

HQPlayer™ Desktop

User Manual

Version 6.2.0

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Streaming partners





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1. Introduction

HQPlayer is a high quality audio player for 64-bit Windows, Linux and macOS.
HQPlayer also features several user selectable high quality resamplers as well as user selectable dither/noise shaping algorithms and delta-sigma modulators.
Some of the more affordable sound cards and D/A converters have suboptimal digital and analog filters, while still having support for higher sampling rates. Effect of this can be reduced by applying high quality upsampling in software before feeding the signal to the audio hardware at higher rate. This moves some of the artifacts of the suboptimal hardware to higher frequencies, away from the audible band. Many of the home-theater amplifiers and digital (room correction) processors also re-sample internally to 48, 96 or 192 kHz, with the HQPlayer, these can be fed at the native rate avoiding lesser quality resampling in the device.
Most modern D/A converters are delta-sigma type. Built-in delta-sigma modulator of HQPlayer allows using DSD-capable converters with this native data format, in many cases bypassing lot of DSP processing in these converters and allowing more direct data path to the conversion stage. For select set of DACs, also correction profiles are available to improve correctness of the output signal.
Resampling also allows playback for high resolution audio files on hardware capable of only lower sampling rates or bit depths. For lower bit depth playback, high quality dither or noise shaping can be employed.
HQPlayer also includes a convolution engine for applying digital room correction filters or other kinds of equalization.
These features ensure the best possible audio quality with the available audio hardware.

1.1. DSDIFF and DSF playback, DSD sources and playback

Playback of DSDIFF and DSF files is supported. In addition, playback from other DSD sources such as ADCs and network streams is supported. In case hardware and drivers support ASIO or ALSA DSD -mode, or one of the “PCM packed” modes, these files can be played back in native format.
For devices capable of only PCM input, PDM (pulse density modulation) content of these files is converted to 176.4 (64*fs*), 352.8 (128*fs*) or 705.6 kHz (256*fs*) PCM (pulse code modulation) format for playback through PCM audio hardware. The playback rate of DSD sources can be further altered by using resampling to chosen rate. Thus, playback rates from 32 to 1536 kHz are possible. Used bit depth is either maximum supported by the playback hardware or lower in case such is requested.
Also multichannel loudspeaker delay- and level-processing is supported in both converted and native modes.

1.2. Architecture and components

HQPlayer Desktop consists of client-server architecture. Both components have some GUI elements, but the server side – the actual HQPlayer Desktop -application contains mostly elements for maintaining configuration and basic playback functionality based on drag-and-drop or simple source folders and audio inputs. Client implements actual

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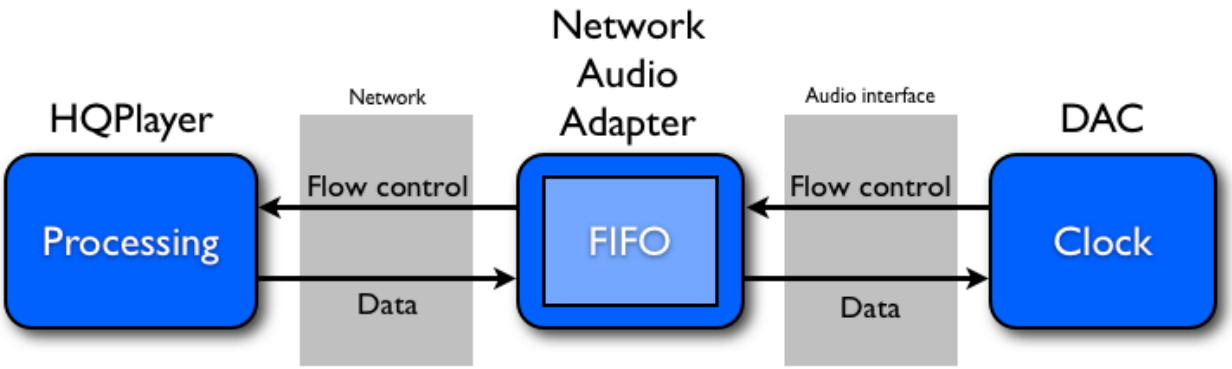
player GUI, but not any standalone playback functionality.

1.3. Starting up the application

First the HQPlayer Desktop main application (server) needs to be started and running somewhere in the network. Then HQPlayer Client can be started either in the same machine, or in some other machine in the network. Available HQPlayer servers (HQPlayer Desktop or HQPlayer Embedded) are shown in the client. Or in case automatic discovery fails, or is not supported, hostname or IP address of the server can be entered in order to connect to the server.

1.4. Network Audio

Network Audio is a way to have remote audio adapters and DACs integrated seamlessly with the player application. All the audio processing is performed at the player application side, and then streamed asynchronously over the network for reproduction.

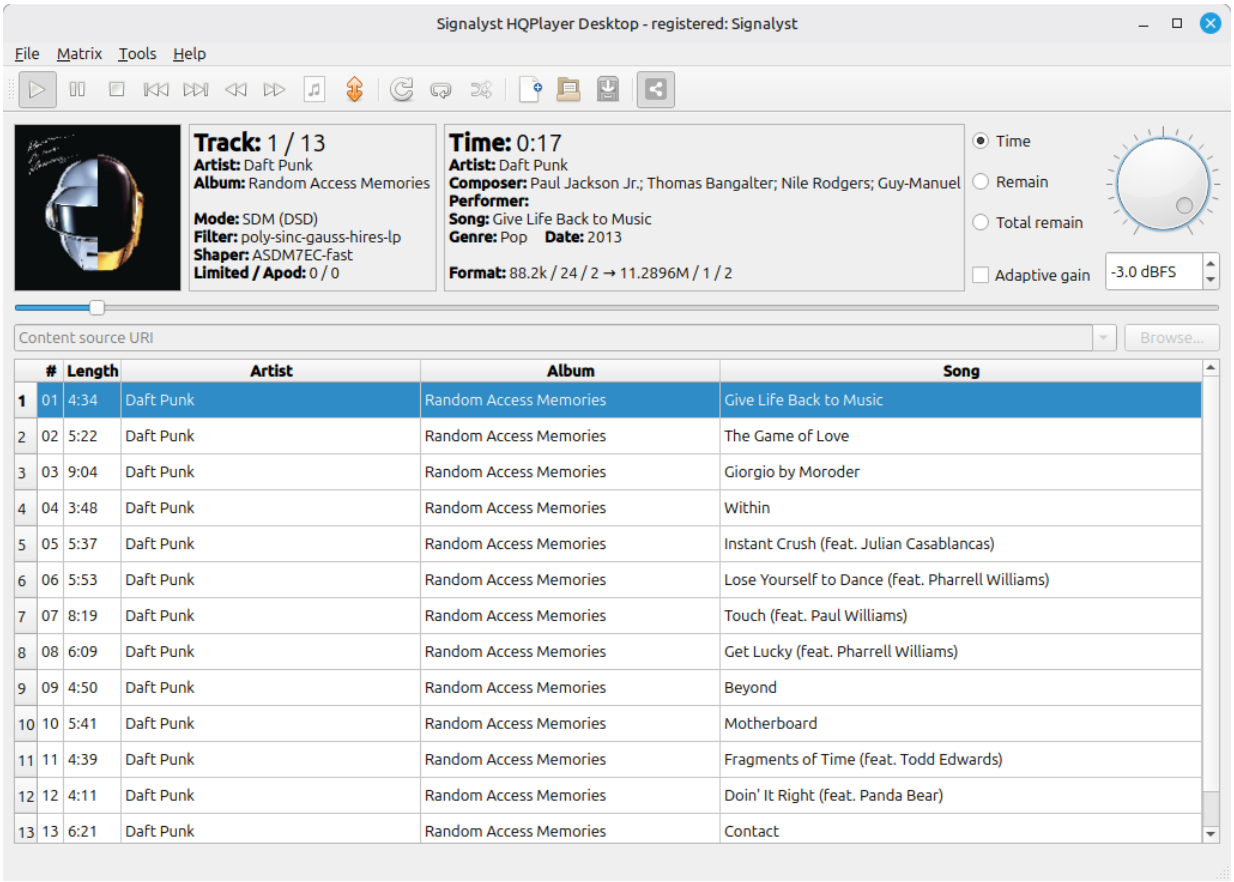


Network Audio system

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2. Main screen

When the application is first started up, main screen is displayed.



Main screen

User interface also supports standard multimedia keys and equivalent remote controls. Tracks, directory trees and playlist files can be added to the current playlist by drag-and-drop from outside of the application.
Note! *In case you experience clicks/pops between DSDIFF/DSF tracks, creating a playlist for the tracks enables special code to reset the modulation state. Playback won't be gapless in this mode.*

2.1. Album / track art

When content includes album or track artwork, it is shown when a track is being played.

2.2. Track display

Current track number and total number of tracks on a transport is shown on this display. For CD, this is the normal track number. For files, track numbering is constructed per directory basis based on embedded track number or file name sorting order. For preferred order, file names should begin with correct zero-prefixed track number.

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2.3. Song display

For CD, this field is used only to display track numbers.
When playing back files, metadata is shown. If metadata is not available a file name is shown.

2.4. Time display

This display shows the selected time information. By default, it is the time from beginning of the track. Other possible values are time from end of the track and total time from end of the album (transport).

2.5. Limiting

HQPlayer contains automatic soft-knee limiter that will reduce volume on respective channels when output exceeds 0 dB level. When limiting is triggered, the “Limited” counter is incremented and volume knob color changes to red. When such happens, it is best to reduce output level and keep the lowered volume level to avoid further triggers of limiting. ***Note!*** *Limiting sensitivity depends on selected filter and upsampling ratio.*

2.6. Apodization

For PCM source content, HQPlayer can detect need for an apodizing filter. This is based on detected errors that originate from the recording ADC or mastering tools. Every time such occasion is detected in source content, the counter is incremented. When such content is detected, especially with higher counts, use of apodizing filter is recommended. This detection is not absolute, but can be used as guidance to decide when non-apodizing filter shouldn’t be used. There is no harm in using apodizing filter for content that doesn’t need one. But there is harm using non-apodizing filter for content that would need one.

2.7. Mode display

Selected time display mode is indicated here. Shown values are “time” for the time from beginning of the track, “remain” for the time from end of the track and “total remain” for the time from end of the album. Display mode can be changed by clicking this box.

2.8. Adaptive gain mode

Apply adaptive gain settings during playback based on metadata or library analysis data. Note that in case metadata includes positive gain values, you may need to provide extra headroom using volume control setting.

2.9. Control buttons

Control buttons can be used to control playback. Clicking “Play” begins the process and “Stop” will stop the process. For normal file playback other buttons can be used as well.

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2.10. Convolution

When this button is depressed, convolution processing is completely bypassed. When this button is pressed, convolution engine is active and the configured impulse responses will be used to process the signal before resampling. Convolution can be enabled and disabled during the playback. This applies only when simple convolution engine is used. Matrix processor settings can be changed on the fly using matrix profiles.
Note! *When source material sampling rate differs from the impulse sampling rate, impulse responses will be scaled to the source material's sampling rate. This can have a huge impact on CPU/GPU load, and with large impulse responses will require significant amount of CPU/GPU processing power.*

2.11. Phase inversion

Absolute phase can be inverted in cases where volume control is available.

2.12. Repeat and Random playback

Current tracklist/playlist can be repeated and played back in random order. It is also possible to repeat a single track. When output is to a file, resulting output matches the playback.

2.13. Playlist management

Clicking the “Clear playlist” -button clears the internal playlist transport. If some other transport (such as album) is active, this doesn't have visible effect until new playlist is created. Playlist can be also loaded and saved using corresponding buttons.
When other transport than playlist is selected, playlist is still in memory. Transport can be switched back to the playlist by clicking the “Activate playlist” button.

2.14. Volume control

Processing volume can be controlled through volume multimedia keys or remote control, or by operating this adjustment wheel. Selected dither/noise-shaping algorithm has significant impact on quality of this adjustment.
When using any resampling, maximum recommended volume level is -3 dBFS to avoid inter-sample overloads, and in case material contains digital clipping/limiting.
Note! *High oversampling ratios can generate high inter-sample overs. Overloading the delta-sigma modulator in SDM mode will also cause audible noises. It is therefore recommended to keep software volume at max -3 dB setting or lower when using PCM to SDM conversion to avoid overloads, especially if the source material contains digital clipping. Maximum modulation depth is monitored and when necessary limited to 50% to retain best possible output fidelity.*

2.15. Position/seek bar

Shows relative playback position of a currently playing song while also allowing seeks to arbitrary position within the song.

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2.16. Source content entry

Source content edit / drop-list contains reference to the source location. References are URI's, just like web browser address bar. When source is a "transport" like a folder or audio device, all content is assumed to belong together and is processed gapless. When source is not a folder, but instead individual files for example dropped on HQPlayer window, pop-prevention processing is employed between DSD tracks, assuming the tracks are independent.
To source audio from a device, "audio:" and "input:" URI schemas are used.
To read content from a CD, "cd:" URI schema is used.
Browse button next to the edit box can be used to browse for a source folder.
Note! *On macOS, in order to enable audio input to HQPlayer, permission to access "Microphone" needs to be granted to HQPlayer in System Preferences → Security & Privacy. On Windows, similarly Desktop applications must be granted access to "Microphone" in order to use input feature.*

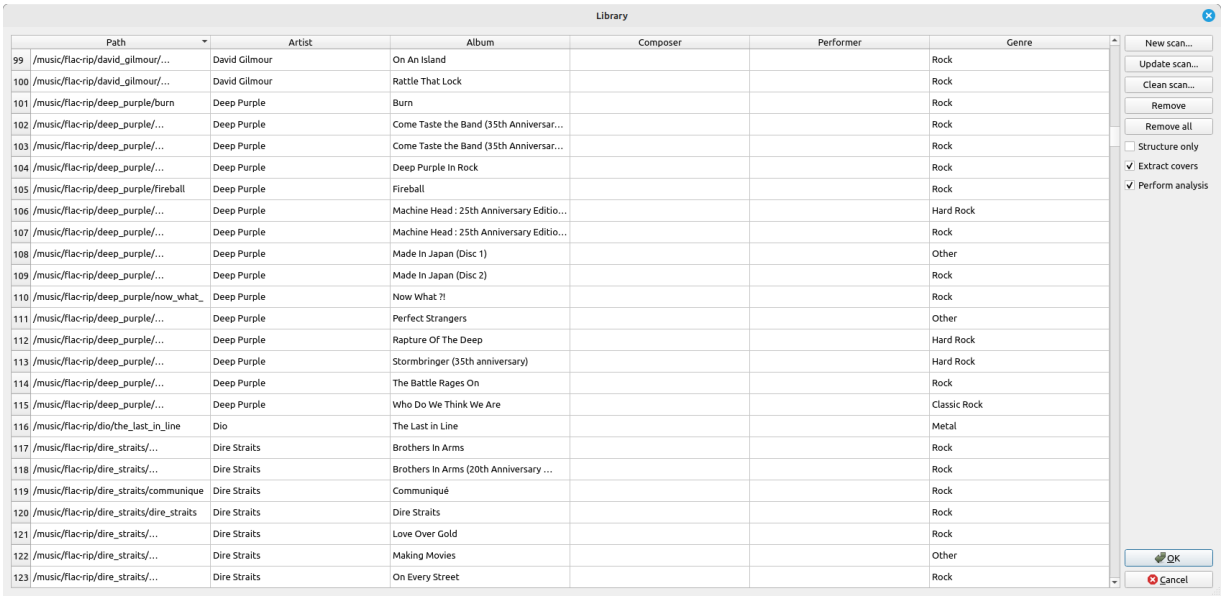
2.17. Playback content table

Table shows content loaded in transport for playback.

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3. Library management

To edit library, open the “File” menu and select “Library...”. Following dialog will be shown.



Library editing dialog

List shown on the left is the list of album locations available on the transport selector. Each path is intended to represent an album consisting of files of same number of channels.
To remove an album from the listing, select the album path and click “Remove” button. Once you are done with the editing, select either “OK” to save your changes or “Cancel” to discard the changes.
It is also possible to edit album metadata by double-clicking a cell.
To scan only for new content and skip the already known material, click “New scan…” and point to the root directory tree node where to start the scan process. This is faster on subsequent scans especially when content analysis is used.
To rescan an entire directory tree or part of it, click “Update scan...” to browse and select base directory of the tree you wish to add and click “OK”. All the nodes of the directory tree with recognized content will be added to the list of available albums and new/changed cover art is recognized. Already known entries are automatically ignored (except for adding missing metadata). If you wish to refresh all the library information, select “Clean scan...” instead. Select “Structure only” to extract metadata solely from the directory tree structure instead of metadata embedded in files. If detached cover art is missing, but embedded cover art exists, “Extract covers” will extract embedded cover art to a detached one making it available in the cover flow view. Cover art cannot be extracted if “Structure only” is selected, as this omits looking into embedded metadata information.
Metadata for each path is loaded when available. If metadata is not available within the file, it is constructed from the full file path, assumed of being in format Artist/Album/Song.

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Note! *Each directory is assumed to contain only one type of supported playback files, the first recognized type will be used and other types of files within the directory will be ignored.*
To clear the list, select “Remove all”, confirmation dialog will appear before the list is cleared.
To perform full analysis on the music content to determine properties such as true peak and RMS levels, loudness and loudness range and error count indicating need for apodizing filter, select also “Perform analysis”.
Note! *Performing analysis will likely take significant amount of time, depending on speed of the computer and storage, and size of the library.*

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4. Settings

To change program's device settings, open the “File” menu and select “Settings...” (on macOS “Preferences...” from the application menu).

Content of the device selection depends on the selected back-end.
For WASAPI and CoreAudio driver types, used audio endpoint (device) can be selected by using the “Device” selection which lists all the available audio endpoints in addition to the default endpoint, which is the one selected in Windows Control Panel or macOS Audio MIDI Setup for the default audio output.
For ASIO driver type, used audio device can be selected by using the “Device” selection which lists all the available ASIO devices. “Ch. offset” can be used to select the channel which is considered to be the first in channel mapping (0-based).
For Network Audio driver type, list of remote audio devices is shown on the “Device” selection. This is always combination of the NAA device plus the hardware device ID.
On Linux, ALSA audio endpoint (device) can be selected by using the “Device” selection which lists all the available hardware audio endpoints.

DSD content can be transferred to/from the audio device by packing it into suitable PCM container, select “DoP” to use the DoP v1.1 standard. The “2wire” setting enables dual-wire channel bonding to achieve 2x higher sampling rates for both PCM and DoP-based DSD on those DACs that support this feature.
Short buffer setting reduces size of audio FIFO buffer to half. This reduces amount of delay for example for volume control. But it also increases likelihood of audio drop-outs.
Channel mapping is following (regardless of driver type):

0. Front Left
1. Front Right
2. Front Center
3. Low Frequency (LFE)
4. Back Left
5. Back Right
6. Side Left
7. Side Right

Length of the hardware audio buffer (in milliseconds) can be changed by using “Buffer time” selection. It is recommended to use “Driver default”, unless audio drop-outs are experienced. When “Driver default” is used, the audio driver defines length of the buffer. In case of WASAPI, this is more or less fixed value of 10 ms. In case of ASIO, this can be usually controlled through ASIO control panel. With ASIO backend it is recommended to leave the value to “Default” and adjust buffer size from the driver Control Panel instead, if available. When ASIO Control Panel is not available, the HQPlayer setting can be used, but it will be capped to the range supported by the driver, and in some cases this means the setting not having any effect if the driver

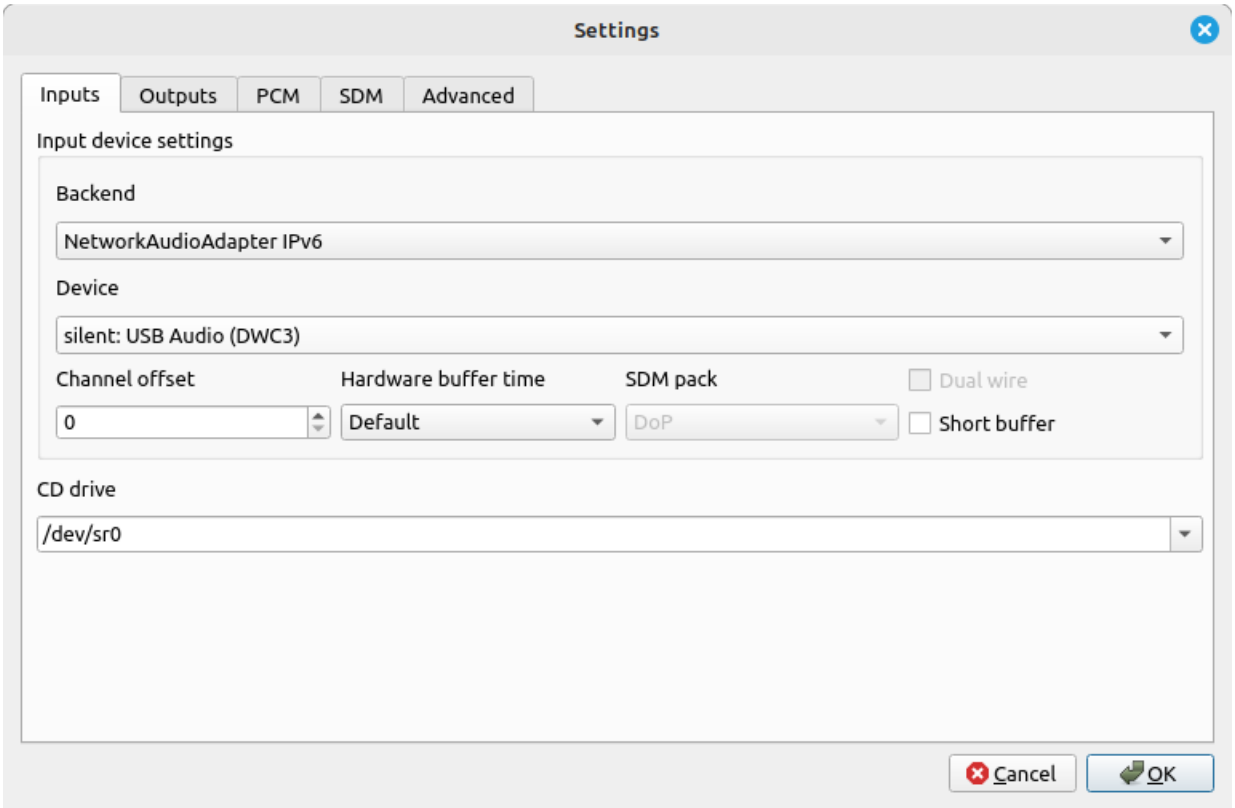
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doesn't allow adjustments at all. Values between 10 and 100 ms are most recommended.

Note! *Due to limitations of ASIO API, ASIO drivers cannot be used for both input and output. Combination of ASIO and WASAPI or ASIO and NAA can be used instead.*

4.1. Inputs

On Inputs tab, different input device related settings can be changed.



Settings dialog, Inputs tab

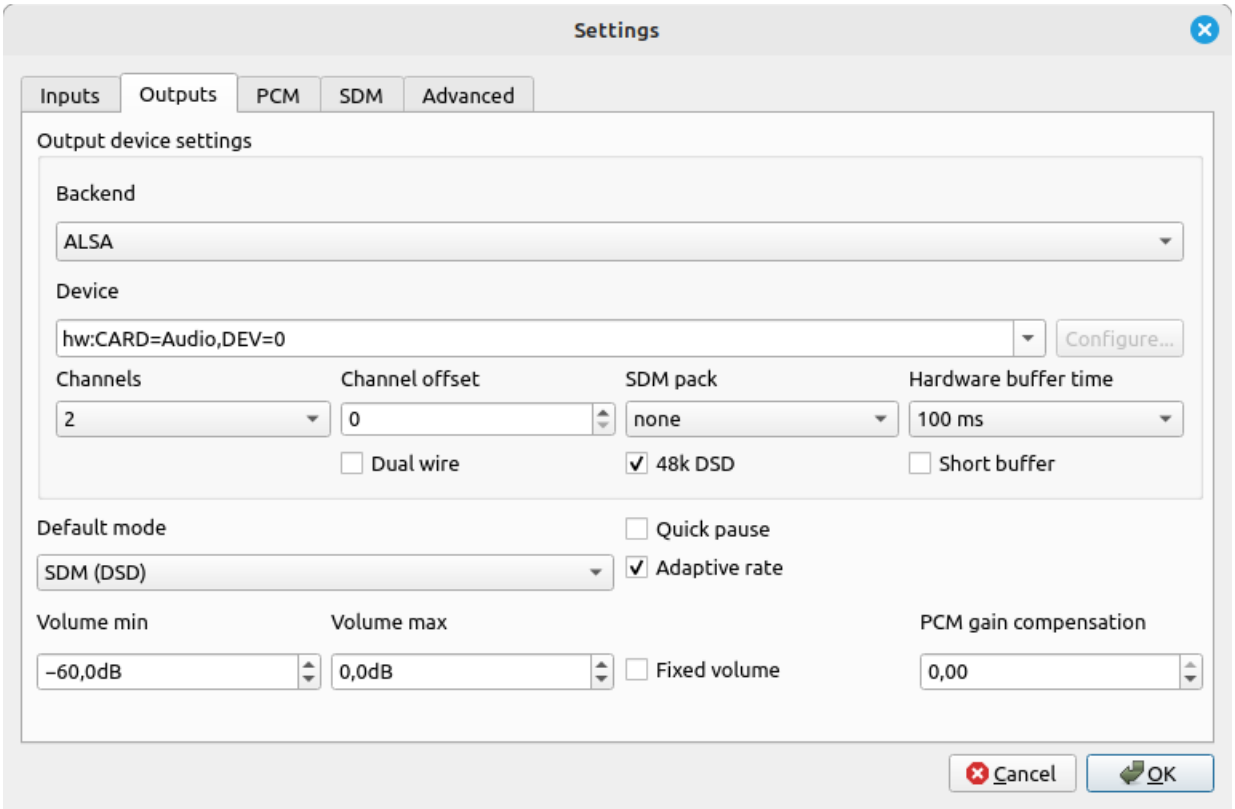
If you don’t have any input device, select “[none]” as the input backend.

On Windows, drive letter for the CD drive can be changed from the “CD drive” selection. On Linux, device node for the CD drive can be entered, this can be typically a symlink such as “/dev/cdrom”. On macOS, device node can be discovered using terminal command “*drutil status*”, where for example if *Name* is “/dev/disk5” the device node to be entered is “/dev/**r**disk5”.

4.2. Outputs

On Outputs tab, audio output device related settings can be changed.

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Settings dialog, Output tab

Number of output channels can be chosen from “Channels” selection, possible choices range from “2” for stereo to 128 output channels, primarily for complex matrix processing cases.

Default output mode

Selects default output mode. When set to “PCM”, all content is played as PCM output. When “SDM (DSD)” is selected, all content is played as SDM output. When “[source]” is selected, PCM content is played as PCM and DSD content is played as SDM. However, using “[source]” usually leads to sub-optimal result with either format since only very few DACs have separate true PCM (R2R) and SDM conversion sections inside. In most cases only either one of the options is optimal for the DAC.

Quick pause

Quick pause changes pause operation to play only basic silence pattern. In some cases this reduces delay when pressing pause. But can cause audible glitches especially when DAC is directly connected to a power amp without intermediate analog volume control.

Adaptive rate

Adaptive output rate makes automatic output rate selection pick sampling rates based on two different rules; grayed selects default or lower rate based on filter and DAC capabilities, while checked selects rate that is multiple of the same base sampling rate as the source. When the setting is not checked, specified output sampling rate is fixed.

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Volume control

It is also possible to configure adjustment range of the volume control. When both values are set to zero (0), volume control is bypassed completely. However, this is not suitable for normal cases since it will cause inter-sample overs and thus limiting either at HQPlayer side or at the DAC side. Fixed volume setting at desired level can be achieved by setting both min and max to the same value.
Fixed volume check box enables special fixed volume with optimized level setting that has enough headroom for most typical inter-sample overs. This is recommended setting when HQPlayer's digital volume control is not needed due to use of some external volume control method. When this setting is checked, volume is roughly -3 dB and when grayed, volume is roughly -6 dB. -6 dB setting is good for music content where normal -3 dB doesn't provide enough headroom, such as heavily clipped content.

PCM gain compensation

Due to nature of DSD, many DACs have different output levels for 0 dBFS PCM vs 0 dB DSD. PCM gain compensation can be used to compensate for this level difference.

DAC type	Compensation (dB)
Asahi Kasei Micro (AKM), AK4490	-3.5
Asahi Kasei Micro (AKM), AK4493	-1 to -3.5 depending on reference level settings
Asahi Kasei Micro (AKM), AK4499	-4.1
Asahi Kasei Micro (AKM), AK4499EX	-3 depending on settings
Cirrus Logic	-3
ESS Sabre	0
ROHM BD34352	-5.25
Texas Instruments / Burr-Brown	Depends on selected AFIR, refer to the datasheet for details
Holo Audio	-6
Denafrips	-3.2
Merging Hapi	-0.6

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4.3. Combo Settings

When “Combo” is selected as an output backend, device selection list is disabled and “Configure…” button is enabled. When the button is clicked, following dialog is shown.

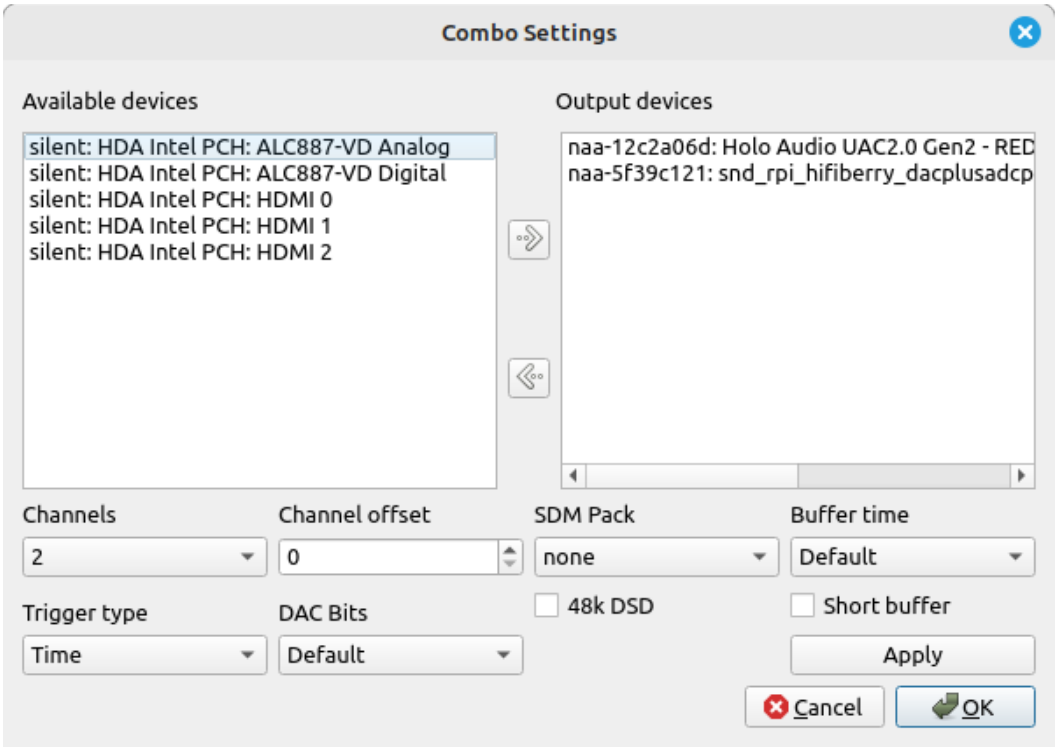


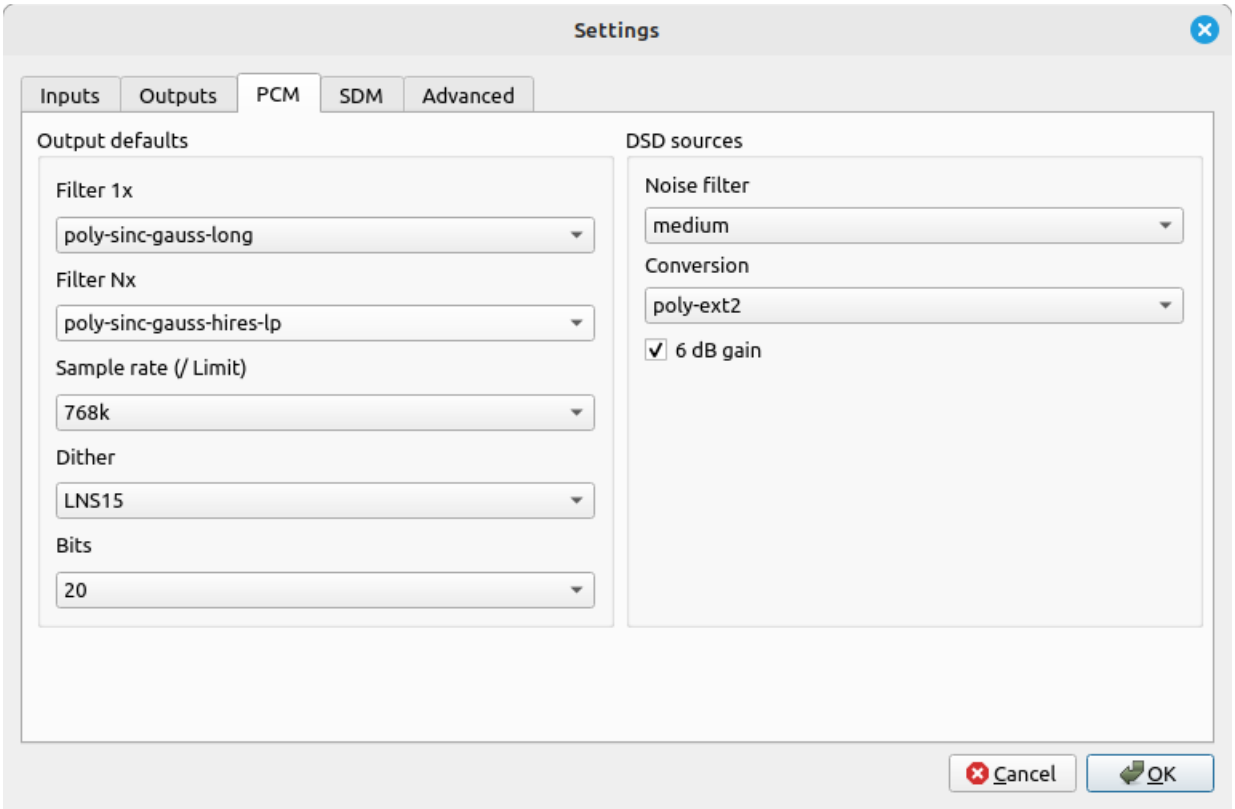
Figure 1: Combo settings dialog

This dialog allows configuration of combined output device that consists of multiple regular networked output devices. Number of channels of combo interface is sum of number of channels of all sub-devices. Channel mapping follows order of the sub-devices. To configure settings of a particular output device, select it from the list, adjust settings and click Apply.

4.4. PCM

On PCM tab, settings for PCM output mode can be controlled.

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Settings dialog, PCM tab

For filter settings, please see a separate section below.

Sample rate / Limit

Output sample rate request, or limit can be set in “Sample rate”. This is the maximum output rate HQPlayer will use.
This selection can be used to switch between supported hardware sample rates. Available choices depend on selected transport and resampling filter type. When default output mode is set to “[source]”, or “Adaptive output rate” is enabled, this is only a default and upper limit for output rate, specific rate is selected by the playback engine during playback time depending on available rates and filter conversion capabilities. When default output mode is PCM, and “Adaptive output rate” is not enabled, the selected rate is static output rate.
Note! *When “none” is selected as resampling algorithm, output sampling rate is adjusted based on source file's sampling rate. For DSD sources, this is 1/16th of the DSD rate.*

Noise-shaping / dither

This selection can be used to switch between different word-length reduction algorithms. It is always recommended to use at least TPDF dither.

NS/Dither	Description
none	No noise-shaping or dithering, only rounding. Mostly suitable for testing cases together with *none* filter selection, where bit-perfect output is needed. *Not recommended.*

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NS/Dither	Description
NS1	Simple first order noise-shaping. Sample values are rounded and the quantization error is shaped such way that the error energy is pushed to the higher frequencies. Suitable mostly for 176.4/192 kHz upsampling.
NS4	Fourth order noise-shaping. Similar in shape as “shaped” dither. Suitable for all rates equal or higher than 88.2 kHz.
NS5	Fifth order noise-shaping. Fairly aggressive noise-shaping designed for 8x and 16x rates (352.8/384/705.6/768 kHz). Not recommended for rates below 192 kHz. (Especially good for PCM1704 at those highest rates.)
NS9	Ninth order noise-shaping. Very aggressive noise-shaping designed especially for 4x rates (176.4/192 kHz) and recommended for these rates. (Especially good for older 16-bit 4x rate capable multibit-DACs like TDA154x etc.)
LNS15	15th order linear noise shaping. Smooth noise-shaping slope designed especially for 16x rates (705.6/768 kHz) and recommended for these higher PCM rates. Can be also used at 8x rates (352.8/384 kHz), but not recommended for rates below.
RPDF	Rectangular Probability Density Function. White noise dither. Computationally light weight, but only suitable for 24-bit or higher output hardware.
TPDF	Triangular Probability Density Function. This is the industry standard simple dither mechanism. Suitable for any rate and recommended if playback rate is 44.1/48 kHz. *Recommended for general purpose use.*
Gauss1	Gaussian Probability Density Function. High quality flat frequency dither recommended for rates at or below 96 kHz where noise-shaping is not suitable.
shaped	Shaped dither. Noise used in this dither has shaped frequency distribution to lower audibility of the dither noise. Suitable for playback rates of 88.2/96 kHz, or higher.

Note! Use of “NS1” with equipment sensitive to ultrasonic noise is not recommended.

Bits

When DAC is connected to a unidirectional interface like S/PDIF, AES/EBU or I2S it is important to select correct number of bits from the “DAC bits” selection. In addition, when a DAC is connected to USB and has something else than 32-bit input resolution, it is recommended to set the actual value here.

Recommended settings for some R2R DACs are shown on the table below. Also when a suitable noise-shaper, such as LNS15, NS9 or NS5 is used in combination with high output rates, linearity errors inherent to all R2R DACs can be corrected. This will lower distortion of especially low level signals and reduce zero-crossing distortions.

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DAC model	Bits
Holo Audio Cyan 2, Spring, Spring 2, Spring 3, May	20
Denafrips	19
LAiV Harmony DAC, Harmony µDAC	18

DSD sources

These settings control DSD to PCM conversion algorithms.

Different types of noise filters for PCM are provided. These reduce amount of ultrasonic noise present in the source data. Standard filtering leaves low level of ultrasonic noise. Some loudspeakers with tweeters of low power handling capability can be sensitive to this noise, especially when higher listening volumes are used. Also some poorly designed, or class-D, amplifiers can misbehave in presence of such ultrasonic content. Therefore more aggressive noise filters can be selected by using “Noise filter” drop list. These filters will also limit bandwidth available for the audio content. Following filters are supported.
When processing output rate of DSD source (assuming DSD64) is 88.2/96 kHz PCM, use of extra noise filtering in addition to “standard” is less important, since most of the noise will be cut out. When processing output rate of DSDIFF or DSF source is 44.1/48 kHz, extra noise filtering in addition to “standard” is not needed and will actually just reduce playback quality.

PCM Noise filter	Description
standard	Standard noise filter will be applied. *Recommended.*
low	Similar to standard, but has lower corner frequency and results in almost flat noise profile in ultrasonic range. *Recommended.*
high-order	High order noise filter designed for material created with high order modulators. *Recommended.*
sac	Sliding average converter.
wec	Weighted element converter.
wec2	Weighted element converter. Optimized to closely match DSD/SACD specification. Non-ringing linear-phase. *Recommended.*
slow-lp	Slow roll-off linear-phase filter.
slow-mp	Slow roll-off minimum-phase filter.

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PCM Noise filter	Description
medium	Medium roll-off linear-phase filter designed to be as gentle as possible while passing minimal amount of out-of-band noise. *Recommended.*
medium-high	Medium roll-off high rate linear-phase filter designed to be as gentle as possible while passing minimal amount of out-of-band noise. Use this instead of “medium” when “none” is selected as PCM Conversion. *Recommended.*
fast-lp	Fast roll-off linear-phase filter.
fast-mp	Fast roll-off minimum-phase filter.
brickwall	Brickwall filter that doesn’t pass any out-of-band noise. Very steep linear phase filter. Cut-off at 25 kHz for DSD64, 50 kHz for DSD128, 100 kHz for DSD256, 200 kHz for DSD512 and 400 kHz for DSD1024.

Type of SDM → PCM conversion can be selected from the “Conversion” drop list. Following conversion types are supported.

PCM Conversion	Description
traditional	Traditional recursive conversion algorithm. Minimizes amount of ringing by using slow roll-off filters.
single-steep	Single-pass conversion algorithm with steep roll-off.
single-short	Single-pass conversion algorithm with normal roll-off. Optimized tradeoff between ringing and wide frequency response.
sinc-S	Linear-phase adaptive length sharp roll-off and high attenuation single pass conversion algorithm. Number of taps is 65536.
sinc-M	Linear-phase million-tap sharp roll-off and high attenuation single pass conversion algorithm.
poly-lp	Linear-phase single-pass conversion algorithm.
poly-mp	Minimum-phase single-pass conversion algorithm.
poly-short-lp	Linear-phase slow roll-off single-pass conversion algorithm. *Recommended.*

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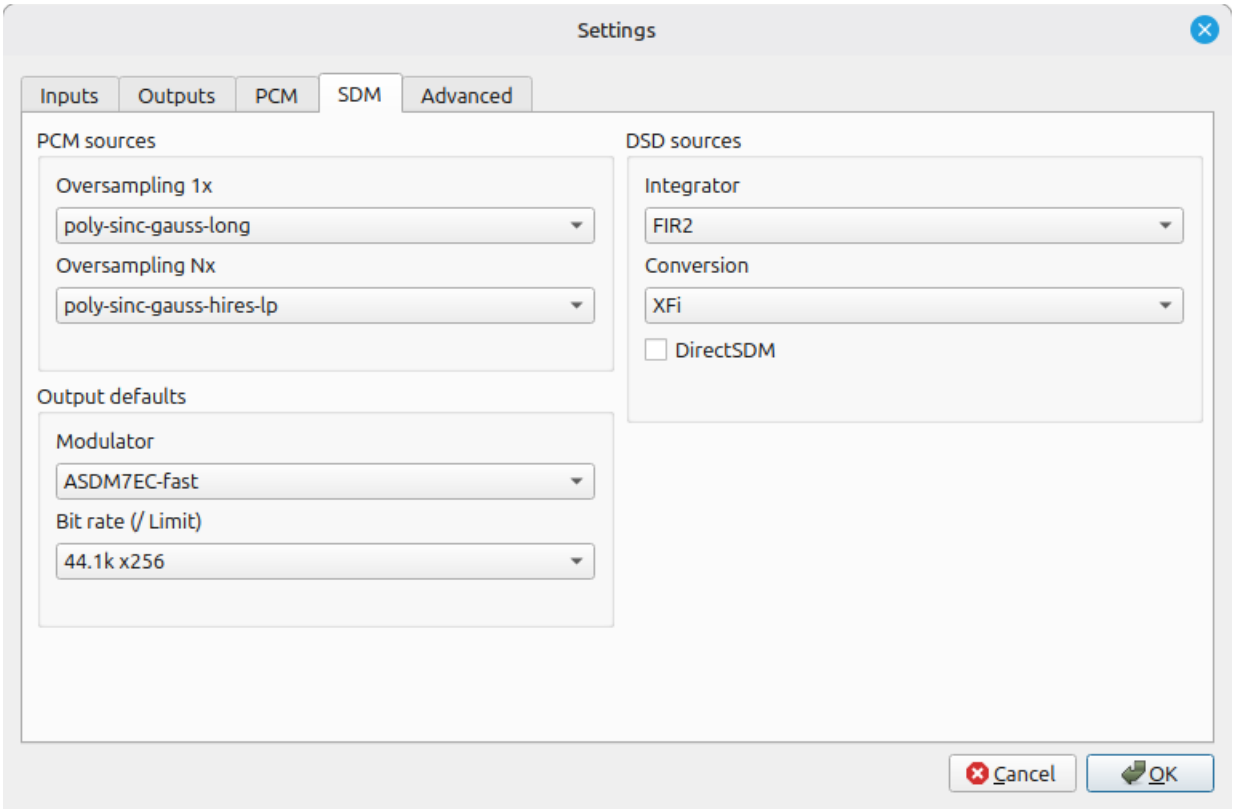
PCM Conversion	Description
poly-short-mp	Minimum-phase slow roll-off single-pass conversion algorithm.
poly-xtr	Linear-phase extreme roll-off and attenuation single-pass conversion algorithm.
poly-xtr-short	Linear-phase extreme roll-off and attenuation single-pass conversion algorithm.
poly-ext2	Linear-phase extended frequency response sharp roll-off and high attenuation single-pass conversion algorithm.
poly-gauss-long	Linear-phase Gaussian extremely high attenuation single-pass conversion algorithm. Optimal time-frequency response.
none	No decimation, intermediate output rate is equal to source DSD rate.

DSDIFF or DSF file should typically have 6 dB of headroom on the signal level. By selecting “6 dB gain” check box, 6 decibels of gain is applied, removing this headroom from the converted signal. This way the normal playback level reaches that of normal PCM. However, this may cause overloads with some source material and may require extra attenuation using volume control.

4.5. SDM

On SDM tab, settings for SDM output mode can be controlled.

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Settings dialog, SDM tab

For oversampling settings, see separate section below.

Modulator

Allows selection of the delta-sigma modulator used to produce SDM output.

Modulator	Description	Generation
DSD5	Rate adaptive fifth order one-bit delta-sigma modulator.	1st
DSD5v2	Revised fifth order one-bit delta-sigma modulator.	2nd
DSD5v2 256+fs	Revised fifth order one-bit delta-sigma modulator optimized for rates >= 10.24 MHz.	2nd
DSD5EC	Rate adaptive fifth order one-bit delta-sigma modulator with extended compensation.	4th
ASDM5	Adaptive fifth order one-bit delta-sigma modulator.	3rd
ASDM5EC	Adaptive fifth order one-bit delta-sigma modulator with extended compensation.	4th
ASDM5ECv2	Second generation of ASDM5EC with minor improvements.	5th
ASDM5ECv3	Third generation of ASDM5EC with minor improvements.	6th

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Modulator	Description	Generation
ASDM5EC-ul	Adaptive fifth order one-bit delta-sigma modulator with extended compensation. Ultralight version.	7th
ASDM5EC-light	Adaptive fifth order one-bit delta-sigma modulator with extended compensation. Light version.	7th
ASDM5EC-fast	Adaptive fifth order one-bit delta-sigma modulator with extended compensation. Transient and load optimized version.	7th
ASDM5EC-super	Adaptive fifth order one-bit delta-sigma modulator with extended compensation. Super version.	7th
ASDM5EC-ul 512+fs	Adaptive fifth order one-bit delta-sigma modulator with extended compensation. Optimized for 512x and higher rates. Ultralight version.	7th
ASDM5EC-light 512+fs	Adaptive fifth order one-bit delta-sigma modulator with extended compensation. Optimized for 512x and higher rates. Light version.	7th
ASDM5EC-fast 512+fs	Adaptive fifth order one-bit delta-sigma modulator with extended compensation. Optimized for 512x and higher rates. Transient and load optimized version.	7th
ASDM5EC-super 512+fs	Adaptive fifth order one-bit delta-sigma modulator with extended compensation. Optimized for 512x and higher rates. Super version.	7th
DSD7	Seventh order one-bit delta-sigma modulator.	2nd
DSD7 256+fs	Seventh order one-bit delta-sigma modulator optimized for rates >= 10.24 MHz.	2nd
ASDM7	Adaptive seventh order one-bit delta-sigma modulator.	3rd
ASDM7EC	Adaptive seventh order one-bit delta-sigma modulator with extended compensation.	4th
ASDM7ECv2	Second generation of ASDM7EC with minor improvements.	5th
ASDM7ECv3	Third generation of ASDM7EC with minor improvements.	6th
ASDM7EC-ul	Adaptive seventh order one-bit delta-sigma modulator with extended compensation. Ultralight version.	7th
ASDM7EC-light	Adaptive seventh order one-bit delta-sigma modulator with extended compensation. Light version.	7th
ASDM7EC-fast	Adaptive seventh order one-bit delta-sigma modulator with extended compensation. Transient and load optimized version.	7th
ASDM7EC-super	Adaptive seventh order one-bit delta-sigma modulator with extended compensation. Super version.	7th

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Modulator	Description	Generation
ASDM7EC-ul 512+fs	Adaptive seventh order one-bit delta-sigma modulator with extended compensation. Optimized for 512x and higher rates. Ultralight version.	7th
ASDM7EC-light 512+fs	Adaptive seventh order one-bit delta-sigma modulator with extended compensation. Optimized for 512x and higher rates. Light version.	7th
ASDM7EC-fast 512+fs	Adaptive seventh order one-bit delta-sigma modulator with extended compensation. Optimized for 512x and higher rates. Transient and load optimized version.	7th
ASDM7EC-super 512+fs	Adaptive seventh order one-bit delta-sigma modulator with extended compensation. Optimized for 512x and higher rates. Super version.	7th
AMSDM7 512+fs	Special adaptive seventh order “pseudo-multi-bit” modulator optimized for rates above >= 20.48 MHz.	3rd
AMSDM7EC 512+fs	Special adaptive seventh order “pseudo-multi-bit” modulator with extended compensation for rates >= 20.48 MHz.	4th
AHM5EC4B	Fifth order 4-bit hybrid modulator with extended compensation. Optimized for rates >= 40.96 MHz. Utilizes 1024x and higher output rates for improved performance with most bit-perfect D/A conversions.	8th
AHM7EC4B	Seventh order 4-bit hybrid modulator with extended compensation. Optimized for rates >= 40.96 MHz. Utilizes 1024x and higher output rates for improved performance with most bit-perfect D/A conversions.	8th
AHM5EC8B	Fifth order 8-bit hybrid modulator with extended compensation. Optimized for rates >= 40.96 MHz. Bandwidth optimized to provide enough flat noise floor bandwidth for practically all hires content.	8th
AHM7EC8B	Seventh order 8-bit hybrid modulator with extended compensation. Optimized for rates >= 40.96 MHz. Bandwidth optimized to provide enough flat noise floor bandwidth for practically all hires content.	8th

Fifth order modulators are more suitable for DACs that have simple analog reconstruction filters. Seventh order modulators provide better technical performance, but also put more demands on the DAC's analog reconstruction filter. Typically this means that fifth order modulators suit DACs that have one switching element while seventh order modulators have potential for better performance on DACs that have multi-element switching arrays. DSD* modulators are fixed configuration ones while ASDM* modulators are adaptive in various ways based on source signal. For ESS Sabre based DACs, fifth order modulators are recommended. For most other DACs, seventh order modulators are optimal.

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Integrator

There are three types of delta-sigma integrators available for different SDM → SDM remodulation schemes. These affect mostly frequency and phase response at highest frequencies. Stated frequencies apply for DSD64 source rate, these frequencies scale as function of source sampling rate.

SDM Integrator	Description
IIR	Normal IIR type integrator structure. 50 kHz audio bandwidth re DSD64.
IIR2	IIR type integrator structure designed to minimize residual noise. 25 kHz audio bandwidth re DSD64.
IIR3	High order IIR type integrator structure. 30 kHz audio bandwidth re DSD64.
FIR	Weighted FIR type integrator structure.
FIR2	Weighted FIR type integrator structure. 50 kHz audio bandwidth re DSD64.
FIR-bl	FIR type integrator structure with band-limiting. 24 kHz audio bandwidth re DSD64 with complete cut by 45 kHz.
FIR-bw	FIR type integrator structure with brickwall band-limiting. 21.5 kHz audio bandwidth re DSD64 with complete cut by 30 kHz.
CIC	Cascade comb type integrator structure.

Conversion

There are different options for SDM → SDM rate conversions. These affect frequency aperture that is assumed to contain useful signal in addition to increasing noise shaping noise. For example piano doesn’t contain high frequency harmonics and for such case “narrow” is suitable, while close miked percussions usually contain high level high frequency content and there “wide” may be more suitable. While “XFi” is suitable for all cases. Default is “XFi”.

SDM Conversion	Description
wide	Wide bandwidth signal
narrow	Narrow bandwidth signal
XFi	Extreme fidelity medium bandwidth

DirectSDM

DirectSDM setting disables all processing when source is DSD content and output format is SDM to a DSD-device or file.

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Note! *Enabling DirectSDM will disable volume control and set PCM volume to fixed -3 dBFS value.*

4.6. Filter / Oversampling selection

This selection can be used to switch between PCM resampling / oversampling filters. This selection has an impact on available hardware sampling rates. Different variants of “poly-sinc” are the most recommended by the author. Filter/oversampling selection for “1x” rates covers source sampling rates below 50 kHz, so called base rates. Filter selection for “Nx” rates covers everything else above the 1x rates. Apodizing filter should be used *at least* when “Apod” counter increments to higher than 10 during any single track.
Where description says “*Only suitable for highest technical quality source materials*”, it means material where Apod counter stays below 10 for the duration of entire track. Quality ratings are not absolute, but a relative guidance regarding technical quality of filter’s output vs other filters in the table.

Filter	Description	Special focus, Quality	Genre	Ratio	Apod
none	No sample rate conversion happens. Only sample depth is changed as needed.	1/5		1:1	N
IIR	This is analog-sounding filter, especially suitable for recordings containing strong transients, long post-ringing is a side effect (not usually audible due to masking). A really steep IIR filter is used. This filter type is similar to analog filters and has no pre-ringing, but has a long post-ringing. Small amount of pass-band ripple is also present. Medium attenuation. IIR filter is applied in time domain.	2/5	Pop, rock, jazz, blues	Integer	Y
IIR2	This is analog-sounding filter, especially suitable for recordings containing strong transients, long post-ringing is a side effect (not usually audible due to masking). A steep IIR filter is used. This filter type is similar to analog filters and has no pre-ringing, but has a long post-ringing. Medium attenuation. No passband ripple. IIR filter is applied in time domain.	4/5	Pop, rock, jazz, blues	Integer	Y

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FIR	Typical “oversampling” digital filter, generally suitable for most uses (slight pre- and post-ringing), but best on classical music recorded in a real world acoustic environment. This is the most ordinary filter type, usually present in hardware. This filter is applied in time-domain. It has average amount of pre- and post-ringing.	3/5	Classical	Integer	Y
asymFIR	Asymmetric FIR, good for jazz/blues, and other music containing transients recorded in real world acoustic environment. Otherwise same as FIR, but with a shorter pre-ringing and longer post-ringing. Modifies phase response, but not as much as minimum phase FIR.	3/5	Jazz, blues	Integer	Y
minphaseFIR	Minimum phase FIR, good for pop/rock/electronic music containing strong transients such as drums and percussion and where recording is made in a studio using multi-track equipment. No pre-ringing, but somewhat long post-ringing.	3/5	Pop, rock, electronic	Integer	Y
FFT	Technically good steep “brickwall” filter, but might have some side effects (pre-ringing) on material containing strong transients. This filter is similar to FIR, but it is applied in frequency-domain and is quite efficient from performance point of view while having rather long impulse response. Length of this filter can be configured separately in Settings dialog.	4/5	Any depending on length	2^x	Y
poly-sinc-lp	Linear phase polyphase sinc filter. Very high quality linear phase resampling filter, can perform most of the typical conversion ratios. Good phase response, but has some amount of pre-ringing. See “FIR” for further details.	Space 4/5	Classical	Any	½

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poly-sinc-mp	Minimum phase polyphase sinc filter, otherwise similar to poly-sinc. Altered phase response, but no pre-ringing. See “minphaseFIR” for further details.	Transients 4/5	Jazz, blues	Any	½
poly-sinc-shrt-lp	Otherwise similar as poly-sinc, but shorter pre- and post-ringings at the expense of filtering quality (not as sharp roll-off).	Space, transients 3/5	Jazz, blues, electronic	Any	½
poly-sinc-shrt-mp	Minimum phase variant of poly-sinc-shrt. Otherwise similar to poly-sinc-mp, but shorter post-ringing. Most optimal transient reproduction.	Transients 3/5	Pop, rock	Any	½
poly-sinc-long-lp	Otherwise similar as poly-sinc, but longer pre- and post-ringings with improved filtering quality (faster roll-off).	Space 4/5	Classical	Any	Y
poly-sinc-long-ip	Intermediate phase version of poly-sinc-long, with small pre-ringing and longer post-ringing with improved filtering quality (faster roll-off).	Space, transients 4/5	Jazz, blues, electronic	Any	Y
poly-sinc-long-mp	Minimum phase variant of poly-sinc-long. Otherwise similar to poly-sinc-mp, but longer post-ringing with improved filtering quality (faster roll-off).	Transients 4/5	Pop, rock	Any	Y
poly-sinc-hb	Linear-phase polyphase half-band filter with steep roll-off and high attenuation. Only suitable for highest technical quality source materials.	4/5	Any	Any	N
poly-sinc-hb-xs	Extremely short linear-phase polyphase half-band filter with slow roll-off and low attenuation. Only suitable for highest technical quality source materials.	2/5	Pop, rock	Any	N
poly-sinc-hb-s	Short linear-phase polyphase half-band filter with slow roll-off and average attenuation. Only suitable for highest technical quality source materials.	3/5	Pop, rock	Any	N

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poly-sinc-hb-m	Medium linear-phase polyphase half-band filter with average roll-off and medium attenuation. Only suitable for highest technical quality source materials.	3/5	Any	Any	N
poly-sinc-hb-l	Long linear-phase polyphase half-band filter with fast roll-off and high attenuation. Only suitable for highest technical quality source materials.	4/5	Classical, jazz, blues	Any	N
poly-sinc-ext	Linear phase polyphase sinc filter with sharper roll-off and somewhat lower stop-band attenuation, while being roughly equal length to poly-sinc.	3/5		Integer	½
poly-sinc-ext2	Linear phase polyphase sinc filter with sharp roll-off and high stop-band attenuation for extended frequency response while completely cutting off by Nyquist frequency. Optimal frequency response and harmonic structure. For SDM outputs, processing is two stages with minimum 16x intermediate rate.	Timbre 5/5	Any	Any	Y
poly-sinc-ext2-short	Linear phase polyphase sinc filter with slow roll-off and high stop-band attenuation for extended frequency response. Optimal frequency response and harmonic structure. For SDM outputs, processing is two stages with minimum 16x intermediate rate.	Timbre 4/5	Pop, rock	Integer up	½
poly-sinc-ext2-medium	Linear phase polyphase sinc filter with fast roll-off and high stop-band attenuation for extended frequency response while completely cutting off by Nyquist frequency. Optimal frequency response and harmonic structure. For SDM outputs, processing is two stages with minimum 16x intermediate rate.	Timbre 4/5	Any	Any	Y

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poly-sinc-ext2-long	Linear phase polyphase sinc filter with very fast roll-off and very high stop-band attenuation for extended frequency response while completely cutting off by Nyquist frequency. Optimal frequency response and harmonic structure. For SDM outputs, processing is two stages with minimum 16x intermediate rate.	Timbre 5/5	Any	Any	Y
poly-sinc-ext2-xla	Very steep 8 times longer version of poly-sinc-ext2-long. For SDM outputs, processing is two stages with minimum 16x intermediate rate.	Timbre 5/5	Classical	Any	Y
poly-sinc-ext2-xl	Very steep 8 times longer non-apodizing version of poly-sinc-ext2-long. Only suitable for highest technical quality source materials. For SDM outputs, processing is two stages with minimum 16x intermediate rate.	Timbre 5/5	Classical	Any	N
poly-sinc-ext2-hires-lp	Linear-phase polyphase sinc filter for HiRes content, with very high stop-band attenuation. Also suitable for playback of lossy compression such as MP3 or MQA.	Timbre 5/5	Any	Any	Y
poly-sinc-ext2-hires-ip	Intermediate-phase polyphase sinc filter for HiRes content, with very high stop-band attenuation. Also suitable for playback of lossy compression such as MP3 or MQA.	Timbre 5/5	Any	Any	Y
poly-sinc-ext2-hires-mp	Minimum-phase polyphase sinc filter for HiRes content, with very high stop-band attenuation. Also suitable for playback of lossy compression such as MP3 or MQA.	Timbre 5/5	Any	Any	Y

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poly-sinc-mqa/mp3-lp	Linear phase polyphase sinc filter optimized for playing back MQA or MP3 encoded content in order to clean up high frequency noise added by the MQA or MP3 encoding. Also suitable for upsampling PCM sources of 88.2 kHz or higher sampling rate, especially for hires PCM recordings of 176.4 kHz or higher sampling rate. Very short ringing. Early slow roll-off.	Transients 4/5	Classical, jazz, blues	PCM: Integer up SDM: Any	Y
poly-sinc-mqa/mp3-mp	Minimum phase variant of poly-sinc-mqa.	Transients 4/5	Pop, rock	PCM: Integer up SDM: Any	Y
poly-sinc-xtr-lp	Linear phase polyphase sinc filter with extreme roll-off and attenuation.	Timbre 5/5	Classical	Any	½
poly-sinc-xtr-mp	Minimum phase polyphase sinc filter with extreme roll-off and attenuation.	Timbre 5/5	Jazz, blues	Any	½
poly-sinc-xtr-short-lp	Short linear phase polyphase sinc filter with extreme roll-off and attenuation.	Timbre, transients 5/5	Electronic, jazz, blues, pop, rock	Any	Y
poly-sinc-xtr-short-mp	Short minimum phase polyphase sinc filter with extreme roll-off and attenuation.	Timbre, transients 5/5	Pop, rock	Any	Y
poly-sinc-gauss-short	Short Gaussian polyphase sinc filter. Optimal time-frequency response. For SDM outputs, processing is two stages with minimum 16x intermediate rate.	Transients 3/5	Electronic, jazz, blues, pop, rock	Integer up	½
poly-sinc-gauss-medium	Gaussian polyphase sinc filter. Optimal time-frequency response. For SDM outputs, processing is two stages with minimum 16x intermediate rate.	Transients, timbre 4/5	Any	Any	Y

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poly-sinc-gauss-long	Long Gaussian polyphase sinc filter with extremely high attenuation. Optimal time-frequency response. For SDM outputs, processing is two stages with minimum 16x intermediate rate.	Transients, timbre, space 5/5	Any	Any	Y
poly-sinc-gauss-xla	Apodizing extra long Gaussian polyphase sinc filter with extremely high attenuation. Optimal time-frequency response. For SDM outputs, processing is two stages with minimum 16x intermediate rate.	Transients, timbre, space 5/5	Classical, jazz, blues	Any	Y
poly-sinc-gauss-xl	Extra long Gaussian polyphase sinc filter with extremely high attenuation. Optimal time-frequency response. For SDM outputs, processing is two stages with minimum 16x intermediate rate.	Transients, timbre, space 5/5	Classical, jazz, blues	Any	N
poly-sinc-gauss-hires-lp	Linear-phase Gaussian filter for HiRes content with extremely high attenuation. Optimal time-frequency response. Also suitable for playback of lossy compression such as MP3 or MQA.	Transients, timbre, space 5/5	Any	Any	Y
poly-sinc-gauss-hires-ip	Intermediate-phase Gaussian filter for HiRes content with extremely high attenuation. Optimal time-frequency response. Also suitable for playback of lossy compression such as MP3 or MQA.	Transients, timbre, space 5/5	Any	Any	Y
poly-sinc-gauss-hires-mp	Minimum-phase Gaussian filter for HiRes content with extremely high attenuation. Optimal time-frequency response. Also suitable for playback of lossy compression such as MP3 or MQA.	Transients, timbre, space 5/5	Any	Any	Y
poly-sinc-gauss-halfband	Linear-phase halfband Gaussian filter. Slightly leaky around Nyquist, but extremely high attenuation. Only suitable for highest technical quality source materials.	Transients, timbre, space 4/5	Any	Any	N

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poly-sinc-gauss-halfband-s	Short linear-phase halfband Gaussian filter. Leaky around Nyquist, but high attenuation. Only suitable for highest technical quality source materials.	Transients, timbre, space 3/5	Any	Any	N
poly-sinc-aclean-gauss	Single stage adaptive cleanup Gaussian polyphase sinc filter. Designed specifically for maximal ADC error correction and distortion reduction. Recommended for 1x and Nx.	Transients, timbre, space 5/5	Any	Any up	Y
poly-sinc-aclean-ext2	Single stage adaptive cleanup Extended 2 polyphase sinc filter. Designed specifically for maximal ADC error correction and distortion reduction. Recommended for 1x and Nx.	Timbre, space 5/5	Classical, jazz, blues	Any up	Y
poly-sinc-aclean-xtr	Single stage adaptive cleanup Extreme polyphase sinc filter. Designed specifically for maximal ADC error correction and distortion reduction. Recommended for 1x and Nx.	5/5		Any up	Y
ASRC	This is a special type of filter, slightly similar to FIR, but with a possibility of asynchronous operation for conversions from any rate to any other rate. Computationally heavy and not recommended.	2/5		Any	N
polynomial-1	Polynomial interpolation. No apparent pre- or post-ringing. Frequency response rolls off slowly in the top octave. Poor stop-band rejection and will thus leak fairly high amount of ultrasonic distortion. These types of filters are sometimes referred to as “non-ringing” by some manufacturers. *Not recommended.*	1/5		Integer up	N
polynomial-2	Similar to polynomial-1, but higher stop-band rejection and only one cycle of pre- and post-ringing. *Not recommended.*	1/5		Integer up	N

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minringFIR-lp	Minimum ringing FIR. Uses special algorithm to create a linear-phase filter that minimizes amount of ringing while providing better frequency-response and attenuation than polynomial interpolators. Performance and ringing are between polynomial and poly-sinc-short.	Transients 2/5		Integer up	N
minringFIR-mp	Minimum phase variant of minringFIR.	Transients 2/5		Integer up	N
closed-form	Closed form interpolation with high number of taps.	3/5		2^x up	N
closed-form-fast	Closed form interpolation with lower CPU load, but also lower precision. Output precision tuned to match about 24-bit PCM.	2/5		2^x up	N
closed-form-M	Closed form interpolation with one million taps.	3/5		2^x up	N
closed-form-16M	Closed form interpolation with 16 million taps.	3/5		2^x up	N
sinc-S	*sinc*-filter with adaptive number of taps. Number of taps is *4096 x conversion ratio*. Very sharp roll-off and high attenuation. Variant of *poly-sinc-ext2-xla*.	Space, timbre 4/5	Any	2^x up	Y
sinc-M	*sinc*-filter with one million taps. Very sharp roll-off and high attenuation. Variant of *poly-sinc-ext2-xla*.	Space, timbre 4/5	Classical, jazz, blues	2^x up	Y
sinc-Mx	Constant time version of sinc-M. Filter length is constant in time, with million taps at 16x PCM output rates. Variant of *poly-sinc-ext2-xla*. (*65536 x conversion ratio)*	Space, timbre 4/5	Classical, jazz, blues	2^x up	Y
sinc-MG	Gaussian constant time filter with million taps at 16x PCM output rates. Extremely high attenuation. Variant of *poly-sinc-gauss-xl*. (*65536 x conversion ratio)*	Transients, timbre, space 4/5	Classical, jazz, blues	2^x up	N

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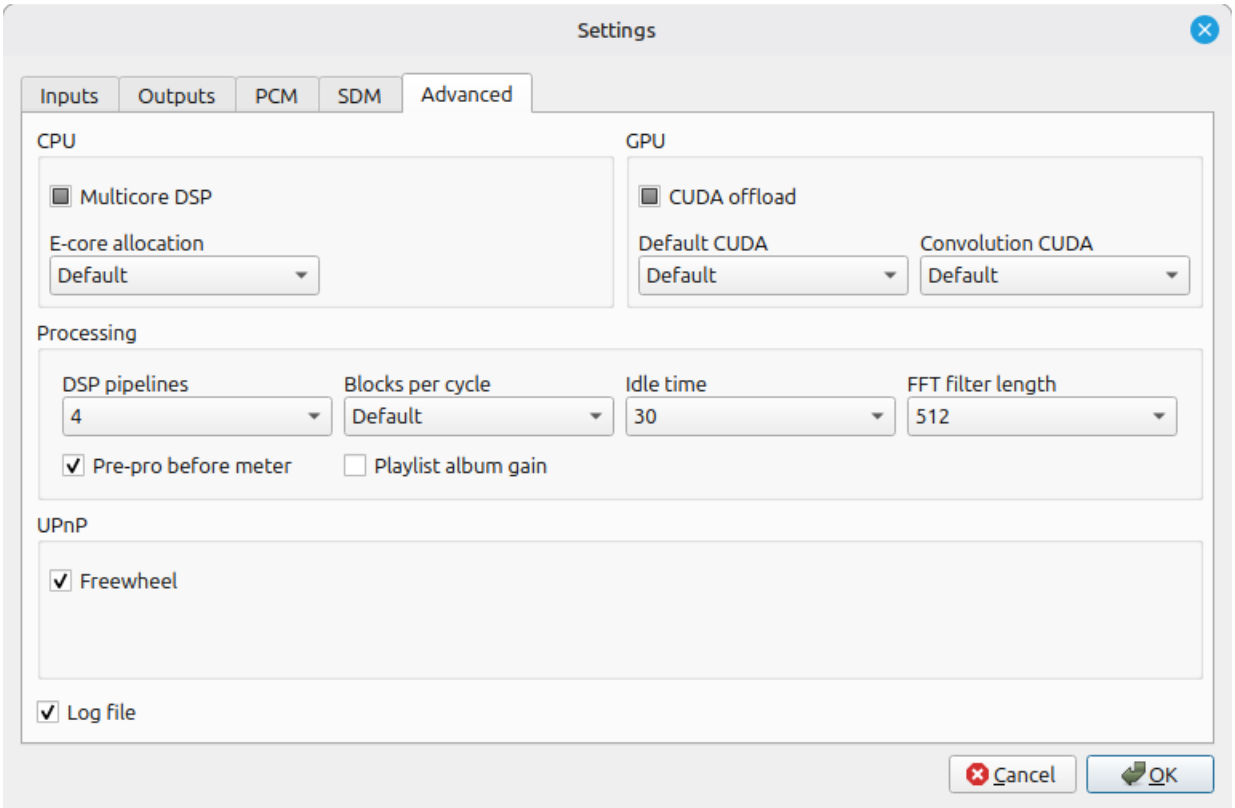
sinc-MGa	Apodizing Gaussian constant time filter with million taps at 16x PCM output rates. Extremely high attenuation. Variant of *poly-sinc-gauss-xla*. (*65536 x conversion ratio)*	Transients, timbre, space 4/5	Classical, jazz, blues	2^x up	Y
sinc-L	*sinc*-filter with adaptive number of taps. Number of taps is *131070 x conversion ratio*. Extremely sharp roll-off and average attenuation.	3/5	Classical	2^x up	N
sinc-Ls	Average attenuation *sinc*-filter with adaptive number of taps (*4096 x conversion ratio*).	2/5	Any	2^x up	N
sinc-Lm	Average attenuation *sinc*-filter with adaptive number of taps (*16384 x conversion ratio*).	2/5	Classical, jazz, blues	2^x up	N
sinc-Ll	Average attenuation *sinc*-filter with adaptive number of taps (*65536 x conversion ratio*).	3/5	Classical	2^x up	N
sinc-Lh	High attenuation *sinc*-filter with adaptive number of taps. (16384 x ratio). Significantly better quality than sinc-L at 1/8th of the load.	4/5	Classical, jazz, blues	2^x up	N
sinc-short	Short average attenuation *sinc*-filter with adaptive number of taps. For SDM outputs, processing is two stages with minimum 16x intermediate rate.	2/5	Any	Any up	N
sinc-medium	Average attenuation *sinc*-filter with adaptive number of taps. For SDM outputs, processing is two stages with minimum 16x intermediate rate.	2/5	Classical, jazz, blues	Any up	N
sinc-long	Long average attenuation *sinc*-filter with adaptive number of taps. For SDM outputs, processing is two stages with minimum 16x intermediate rate.	3/5	Classical	Any up	N
sinc-long-h	Long high attenuation *sinc*-filter with adaptive number of taps. For SDM outputs, processing is two stages with minimum 16x intermediate rate.	4/5	Classical, jazz, blues	Any	N

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*-2s	Two stage oversampling. First stage rate conversion is performed by at least a factor of 8 using the selected algorithm. And further converted to the final rate using algorithm optimized for conversion of content that has already been processed to at least 8x rate. This lowers the overall CPU load, while preserving the same conversion quality. Especially useful for highest output rates.			Same as the base filter	O

4.7. Advanced

In the Advanced tab, various advanced settings such as hardware related optimizations can be adjusted.



Settings dialog, Advanced tab

Multicore DSP

Multicore DSP increases parallelization of various DSP operations.
When the selection box is grayed, automatic detection and configuration is active and can utilize any number of cores. For best performance it is recommended to use the

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auto-detection.
When the box is cleared, processing is optimized for cases where number of cores is equal or less than number of output channels. Such as dual-core CPUs when output is stereo.
When the box is checked, processing is optimized for modern multi-core CPUs with much higher core count than number of output channels. Since this parallelization increases processing overhead, it will increase total CPU time consumption. If there are performance problems with “auto” setting, it is typically useful to try this option.

E-core allocation

On newer CPUs that have both performance and efficiency cores, efficiency cores can be allocated as offload processors instead of normal (default) use. These e-cores can be allocated either for processing resampling filters, or for a generic DSP pool for performing other tasks such as convolution.

CUDA offload

“CUDA offload” can utilize nVidia GPU to partially offload the processing from CPU to GPU. CUDA offload requires nVidia GPU with minimum Compute Capability level 5.2, 4 GB of graphics RAM and latest official nVidia drivers. When offload is enabled and suitable GPU is available, message about the offload is briefly shown at the beginning of playback of each track. When CUDA offload is enabled, also Multicore DSP should be enabled, or left at automatic setting to achieve best performance.
When CUDA offload checkbox is grayed, only convolution algorithms are offloaded to GPU.
Same or different GPU can be selected separately for performing filters and other DSP tasks, and convolution and other large operations. This allows to split the workload across two separate GPUs.

DSP pipelines

Specify number of DSP pipelines available. This is both number of matrix pipelines, and total number of possible input or output channels. Using suitably low value reduces resource consumption, such as RAM usage to some extent.

Blocks per cycle

Number of blocks to process at once. This setting can be used to fine tune CPU/GPU load to lowest possible figure. When set to “Default” the value is auto-configured based on detected amount of CPU cache etc. Processing more blocks at once reduces overhead, especially when GPU is used. While processing fewer blocks at once helps keeping most of the data in CPU cache. Higher values are better suited for processors with large cache, such as AMD 3D-series and some Intel Xeon models, or systems with high speed RAM. While smaller values are better suited for CPUs with small cache, or systems with slower RAM.

Idle time

Defines amount of time the engine is left idling after playback of current content has ended. This allows faster playback restart within the idle period.

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FFT filter length

This option specifies length of the FFT filter. Default value is 512. Length affects steepness of the filter, shorter lengths result in slower (gentler) roll-off, while higher lengths result in faster (steeper) roll-off. This setting is per each 2x cascade filter, thus it is not conversion ratio dependent.

Pre-process before metering

When enabled, pre-processing, such as 20 kHz filter, is run before metering. This allows one to see effect of the pre-process. But it may make it harder to detect when to disable 20 kHz filter again.

Playlist album gain

When enabled, album gain is used for playlist items instead of track gain. This can be desirable on playlists with multiple complete albums.

UPnP

For UPnP controlled sources, freewheel can be enabled. This allows the entire track to be (pre-)fetched to memory at full network speed, when track size is known. This can add increased resource load spike when fetch happens. It also has memory usage implications.

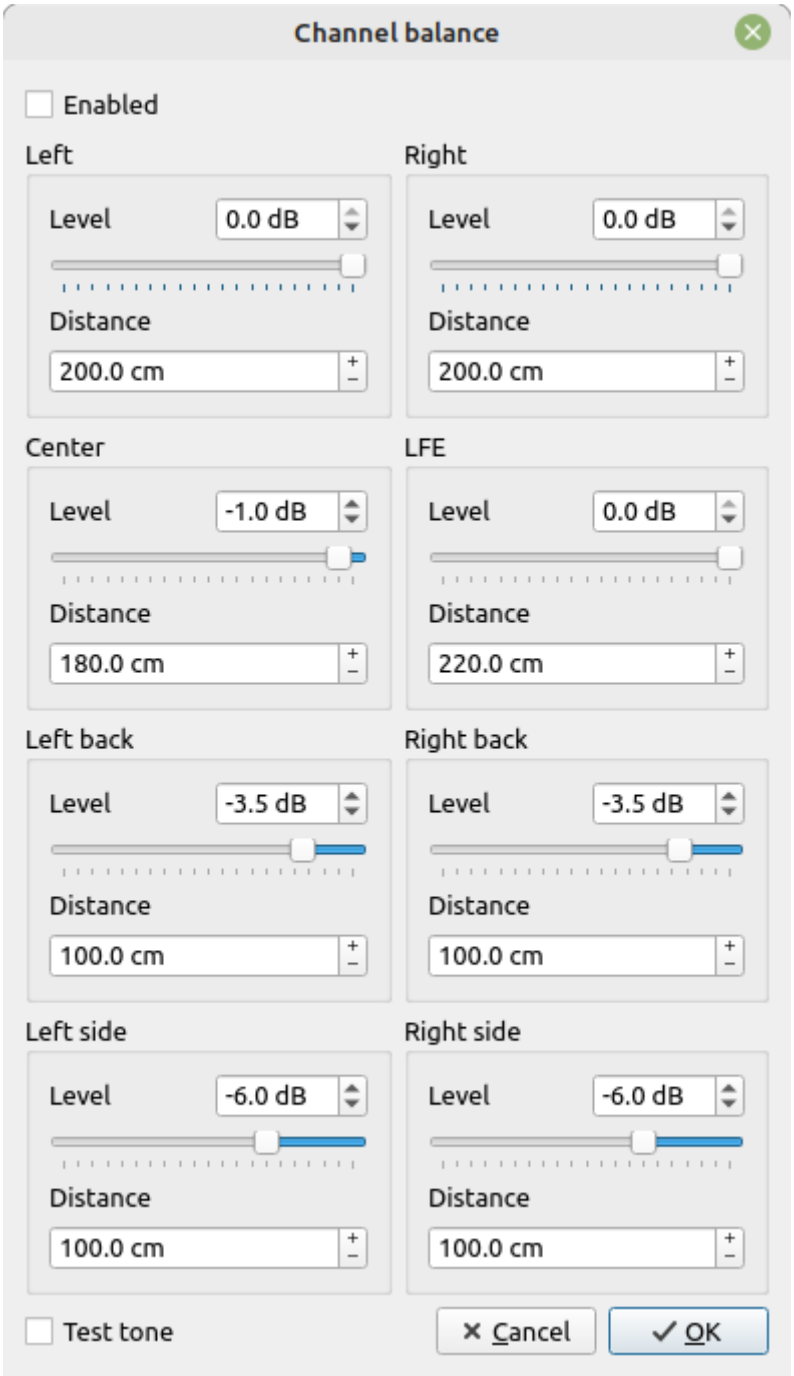
Log file

Log file can be enabled for troubleshooting purposes. After changing this setting, restart HQPlayer for the change to take full effect. Log can be viewed / exported through the Tools menu options.

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5. Channel balance

For multichannel processing, speaker/microphone levels and distances can be configured using Channel balance dialog. This dialog can be reached by opening “Tools” menu and selecting “Channel balance...”.



Speaker setup dialog

In this dialog, distance to each individual speaker can be set in centimeters. Level of the channel can be set using the volume slider and unit shown in upper right corner is in dB. Allowed adjustment range using spinbox input is wider than range of the slider.

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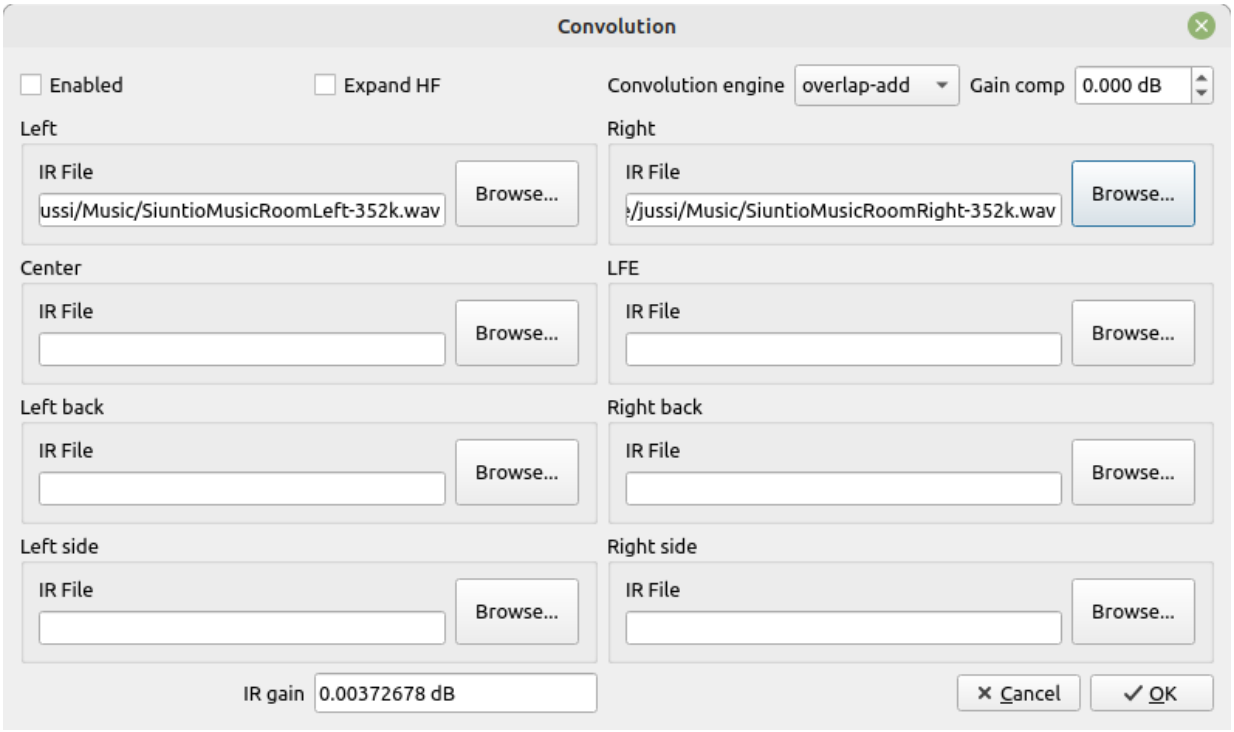
For adjusting speakers, pink noise test tone can be played by selecting the “Test tone” box. When the box is checked, tone will be played in all channels, thus making it easy to adjust all levels in such a way that they sound equal. When the box is grayed, tone will be played one channel at the time in rotating manner, making it easy to adjust all levels using an SPL meter.
Multichannel delay processing is done in target sampling rate. This increases processing accuracy as the output sampling rate is increased.
This method is suitable for simplest per-channel level adjustment and is processed in simpler and lighter way than full pipeline matrix.

Note! *Distance processing is available also for bit-perfect pass-through of DSD when Direct SDM is enabled!*

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6. Convolution engine

Convolution engine can be configured through the “Convolution” menu and selecting “Engine setup...”.
Following dialog will be shown.



Convolution engine setup dialog

When “Enabled” option is checked, convolution engine is enabled at the application level and enabled by default at the startup time. Enable this selection only after selecting suitable impulse response files and if you are certain that your files contain intended impulse response data.
Convolution algorithm can be changed from the “Convolution engine” selection. There are two possibilities, “overlap-add” consumes less CPU power and is recommended. Another alternative is “overlap-save” which consumes more CPU power.
To select impulse response files, “Browse...” button can be used. A normal file selection dialog will be shown. After a file is selected, some preliminary checks are done for the suitability is done and an error message is displayed in case of incorrect file details. Left and right channels can have independent files.
When an impulse response file is selected through “Browse...”, its estimated gain function is calculated and displayed in “IR gain” box. This can help choosing suitable value for “Gain compensation”. Also the default convolution engine can be selected.
When positive gain compensation is chosen, it is applied as negative gain when convolution is disabled from the main screen. This makes it easier to compare impact of the particular convolution setup.
When provided impulse response is lower sampling rate than source material, its high frequency response can be expanded to cover the new bandwidth by selecting “Expand HF” setting.

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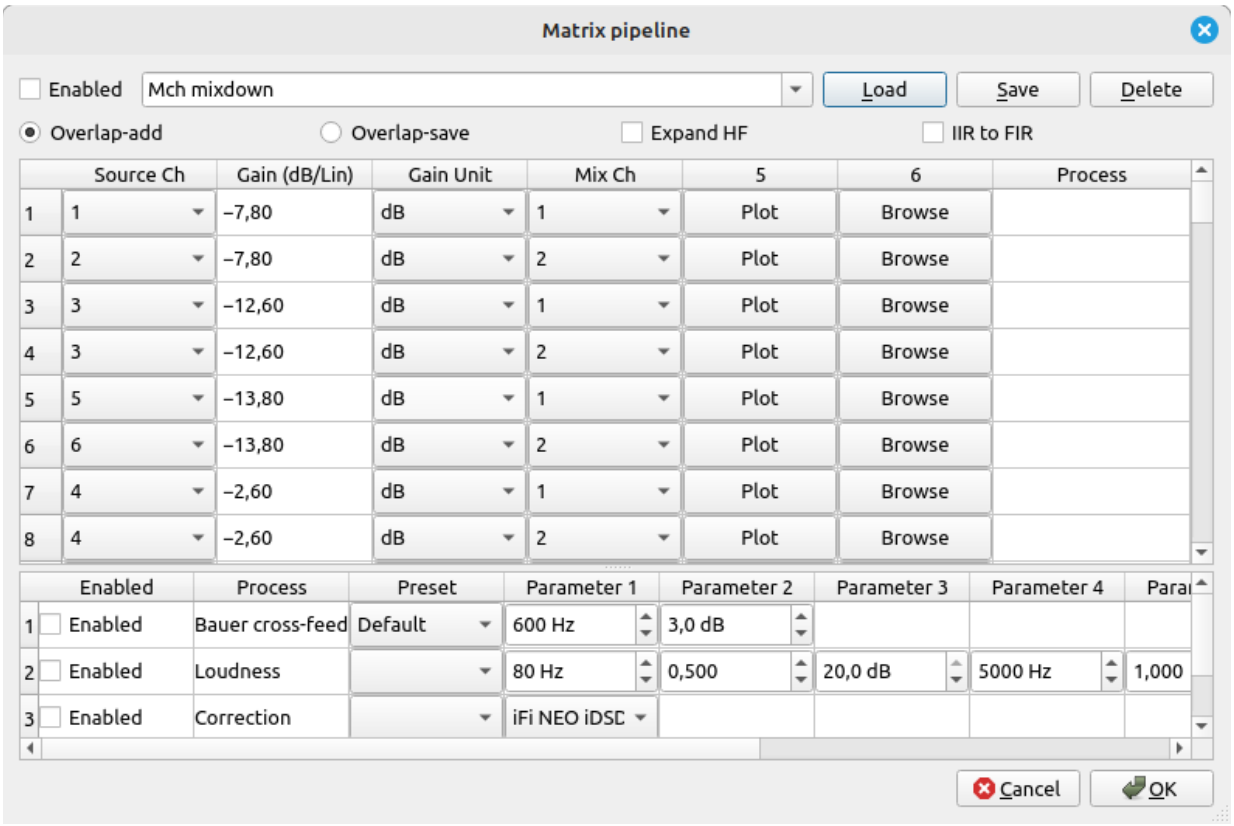
Clicking “OK” will save the setting to the configuration file and the settings are ready for use.
Convolution engine requires impulse responses to be mono RIFF (WAV) format files. If some of the channels don't need processing, or are not used, clearing the filename will disable convolution engine for those channels.
For most optimal case for all kinds of source material, use extended frequency response convolution filters with 352.8 kHz sampling rate. When such are used, Expand HF can be left disabled for all cases.
For example popular Room EQ Wizard can export suitable impulse responses after designing for “Generic” equalizer by selecting File → Export → Filters Impulse Response as WAV. Or more advanced tools like rePhase that can utilize Room EQ Wizard measurements. Expert users can also use open source DRC tool for designing even suitable full-band correction filters.
Note! *Use of long convolution filters for all eight channels of audio for hires audio files will need substantial amount of CPU/GPU processing power!*
Note! *It is not recommended to use convolution engine together with matrix processor. When matrix processor is enabled, it is recommended to do convolution there instead.*

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7. Matrix processing

Matrix processing offers a way to copy, route, filter and mix down channels with specified gains. Matrix processing consists of maximum 128 virtual channels – pipelines, number of active pipelines can be configured through advanced settings.

Note! It is not recommended to have both simple convolution engine (section 6) and matrix processor active simultaneously. If convolution is needed with matrix processing, it is recommended to configure convolution here.



Matrix configuration dialog

For example the configuration shown above can be used to mix down 5.0/5.1 channel material to stereo. As an example it also contains additional parametric equalizer giving -3 dB attenuation of main channels above 200 Hz (not needed for multichannel mix down use).

Combo box at the top is used to specify and select saved matrix configuration profiles. These profiles can be also selected remotely. Currently shown values are saved as default profile when OK is clicked.

Convolution engine applicable to filter(s) defined in *Process*, can be selected from two choices, *Overlap-add* which is the default and recommended and *Overlap-save* which is alternative method. For filters using low sampling rate, frequency response of the filter can be extended beyond Nyquist frequency of the filter's sampling rate by selecting *Expand HF*. In addition, various plugin instances with parameters can be

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specified in the *Process* item, as a comma-separated list.

Using the *IIR to FIR* it is possible to choose whether parametric EQs are converted to a convolution EQ. When the box is grayed, the conversion is a direct conversion and retains original minimum phase response. When the box is checked, the EQ filter is converted to a linear phase one.
In some cases, like GPU offloading, it may be more efficient to compute set of parametric EQs as a convolution filter instead.
Note! Conversion to linear phase will introduce some amount of unnatural pre-ringing in the audio band. This is why EQ filters are typically minimum-phase. Higher the parametric filter's Q and dB values are, more pre-ringing it will also introduce for linear-phase. So the linear phase conversion works best with rather gentle EQ setups.

The “Source Ch” specifies the channel which is used as a source for the virtual channel. “Gain” is overall gain applied for the virtual channel. And “Mix Ch” is the logical output channel. When multiple virtual channels have the same target channel, outputs of the virtual channels are mixed together to the target output channel.
“Process” can define external filter impulse response(s) WAV file for convolution, parametric equalizer specification in RoomEqWizard text output format, and parametric filter specifications (see Plugins section later). “*Browse*” button can be used to select WAV and TXT files.
Gain can be applied in both dB scale or linear scale, as selected in the corresponding column. Linear scale factors can be also negative to perform phase inversion, this allows for example M/S processing.
When choosing format for convolution filters, for most optimal case for all kinds of source material, use extended frequency response convolution filters with 352.8 kHz sampling rate. When such are used, Expand HF can be left disabled for all cases.

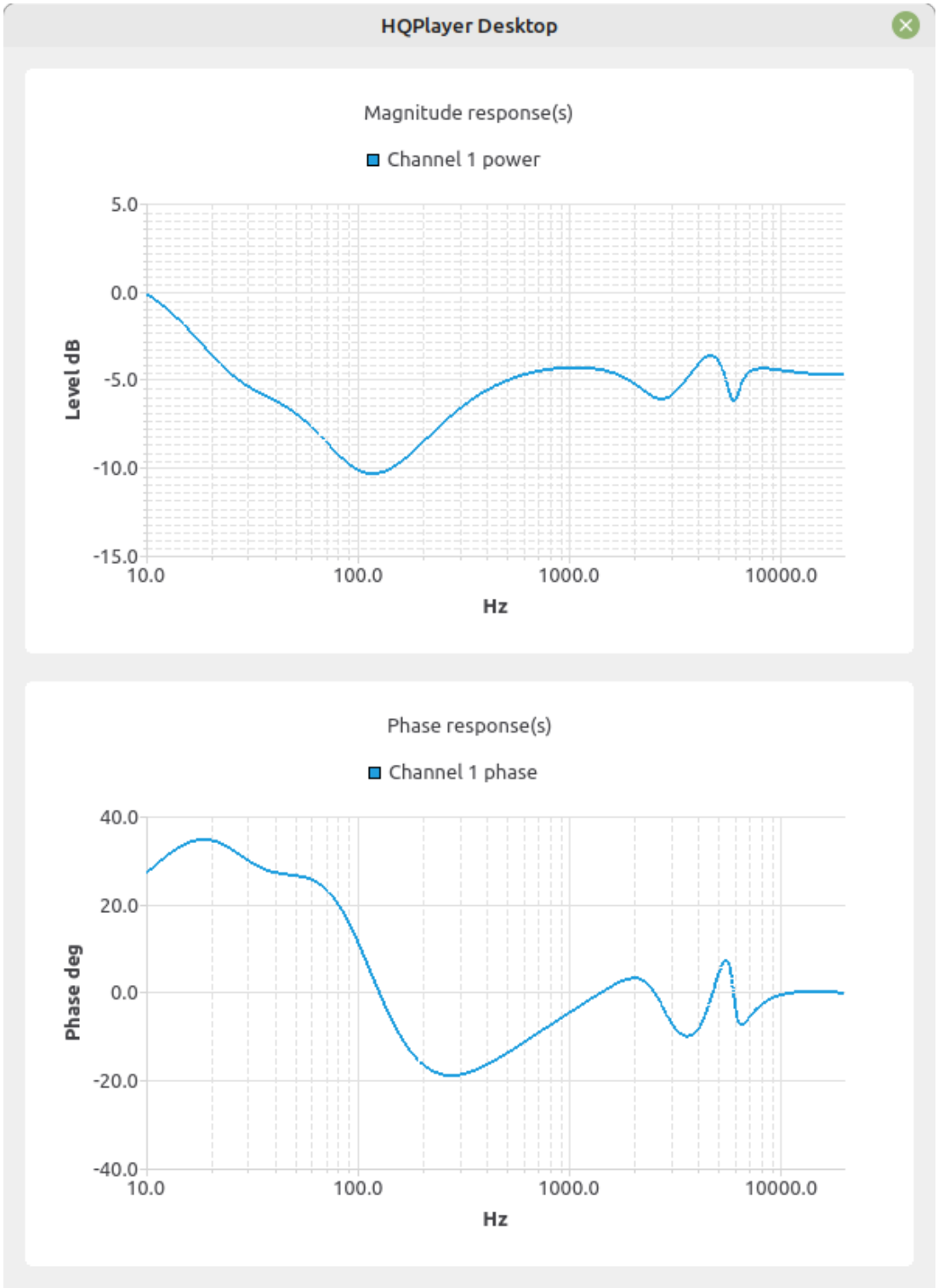
Various headphone equalization files can be found from AutoEq. Scroll down of recommended results for different headphone models. Choose the ParametricEQ txt file. This can be directly used in HQPlayer matrix processor without modifications and also includes gain compensation data. With IIR to FIR setting, this can be also processed as a FIR filter and if desired as a linear phase correction as well (default is minimum phase).

Post-processing algorithms can be enabled and configured in the table below the pipeline routing matrix. These are applied to the output mix bus, meaning output channels after the matrix processing.
Bauer cross-feed is processing for headphones that is intended to make the listening experience more natural and spacious. This is very simple model, with three presets. When custom preset is selected, cross-feed filter frequency (*Parameter 1*) and level (*Parameter 2*) can be entered respectively.
Loudness is a volume-adaptive loudness control with adjustable parameters. For bass and treble controls, the corner frequency, slope factor (see IIR plugin) and level can be adjusted (*Parameter 1* to *Parameter 6* respectively). Lower bound (*Parameter 7*) is volume setting where at or below, set maximum loudness value is reached. Higher bound (*Parameter 8*) is volume setting where at and above, loudness value reaches 0 dB.
Correction performs corrections for the output signal of selected DAC. These

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corrections are specific to a DAC model and output rate. When Combo-backend is used, there may be multiple corrections available for each corresponding DAC.

Clicking “*Plot*” button opens magnitude- and phase response plots for the corresponding pipeline. Multiple plots can be added by clicking the buttons as long as the graph dialog is open. To reset the view, close the graph dialog.



Magnitude and phase response plot

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7.1. Plugins

Each plugin description begins with plugin name, followed by colon. After the colon, semicolon separated list of parameters can be specified.

Syntax is as follows:
<plugin>:[arg1[=val]];[arg2[=val]];...;[argn[=val]]

7.2. “delay” plugin

Delay plugin provides specified amount of delay.

Argument	**Description**
s	Delay in number of samples at source rate
t	Delay in time, number of seconds
d	Delay in distance, number of meters
v	Velocity of sound, in m/s, default 343.956

7.3. “iir” plugin

IIR plugin provides parametric EQ based on IIR biquad filters.

Type	**Description**	**Arguments**
lp	Low-pass filter	f=*frequency* q=*Q* **OR** s=*slope*
lp1	1st order low-pass filter	f=*frequency*
hp	High-pass filter	f=*frequency* q=*Q* **OR** s=*slope*
hp1	1st order high-pass filter	f=*frequency*
bp	Band-pass filter	f=*frequency* q=*Q* **OR** bw=*bandwidth*
ap	All-pass filter	f=*frequency* q=*Q* **OR** bw=*bandwidth*
notch	Notch filter	f=*frequency* q=*Q* **OR** bw=*bandwidth*
peak	Peaking filter	f=*frequency* q=*Q* **OR** bw=*bandwidth* g=*gain*
lshelf	Low-shelf filter	f=*frequency* q=Q **OR** s=*slope* g=*gain*
hshelf	High-shelf filter	f=*frequency* q=Q **OR** s=*slope* g=*gain*

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Type	Description	Arguments
biquad	Raw biquad filter	b0=*b0* b1=*b1* b2=*b2* a0=*a0* a1=*a1* a2=*a2*

Where *f* is in Hz, *BW* is factor, *s* is factor (1 maximum steepness) and *g* is in dB.
Note! Syntax is case sensitive!

7.4. “riaa” plugin

RIAA plugin provides RIAA EQ curve correction for vinyl playback. Currently plugin provides only one adjustable parameter “*subsonic*” which is additional 20 Hz subsonic filter pole, with values of “*1*” (enabled) and “*0*” (disabled). For best performance and accuracy, use input sampling rate of 192 kHz or higher. Minimum recommended input sampling rate is 96 kHz.

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8. HQPlayer Client

When HQPlayer Client is started, following kind of screen is shown.



Play view

This display is optimized for touch-screens, but can be also used with a mouse or other suitable pointing device.
On top left corner is server name/address entry and list, to connect to a server, either select a server shown on the list or type in either hostname or IP address of the server.
On top right corner is list of inputs available on the server, such as CD transport and possible ADC's or digital inputs. This input field can be also used to send URLs for playback, such as internet radio playlist or stream URLs.
Cover art of the current track is shown as a background image, when available. If track doesn't have embedded cover art, album/folder cover image is used instead. If no cover image is available, default background image is shown. Information about current server, input and output buffer levels and current track is shown in the top left corner of the screen.
Track-listing is shown in top-right corner of the screen with high-light of current track and possibility to directly select a track. Volume adjustment is in the lower right corner and can be adjusted by dragging from the adjustment either up or down, or alternatively by through slider control that can be opened by double-clicking the volume control. Volume can be also operated using a mouse wheel when mouse pointer is within the control.
Star-button can be used to control favorite status of the currently playing track.
Clicking on the cover image will display associated PDF booklet when available.

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Hotkeys

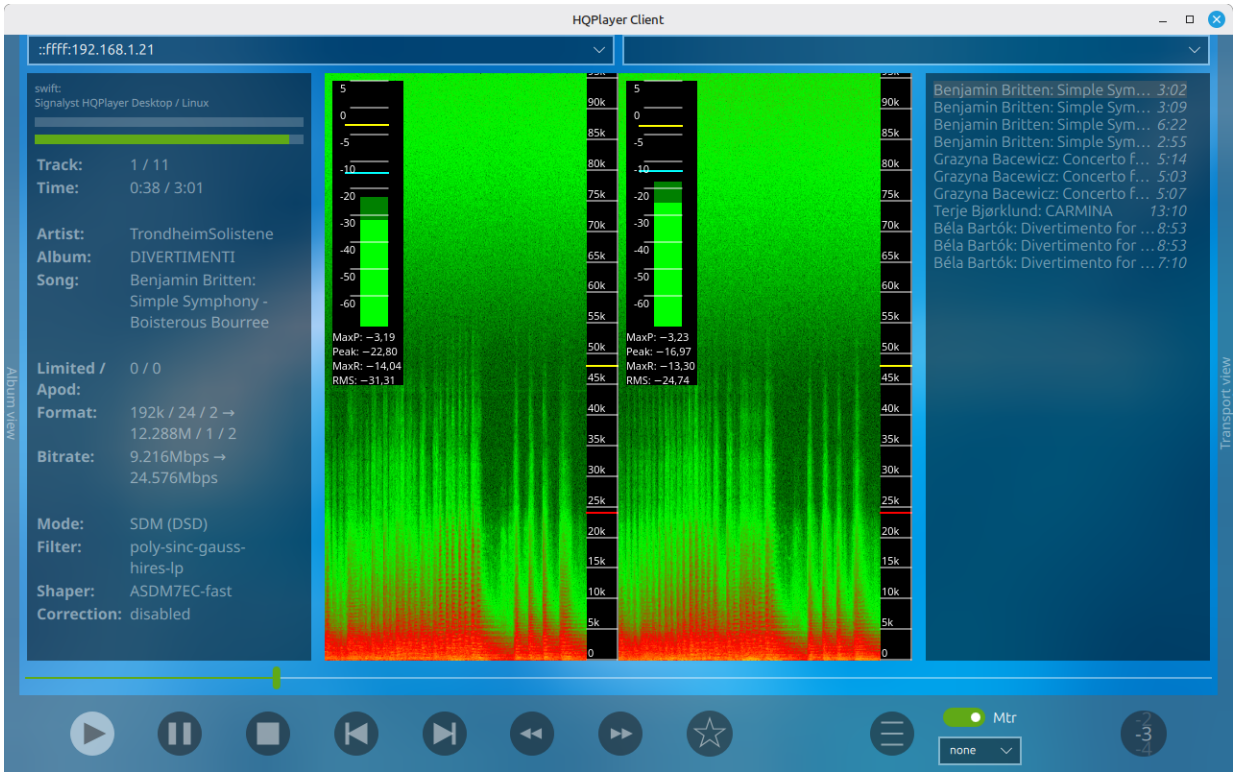
Client also supports various hotkeys for controlling playback and operation.

Key	Description
Play PlayPause Space F8	Play/pause
Stop F6	Stop
Previous Left F7	Previous track
Next Right F9	Next track
Rewind	Jump back 10 seconds
Forward	Jump forward 10 seconds
RandomPlay	Toggle random
Repeat	Toggle repeat
VolumeUp Up F12	Volume up +1
VolumeDown Down F11	Volume down -1
1 F1	Album view
2 F2	Play view
3 F3	Transport view (playlist editor)

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Metering

Metering can be enabled using the M switch in the bottom toolbar. This will replace the cover image view with meter display.



Play view, metering

Left channel meters are shown on the left side, right channel meters on the right side. In top left corner of each channel is shown *output level* meters, where yellow line indicates peak hold level, dark green peak-to-peak level and light green RMS level. Numerical values are also shown.
Underneath is spectrogram display with time on horizontal axis and frequency on vertical axis. Color coding is used to display signal level (in dB) in the time-frequency space. Highest level in yellow and lowest in black. Red marker in frequency scale shows base rate's Nyquist frequency (highest possible), while yellow markers show subsequent sub-band Nyquist frequencies.

Spectrogram color scheme can be changed from the control panel (see control panel description below).

Playback filter

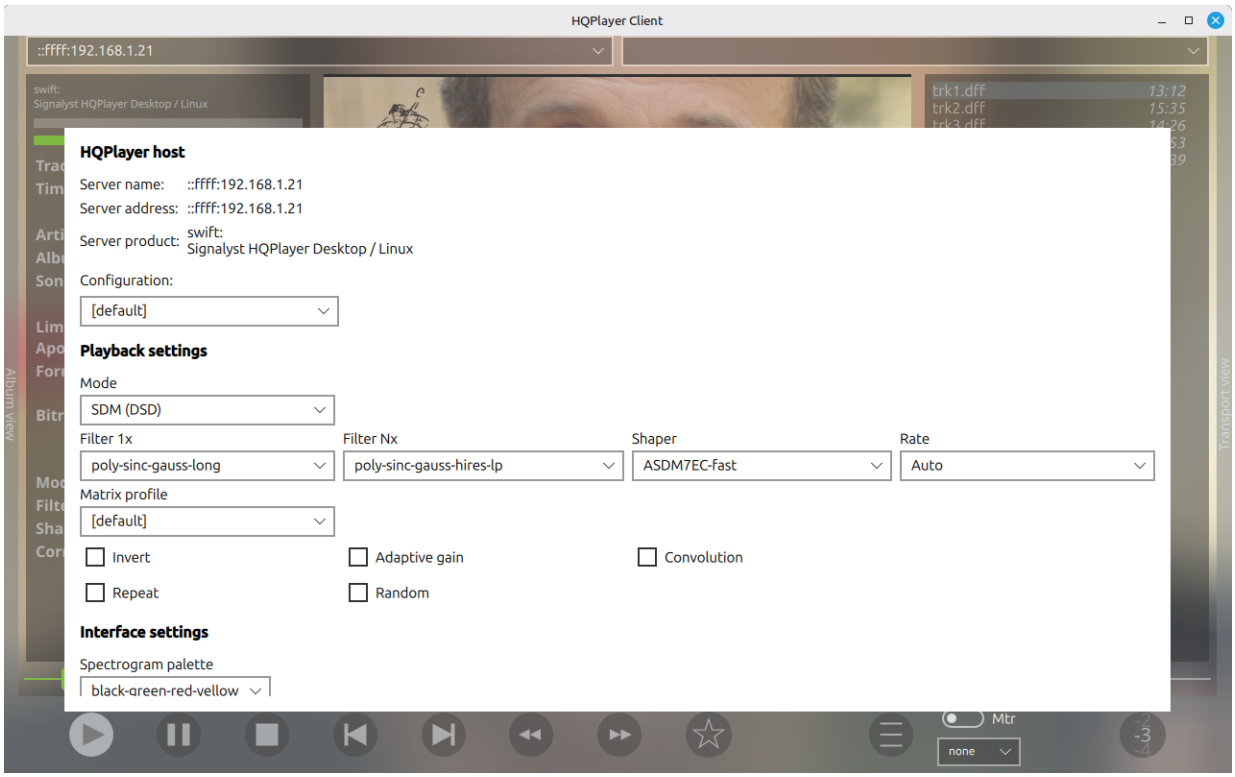
Various playback filters are provided to deal with noise, errors and distortion. For example in bad quality or fake hires source. These filters can be switched at any time during playback. To see their effect on the source, pre-process before metering can be enabled from Advanced tab in Settings.

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Filter	Description
20k	20 kHz filter is useful for cleaning up fake high-res content when such is observed through metering. It will place a sharp roll-off filter at 20 kHz.
30k	30 kHz filter is a slow roll-off filter for removing high frequency disturbances unrelated to the music, while keeping optimal transient response. This can be used for certain hires recordings that are for example transfers from analog tape.
40k	40 kHz filter is a slow roll-off filter for removing high frequency disturbances unrelated to the music, while keeping optimal transient response. This can be used for certain hires recordings that are for example transfers from analog tape.
50k	50 kHz filter is a very slow roll-off filter for cleaning out for example excessive noise shaping from certain ADCs and DSD to PCM conversions.
2x	Places steep cut-off at *2x* of the base rate.
4x	Places steep cut-off at *4x* of the base rate.
8x	Places steep cut-off at *8x* of the base rate.

Control panel

Additional settings can be accessed through menu-button (three horizontal lines) to see following kind of popup.



Control panel

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At the top of the control panel view, there is information about the server connection. In addition list of configuration profiles is shown and the active configuration profile can be changed. Changing the configuration profile will typically take several seconds.

Currently active mode, filter, dither/noise-shaper and sampling rate settings can be viewed and changed using the settings popup dialog. These can be changed at any time. Matrix profile can be switched at any time during playback as well.
In addition, phase inversion, adaptive gain, repeat (checked = current track, grayed = all tracks), random playback order and currently active matrix profile can be changed.

Verbose metadata replaces some of the output format information with more elaborate metadata display of current track. Elide left (default) allows track names to be shortened from left instead of right when they don’t completely fit in the view. Album view can be sorted in ascending (grayed) or descending (checked) release year order. In addition, transparency of album and transport views can be controlled. Background option generates color themed background image based on the current cover picture.

Prefetch (default) begins streaming next track before currently playing track finishes to ensure gapless playback. Freewheel (default) will fetch entire track at full speed. It is not always optimal on slower network links because it causes sharp network traffic floods that may interfere with other things.

Auto play is used on Qobuz service to continue playing similar content automatically, after the currently queued items have been played.
Login credentials of currently active service can be cleared using the button at the bottom of the view.

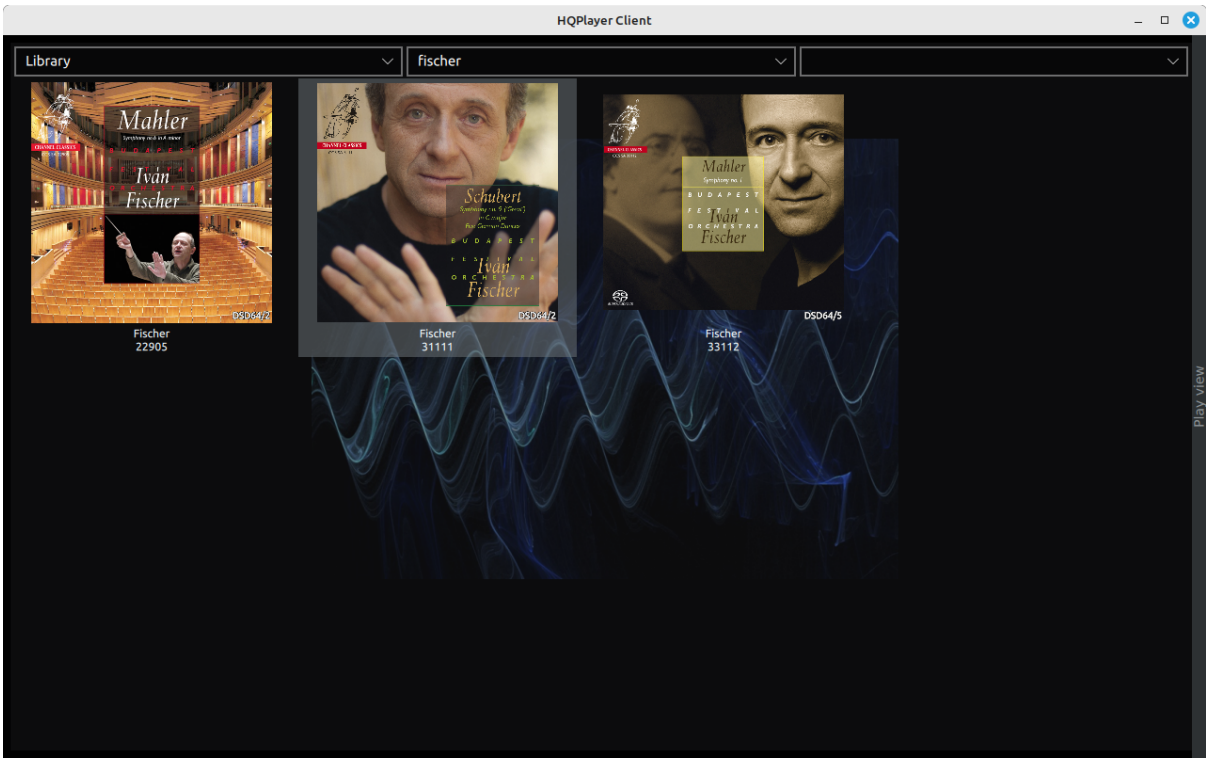
8.1. Switching views

HQPlayer Client has three parallel views in horizontal direction. The view described above is the middle one. Views can be switched by flick gestures to left or right, clicking the title bars on left or right with a mouse cursor, or by keyboard shortcut keys 1, 2, 3.

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8.2. Album selection view

On the left, there is an album selection view with cover flow. On top of the album view there's library selection, search / category selection, and genre selection.



Album selection view

Library selection allows selection of backing music library. You can also select one of the supported streaming services to browse and play music from a streaming service. Currently supported streaming services are Qobuz and HRA Streaming by highresaudio.com. Default view is My Albums, or new releases / top 50 in case My Albums is empty. Search allows searching for content. On HRA Streaming service, prefixing search string with character '?' performs quick search action.

For local library, search strings of three characters or less are considered "*begins with*" matches. Search strings longer than three are considered "*contains*" matches. Default matching logic is logical "OR" operation, where match on any of the terms results in positive match. By prefixing a word with a '+' character, logic is changed to logical "AND" operation for that item.

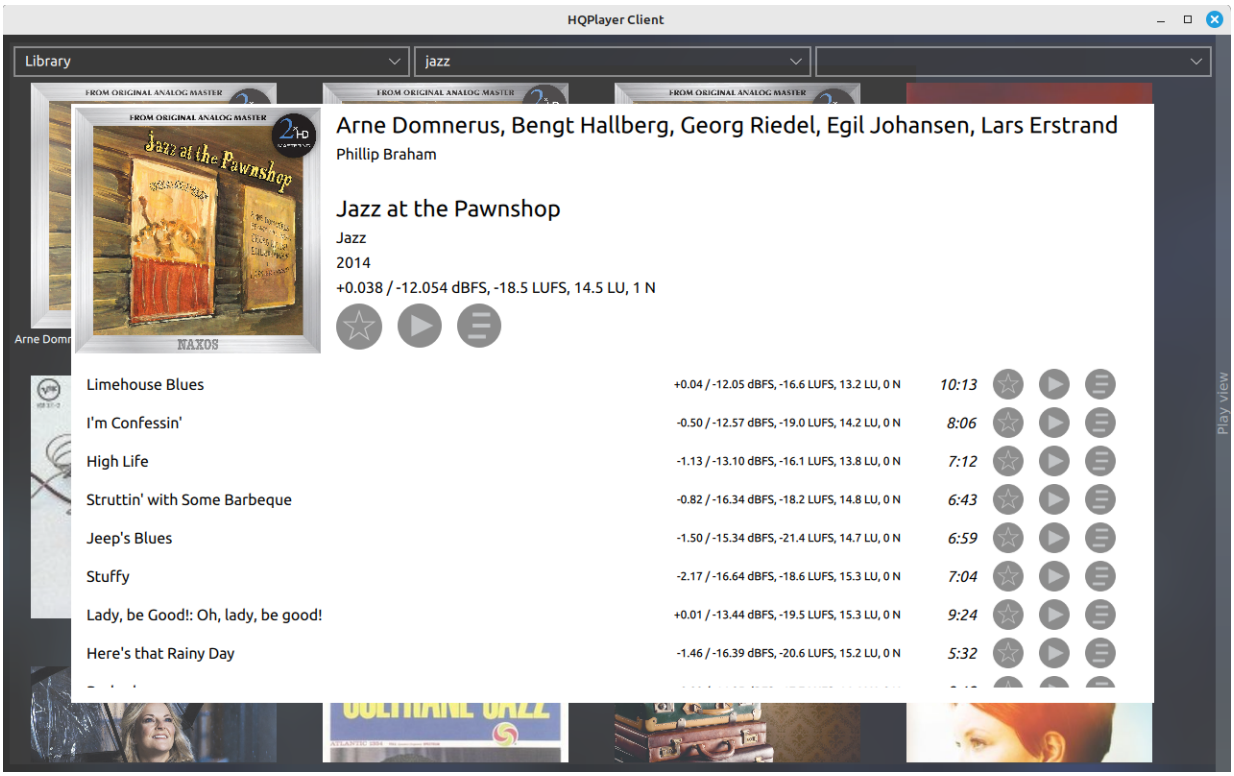
It is also possible to search for a specific type of item. Search string can be prefixed with "album:", "artist:", "playlist:" or "track:".

Genre selection works on local library, and on some selected streaming service categories depending on service's functionality.

Clicking or tapping an album will select it for playback and return the view back to the normal middle-view.

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Double-click of an album will both load the album in transport and start playback.
Long-press on an album opens up following kind of popup window on album details.



Album details view

From the album details view, album can be played immediately or queued for later playback. Individual tracks can be also played immediately or queued for later playback.
Album or track favorite status can be seen on, and changed, using the star button.
Clicking or tapping outside of the popup closes the popup window. Also Esc button on keyboard closes the popup.
Analysis results, when it has been performed at library scan time, are shown for the entire album and for each track. First value is true peak level and second value is true peak RMS, both in dBFS. Third value is integrated loudness in LUFS, and fourth is loudness range in LU. The last value, is error counter value indicating number of incidents that could be potentially fixed by using an apodizing upsampling filter during playback. When figures higher than 10 per track are reached, it is strongly recommended to use an apodizing filter for playback.
Clicking on the cover image displays associated PDF booklet when available.

Qobuz dynamic content

To trigger playback of similar content based on entire album, long press of album play button can be used. Long press of the album queue button will queue similar content for later playback. Similarly, long press of a track play button will trigger playback of similar content. And long press of a track queue button will queue similar content.

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8.3. Playlist edit view

On the right, there is a playlist edit view shown below.



Playlist edit view

Going from left to right, there's an artist selection column, album selection column and track selection column. The right-most column is the current playlist. Tracks can be added or removed using the **+** and **–** keys below. Tracks can be also reordered by using the up/down arrow buttons. Or the playlist can be cleared using the **C** button.
On top of the artist list, there's a search entry for performing searches same way as with the one in the album selection view.
Double-clicking an album, song or playlist entry will initiate immediate playback of the item.

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9. Registering your copy

After purchasing a license key file will be provided. Store this file in a safe place. This file can be installed by selecting “Register...” from the “Help” menu. Standard file open dialog will appear asking to locate and select the license key file. Once the license key file has been successfully installed, restart HQPlayer for the license to fully take effect. ***Note!*** *Remember to back-up the license key file!*

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10. Troubleshooting

This chapter explains some known workarounds and things which you can try, in case of problems.

10.1. Reporting bugs

In case you discover bugs, please enable log file functionality from the settings dialog, try to reproduce the bug and send the log file together with a screen shot (PrtScn button) to our support email address support@signalyst.com . On Windows the log file can be found from %LOCALAPPDATA%\HQPlayer directory of the system drive. You need to enter this path on the address bar of the Windows File Explorer as these are not shown by default. On Linux and macOS, log file can be found from the hidden ~/.hqplayer directory, where “~” denotes user's home directory. The log file is called HQPlayer5Desktop.log.

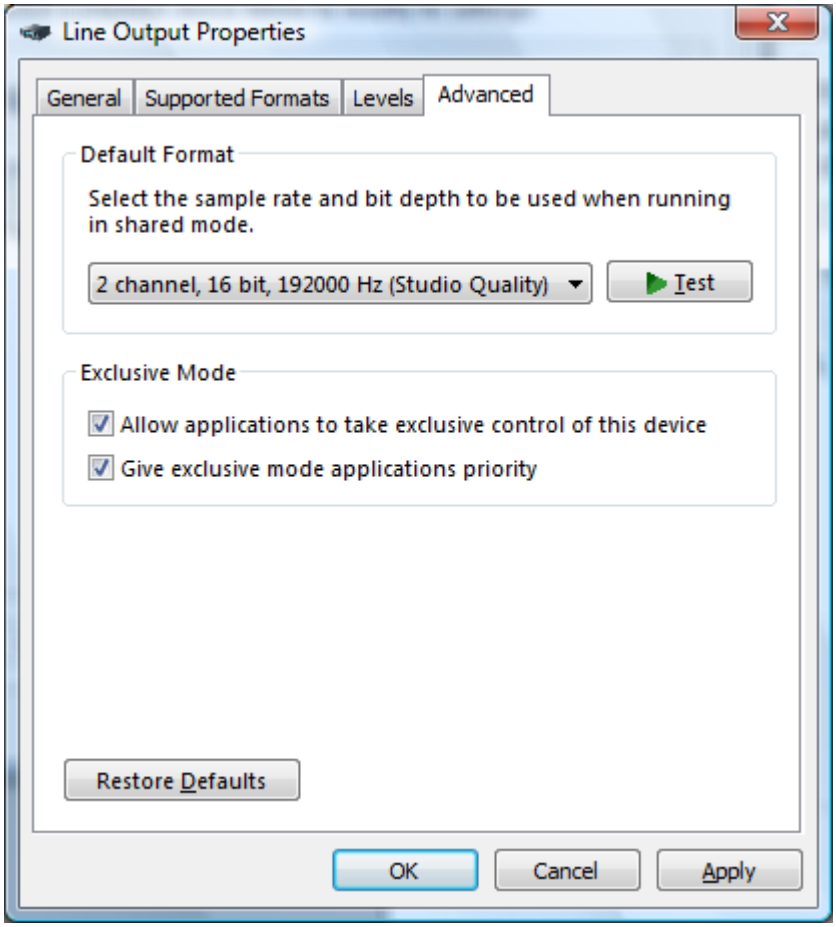
10.2. Sound problems with USB audio device

Default buffer size for USB audio devices is fairly small (10 ms) in Windows. This sometimes causes various audio issues when some other process is loading the system. If you experience such problems, increase size of the audio buffer by changing the “Buffer time” setting in Settings-dialog. Good starting point is 100 ms.

10.3. Generic

You might want to check that your selected sound device has exclusive mode enabled in endpoint properties.

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Endpoint properties dialog

With some buggy drivers it might also be necessary to change default format to match the sample rate you are trying to use with HQPlayer.

10.4. No rates available

In some cases, rate selection may stay empty. This means that there are no suitable hardware sampling rates available for selected source material and resampling filter combination. In this case, try selecting different resampling filter, such as “sinc”, “minphase-sinc” or “none”.

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11.1. HQPlayer

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