HQPlayer Desktop

User Manual

Version 3.14.0

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1. Introduction

HQPlayer is a high quality audio player for Windows Vista, Windows 7/8/8.1/10 and Linux and Mac OS X. 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 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 also re-sample internally to 48, 96 or 192 kHz, with the HQPlayer, these can be fed at the native rate.

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.

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

Playback of DSDIFF and DSF files is supported. In case hardware and drivers support ASIO 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 (64fs), 352.8 (128fs) or 705.6 kHz (256fs) PCM (pulse code modulation) format for playback through PCM audio hardware.The playback rate of DSDIFF and DSF files can be further altered by using resampling to chosen rate. Thus, playback rates from 32 to 1536 kHz are possible. Used bit depth is always maximum supported by the playback hardware.

Also multichannel loudspeaker delay- and level-processing is supported in both converted and native modes.

1.2. 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.
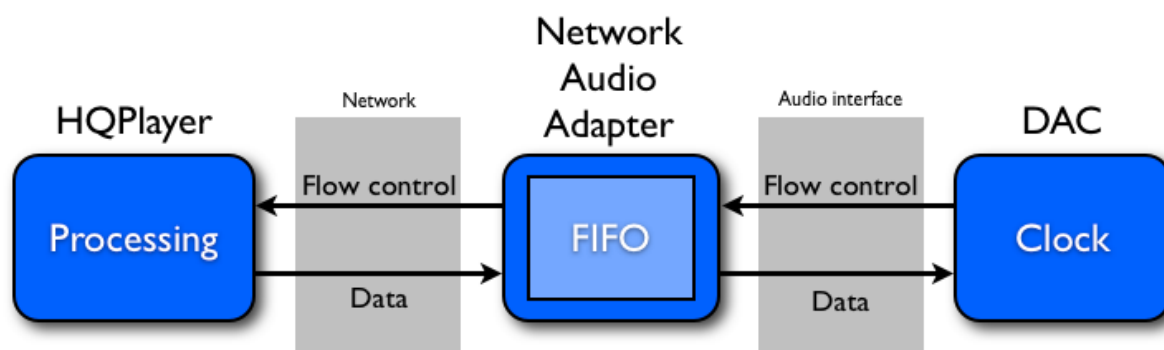
Network Audio system

2. Main screen

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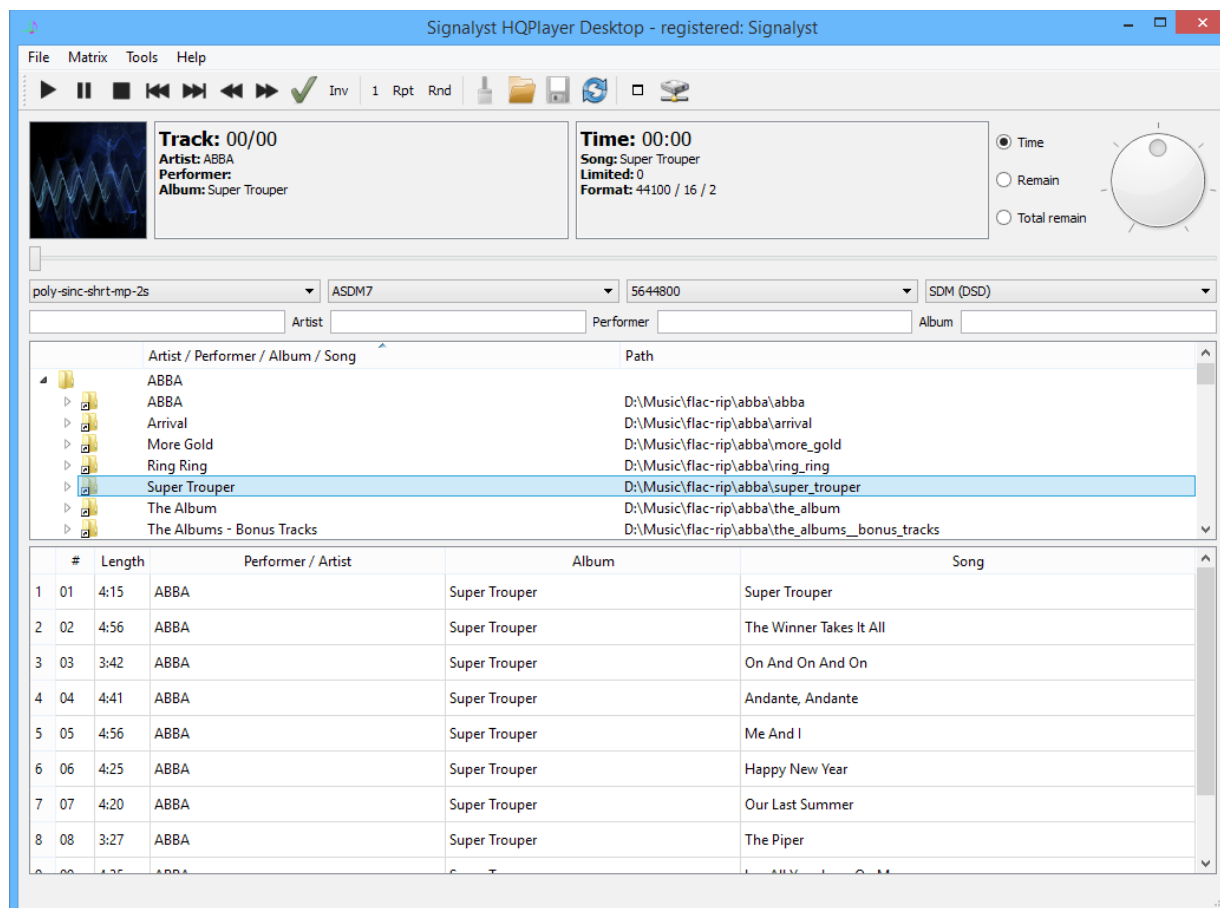
Main screen

User interface also supports standard multimedia keys and equivalent remote controls. It also features two modes, the traditional desktop application mode shown above, and new touch-optimized full-screen mode available through the right-most button in the toolbar.

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. Transport selection (library)

Transport is the playback source location. It lists all configured FLAC-, DSDIFF-, DSF- , WAV- and AIFF- file paths as well as playlist files and a CD-drive. For file path, each sub-directory is expected to be an album consisting of files in one of the supported formats. Transports are structured in a tree model in Artist / Performer / Album / Song order.

Selecting a new transport represents similar action as changing a CD in physical drive.

When album node is expanded, tracks of the album are listed. Selecting tracks switches transport into playlist mode, where individual tracks (from same or different album) can be added.

2.2. Transport search and filters

There is one search entry and three filter entries (Artist, Performer and Album). Search entry is typical case insensitive "contains" search. Except for search strings with length equal to, or shorter than three characters, in which case search criteria is "begins with".
Transport view can also be filtered by entering filtering rules. Rules can be in traditional "wildcard" ('*') format, or in Perl-style regular expression format, when prefixed with '#'.

2.3. Song display

For CD, this field is used only to display track numbers.
When playing back FLAC-files, metadata is shown. If metadata is not available a file name is shown. For DSDIFF-, DSF-, WAV- and AIFF- files, file name is displayed.

2.4. 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 FLAC/DSDIFF/DSF/WAV/AIFF-files, track numbering is constructed per directory basis based on file name sorting order. For preferred order, file names should begin with correct zero-prefixed track number.

2.5. 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).

2.6. 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.

2.7. 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.
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 load, and with large impulse responses will require significant amount of CPU processing power.*

2.8. Phase inversion

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

2.9. 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.

2.10. 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.
Playlist playback of DSDIFF/DSF files is not gapless, but a special code is enabled in order to avoid clicks/pops between unrelated tracks.

2.11. Allow remote control from network

When this button is pressed, remote control of the player is allowed also from network. When the button is depressed, remote control is only allowed from the same machine.

2.12. Filter / oversampling selection

This selection can be used to switch between resampling filters. This selection has an impact on available hardware sampling rates. This selection cannot be changed during playback. Different variants of "poly-sinc" are the most recommended by the author.

Filter	Description
none	No sample rate conversion happens. Only sample depth is changed as needed.
IIR	This is analog-sounding filter, especially suitable for recordings containing strong transients, long post-echo 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-echo, but has a long post-echo. Small amount of pass-band ripple is also present. IIR filter is applied in time-domain. This filter performs better when resampling with only factor of 2 or 3 and somewhat worse at factor of 4 and higher.
FIR	Typical "oversampling" digital filter, generally suitable for most uses (slight pre- and post-echo), but best on classical music recorded in a real world acoustic environment such as concert hall. 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-echo.
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-echo and longer post-echo. Modifies phase response, but not as much as minimum phase FIR.
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-echo, but somewhat long post-echo.
FFT	Technically good steep "brickwall" filter, but might have some side effects (pre-echo) 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.
poly-sinc / *better space*	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-echo. See "FIR" for further details.
poly-sinc-mp / *better transients*	Minimum phase polyphase sinc filter, otherwise similar to poly-sinc. Altered phase response, but no pre-echo. See "minphaseFIR" for further details.
poly-sinc-shrt	Otherwise similar as poly-sinc, but shorter pre- and post-echos at the expense of filtering quality (not as sharp roll-off and reduced stop-band attenuation).
poly-sinc-shrt-mp	Minimum phase variant of poly-sinc-shrt. Otherwise similar to poly-sinc-mp, but shorter post-echo. Most optimal transient reproduction.

Filter	Description
poly-sinc-hb	Linear-phase polyphase half-band filter with steep cut-off and high attenuation.
poly-sinc-ext	Linear phase polyphase sinc filter with sharper cut-off and somewhat lower stop-band attenuation, while being roughly equal length to poly-sinc.
poly-sinc-mqa	Linear phase polyphase sinc filter optimized for playing back MQA encoded content in order to clean up noise added by the MQA encoding.
sinc	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.
polynomial-1	Polynomial interpolation. Most natural polynomial interpolation for audio. Only two samples of pre- and post-echo. Frequency response rolls off slowly in the top octave. Poor stop-band rejection and will thus leak fairly high amount of ultrasonic noise. These type of filters are sometimes referred to as "non-ringing" by some manufacturers. *Not recommended.*
polynomial-2	Similar to polynomial-1, but higher stop-band rejection with the expense of being a bit less natural for audio. *Not recommended.*
minringFIR	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 is between polynomial and poly-sinc-short.
closed-form	Closed form interpolation with high number of taps.
closed-form-fast	Closed form interpolation with lower CPU load, but also lower precision. Output precision tuned to match about 24-bit PCM.
*-2s	Two stage oversampling. First stage rate conversion is performed by at least by 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.

2.13. Noise-shaping / dither / modulator selection

This selection can be used to switch between different word-length reduction algorithms. Especially important when playback hardware supports less than 24 bits. This selection cannot be changed during playback.

NS/Dither	Description
none	No noise-shaping or dithering, only rounding. Mostly suitable for 32-bit output hardware where quantization error is buried in thermal noise. *Not recommended.*
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.)
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.
DSD5	Rate adaptive fifth order one-bit delta-sigma modulator.
DSD5v2	Revised fifth order one-bit delta-sigma modulator.
DSD7	Seventh order one-bit delta-sigma modulator.
DSD5v2 256+fs	Revised fifth order one-bit delta-sigma modulator optimized for rates >= 10.24 MHz.
DSD7 256+fs	Seventh order one-bit delta-sigma modulator optimized for rates >= 10.24 MHz.
ASDM5	Adaptive fifth order one-bit delta-sigma modulator.
ASDM7	Adaptive seventh order one-bit delta-sigma modulator.

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

2.14. Sample rate selection

This selection can be used to switch between supported hardware sample rates. Available choices depend on selected transport and resampling filter type. This selection cannot be changed during playback. *Auto* setting automatically selects a highest rate that is equal or less than the limit set in the Settings dialog and is possible using the chosen filter.
Note! *When "none" is selected as resampling algorithm, output sampling rate is adjusted based on source file's sampling rate.*

2.15. PCM / SDM (DSD) output mode selection

In case playback hardware supports both modes, the mode can be changed using this selection. In PCM mode, when source content is DSD it will be converted to PCM for playback. In SDM mode, when source content is PCM it will be converted to SDM for playback. *Auto* setting automatically switches between PCM and SDM modes to match the source content type.
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. Maximum modulation depth is monitored and when necessary limited to 50% as per SACD specification.*

2.16. Volume control

Playback 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 -2 dBFS to avoid inter-sample overloads, and in case material contains digital clipping/limiting.
**Note!** *Never use software based volume control as sole mean for adjusting volume. Always limit the maximum possible volume to reasonable levels by using suitable hardware adjustment!*

2.17. Position/seek bar

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

3. Full-screen mode

When full-screen button is clicked in the main window toolbar, following kind of screen is shown either in maximized window or in full-screen, depending on corresponding setting in Settings-dialog.
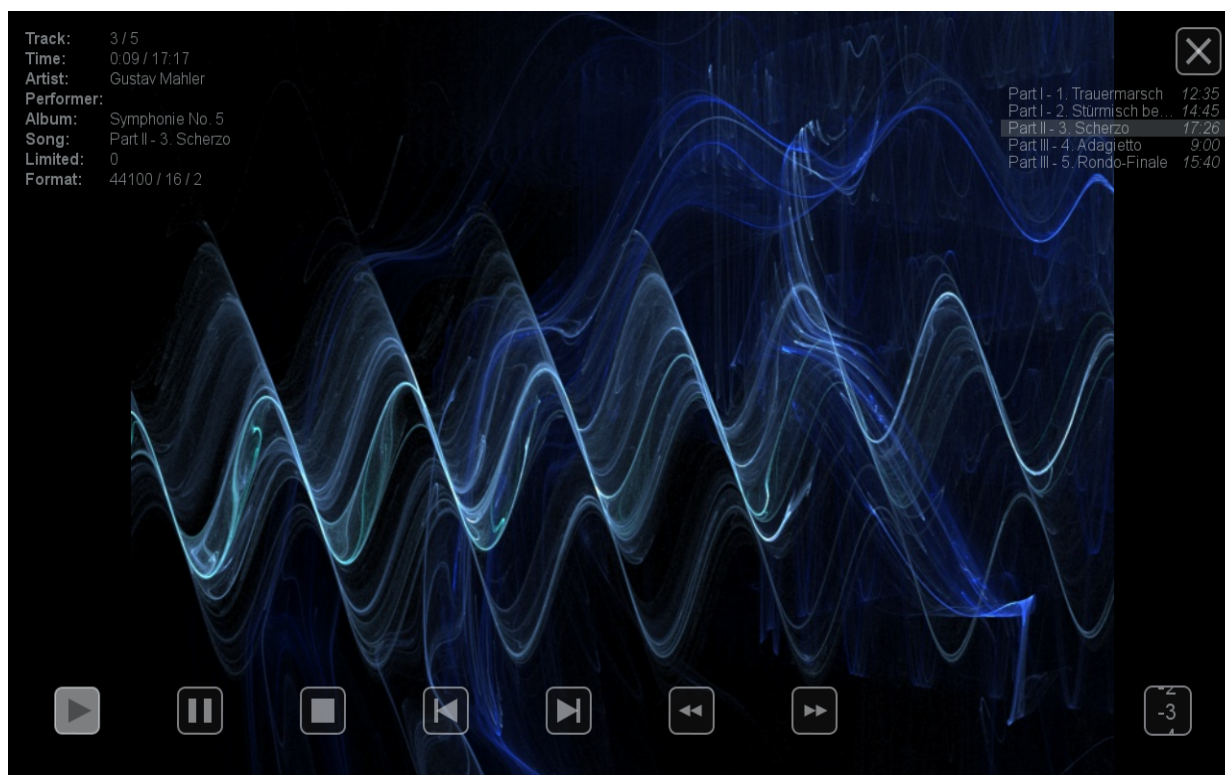
Full-screen mode

This display is optimized for touch-screens, but can be also used with a mouse or other suitable pointing device.
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 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 the adjustment either up or down. Volume can be also operated using a mouse wheel when mouse pointer is within the control.

3.1. Switching views

Full-screen mode 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.

3.2. Album selection view

On the left, there is an album selection view with cover flow and search on the top of the screen. When the search entry at the top of the screen is clicked or tapped, virtual

keyboard is shown (Windows only). Search strings of three characters or less are considered "*begins with*" matches. Search strings longer than three are considered "*contains*" matches. Search strings with either or both '*' and '*?*' wildcard characters are handled as normal wildcard matches, where '*' means any number of any character and '*?*' means any single character. Clicking or tapping an album will select it for playback and return the view back to the normal middle-view.

3.3. 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.

4. Library management

To edit library, open the "File" menu and select "Library...". Following dialog will be shown.
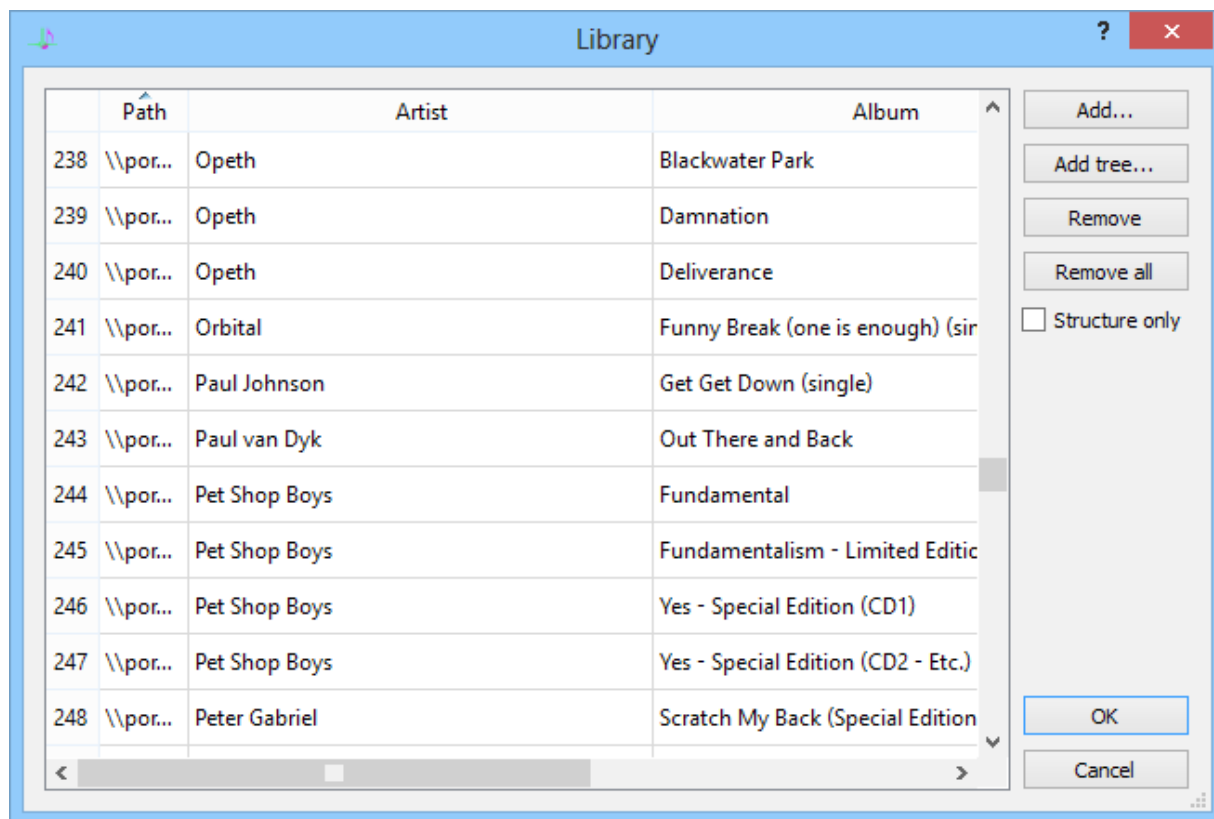
Transport editing dialog

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 add a specific album path, click "Add..." to browse and select an album folder you wish to add and click "OK". Now the path to the selected file will be added to the list of available albums. To add an entire directory tree, click "Add tree..." 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 added folder cover art). Select "Structure only" to extract metadata solely from the directory tree structure instead of metadata embedded in files.

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.

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.

5. Settings

To change program's device settings, open the "File" menu and select "Settings...".
Following dialog will be shown.
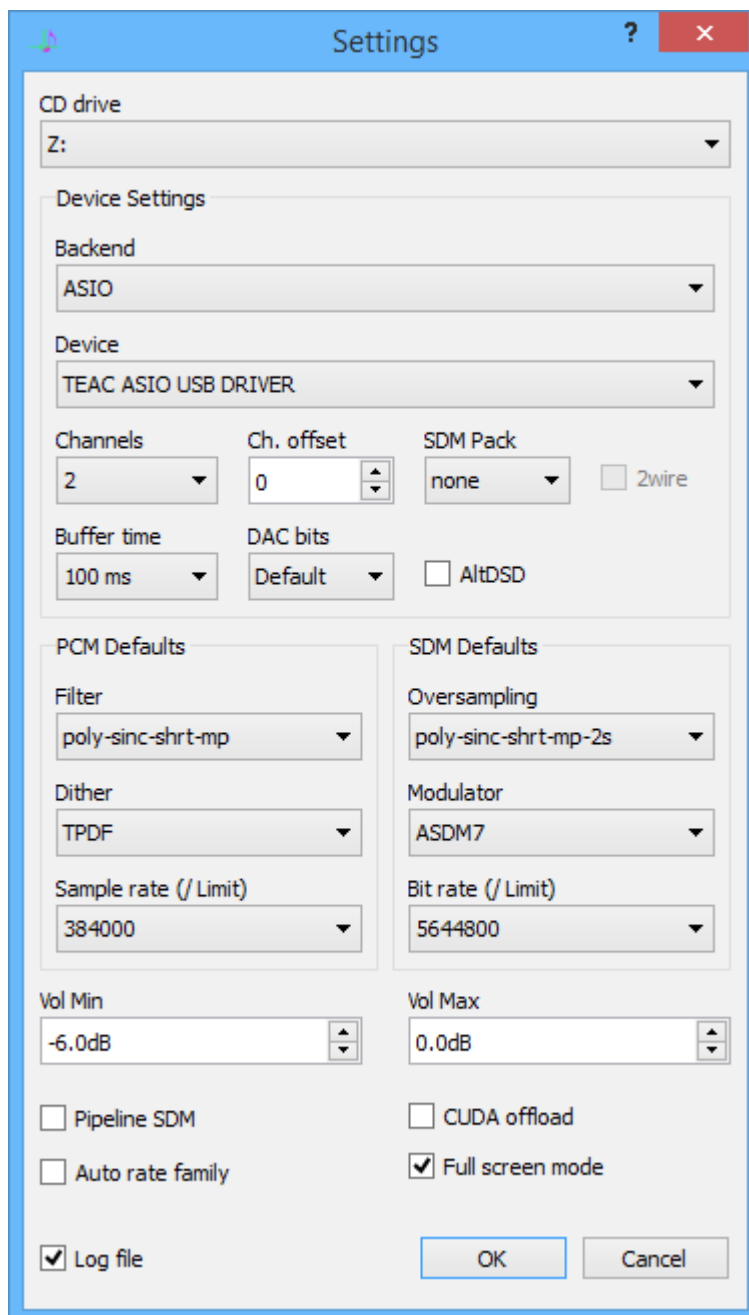
Settings dialog

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 OS X 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.
On Linux, ALSA audio endpoint (device) can be selected by using the "Device" selection which lists all the available hardware audio endpoints.
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.
Default audio processing settings can be changed from the "PCM Defaults" and "SDM Defaults" boxes.
Number of output channels can be chosen from "Channels" selection, possible choices are "2" for stereo, "5.1" for normal multichannel, "7.1" for extended multichannel and "16"/"32" primarily for complex matrix processing cases.
DSD content can be transferred to 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. For ASIO on Windows, "AltDSD" changes units of the DSD sample from bit to a byte, use this only when experiencing incorrect DSD playback, incorrectly enabling this will make the application crash (currently known devices to need this setting are Playback Designs and Merging).
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. In case of ASIO, this can be usually controlled through ASIO control panel, which can be launched by selecting "ASIO Control Panel..." from the "Tools" menu. Values between 10 and 100 ms are most recommended.
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.
When DAC with other bit depth than 24-bit is connected to a unidirectional interface like S/PDIF or AES/EBU it is recommended to select correct number of bits from the "DAC bits" selection. If measurement results for the DAC are available, a value corresponding to linear range of the DAC can be used.

Pipeline SDM increases parallelization of PCM-to-SDM conversion. Mostly useful on computers with multiple slower (real) CPU cores or when using CUDA offload. Since this parallelization increases processing overhead, it will increase total CPU time consumption. "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 3.0, 2 GB of graphics RAM and 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.

Auto rate family, when used in conjunction with Auto output rate setting in main window makes automatic output rate selection pick sampling rates that are multiples of the source content sampling rate.

Full screen mode setting makes the full screen touch-oriented interface truly full screen instead of maximized window.

To save the settings, click the "OK" button and to discard any setting changes click the "Cancel" button.

Note! *Using one of the heavy algorithms for upsampling multichannel audio files can require significant amount of CPU processing power!*

Note! *Do NOT select anything other than "None" as SDM packing, unless you are sure your hardware supports the particular special packed mode!*

5.1. DSDIFF/DSF settings

Settings for the PDM → PCM conversion process can be changed through a settings dialog available from the "File" menu by selecting "DSDIFF/DSF settings...". Following dialog will be displayed.
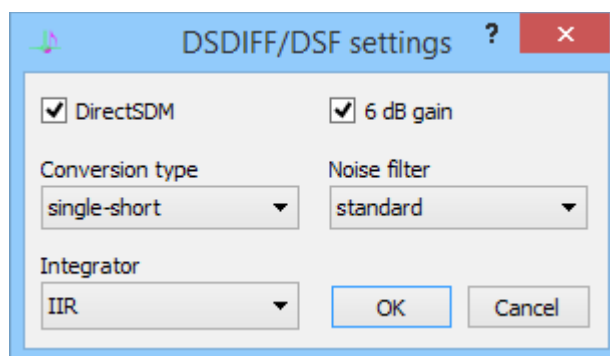
DSDIFF/DSF settings dialog

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

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 CD, FLAC, WAV or AIFF. However, this may cause digital clipping with some source material and thus use of 6 dB gain is *not recommended*!

Type of conversion can be selected from the "Conversion type" drop list. Following conversion types are supported.

Conversion type	Description
traditional	Traditional recursive conversion algorithm. Minimizes amount of ringing by using slow roll-off filters.
single-steep	Single-pass conversion algorithm with steep cut-off.
single-short	Single-pass conversion algorithm with normal cut-off. Optimized tradeoff between ringing and wide frequency response.
poly-lin	Linear-phase single-pass conversion algorithm.
poly-mp	Minimum-phase single-pass conversion algorithm.
poly-short-lin	Linear-phase slow roll-off single-pass conversion algorithm. *Recommended.*
poly-short-mp	Minimum-phase slow roll-off single-pass conversion algorithm.

Different types of noise filters 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 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 playback rate of DSDIFF or DSF is 88.2/96 kHz, use of extra noise filtering in addition to "standard" is less important, since most of the noise will be cut out. When playback rate of DSDIFF or DSF is 44.1/48 kHz, extra noise filtering in addition to "standard" is not needed and will actually just reduce playback quality.

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.*
sac	Sliding average converter.
wec	Weighted element converter.
slow-lin	Slow roll-off linear-phase filter.
slow-mp	Slow roll-off minimum-phase filter.
fast-lin	Fast roll-off linear-phase filter.
fast-mp	Fast roll-off minimum-phase filter.

There are three types of delta-sigma integrators available.

Integrator	Description
IIR	Normal IIR type integrator structure.
FIR	Weighted FIR type integrator structure.
CIC	Cascade comb type integrator structure.

5.2.   Speaker setup

For multichannel playback, speaker distances and playback levels can be configured using speaker setup dialog. This dialog can be reached by opening "Tools" menu and selecting "Speaker Setup...".
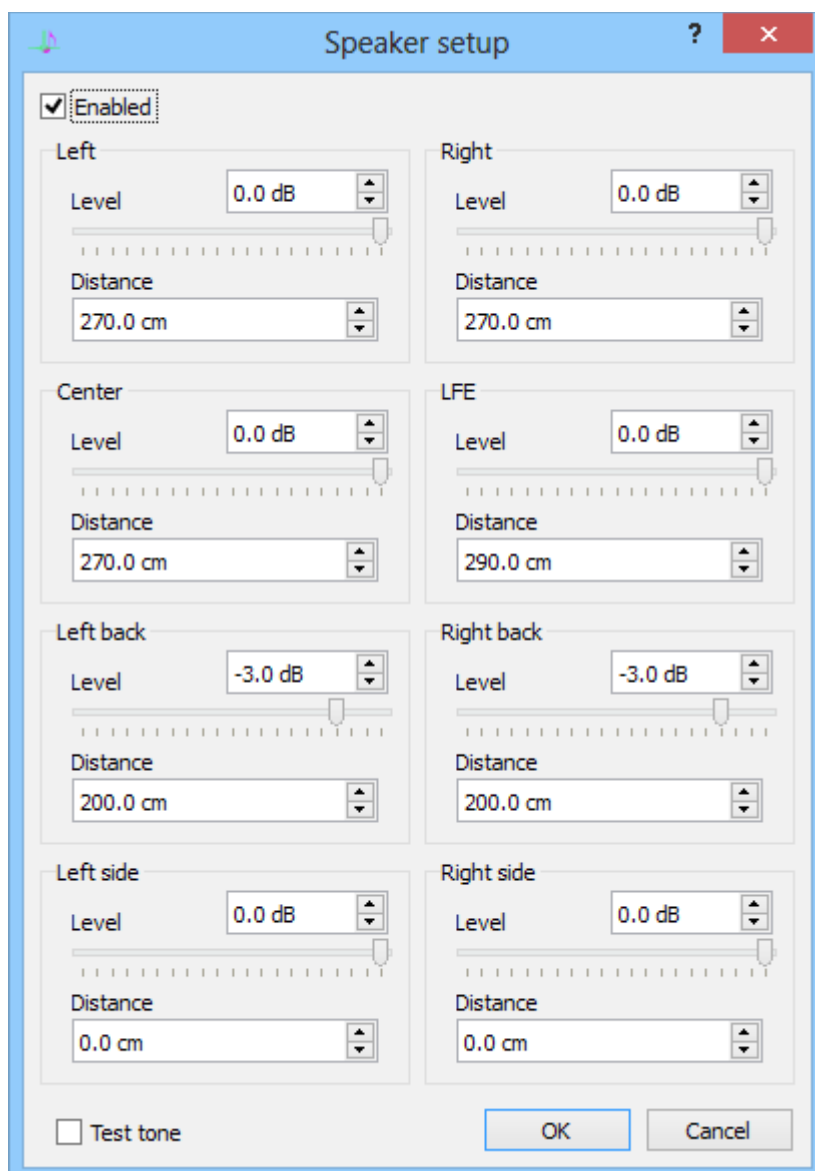
Speaker setup dialog

In this dialog, distance to each individual speaker can be set in centimeters. Playback of the speaker can be set using the volume slider and unit shown in upper right corner is in dB.

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 way that the 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.

5.3. Network Audio Adapter naming

Network Audio Adapters can be assigned a non-default name using Tools / Network

naming menu selection and following dialog.
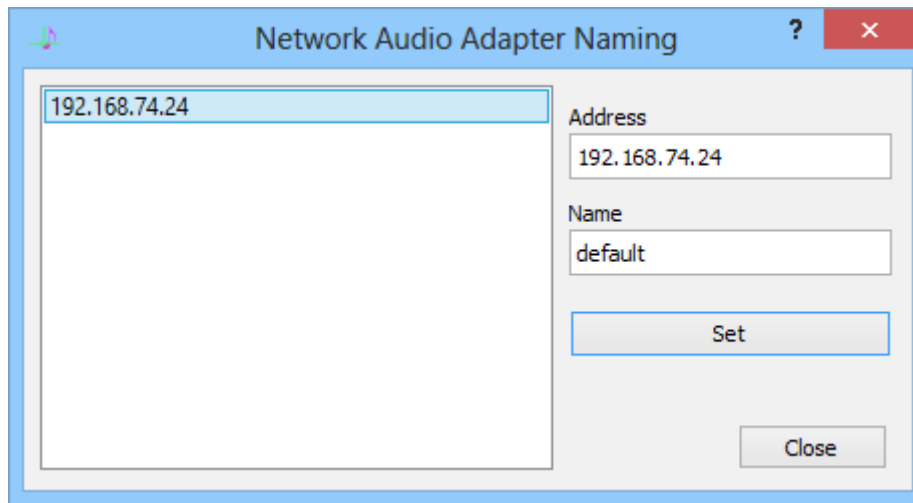
Network Audio Adapter naming dialog

When this dialog is opened, a NAA discovery is run and a list of discovered adapters appears in a few seconds. Selecting an entry from the list displays the selected address and it's associated name on the right. Name can be modified and sent to the device by clicking "Set" button. After each modification a re-discovery is run.
This way, different names can be assigned for different adapters for example for multi-room use, so that different adapters can be recognized and found even if their assigned IP address changes over time.

5.4.   ASIO channel mapping

When using ASIO, in addition to just using default channel mapping it is also possible to assign custom channel mappings. This can be done by selecting "ASIO Channel Mappings..." from the "Tools" menu.
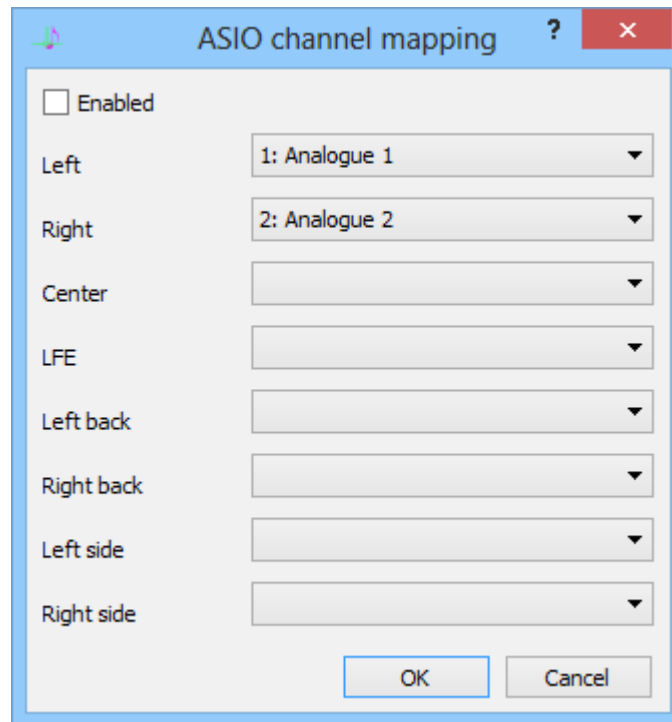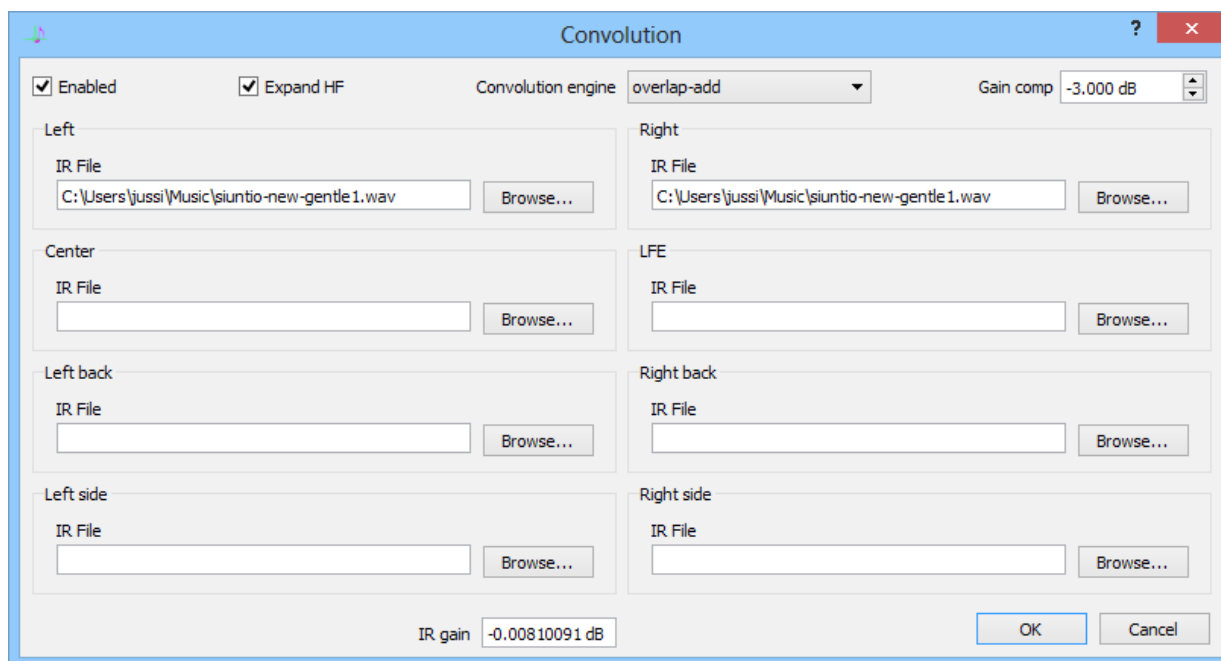
ASIO channel mapping dialog

In this dialog, left hand side shows the source channel and the channels reported by the ASIO driver can be selected as destination channels from the drop lists on the right hand side.

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 "Engine 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 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...", it's 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, it's high frequency response can be expanded to cover the new bandwidth by selecting "Expand HF" setting.
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 example popular Room EQ Wizard can export suitable impulse responses after designing for "Generic" equalizer by selecting File → Export → Filters Impulse Response as WAV. 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 processing power!*

7. Matrix processing

Matrix processing offers a way to copy, route, filter and mix down channels with specified gains. Matrix processing consists of 32 virtual channels.
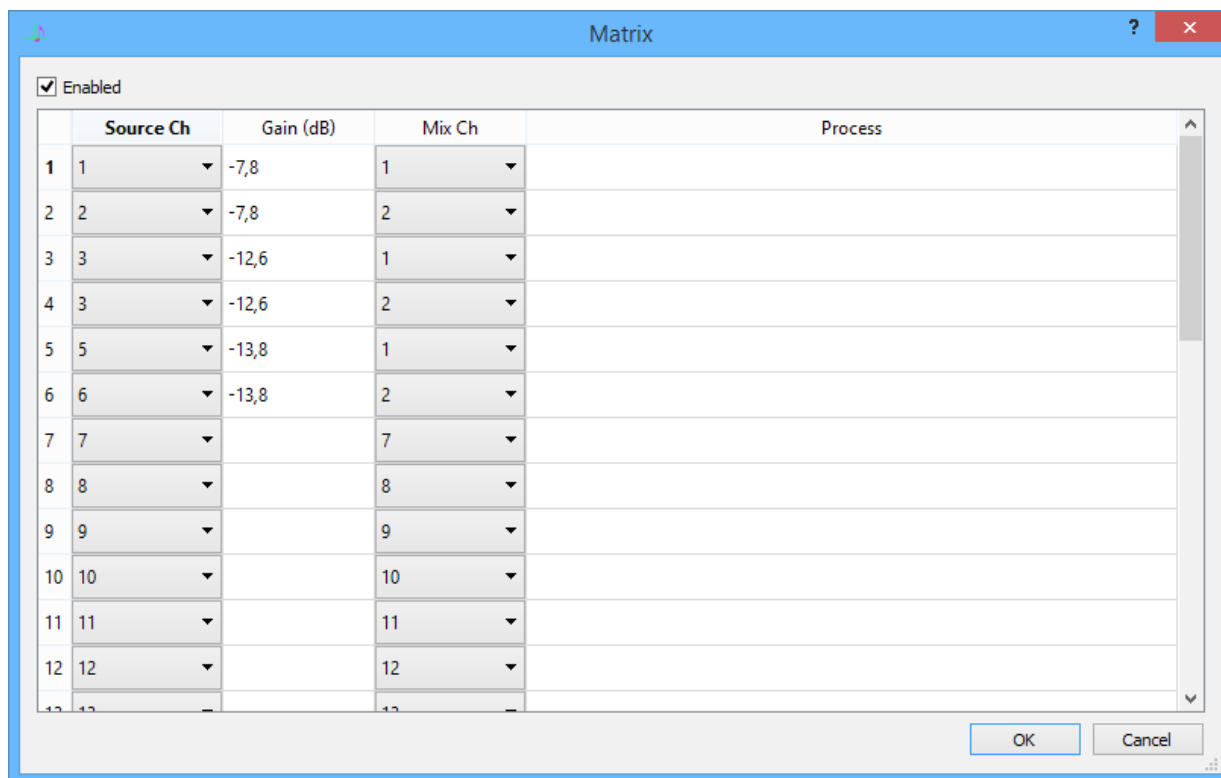
Matrix configuration dialog

For example the configuration shown above can be used to mix down 5.0/5.1 channel material to stereo.

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.
At the moment "Process" can define external filter impulse response WAV file that is applied as part of the processing.

Note! *Convolution and Matrix processing are mutually exclusive, so only one of the two should be enabled at a time. However, the Convolution setup dialog can be used for easier selection of filter files for the first eight channels.*

8. Registering your copy

Activation code can be entered by opening "Help" menu and choosing "Register...".
Dialog will appear asking for the code. If your code was shipped with a name, fill also
name field, otherwise leave it empty.

Code can be also entered after, or close to trial period closing when a dialog asking for
the code appears at startup time.

Note! *Especially on Windows 7 it may be necessary to start the application as
Administrator in order to successfully store the registration code. To do it, right-click on
the application's Start-menu entry and select "Run as administrator" from the menu.*

8.1. Registering Linux / Mac OS X version

After purchasing a license for Linux or Mac OS X version, key file will be provided.
This file can be installed by selecting "Register..." from the "Help" menu.

9. Troubleshooting

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

9.1. 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 \Users\<username>\AppData\Local\HQPlayer directory of the system drive. You may need to complete this path on the address bar of the Windows Explorer as these are not shown by default. On Linux and Mac OS X, log file can be found from the hidden ~/.hqplayer directory, where "~" denotes user's home directory. The log file is called HQPlayer.log.

9.2. 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 playback 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.

9.3. Generic

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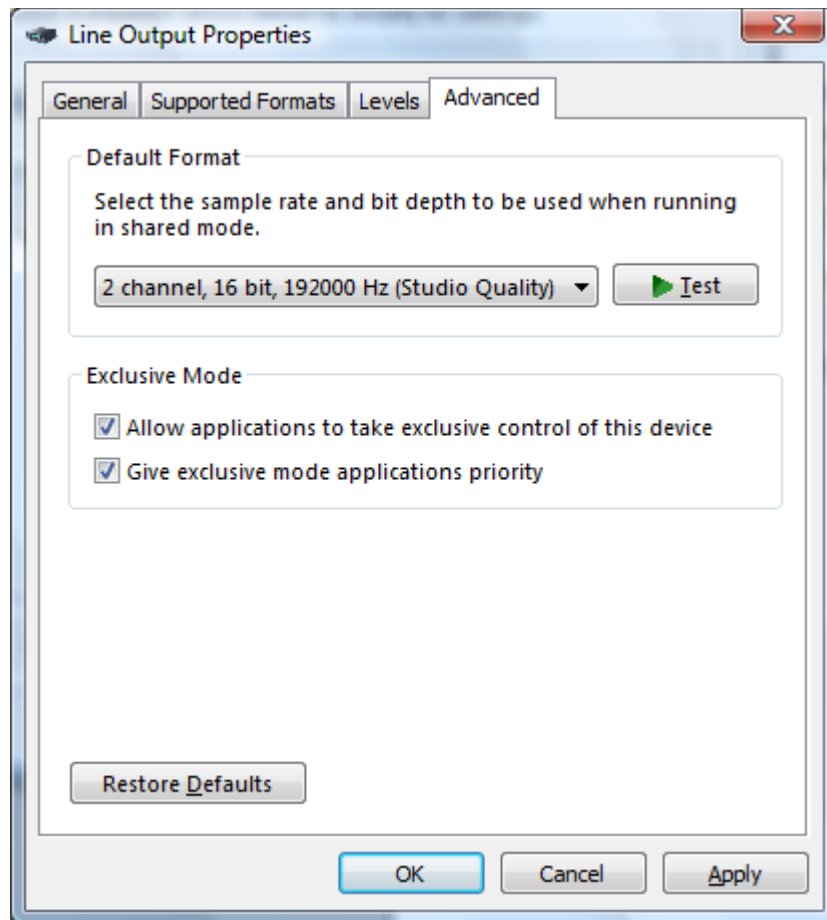
Endpoint properties dialog

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.

9.4. Channel mapping

Please note, that depending on sound device, channel mappings may have playback-software independent implications. For example some cards perform internal downmix of multichannel playback depending on speaker layout configured in their respective control panels.
Here is example of speaker setup found in Windows Vista in Control Panel → Sound / Playback → Configure.
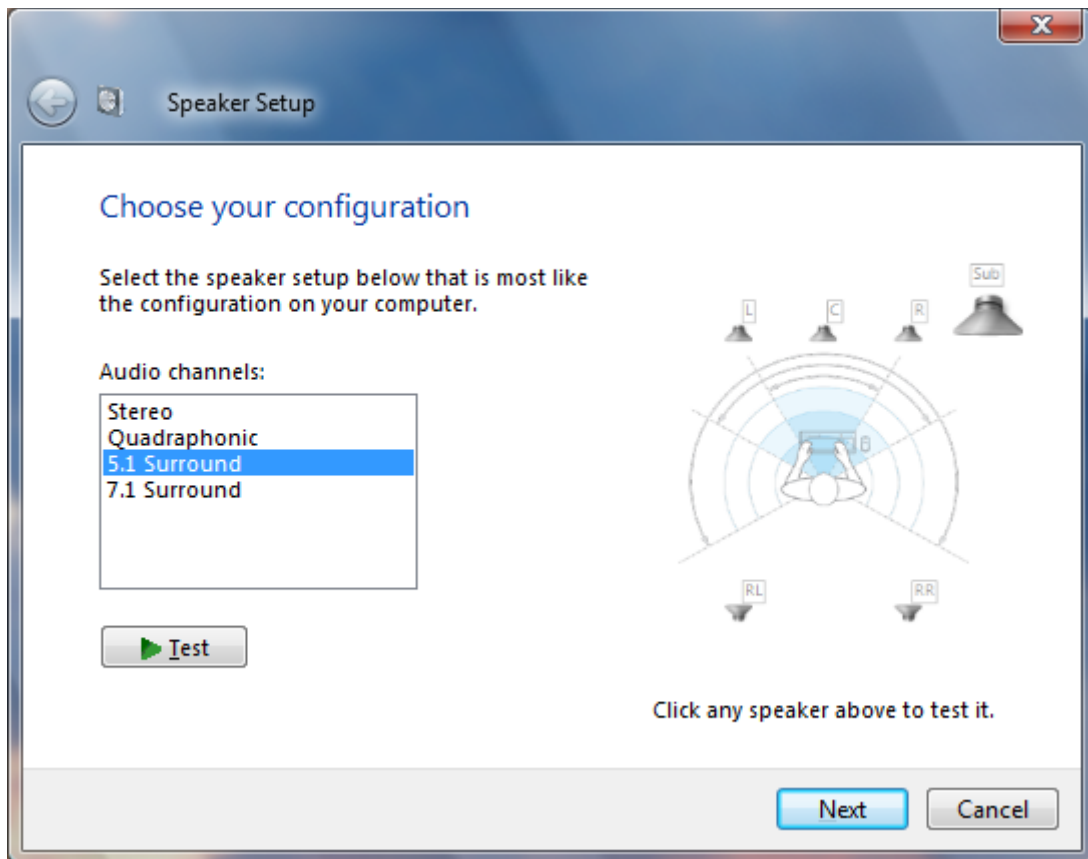
Windows Vista Speaker Setup dialog

And here is example of Creative Audio Console speaker setup.
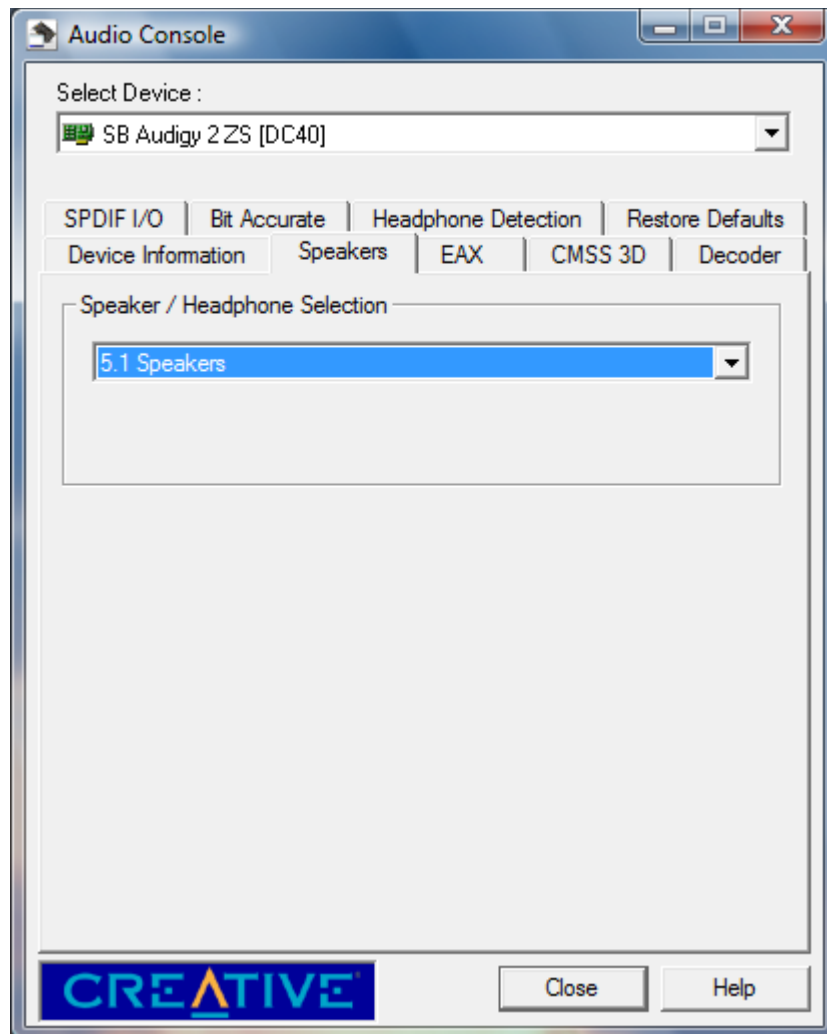
Creative Audio Console

When using ASIO channel mapping with Creative driver, it is advisable to select maximum number of channels (7.1 Speakers) here in order to avoid conflicts between downmixing done by the Creative hardware/drivers and channel mapping / speaker setup processing done by the HQPlayer.

9.5. 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".

9.6. Known workarounds

E-MU 0202 USB, 0404 USB and Tracker Pre USB

Driver may use fixed rate which can be changed from the E-MU control panel (icon can be found from the system tray).
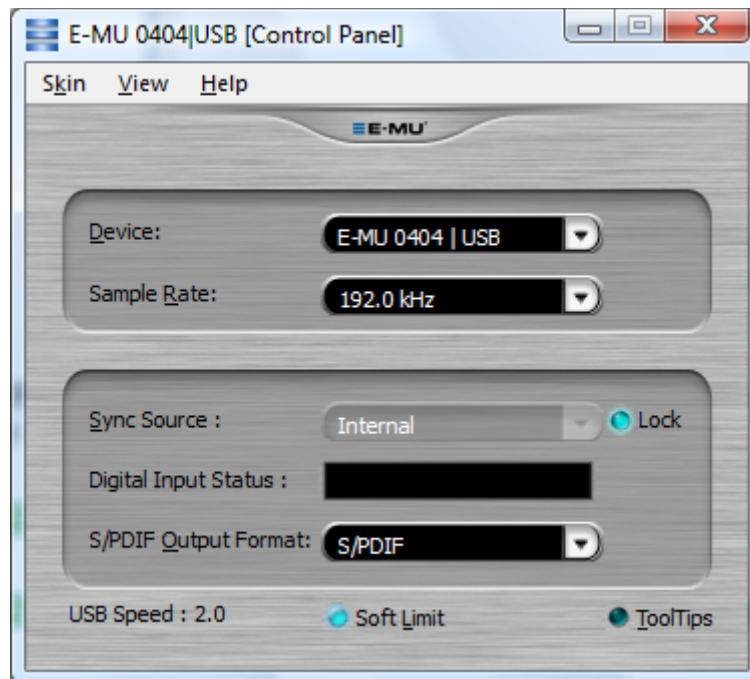
E-MU Control Panel

Usually selecting 44.1 kHz rate and internal sync source allows rate control by the software. If not, sample rate needs to be changed from this control panel before clicking "Play" in HQPlayer, otherwise starting the device may fail. If this happens anyway, you can click "Stop", then correct the sampling rate in E-MU control panel and click "Play" again.

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10.1. HQPlayer

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10.6. Mac OS X ASIO loader

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