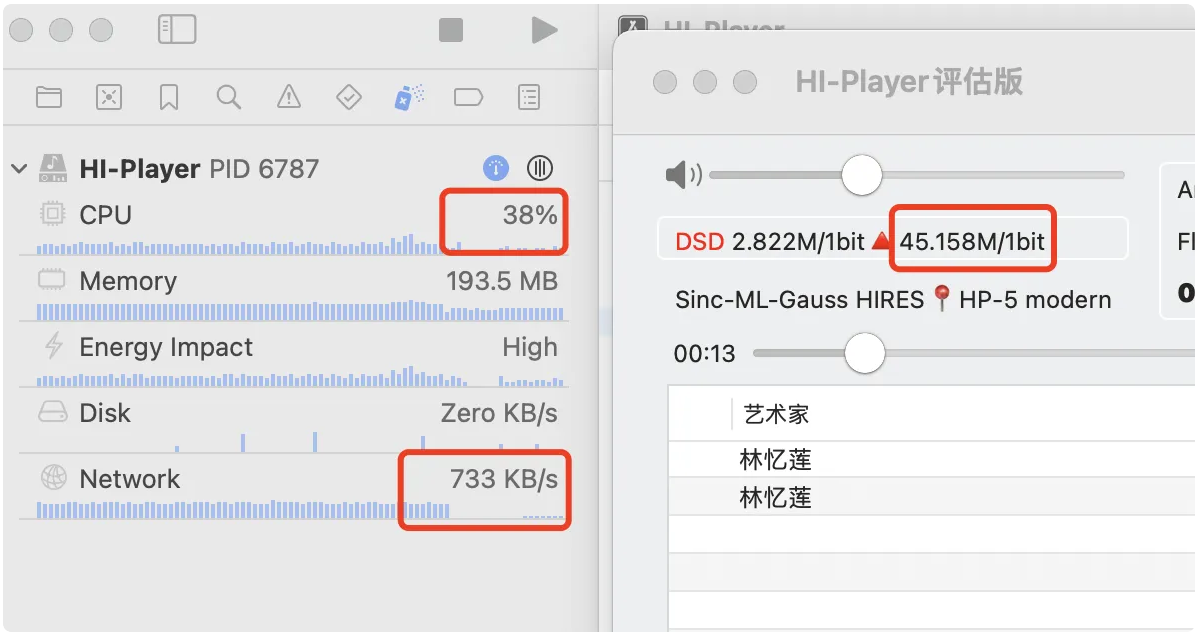


DSD64 UpResample to DSD512



DSD1024

DSD64 UpResample to DSD1024



HI–Player adopts DSD compression algorithm and can support DSD1024 with a network bridge at a very low bandwidth occupation.

PCM upscaling requires a high bandwidth, and our WI–FI is automatically throlimited after 4M, so it cannot actually operate continuously;

Installing the network bridge

Currently, the first version of the bridge supports the official Raspberry PI 4B os and Ubuntu24 version;

Executable program in

https://github.com/sunhuino1/HI–Player/tree/main/audioNetworkBridge

Method 1

1, SSH the audioNetworkBridge to any directory of the Raspberry PI,

Run chmod a+x. /audioNetworkBridge in your terminal

3, after execution, execute./audioNetworkBridge to run the bridge

Method 2

1, SSH the audioNetworkBridge to any directory of the Raspberry PI,

2,In the terminal, execute chmod a+x ./startANB.sh

3.The terminal executes chmod a+x ./stopANB.sh

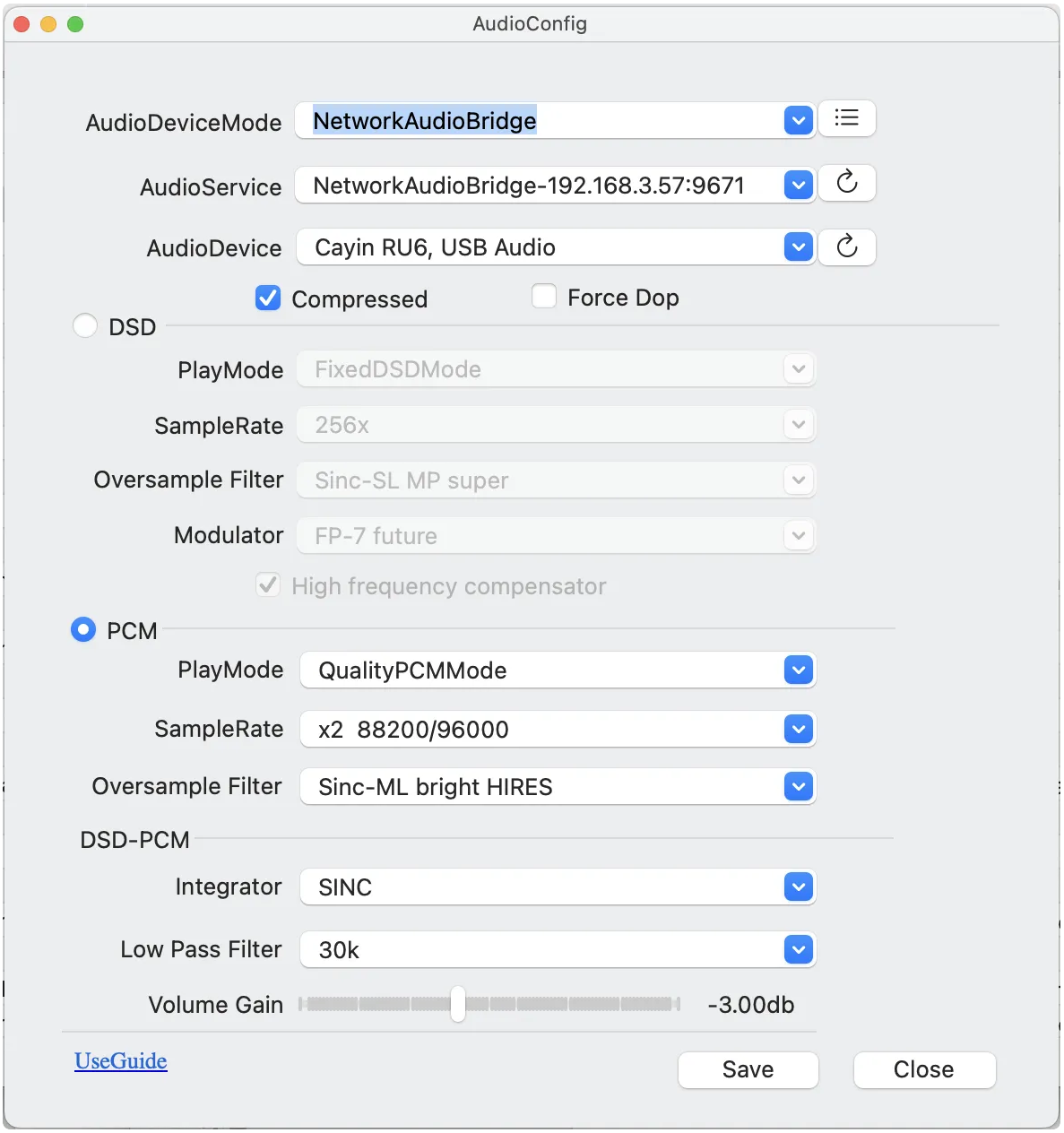
4,The terminal executes chmod a+x ./audioNetworkBridge in the terminal

5, start to execute ./startANB.sh &

if not with & after the return, the program automatically exits, with & can reside in the background to run

6, close the execution ./stopANB.sh, type y below to close

Option



Compressed

After clicking, open the bridge compression data function, HI–Player will use lossless compression method to compress data, reduce bandwidth occupation, PCM compression rate is about 80%, DSD. Format about 65 ~ 75%;

Force Dop

We found that some Dacs claim to support nativeDSD, but they are actually silent, and they need to manually switch to DOP mode to have sound. HI–Player is nativeDSD and DOP mode adaptive,

if you encounter this situation, click Force DOP to make the bridge directly use DOP mode to avoid silence.

Suggestions

1. Raspberry PI is recommended to use an Internet cable connection, but WI-FI is also acceptable if high-bandwidth WI-FI is available;

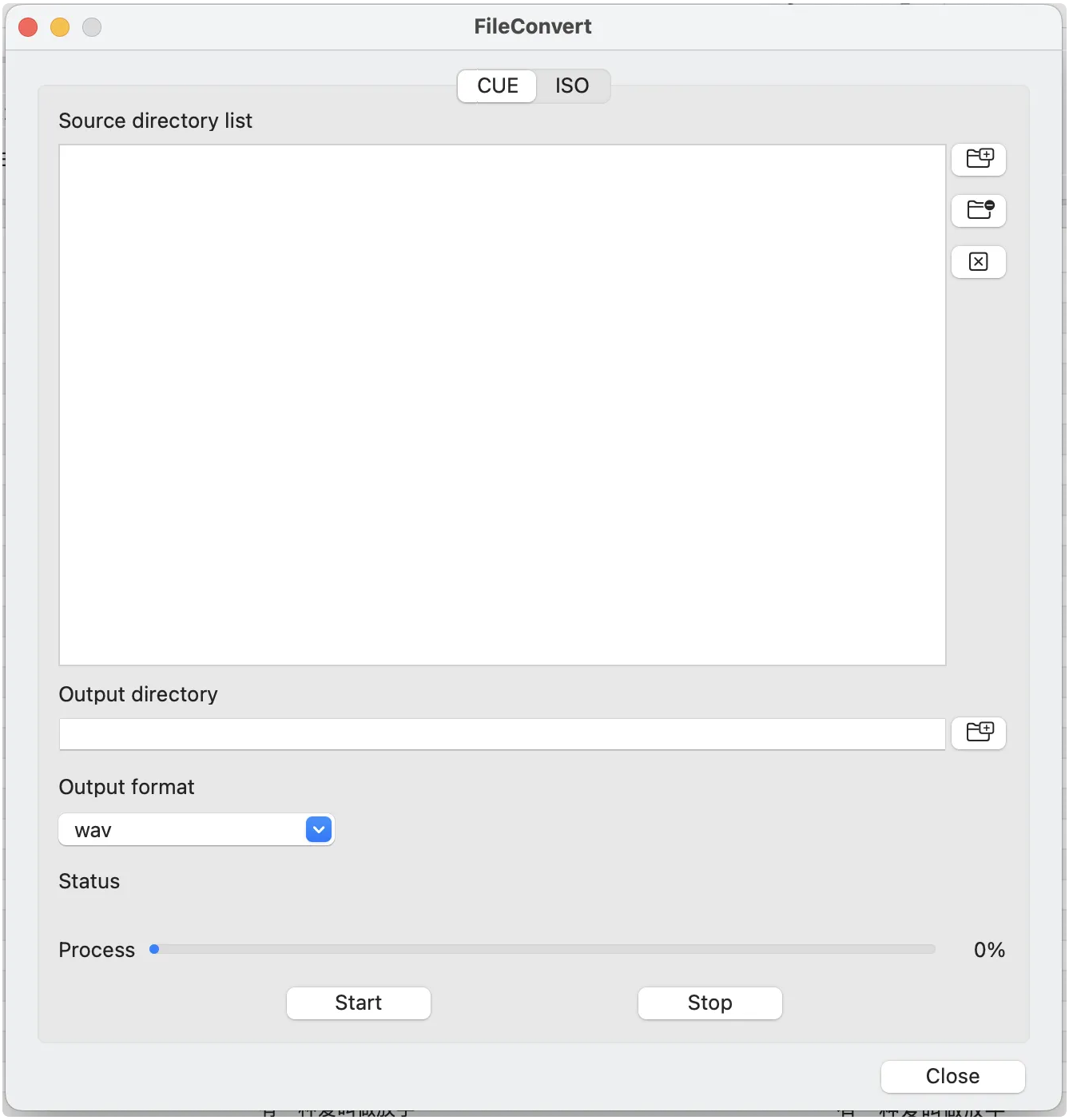
2. If the Mac side uses WI-FI, it is recommended to use high-performance WI-FI devices that support WiFi6/5, and the DSD upgrade frequency should not exceed DSD512; DSD file frequency upgrade can use DSD1024;

3, Mac terminal use network cable, theoretically will be more stable, it is recommended to use 1000M network; HI-Player has special compression for DSD, and it can also run stably on 100M network, but it will affect the network communication of other devices in the network, so it is recommended to use 1000M network.

4. Since Mac default Settings will automatically disconnect the network after the display is turned off (power adapter is recommended), it is recommended to use the option shown below to avoid network disconnection after the display is turned off and normal communication cannot be performed;

5.The current HI-Player version has a BUG. After the Raspberry PI decoder is updated, it is recommended to close the device Settings interface before re-entering. The 3.9 version will fix this BUG.

7, File Convert /Split track



Instructions

1, the cue corresponding wav/ape/flac file is divided into single files according to the cue track information, and the id3v2 information is automatically added to the file after the track according to the cue file information and the cover file is automatically copied;

2, the ISO file is divided into dsf/dff files according to the built-in ISO track information, and the id3v2 information is automatically added to the file after the track is divided according to the ISO file information and the cover file is automatically copied;

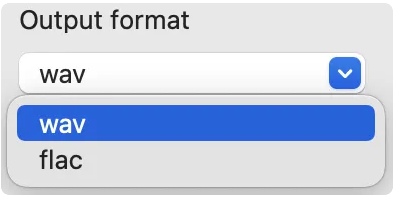
The program did limit the maximum number of tracks of cue/ISO single file to 99;

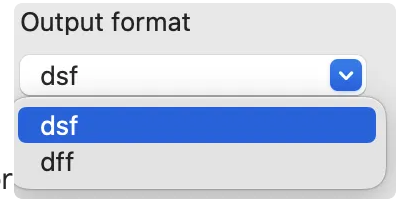
Instructions for use

CUE/ISO track functions are basically the same;

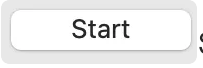
1, Click next to the source directory list  Add a directory containing cue/ISO files, The added file directories will be added to the folder list. If you want to delete some directories, you can select them first, 然后点击  Then click, Or click  Clear all directories in the source directory;

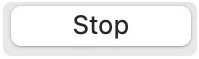
2, Click next to the destination directory  , Set the address of the directory to be exported. A new directory is created according to the cue/ISO file name. All files of the subtrack are poured into this directory;

3, choose 

or 

Set the output format;

4, click 

Start the split track, If you want to pause in the middle, you can click on it 

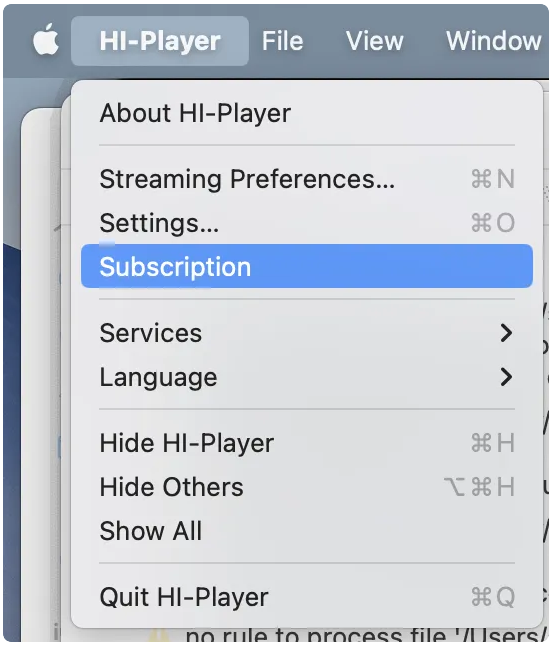
;

After the file split is completed, the user will be prompted by a pop-up window, and the folder of the split will be opened;

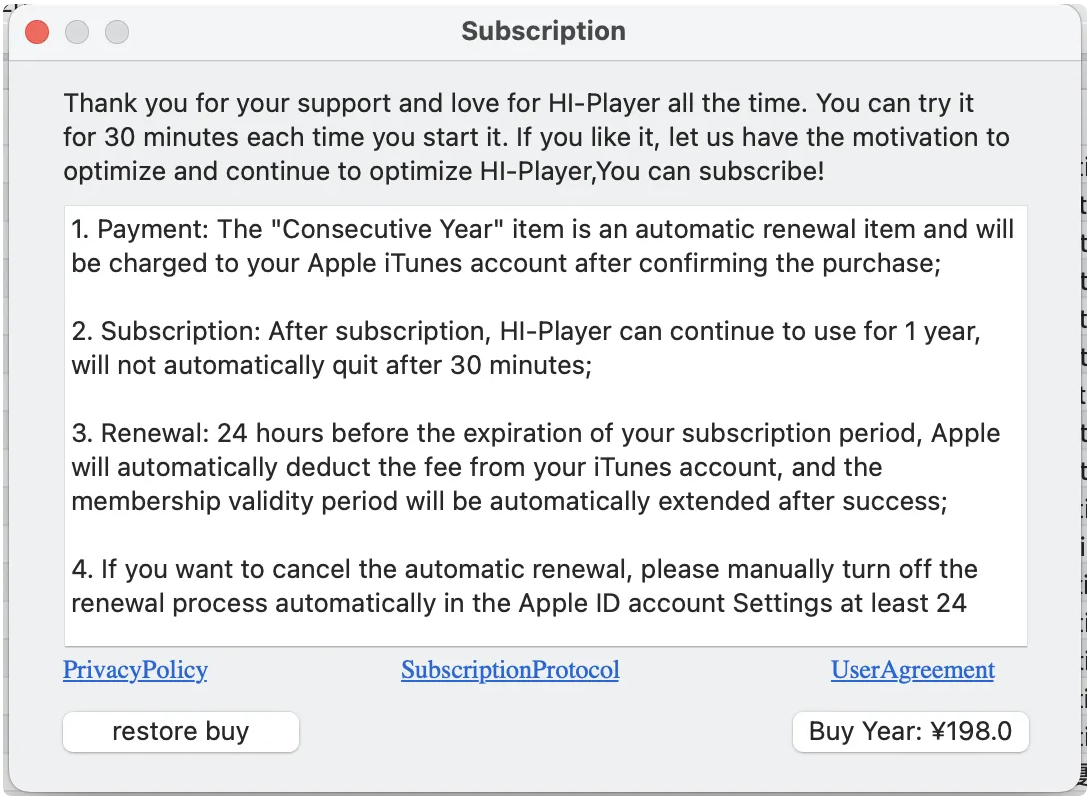
8, Subscription services

On the App store, it's a 30-minute trial version, and you'll have to subscribe to it before you can use it indefinitely. Here's how:

1.Click Subscribe Services under HI-Player in the menu bar.



2， The subscription window will pop up, select the Buy year button, followed by apple's subscription process. You can complete the subscription after entering the apple account to confirm. After the subscription, it will not automatically quit in 30 minutes, but will be used for unlimited time within the subscription period.



Test report

Power consumption

DSD frequency up

Test file

44100/2 track FLAC file

There is a high CPU usage a few seconds before the file starts playback, the test value is the stable value after 30 seconds after the start of playback, and the minimum and maximum values are selected after continuous observation for 1 ~ 2 minutes;

Test device

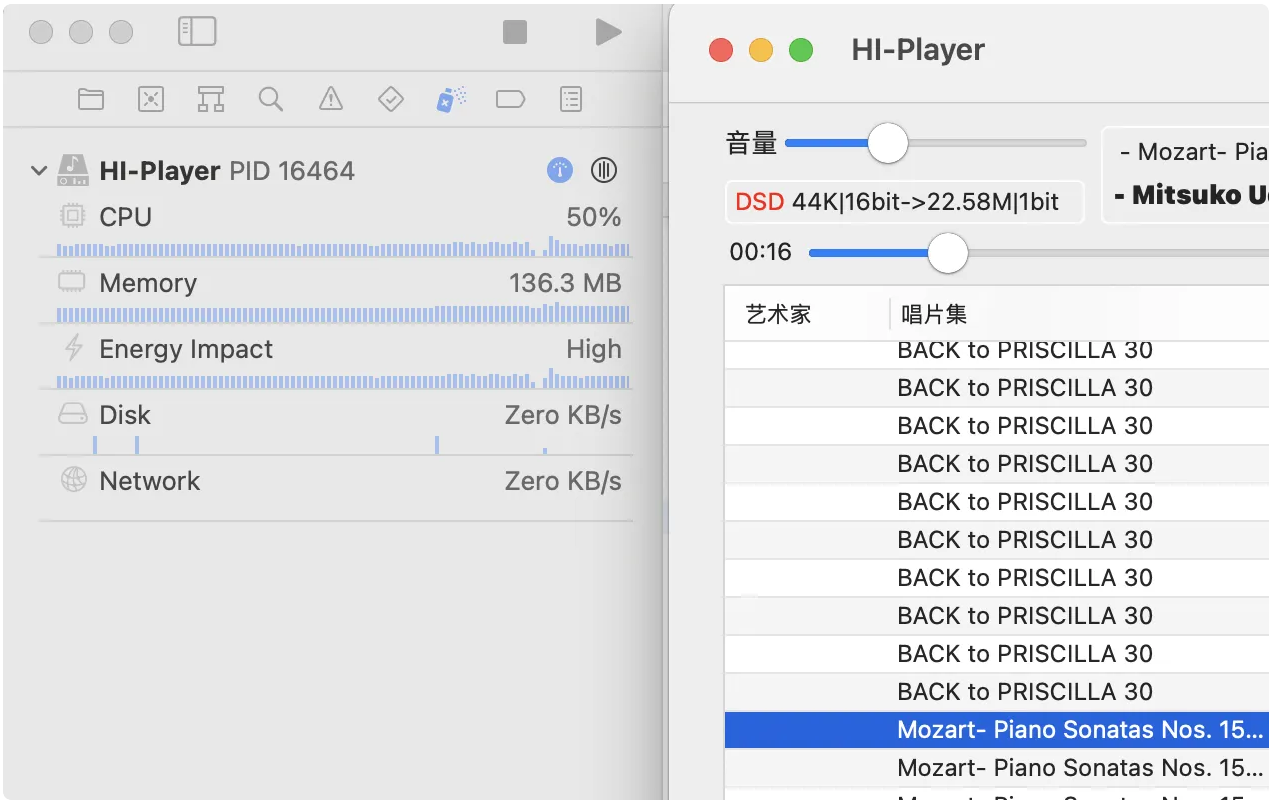
MacBook Pro (Retina, 13-inch, Early 2015)	10.14.6	2.7 GHz Intel Core i5	8 GB 1867 MHz DDR3

For DSD64~ DSD256, we use the above MacBook Pro 2015 device for testing;

MacBook Pro (13-inch,2023)	13.0	Apple M2 Pro 10核心（6性能和4能效）	16GB

For DSD512 format, we use the above M2 Pro for testing, M2 Pro can be extremely stable upscaling DSD512 playback;

We found that the M2 Pro based on the TypeC interface has very good compatibility;



Low power consumption group

Sampling rate	Interpolation filter	Modulator	Test value
44100*64	Poly–Sinc–Gauss sharp LP	HP–5 modern	13.6%～14.1%
44100*128	Poly–Sinc–Gauss sharp LP	HP–5 modern	20.1～20.8%
44100*256	Poly–Sinc–Gauss sharp LP	HP–5 modern	31.2%～32.9%
44100*512	Poly–Sinc–Gauss sharp LP	HP–5 modern	51.2%～56.1% for M2–Pro test

Medium power consumption group

Sampling rate	Interpolation filter	Modulator	Test value
44100*64	Sinc–HB–Gauss classic	HP–7 modern	14.4%～15.7%
44100*128	Sinc–HB–Gauss classic	HP–7 modern	21.1%～22.1%
44100*256	Sinc–HB–Gauss classic	HP–7 modern	33.9%～35.7%

44100*512	Sinc-HB-Gauss classic	HP-7 modern	60%~67.3% for M2-Pro test

Medium to high power consumption group

Sampling rate	Interpolation filter	Modulator	Test value
44100*64	Sinc-HB super	HP-5 modern	17.1%～18.5%
44100*128	Sinc-HB super	HP-5 modern	24.4%～25.2%
44100*256	Sinc-HB super	HP-5 modern	35.7%～38.5%
44100*512	Sinc-HB super	HP-5 modern	58.1~68.2% for M2-Pro test

High power consumption group

Sampling rate	Interpolation filter	Modulator	Test value
44100*64	Sinc-HB super	MAD-8	23.7%～25.2%
44100*128	Sinc-HB super	MAD-8	34.2%～36.2%
44100*256	Sinc-HB super	MAD-8	45.1%～46.3%
44100*512	Sinc-HB super	MAD-8	58.3%～63.2% for M2-Pro test

Memory

For high power consumption group, under the condition of the highest power consumption, memory occupies about 78M~ 100M, similar software in the same case occupies about 190M memory, HI-Player is much lower than similar software, and the memory remains stable throughout playback, so the memory occupation of different power consumption groups is not listed in detail here;

From the test results, it can be seen that compared with similar software, HI-Player has lower power consumption on the 2015 Mac Book Pro, and we are also in the deep optimization

algorithm, the high-time-consuming algorithm will be finely optimized, and the optimization will be fed back to everyone in time, everyone is responsible for energy emission reduction!

Test device

USB2.0 bandwidth

USB2.0 limit bandwidth (480M bps/s);

DSD256 is 44100*256/8 *8=11289600bps(11.289M), and DOP 24bit calculation requires about 33868800(33.9M).

DSD512 is 44100*512/8 *8=22579200bps(22.578M), and DOP 24bit calculation requires about 67737600(67.7M).

DSD1024 is 44100*1024/8 *8=45158400bps(45.16M), DOP 24bit calculation requires about 135475200(125.48M),

USB2.0 can theoretically support up to DSD1024;

At present, Mac can definitely support it, and our test results are in line with expectations;

The Mac BookPro intel 2015 model we tested supports DSD256 and runs smoothly;

Mac Apple M2Pro supports DSD512 smoothly, without any lag or noise, test data see above;

The following mainly introduces the test situation of testing DSD frequency up, PCM mode is supported by all devices, due to the differences in the PCM support format of each device, it can basically support the USB2.0 bandwidth limit;

USB cable

We tested a variety of decoders, including desktop decoders and portable decoders, little tails and other devices, USB cable is a more discussed topic in HIFI circle, the actual test results show that we have a number of USB cables under DSD64/128, which are stable, with better sound quality, no noise and stuttering, individual devices have problems, we have introduction and solutions in the test results, very few USB cables have intermittent stuttering under DSD256,

The test also has the same problem with other upscaling software, stable after changing a certain brand of USB cable (lower price), portable player and small tail test, using USB short line;

In the latest DSD512 test, we used a large brand (home grade) TypeC cable to connect Apple M2 Pro, the test is extremely stable;

Desktop decoder

Decoder name	DAC	Manufacturer	Test results and suggestions
TERMINATOR–PLUS 12th	R2R	Denafrips	DSD64/128/256/512 is stable, our daily use of decoder, DSD for hardware decoding, with good sound quality, excellent low frequency and good atmosphere; Support 48K DSD standard;
PONTUS II 12th	R2R	Denafrips	DSD64/128/256/512 is stable, the sound has a good texture, with good low frequency; Support 48K DSD standard;
May	R2R	holo audio	DSD64/128/256 is available, with large sound stage, fresh and natural style; **DSD512 is not tested; The 48K DSD standard is not tested;
element X2 pure	ESS 9039Pro x2	Matrix	DSD64/128/256 is stable and available, the device is asynchronous by default, you need to switch to the synchronous mode, otherwise the music is easy to be intermittent, this device uses a newer DAC chip, there are good technical indicators; **DSD512 is not tested;

element X2	ESS 9038Pro	Matrix	DSD64/128/256 is stable for use, there is no need to modify the mode on this device and the Mac M2 Pro, and the device has a softer sound; **DSD512 is not tested;
DAC-R26	R2R	GUSTARD	DSD64/128 is available, DSD256 occasionally short temporary relay abnormal sound when cutting songs, the factory will solve the problem, with good medium and high frequency, suitable for female voice or medium and high frequency music requirements, not so good in low frequency;
DAC50	AKM 4490	Accuphase	DSD64/128 is stable, great improvement after DSD frequency up, low frequency and atmosphere are greatly improved, technical indicators are average, high frequency is gorgeous but enduring; **Only support DSD64/128;

Some of the data is early testing, only DSD64/128/256 was tested;

After our optimization, the latest HI-Player can support DSD512, which can play DSD512 continuously and smoothly on supported devices;

Test device

Audio Power Amplifier：**Accuphase** E650 Class **A** ;

Speaker：**Harbeth** M30.1 ;

XLR cable：**VandenHul** XLR cable ;

Speaker cable：**VandenHul** MagnumMKII ;

Portable player

Recently, domestic HIFI manufacturers have begun to use discrete components to build DAC modules, cayin is one of the domestic pioneers (foreign Esoteric, dCS, etc. have similar products), cayin N7/RU7 are delta SIGMA DACs built with discrete components, all audio will be converted into DSD format and then converted into analog signals, it seems that DSD DAC is a good direction;

We collected representative portable players and small tail products on the market to do a test, the specific conclusions are shown below;

Player name	DAC	Manufacturer	Suggeastions and remark
PAW GOLD Touch	AKM4497	Iotoo	DSD64/128/256 is available, Mac BookPro 2015 model uses DSD256 cable, stable after using Sony wiring, it is recommended to use mobile phone mode (select USB DAC (mobile phone)) to access the DAC, and use computer DAC mode to have audible current sound and instability; Use with Apple M2 Pro, DSD256 is extremely stable, delicate and smooth; It is not very stable with the 2015 Mac Book Pro, and often drops, which may be a sampling rate compatibility issue due to UAC feedback feedback according to experience; 48K DSD standard is not supported;
N7	DeltaSIGMA DAC	Cayin	This device is a discrete element DS architecture DAC, DSD signal pass-through DAC decoding; DSD64/128 is stable, DSD256 is occasionally intermittent; 48K DSD standard is supported;

N8 II	ROHIM BD34301 x2	Cayin	This device is the most high-end portable player of Cayin. DSD64/128/256 is stable and available, and the sound quality is suitable for human voice 48K DSD standard is not supported;
RU6	R2R	Cayin	This device is a discrete element R2R DAC, DSD64/128 is supported, with good sound quality, internally convert to PCM through audio bridge and then convert analog signal through R2R, I commonly used test equipment, the equipment is extremely stable, excellent compatibility; 48K DSD standard is supported;
RU7	DeltaSIGMA DAC	Cayin	The device is a discrete element DS architecture DAC, DSD64/128 is supported, internally, the audio bridge is converted into DSD and then the analog signal is converted through the filter bank. DSD inputs are treated the same way because there are additional DSD format conversion operations, it is recommended to use PCM upscaling mode, the use of DSD frequency up mode does not sound good;

Earphones

SONY IER M7 **4.4 balance port**

Westone W80 V3 3.5

FiiO EM5/FF5 **4.4 balance port**

Audio Techinica CM2000Ti **4.4 balance port**

Beyerdynamic DT900ProX 3.5

OS and hardware

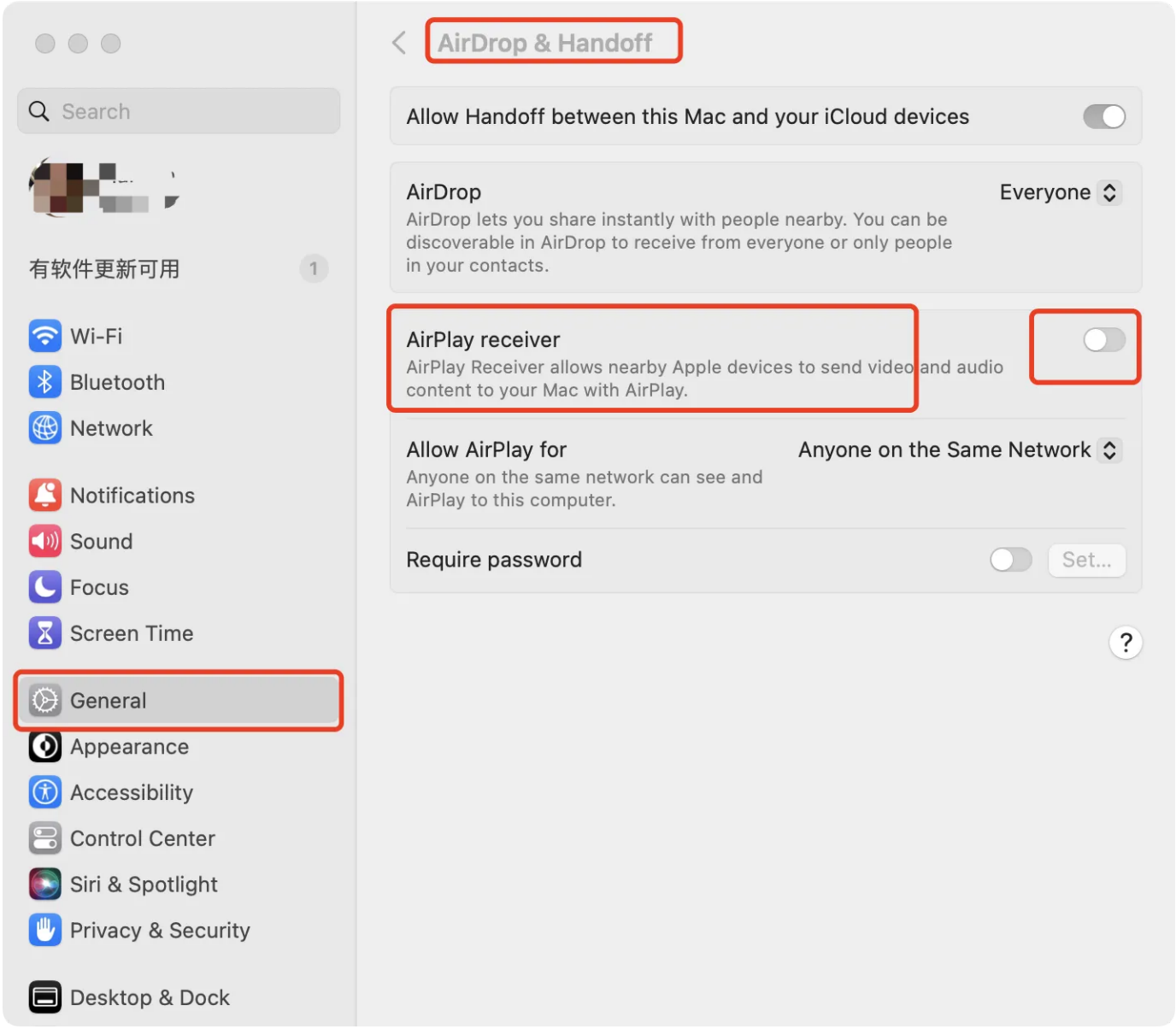
Four Mac hosts tested, all of which are MacBook Pro notebooks with the following configuration:

Test device name	OS	CPU	memory
MacBook Pro (16-inch,2019)	10.15.7	2.6GHz 6 Intel Core i7	16G 2667 MHZ DDR
MacBook Pro (Retina, 13-inch, Early 2015)	10.14.6	2.7 GHz Intel Core i5	8 GB 1867 MHz DDR3
MacBook Pro (13-inch,2016,Four Thunderbolt 3 Ports)	12.6.7	2.9 GHz Intel Core i5	8GB 2133 MHZ LPDDR3
MacBook Pro (13-inch,2023)	13.0	Apple M2 Pro 10 CPU（6+4)	16GB

Precautions

一，Mac AirPlay

Because the higher version of Mac comes with the AirPlay receiver capability, and it is turned on by default, it will cause trouble to the receiving ability of HI-Player's AirPlay, you can manually turn off the function, restart HI-Player, then you can use HI-Player's AirPlay receiving function;



二，Installation package is damaged

参考

https://blog.csdn.net/CharlesYooSky/article/details/128954049

1、Turn on “Allow any source”

2、If it still shows “Damaged, cannot be opened”, you can enter the following command

```
xattr –cr
```

```
xattr –cr /Applications/HI–Player.app
```

Reason: HI–Player is currently in the testing stage and has not yet been released. It will be normal when it is released, you can update it directly in the app store;

Get test version

To obtain the test version, please join QQ group:：487580030；

EMail：hi_playerdsd@163.com；