SounDevil music player

Manual 1.1

Introduction

Brief

SoundDevil is a modern music player of Windows platform.
Focus on high-quality music playback, pursue ultimate sound quality and high reliability.
Optimize for modern hardware and software, and discard compatibility with older systems and low - end devices.

Audio engine is Surreal Engine

Input file format, supports DSD, WAV, FLAC, APE, MP3.
Output interface, supports ASIO, WASAPI exclusive mode.
Output format, supports PCM, DSD native, DoP.
DSP uses 64bit floating-point number, adopts modern high-quality algorithm, supports modern high-performance instruction set, has the functions of channel conversion, sampling rate conversion (software up-sampling), coding conversion, parameter filter, convolution filter, etc.

User interface is DiVect UI (Direct Vector User Interface)

Using Direct2d as the rendering engine and the graphics card for acceleration, which is much faster than the traditional GDI / GDI +, and the image quality is also better.
Floating point coordinates and vector graphics are adopted to support lossless scaling at any scale.
With display synchronous refresh technology, it can achieve the best performance and energy efficiency.
Supports very high refresh rate.
Supports any high resolution, automatically adapts to screen DPI, and supports application scaling.
Supports multiple languages.

System requirements

	Minimum	Recommended
operating system	Windows 7	Windows 10 / 11
CPU	4 threads, with AVX	
memory	available memory 50M	
hard disk	free space 50M	
display	resolution 800 x 600	height DPI screen

sound card	44100Hz or 48000Hz	44100Hz and 48000Hz, 24bits

Install and Run

The software has two version: install and portable.
For the install version, run the installation program. After installation, a shortcut will be generated on the desktop and start menu. Run SounDevil.
The portable version does not need to be installed. Unzip it, enter the generated folder, and run SounDevil.exe

Software components

The software mainly includes executable program, configuration file and user manual.
Executable program mainly includes:
Soundevil.exe - main program,
dui.dll - user interface library,
surreal.dll - playback engine library.
Configuration file mainly includes:
User interface related files, including XML files, SVG graphics and language files.
User files (excluding data).

Data flow

Source -> Input ->DSP-> Output -> Endpoint

Source is usually a file, such as “music .wav"

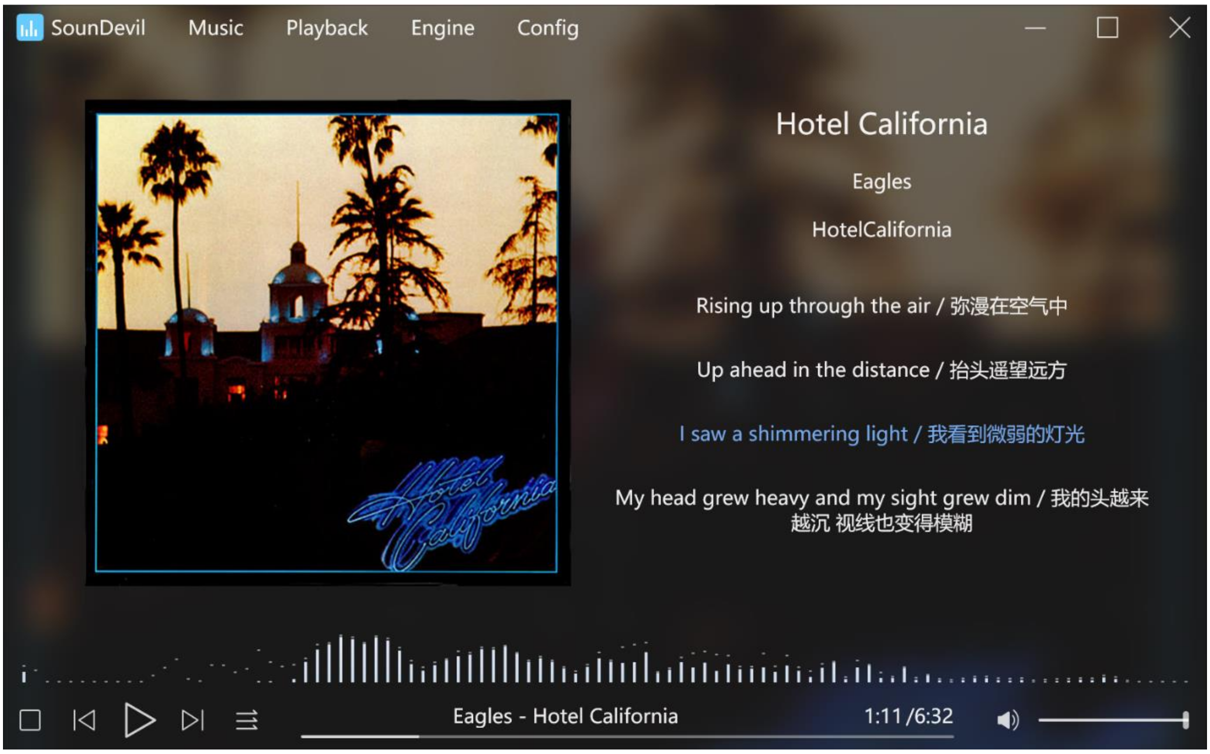
Input gets data from the source and convert it to PCM format.

DSP processes the data, such as sampling rate conversion, volume adjustment, format conversion.

Output transmits data to the endpoint through API (such as ASIO, WASAPI).

Endpoint is usually a sound card and its connected devices, such as "Speakers (Realtek (R) Audio)", including the sound card "Realtek (R) Audio" and its connected speakers.

User interface



The main interface is divided into three parts, from top to bottom is the title bar, page, playback control bar.

Title bar

On the left is the program icon and name, in the middle is the page label, and on the right is the window control button.

Hold down the left mouse button and move the mouse to drag the window position.

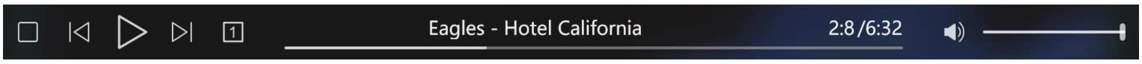
Double click the mouse to maximize or restore the window.

Left click on the lower right corner of the window and drag to change the size and shape of the window.

Click the tab to switch the page.

Click the button on the right to minimize, maximize / restore the window and close the application.

Playback control bar



The buttons on the left are stop, previous, play / pause, next and playback order.

Click playback order to select other order.

The artist and title, playback position and track length are displayed in the middle.

The middle and lower part is the playback progress bar. Click the progress bar or drag the slider to change the playback position.

On the right, click the speaker button to control the mute switch, and click / drag the volume slider to control the volume.

Music page

Used to select the source.

On the left is the lists, and on the right is the list content.

The lists include Default List, Recently Played and user created lists.

Left click one of them, its content will be displayed on the right, and right click will display the menu.

The yellow mark on the left of the list is the currently displayed list, and the green mark is the current playlist.

Click the plus button on the right of My Lists to create a list.

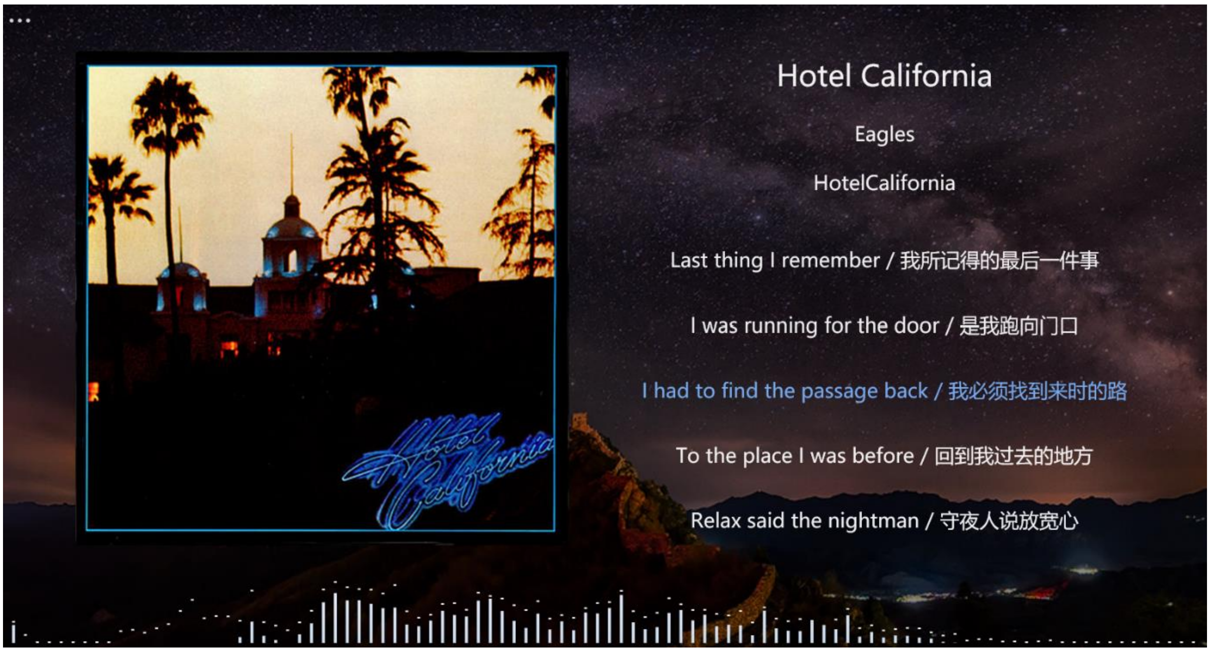
The list name is displayed at the top of the right.

The toolbar button can add files and folders to the list and empty the list.

Select files (folders) from Windows Explorer (multiple files can be selected at the same time), drag and drop to this window to add tracks to the currently displayed list.

Left click the track to start playing, right click to pop up the menu.

Playback page



Displays information about the current track.

On the left is the album cover, on the right is the tags and lyrics.

At the bottom is the mini visualization.

Cover

Supported formats: JPEG, BMP, PNG

Search order (the following example assumes that the track path is D:\album\track1.wav)

1 Embedded picture.

2 The picture file has the same path as the track file (e.g., D:\album\track1.jpg).

3. The picture file has same directory as the track file, and the file name is the parent folder (e.g., D:\album\album.jpg).

4. The picture file has same directory as the track file, and the file name is “cover” (e.g., D:\album\cover.jpg).

Lyrics

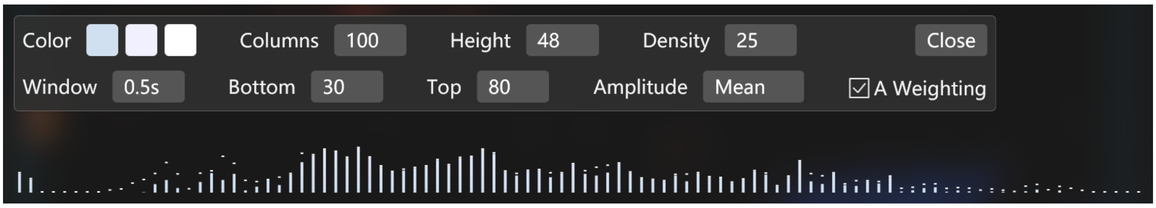
Supported format LRC (.lrc)

Search order

1 Embedded lyrics.

2 The lyrics file has the same path as the track file (e.g., D:\album\track1.lrc).

Mini visualization



Displays the spectrum of the live music.

The abscissa is the frequency and the ordinate is the volume level.

The frequency range of each column is basically divided according to the "Twelve-Tone Temperament".

Hovering the mouse over the column will display its frequency range.

Right click to display the menu. Click the menu item to show config and hide the mini visualization.

After the mini visualization is hidden, an icon will appear in the lower left corner. Click it to redisplay the mini visualization.

Config:

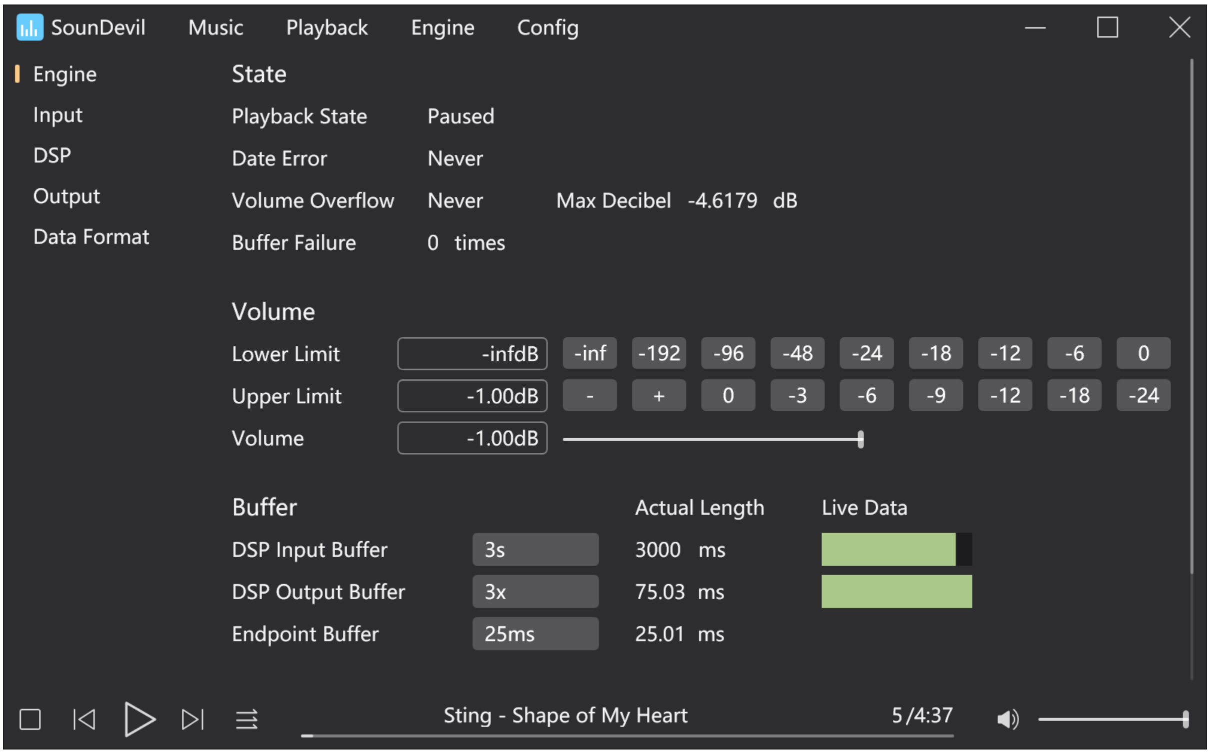
Click the color button to select the color of the bottom, top and peak of the column.

You can also set the number, height and density of columns.

Window duration refers to the time length of DFT. The shorter it is, the faster it changes, it is suitable for fast music rhythm. The longer it is, the smoother it is, and it is suitable for slow music.

For A-weighting, in short, the low frequency and high frequency are attenuated, and the medium frequency is slightly increased to make the vision closer to people's auditory feeling. Without A-weighting, the most obvious thing is that the low frequency looks louder than it sounds.

Engine page



Display playback engine status and set engine parameters.

Engine

Status

Data error means that the data of source has error, or the source format is not recognized.

Volume overflow means that the volume is out of range for the output data type.

Overflow occurs when the maximum decibel exceeds 0dB.

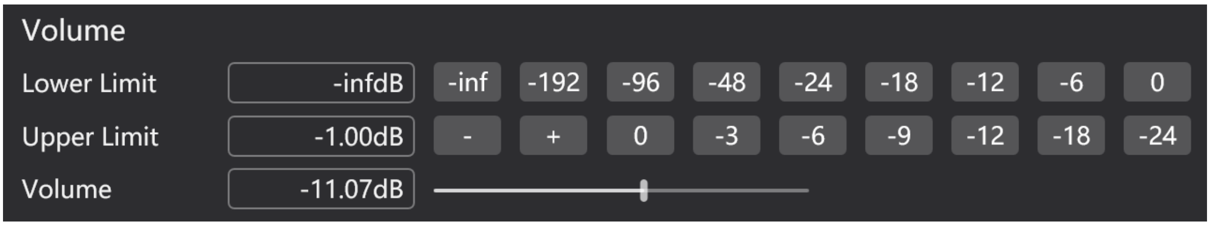
Due to the anti overflow function, the actual output will not exceed the range, but roof cutting will occur, so it should be avoided.

Overflow is usually caused by DSP. For example, the parameter filter does not set the preamp correctly. In this case, the filter setting should be modified.

It is known that SoXR algorithm may produce a gain of a few tenths of dB. When using this algorithm, the upper volume limit should be set below -1dB.

Buffer failure means that the engine fails to provide data in time when the endpoint needs data. Generally, using a larger buffer can reduce the probability of failure.

Volume



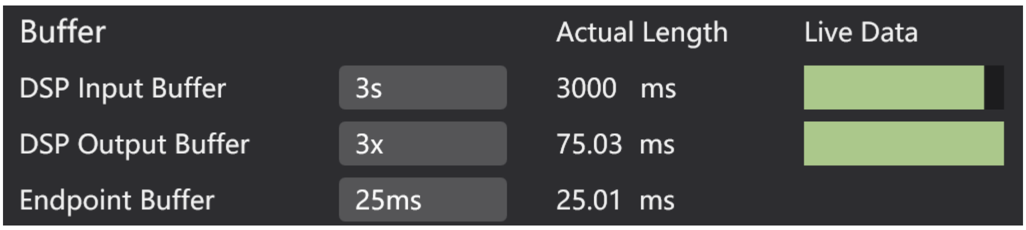
Set the player volume.

You can enter a value in the edit box, or click the button on the right / drag the slider to select a value.

The upper limit has two main functions: one is to provide room for DSP, and the other is to avoid accidentally setting the volume too loud.

Output = Input * ReplayGain * Volume

Buffer



There is a three-level buffer inside the playback engine:

The source data is converted into PCM through input, and then stored in the primary buffer after channel conversion and filtering in DSP;

The data is read from the primary buffer and stored in the secondary buffer after volume adjustment, sample rate conversion and coding conversion;

The data is read from the secondary buffer and stored in the tertiary buffer for exchanging data with the endpoint.

Too small buffer will cause broken sound, too big buffer will waste memory. As long as there is no broken sound, the smaller the better.

The primary buffer is 6 seconds by default, and 2 to 6 seconds is recommended. It is recommended to use a larger value for Hard Drive Disk and a smaller value for Solid State Disk.

The secondary buffer is an integer multiple of the tertiary buffer. The default value is 4 times. It is recommended to be 2 to 4 times. For computers with good performance and low load, a smaller value can be used.

The default value of tertiary buffer is 50ms, and the recommended value is 25ms to 100ms. The actual value depends on the endpoint.

If there are repeated buffer failures, you can try to increase the buffer.

If the live data is always full, you can try to reduce the buffer.

Input

Source format

In principle, only lossless format is supported.

currently supported

APE(Monkey's audio), *.ape *.mac

Cue sheet, *.cue

FLAC(Free Lossless Audio Codec), *.flac *.fla

MP3(MPEG Audio Layer 3), *.mp3

SACD(Super Audio CD), *.dff *.dsf *.iso *.dat

WAV, *.wav *.wave

Replay gain

At present, only DSD to PCM gain is supported.

According to the standard, the gain from DSD to PCM should be 6dB (otherwise the volume is too low), but some sources do not meet the standard and overflow may occur.

DSD to PCM

Why convert?

Because

1. DSD does not support volume, EQ, and other processing. Compared with the small disadvantages

caused by transcoding, these processing obtain greater benefits.

2. Many output devices do not support DSD and need soft decoding.

3. Many devices that support DSD hard decoding are internally converted to PCM, and the quality of soft decoding can be higher.

What did the conversion lose?

DSD to PCM is generally lossy. The so-called lossy means that DSD cannot be completely restored with the converted PCM.

DSD uses ultra-high sampling rate, not to store ultrasonic wave, but to better store audible frequency band, because its bit depth of only 1 bit will produce great sampling error. Most noise is placed in ultra-high frequency band through ultra-high sampling rate and noise shaping.

SACD standard stipulates that the playback DSD signal must pass through a low-pass filter to filter out the noise in the UHF band. Basically, the software conversion will also use the low-pass filter. Of course, the obtained PCM cannot completely restore the original DSD.

Lossless conversion is possible. For example, the sampling rate of PCM is the same as that of DSD, each sample uses 32bit, without low pass filter. However, retaining this UHF noise is not good for subsequent processing.

Our DSD filters

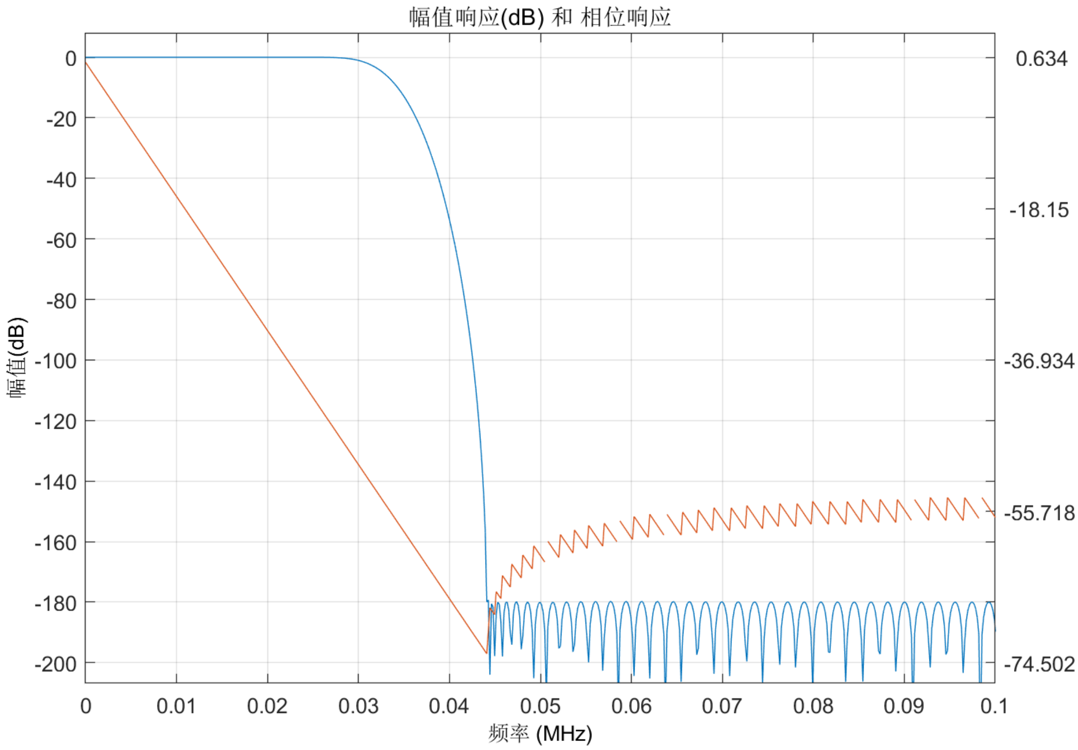
are linear phase FIR filters with a length of about 1500. Targeted optimized parameters are used for each input frequency.

The performance has far exceeded the actual needs and is better than SACD standard filter.

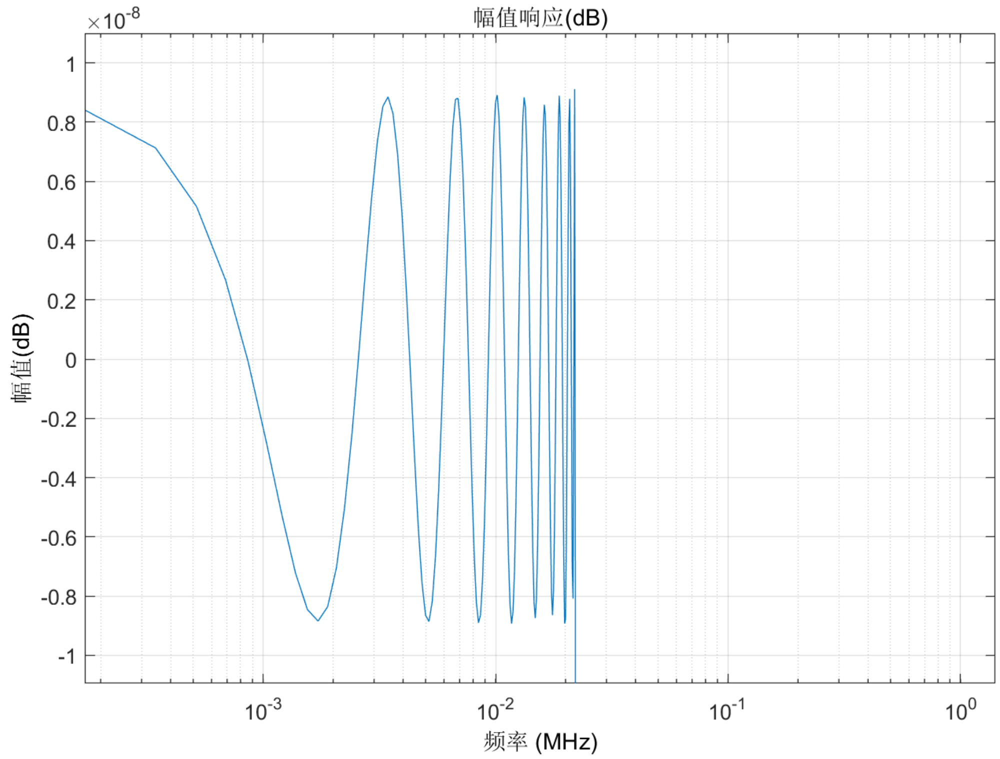
For example, the filter used for DSD64 to PCM conversion:

```
/*
*Discrete time FIR filter (real number)
* ----------------

*Filter structure: direct form fir
*Filter length: 1496
*Stable: Yes
*Linear phase: Yes (type 2)
* Fs:2822400
* Fpass:22050
* Fstop:44100
*Apass: < 0.00000001db, passband ripple less than 1 / 100 million
*Astop: > 180db, stopband response less than 1 / 100 million
*/
```



Amplitude response and phase response



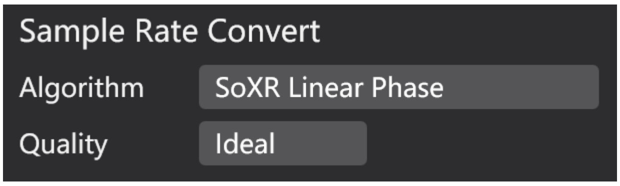
Passband amplitude response

DSP

Filter

Go to [Filter]

Sample rate convert



The engine will convert the sample rate according to the needs of input and output format.

Supports the conversion between 44100Hz and 1536000Hz.

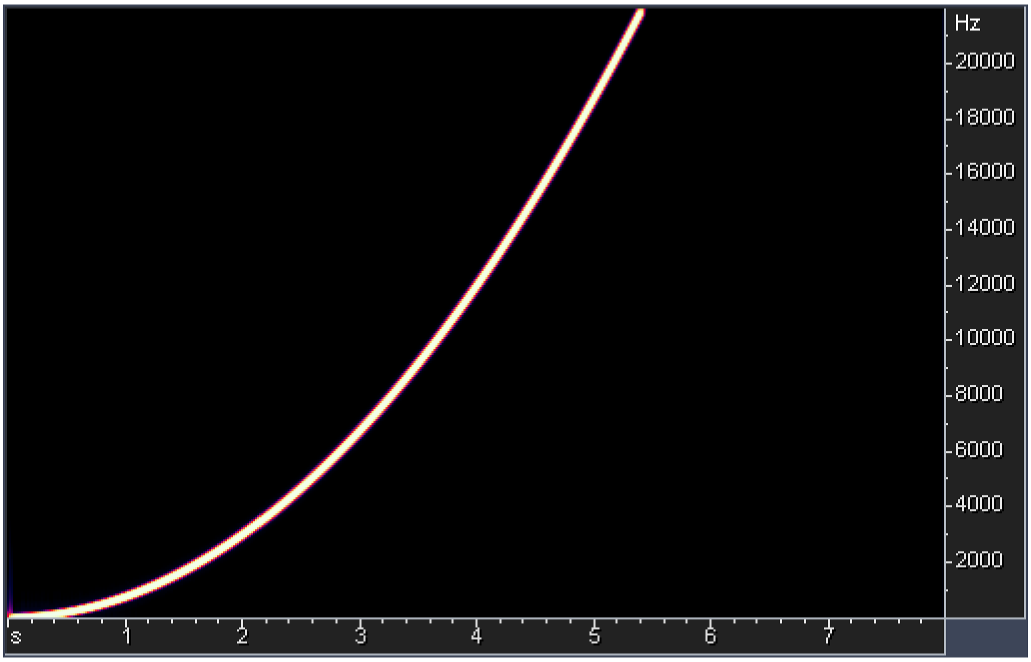
The choice of algorithm, if you can hear good or bad, choose your favorite.

The difference between quality, Fine and Ideal may be heard, Ideal and Ultimate cannot be heard in theory. Generally, choose Ideal.

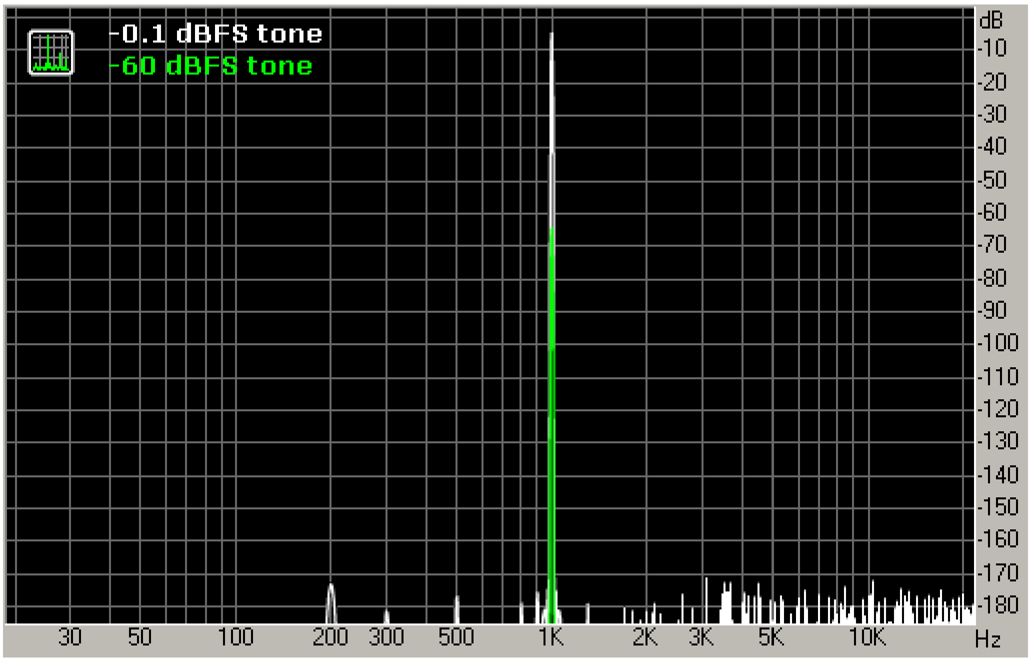
The changes will not take effect until the next playback.

The software algorithm can be superior to most hardware built-in algorithms and can be used for software up-sampling.

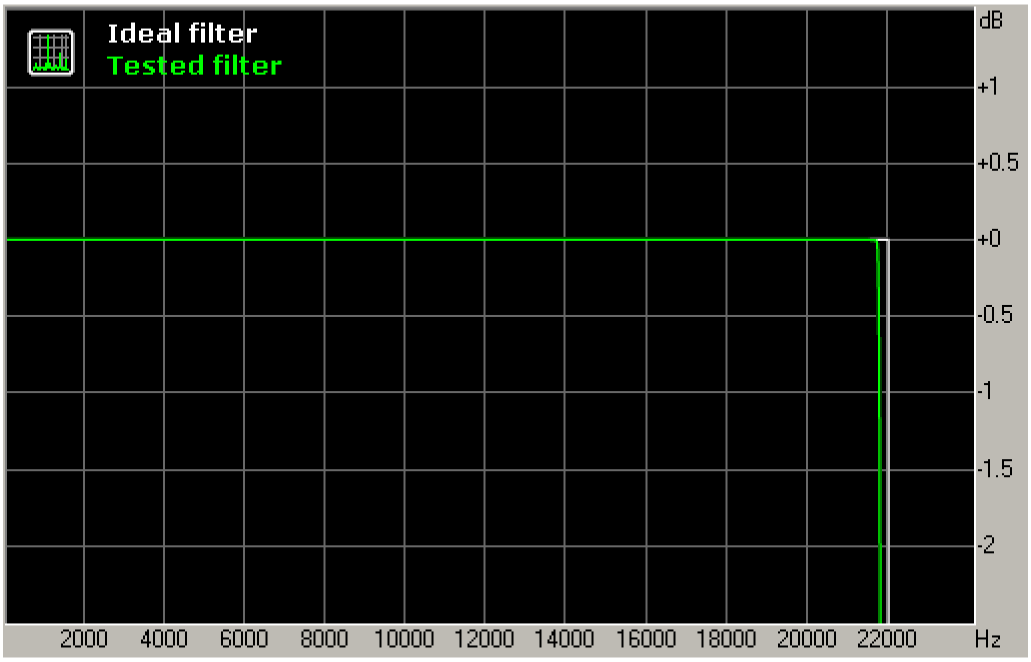
The following are the test results of 96KHz to 44.1KHz, linear phase (from SRC Comparisons (infinitewave.ca)):



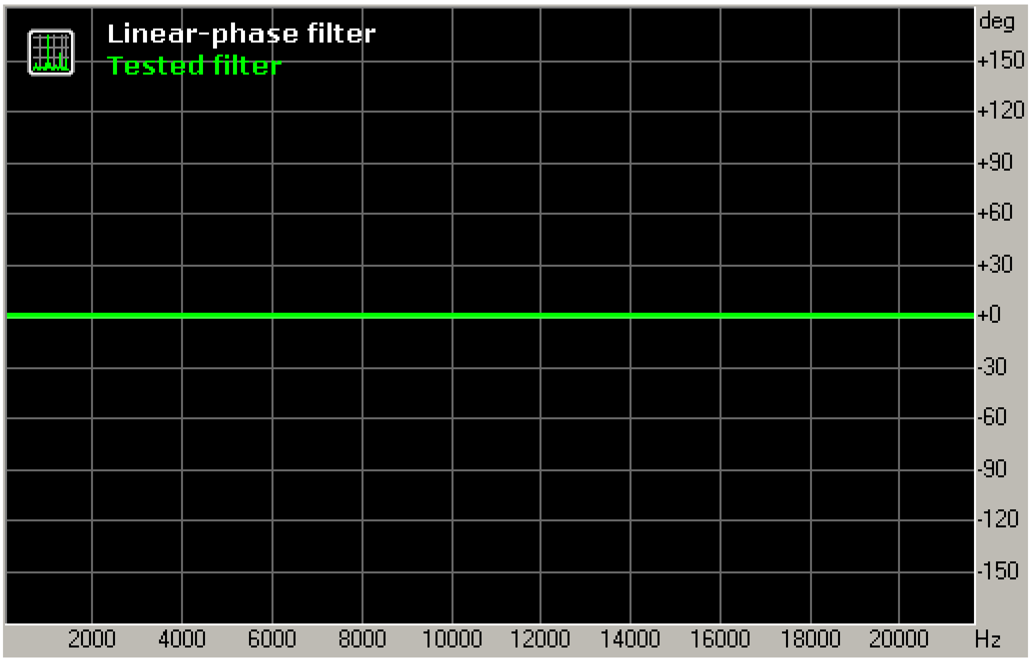
Sweep



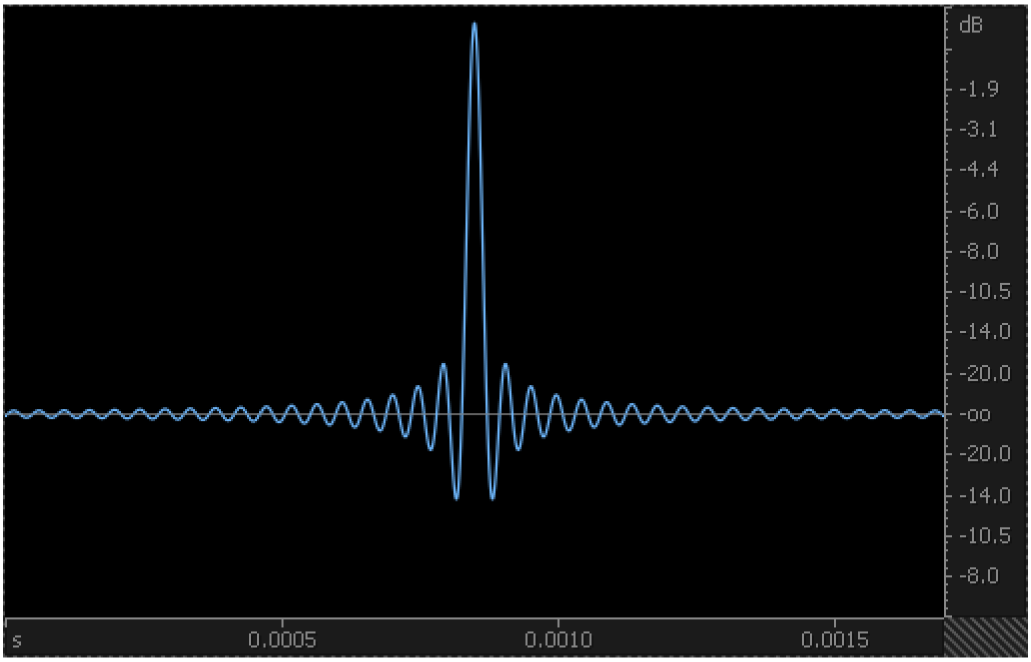
1kHz Tone



Passband

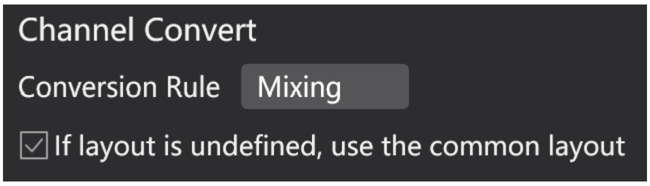


Phase



Impulse

Channel convert



The engine will do channel conversion as needed. The user can select the conversion rules.

If the channel layout of input or output (such as ASIO) is not defined, it will be set as a common layout according to the number of channels. For example, two channels will be treated as stereo.

The changes will not take effect until the next playback.

Channel definition

Channel name		Abbreviation
Left		L
Right		R
Center		C
Low Frequency		LF
Back Left		BF
Back Right		BR
Side Left		SL
Side Right		SR

Layout definition

Layout name		Included channels
Mono		C
Stereo		L,R
Quad		L,R,BL,BR
5.1 Surround		L,R,C,LF,BL,BR
7.1 Surround		L,R,C,LF,BL,BR,SL,SR

There are two conversion rules:

Discrete

Take the smaller value of the number of input channels and the number of output channels as m, and copy the first m input channels to the first m output channels,

If the number of input channels is more than m, the extra channels are discarded,

If the number of output channels is more than m, the extra channels are filled with silent.

Mixing

Input	Output	Rules
Mono	Stereo	output.L = input.C output.R = input.C
Mono	Quad	output.L = input.C output.R = input.C output.BL = 0 output.BR = 0
Mono	5.1	output.L = 0 output.R = 0 output.C = input.C output.LFE = 0 output.BL = 0

		output.BR = 0
Stereo	Mono	output.C = 0.5 * (input.L + input.R)
Stereo	Quad	output.L = input.L output.R = input.R output.BL = 0 output.BR = 0
Stereo	5.1	output.L = input.L output.R = input.R output.C = 0 output.LFE = 0 output.BL = 0 output.BR = 0
Quad	Mono	output.C = 0.25 * (input.L + input.R + input.BL + input.BR)
Quad	Stereo	output.L = 0.5 * (input.L + input.BL) output.R = 0.5 * (input.R + input.BR)
Quad	5.1	output.L = input.L output.R = input.R output.C = 0 output.LFE = 0 output.BL = input.BL output.BR = input.BR
5.1	Mono	output.C = 0.7071 * (input.L + input.R) + input.C + 0.5 * (input.BL + input.BR)
5.1	Stereo	output.L = input.L + 0.7071 * (input.C + input.BL) output.R = input.R + 0.7071 * (input.C + input.BR)
5.1	Quad	output.L = input.L + 0.7071 * input.C output.R = input.R + 0.7071 * input.C output.BL = input.BL output.BR = input.BR
Others	Others	As discrete

Code convert

Supported output:

PCM 16 – 32 bit integer, 32 or 64 bit floating point number, 44100HZ - 1536000hz

DSD 64 – 2048, 1-bit

DOP 64 – 2048, 1-bit

Misc.

Anti-overflow

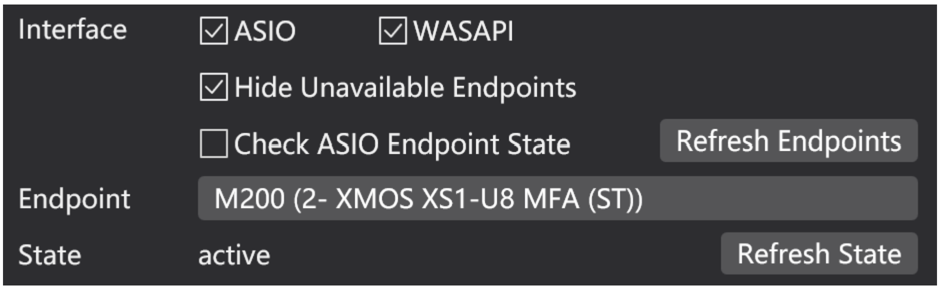
is to prevent the volume from exceeding the range of the output data type.

MMCSS

In short, more computing resources are used to achieve lower latency and higher stability.

Output

Select endpoint



The endpoint combo displays the name of the current endpoint. Click to pop up the endpoint list, and you can select another endpoint.

The first line selects which interface endpoints are displayed. For example, if you want to display only ASIO endpoints, check ASIO and uncheck other interfaces.

WASAPI has a mechanism to monitor the endpoint state without manual refresh.

Since ASIO does not have a mechanism to monitor the endpoint state, it needs to be detected manually, and the detection needs to occupy the device for a short time. If the device is in use, it may cause short-term sound loss or other consequences.

If "Check ASIO endpoint state" is not checked, the state in the endpoint list will display "unknown", and its state will be detected only when it is selected as the current endpoint.

If "Hide unavailable endpoints" is checked, only endpoints in "active" or "unknown" state will be displayed in the endpoint list. This is useful for WASAPI, because this interface may have dozens of unavailable devices.

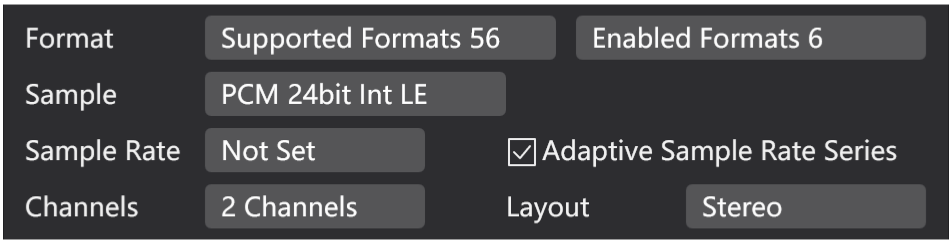
Meaning of endpoint state:

state	meaning
unknow	For ASIO, it refers to the undetected state.
active	The audio adapter that connects to the endpoint device is present and enabled, if the endpoint device plugs into a jack on the adapter, then the endpoint device is plugged in. Only active endpoints are available. However, the active state may not be available because it may be occupied by other programs.
disabled	For WASAPI, it means that the endpoint has been disabled. It can be re enabled in Device Manager.
not present	For WASAPI, it means that endpoint is not present because the audio adapter that connects to the endpoint device has been removed from the system, or the user

	has disabled the adapter device in Device Manager. For ASIO, it refers to other states that are not active.
unplugged	For WASAPI, it means that the endpoint is unplugged.

For the current endpoint, the supported data formats will be further obtained.
If everything goes well, its state will be displayed as "ready" and can start playing.
If something goes wrong, such as the device is unavailable, other state will be displayed and playback cannot start.
For WASAPI endpoint, when the endpoint becomes available again (or becomes unavailable), the state will be updated automatically without manual refresh.
For ASIO endpoint, when the endpoint state changes, you need to manually click the "refresh endpoint state" button to obtain the latest state.
It is recommended not to use the device used for music playback as the default output device of the operating system, especially when ASIO is used, because ASIO does not have a good communication mechanism with the operating system, various strange faults may occur.

Set format



Format refers to a combination of sample type, sample rate, channel number and channel layout.
Supported formats refer to all formats accepted by the endpoint. Click to pop up a list for viewing.
Enabled format refers to the format allowed by the user. It is a subset of supported formats. Click to pop up a list.
The user can specify the format partially or completely.
For unspecified parts, the engine will automatically select according to the input format and endpoint support format.
Our engine has complete rules, it can use the best format dynamically, even if the user does not specify the format at all.
The changed settings will not take effect until the next playback.

Sample type

If not specified, the PCM with the maximum bit depth is used.

Code

Generally, PCM should be used because the DSP of the player uses PCM format.
DSD is considered only when the device supports "DSD pass through" (it will not be converted to PCM internally).
Native DSD takes precedence over DoP because it is more efficient and usually supports higher

sample rate.

In the DoP mode, it is necessary to set the device volume to the maximum value (in the system settings), because the DoP uses special flag bits, if the volume is not 100%, these flags will be changed, so that the device cannot recognize it as DoP and will treat it as PCM. In DoP mode, you can use this software to control the volume.

Some devices cannot switch from DoP mode to PCM mode immediately. If there is a problem, please stop playing for a few seconds.

Bit depth

Always use the maximum bit depth supported by the device.

It is strongly recommended to use an output device that supports 24 bits or more.

Why no dither

Because no dither is required for more than 24 bits.

Dither improves 16 bits, but it is still far less than 24 bits.

If you have a desire for sound quality, especially when using software to control the volume, please use a device with 24 bits or more.

Sample rate

If not using up-sampling, select "not set" and the output sample rate closest to the input sample rate will be automatically selected. (The actual rules are complicated, in short, the most suitable sample rate will be selected)

If using up-sampling, specify a sample rate, usually the highest sample rate that the device supported, and check "adaptive sample rate series".

"Adaptive sample rate series" will automatically select the same level as the specified output sample rate and the same series as the input sample rate as the output sample rate, because the conversion between the same series is the best. For example, if input is 44.1kHz and output 192kHz is specified, the actual output is 176.4kHz.

Channel

The channel should be selected according to the actual speaker configuration.

Specifying a layout facilitates channel conversion.

If "not set", the engine automatically selects the output channel configuration closest to the input.

Exclusive Mode

ASIO has exclusive mode only.

WASAPI has two modes: exclusive and shared. We only support exclusive mode.

Sharing mode needs to convert audio data into the same sharing format, then mix and output.

First, this increases the delay.

Secondly, other programs cannot be prevented from sounding.

The sharing format is not easy to select.

Music is usually 44.1KHz, and computer and network video are usually 48Khz.

Finally, the operating system shall be suitable for low-end device, and its algorithm quality can be imagined that it will not be very good.
If the exclusive device is the default output device of the system, other software using the device may be abnormal, for example, the video web page stops playing. The solution is not to use it at the same time. After stopping playing, refresh the web page or restart the software.

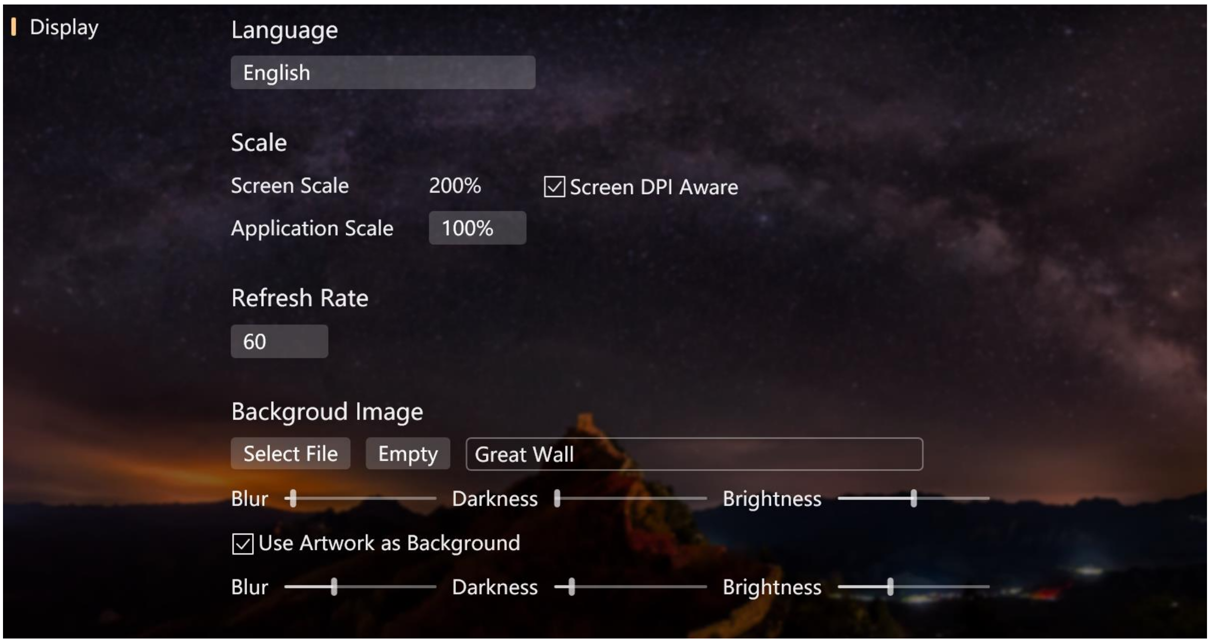
Data format

Display the data format of each part.

	Sample	Sample Rate	Channels	Layout
Source	FLAC 16bit	44.1kHz	2 Channels	Stereo
Input	PCM 64bit Float LE	44.1kHz	2 Channels	Stereo
DSP	PCM 24bit Int LE	44.1kHz	2 Channels	Not Set
Output	PCM 24bit Int LE	44.1kHz	2 Channels	Not Set
Endpoint				

Config page

Display



Language

Select the language to display.

Scale

Controls the display size of text, icons, etc.

The actual rendering scale (zoom rate) is equal to the product of the display scale and the application scale.

If DPI adaptation is checked, the display scale will change when the DPI of the display changes. If it is not checked, the display scale is equal to 100%.

Refresh rate

Controls the maximum number of frames updated per second by the user interface.

When the content of the interface does not change, it will not be updated.

The higher the frame rate, the smoother the animation, but the greater the load of CPU and GPU.

Generally, 60 frames to 120 frames are relatively moderate.

Background image

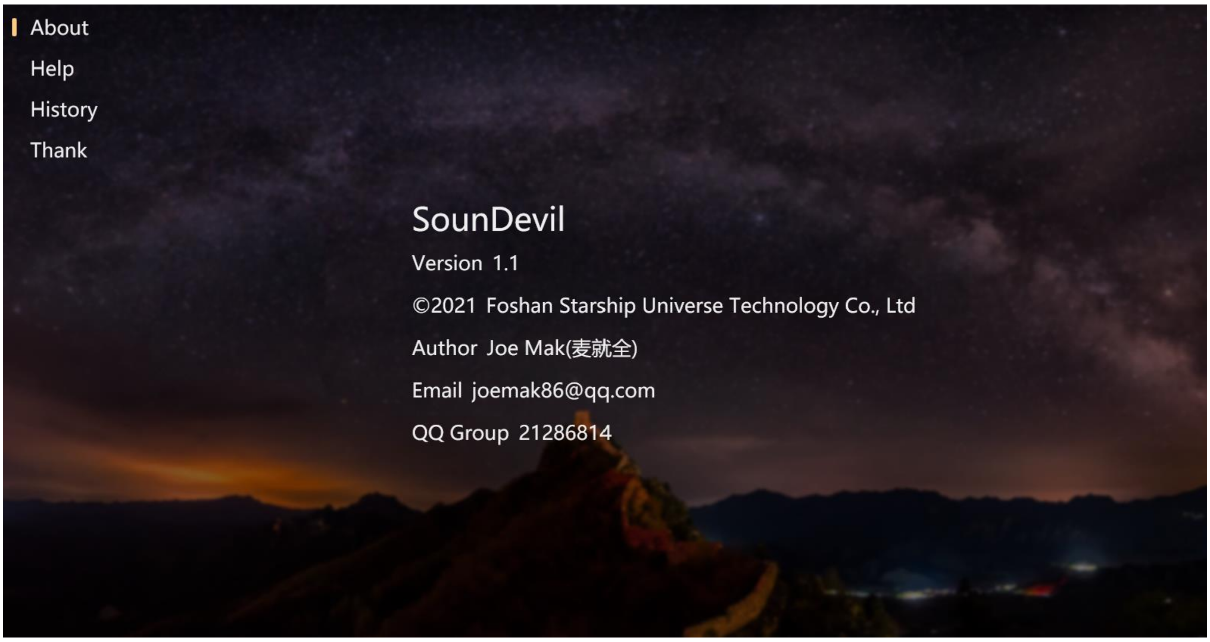
Controls the background of the main window.

In order not to conflict with other interface elements, the background image can be blurred, and brightness adjusted.

If you select "Use artwork as background" and the current track has a cover image, it will be used as the background image.

To adapt to any background image, each part of the main window has no border and uses a unified background color, which is a compromise. It is recommended to use a background image. Of course, you can also display the nude color.

About page



Display software related information, help, update history, thank.

Error message

When an error occurs, detailed and practical information is displayed.

The content includes current task, error cause, impact (effect), solution...

Technical information is for developers.

If you can't solve the problem, you can click the copy button to send the information to us by email or discussion group.

It is recommended that you use a newer version of the software, because the problems found in the old version may be corrected in the new version.

Filter

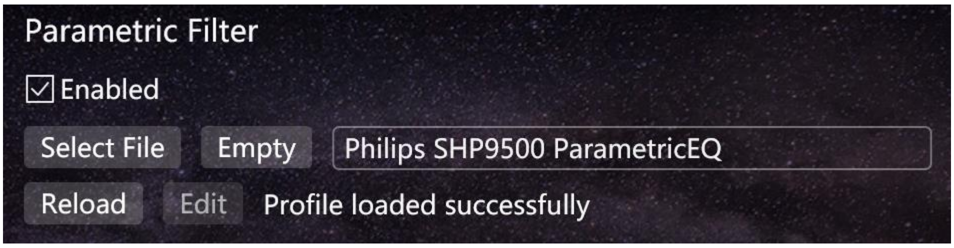
Filters can change the amplitude of different frequencies of audio signals.

For room acoustic correction, headphone frequency response adjustment, channel balance, electronic frequency division...

Parameter Filter

Supports all 8 kinds of filters, unlimited number.

All channels can be configured in common, or each channel can be configured independently.



The "Enabled" checkbox controls whether the filter takes effect. It can be switched on and off during playback and take effect after a short delay (depending on the buffer length).

Click the "Select file" button to select a profile.

The "Empty" button clears the path of the profile and does not empty the contents of the file.
The "Profile name" edit box displays the name of the profile. Click to edit the file path.
"Reload" button to re-read the contents of the file when the file path remains unchanged. After editing and saving the profile externally, click "Reload" to update the configuration.
The "Edit" button is not yet available.

Profile

It is a text file(.txt), which can be edited manually or generated by third-party software.
There are two types: one is that all channels use the common configuration, and the other is that each channel is configured independently. A configuration file can only be one of the two and cannot be mixed.
In the file, the line begins with # is a comment, which has no practical effect.

Common configuration:

Do not need to specify a channel, that is, do not need the "Channel" line.
Example:
Preamp: -6 dB
Filter: ON PK Fc 100 Hz Gain 0 dB Q 10
Filter: ON LPQ Fc 100 Hz Q 0.8071

Independent configuration:

You need to specify the channel, that is, you need a "Channel" line.
Example:
#This is a two-channel independent configuration
#Configuration of channel 1
Channel: 1
Preamp: -3 dB
Filter 1: ON PK Fc 50 Hz Gain 3 dB Q 0.48
Filter 2: ON PK Fc 100 Hz Gain -3 dB Q 0.84
#Configuration of channel 2
Channel: 2
Preamp: -12 dB
Filter 1: ON PK Fc 50 Hz Gain 2 dB Q 0.48
Filter 2: ON PK Fc 100 Hz Gain -2 dB Q 0.84

Configuration items

Including preamp and several filters.
In principle, if the filter has gain on the signal, you need to use preamp to reduce the overall volume level to avoid overflow. For example, if the filter with the largest gain is 3dB, the preamp should be set to - 3dB.
Examples of various filters:
#Preamp

Preamp: -6 dB

#Peaking
#Q
Filter: ON PK Fc 100 Hz Gain 0 dB Q 10
#Bandwidth
Filter: ON PK Fc 100 Hz Gain -6 dB BW Oct 0.1442

#Lowpass
#Default Q=0.7071
Filter: ON LP Fc 100 Hz
#Q, type name with "Q"
Filter: ON LPQ Fc 100 Hz Q 0.8071

#Highpass
#Default Q=0.7071
Filter: ON HP Fc 100 Hz
#Q, type name with "Q"
Filter: ON HPQ Fc 100 Hz Q 0.5071

#Bandpass
#Default Q=0.7071
Filter: ON BP Fc 100 Hz
#Q
Filter: ON BP Fc 100 Hz Q 0.9071

#Lowshelf
#Default S=0.9
Filter: ON LS Fc 100 Hz Gain 4 dB
#Slope
Filter: ON LSC 12 dB Fc 100 Hz Gain -5 dB
#Q
Filter: ON LSC Fc 100 Hz Gain 3.2 dB Q 0.6708

#Highshelf
#Default S=0.9
Filter: ON HS Fc 100 Hz Gain -0.5 dB
#Slope
Filter: ON HSC 3.0399 dB Fc 100 Hz Gain 4 dB

#Q
Filter: ON HSC Fc 100 Hz Gain -2 dB Q 0.6071

#Notch
#Default Q=30
Filter: ON NO Fc 100 Hz
#Q
Filter: ON NO Fc 100 Hz Q 12.1582

#Allpass
Filter: ON AP Fc 100 Hz Q 6.4397

Third party software compatibility

This software supports reading:

ParametricEQ generated by AutoEq.

Generic equalizer generated by Room EQ Wizard.

Profile generated by Equalizer APO that contains only parametric filters, but:

It is not supported to select multiple channels at the same time, such as "channel: 1, 2". It needs to be separated. Only one channel can be selected at a time.

The channel name is not supported. For example, "Channel: L" needs to be changed to "Channel: 1".

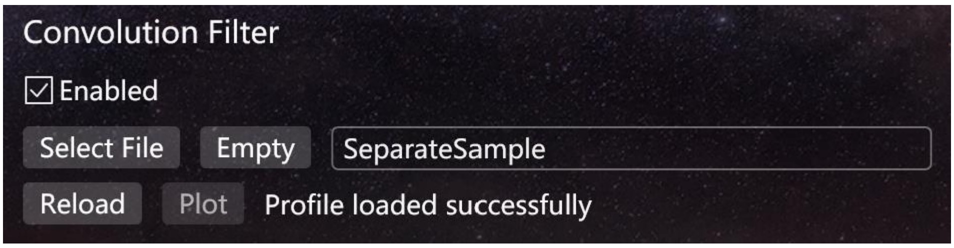
"Settings shared by all channels" and "settings for some channels" are not supported to appear in one file at the same time, such as

Channel: all
Filter 1: …
Channel: 1
Filter 2: …

It needs to be changed to either share the whole file for all channels or set independently for each channel.

Convolution filter

All channels can be configured in common, or each channel can be configured independently.
Support automatic switching of convolution kernel according to audio data sampling rate.



The "Enabled" checkbox controls whether the filter takes effect. It can be switched on and off during playback and take effect after a short delay (depending on the buffer length and reading the convolution kernel definition file).

Click the "Select file" button to select a profile.

The "Empty" button clears the path of the profile and does not empty the contents of the file.

The "Profile name" edit box displays the name of the profile. Click to edit the file path.

"Reload" button to re-read the contents of the file when the file path remains unchanged. After editing and saving the profile externally, click "Reload" to update the configuration.

The "Plot" button is not yet available.

Profile

Convolution kernel definition file is usually a wave file(.wav), which can be generated by third-party software. Note that it must be mono.

Each kernel definition file is only applicable to the data of one sampling rate, so the corresponding convolution kernel needs to be defined for each input data sampling rate.

Convolution filter profile is a text file(.txt), which can be edited manually.

There are two kinds of profiles. One is that all channels use the common configuration, and the other is that each channel is configured independently. One profile can only be one of the two and cannot be mixed.

In the file, the line begins with # is a comment, which has no practical effect.

Common configuration:

Do not need to specify a channel, that is, do not need the "Channel" line.

Example:

Convolution: 44100 44100.wav

Convolution: 48000 48000.wav

Independent configuration:

You need to specify the channel, that is, you need a "Channel" line.

Example:

#This is a two-channel independent configuration

#Configuration of channel 1

Channel: 1

Convolution: 44100 FL44100.wav

Convolution: 48000 FL48000.wav

Configuration of channel 2

Channel: 2

Convolution: 44100 FR44100.wav

Convolution: 48000 FR48000.wav

Configuration items

Take "Convolution: 44100 FL44100.wav" for example,

"Convolution" indicates that this is a convolution filter,
"44100" indicates that the filter is suitable for 44100Hz audio data,
"FL44100.wav" represents the path of kernel definition file,
If the kernel definition file is in the same directory as the filter profile, you can write only the file name (including the extension) instead of the full path.
If they are not in the same directory, you need to write a complete path, such as "Convolution: 44100 D:\Filter\FL44100.wav”.

Third party software compatibility

Remind again: convolution kernel definition file only supports mono.
This software supports reading:

Convolution EQ WAV generated by AutoEq.

Filters impulse response WAV exported by Room EQ Wizard.