

SoX Plugins

*A Reimplementation of the SoX Commandline
Processor as DAW Plugins
— Version: VST3 on Windows x64 —*

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Chapter 1

Introduction

1.1 Overview

The SoX-Plugins software package provides plugins for being used in digital audio workstations (DAWs); they implement some of the more prominent audio processing effects from SoX as DAW audio plugins.

SoX [SoXDOC] is a command line audio processing tool for Unix, Windows and Mac OS that transforms source audio files in several formats into other audio files. It provides several standard audio effects (like e.g. filters or reverb) in good quality and with a transparent, open-source implementation.

The plugin implementation is completely platform-neutral based on the JUCE audio framework [JUCE]. Currently only a build for VST3 plugins under Windows 10 is provided, but porting to other targets should be straightforward, since building is supported by a platform-neutral CMAKE build file (see chapter 5.2).

The effects provided here are a complete rewrite in C++ of the original SoX algorithms, but still aiming at producing (*bit-exact*) *identical* renderings in the DAW. This goal is achieved. When rendering some audio externally via SoX and internally with the plugins, there is almost no difference. After rendering and subtracting the results (see chapter 4) the results cancel out with typically a residual noise of less than -120dBFS due to rounding or precision errors (SoX often uses 32bit integer processing, while SoX-Plugins always uses double floating point processing).

The main motivation for this package is to be able to play around with effects in a DAW and be sure that any external rendering by SoX will produce exactly the same results. Although SoX does not always provide the "best" effects, it still is a reliable and well-defined audio tool.

Only a selection of SoX effects has been reimplemented as plugins, but those are the ones that are — in my opinion — the more prominent effects in that

suite.

Because SoX very often uses rich command line options for its effects, not every effect configuration from SoX can be fully transported into the slider oriented GUI of the SoX-Plugins. For example, the compander of SoX allows the definition of a transfer function having multiple segments. Although the internal engine of the SoX-Plugins compander implements exactly the same internal segment logic of SoX, the user interface only allows the typical definition of a threshold and a compression ratio (leading to a transfer function with three segments).

Note also that a spiffy user interface is *not at all* a priority for this package. Also the parameter ranges are somewhat debatable, because they simply reflect the SoX command parameters.

The redesign and restructuring has also been done for easier maintenance, because there is some redundancy and unnecessary complexity in the original source due to its several contributors. Nevertheless — as pointed out — the effects provided here faithfully model the SoX command-line processing.

All the code is open-source; hence you can check and adapt it to your needs (see chapter 5).

1.2 Available Effects

The following effects are supported:

allpass:

a biquad allpass filter two-poled with filter frequency and the filter bandwidth (in several units)

band:

a biquad band filter with center filter frequency and the filter bandwidth (in several units) and an option for unpitched audio

bandpass:

a biquad filter for bandpass with center filter frequency and the filter bandwidth (in several units)

bandreject:

a biquad filter for bandreject with center filter frequency and the filter bandwidth (in several units)

bass:

a biquad filter for boosting or cutting bass with a shelving characteristics with settings for filter frequency and the filter bandwidth (in several units)

biquad:

a generic biquad (iir) filter with 6 coefficients b0, b1, b2, a0, a1 and a2

compand:

a compander with attack, decay, input gain shift, threshold and compression and soft knee; this is a reduced version of SoX compand with only a simple transfer function and a combined attack/decay setting

equalizer:

a biquad filter for equalizing with settings for the pole count, the filter frequency and the filter bandwidth (in several units)

gain:

a volume changer by *exact* decibels...

highpass:

a biquad filter for highpass with settings for the pole count, the filter frequency and the filter bandwidth (in several units)

mcompand:

a multiband compander with a Linkwitz-Riley crossover filter and for every band a compander with attack, decay, input gain shift, threshold and compression and soft knee; again the companders only allow a simple transfer function and a combined attack/decay setting

lowpass:

a biquad filter for lowpass with settings for the pole count, the filter frequency and the filter bandwidth (in several units)

overdrive:

a simple tanh distortion with gain and colour specification

phaser:

a phaser effect with sine or triangle modulation

reverb:

a reverb effect (based on Freeverb) with several parameters for the room (like size and HF damping) as well as a possible predelay

treble:

a biquad filter for boosting or cutting treble with a shelving characteristics with settings for filter frequency and the filter bandwidth (in several units)

tremolo:

a tremolo effect with sine modulation using a double-sideband suppressed carrier modulation

Chapter 2

Installation of the SoX-Plugins Effects

The installation is as follows:

1. Copy the plugins from the `_DISTRIBUTION` directory in `[SoXVST]` into the directory for VST plugins of your DAW.
2. The distribution also contains this documentation pdf file in subdirectory `doc` and test files in subdirectory `test` (see section 4).
3. Restart your DAW and rescan the VST plugins. You should now be able to select all the SoX-Plugins(they are all prefixed by "SoX").

Chapter 3

Description of the Effects in SoX-Plugins

3.1 General Remarks

As mentioned in the introduction this package provides several audio tools written in C++ for emulating SoX bit-exactly.

This goal is reached up to a certain precision (of about -120dBFS), because SoX often uses 32bit integer processing while SoX-Plugins uses 64bit float processing.

Where noted in the following description, some simplifications have been done to take care of the limited user interface and also some parameters were omitted.

Note again that the focus of this toolset is the faithful reimplementation and somehow a redesign of SoX; a spiffy user interface is *not at all* a priority in this project.

For the same reasons none of the effects of SoX-Plugins displays anything; they just process audio parametrized by their slider settings.

All effects of SoX-Plugins are discussed in alphabetical order in the following chapter. Note that the effects description is mostly taken from the SoX documentation [SoXDOC] except for specifics of the SoX-Plugins effects.

3.2 SoX Allpass Filter

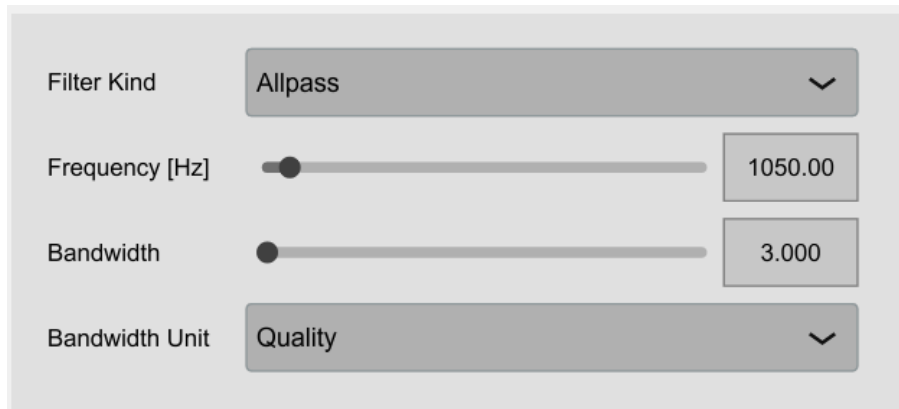


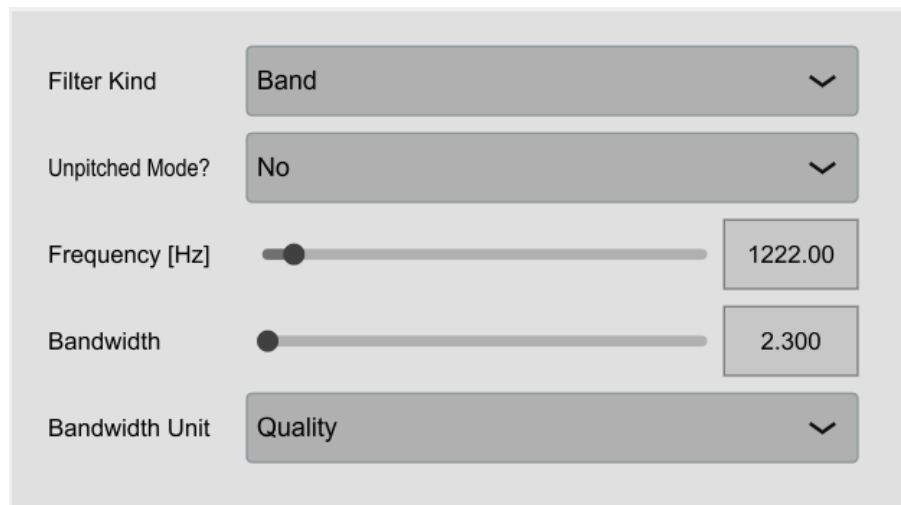
Figure 3.1: Panel for SoX Plugin Allpass

Parameter	Description	Unit
Filter Kind	the kind of the filter (here: Allpass)	Allpass / Band / Bass / BandPass / BandReject / Biquad / Equalizer / HighPass / LowPass / Treble
Frequency	the center frequency of the filter	Hz
Bandwidth	the bandwidth modulus of the filter	—
Bandwidth Unit	the bandwidth unit of the filter	Frequency / Octaves / Quality / Butterworth

This effect is a variant of the plugin `SoxFilter` and implements a two-pole all-pass filter with **Frequency** as center frequency and filter-width **Bandwidth** with unit **Bandwidth Unit**. The bandwidth kinds are a relative *Frequency*, a specification of *octaves*, the filter *quality* or the *butterworth* quality (with fixed quality $q = \sqrt{2}/2$).

An all-pass filter changes the audio's frequency-to-phase relationship without changing its frequency-to-amplitude relationship. The detailed filter description can be found in [RBJFILT].

3.3 SoX Band Filter



The image shows a graphical user interface for the SoX Band filter. It consists of five rows of controls:

- Filter Kind:** A dropdown menu currently showing 'Band'.
- Unpitched Mode?:** A dropdown menu currently showing 'No'.
- Frequency [Hz]:** A horizontal slider with a value of 1222.00 displayed on the right.
- Bandwidth:** A horizontal slider with a value of 2.300 displayed on the right.
- Bandwidth Unit:** A dropdown menu currently showing 'Quality'.

Figure 3.2: Panel for SoX Plugin Band

Parameter	Description	Unit
Filter Kind	the kind of the filter (here: Band)	Allpass / Band / Bass / BandPass / BandReject / Biquad / Equalizer / HighPass / LowPass / Treble
Unpitched Mode?	flag to tell whether special processing for unpitched audio is applied	Boolean
Frequency	the center frequency of the filter	Hz
Bandwidth	the bandwidth modulus of the filter	—
Bandwidth Unit	the bandwidth unit of the filter	Frequency / Octaves / Quality / Butterworth

This effect is a variant of the plugin `SoxFilter` and implements a band-pass filter. The frequency response drops logarithmically around **Frequency**, the **Bandwidth** and **Bandwidth Unit** parameters gives the slope of the drop; frequencies at *frequency+width* and *frequency-width* will be half of their original amplitudes. The effect defaults to a mode that is oriented to pitched audio, i.e. voice, singing, or instrumental music.

When the option **Unpitched Mode?** is set, an alternate mode for un-pitched audio (e.g. percussion) is applied. Note that this option introduces a power-gain of about 11dB in the filter, so beware of output clipping; the option introduces noise in the shape of the filter, i.e. peaking at the center frequency and settling around it.

3.4 SoX Bandpass Filter

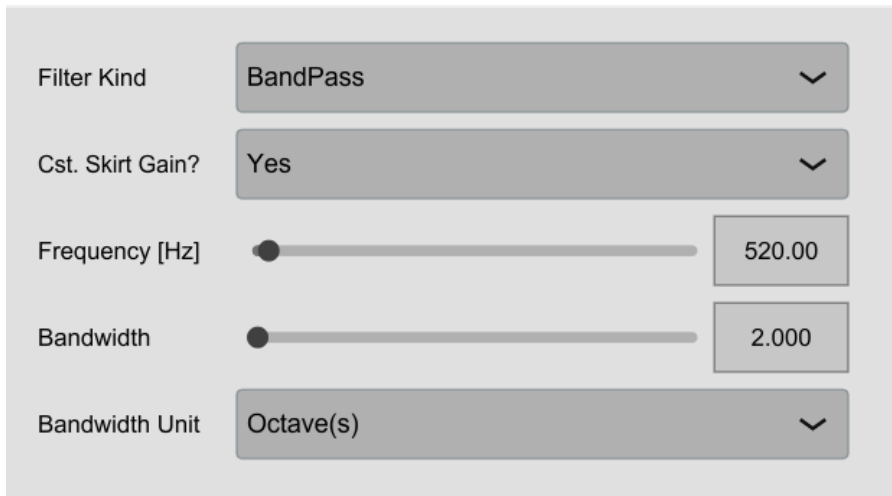


Figure 3.3: Panel for SoX Plugin Bandpass

Parameter	Description	Unit
Filter Kind	the kind of the filter (here: Band-Pass)	Allpass / Band / Bass / BandPass / BandReject / Biquad / Equalizer / HighPass / LowPass / Treble
Cst. Skirt Gain?	flag to tell whether a constant skirt gain is applied	Boolean
Frequency	the center frequency of the filter	Hz
Bandwidth	the bandwidth modulus of the filter	—
Bandwidth Unit	the bandwidth unit of the filter	Frequency / Octaves / Quality / Butterworth / Slope

This effect is a variant of the plugin `SoxFilter`; by selecting the appropriate **Filter Kind**, this effect is a two-pole Butterworth band-pass filter with **Frequency** as central frequency, and (3dB-point) band-width given by **Bandwidth** and **Bandwidth Unit**. The **Cst. Skirt Gain?** option selects a constant skirt gain (peak gain = Q) instead of the default, which is a constant 0dB peak gain. The filters roll off at 6dB per octave (20dB per decade).

The detailed filter description can be found in [RBJFILT].

3.5 SoX Bandreject Filter

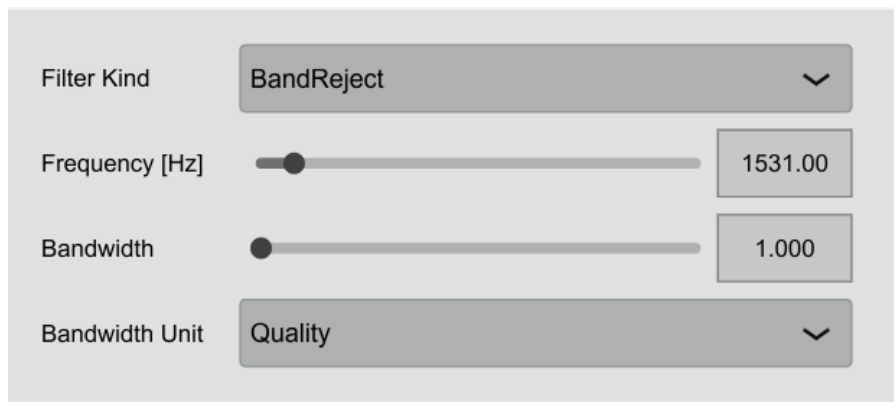


Figure 3.4: Panel for SoX Plugin Bandreject

Parameter	Description	Unit
Filter Kind	the kind of the filter (here: BandReject)	Allpass / Band / Bass / BandPass / BandReject / Biquad / Equalizer / HighPass / LowPass / Treble
Frequency	the center frequency of the filter	Hz
Bandwidth	the bandwidth modulus of the filter	—
Bandwidth Unit	the bandwidth unit of the filter	Frequency / Octaves / Quality / Butterworth / Slope

This effect is a variant of the plugin `SoxFILTER`; by selecting the appropriate **Filter Kind**, this effect is a two-pole Butterworth band-reject filter with **Frequency** as central frequency, and (3dB-point) band-width given by **Bandwidth** and **Bandwidth Unit**.

The detailed filter description can be found in [RBJFILT].

3.6 SoX Bass Filter

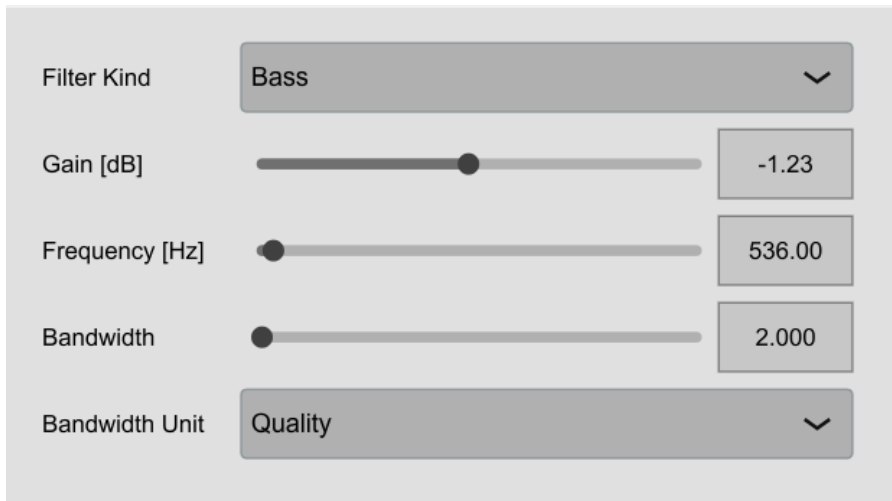


Figure 3.5: Panel for SoX Plugin Bass

Parameter	Description	Unit
Filter Kind	the kind of the filter (here: Bass)	Allpass / Band / Bass / BandPass / BandReject / Biquad / Equalizer / HighPass / LowPass / Treble
Gain	gain of filter at 0Hz	dB
Frequency	the center frequency of the filter	Hz
Bandwidth	the bandwidth modulus of the filter	—
Bandwidth Unit	the bandwidth unit of the filter	Frequency / Octaves / Quality / Butterworth

This effect is a variant of the plugin `SoxFilter`; by selecting the appropriate **Filter Kind**, this effect boosts or cuts the bass (lower) frequencies of the audio using a two-pole shelving filter with a response similar to that of a standard hi-fi's tone-controls. This is also known as shelving equalisation (EQ).

The parameters are as follows:

- **Gain** gives the gain at 0Hz. Its useful range is about -20 (for a large cut) to +20 (for a large boost). Beware of clipping when using a positive gain.
- **Frequency** sets the filter's central frequency and can be used to extend or reduce the frequency range to be boosted or cut.
- The band-width given by parameters **Bandwidth** and **Bandwidth Unit** determines how steep is the filter's shelf transition. In addition to the

common width specification methods described above, “slope” may be used. The useful range of slope is about 0.3, for a gentle slope, to 1 (the maximum), for a steep slope.

The detailed filter description can be found in [RBJFILT].

3.7 SoX Biquad Filter

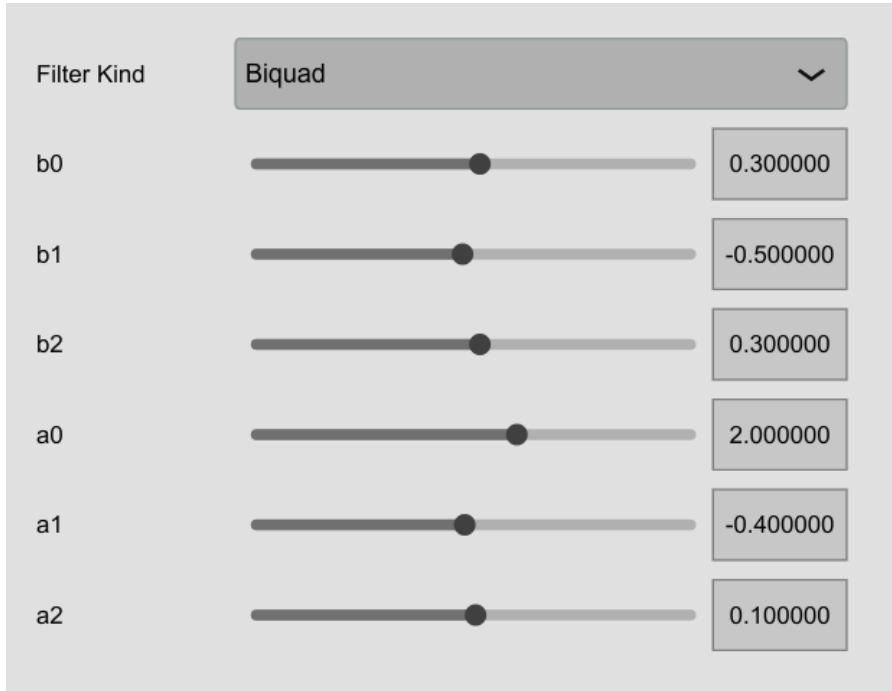


Figure 3.6: Panel for SoX Plugin Biquad

Parameter	Description	Unit
Filter Kind	the kind of the filter (here: Biquad)	Allpass / Band / Bass / BandPass / BandReject / Biquad / Equalizer / HighPass / LowPass / Treble
b0	coefficient for x_n	—
b1	coefficient for x_{n-1}	—
b2	coefficient for x_{n-2}	—
a0	coefficient for y_n	—
a1	coefficient for y_{n-1}	—
a2	coefficient for y_{n-2}	—

This effect is a variant of the plugin `SoxFilter` and implements a biquad IIR filter with the given coefficients (see [DBIQFILT]). It implements the (direct form) function

$$y_n = \sum_{i=0}^2 b_i x_{n-i} - \sum_{i=1}^2 a_i y_{n-i}$$

and is the basis for the other biquad filters (like e.g. the “SoX Equalizer”).

3.8 SoX Compand

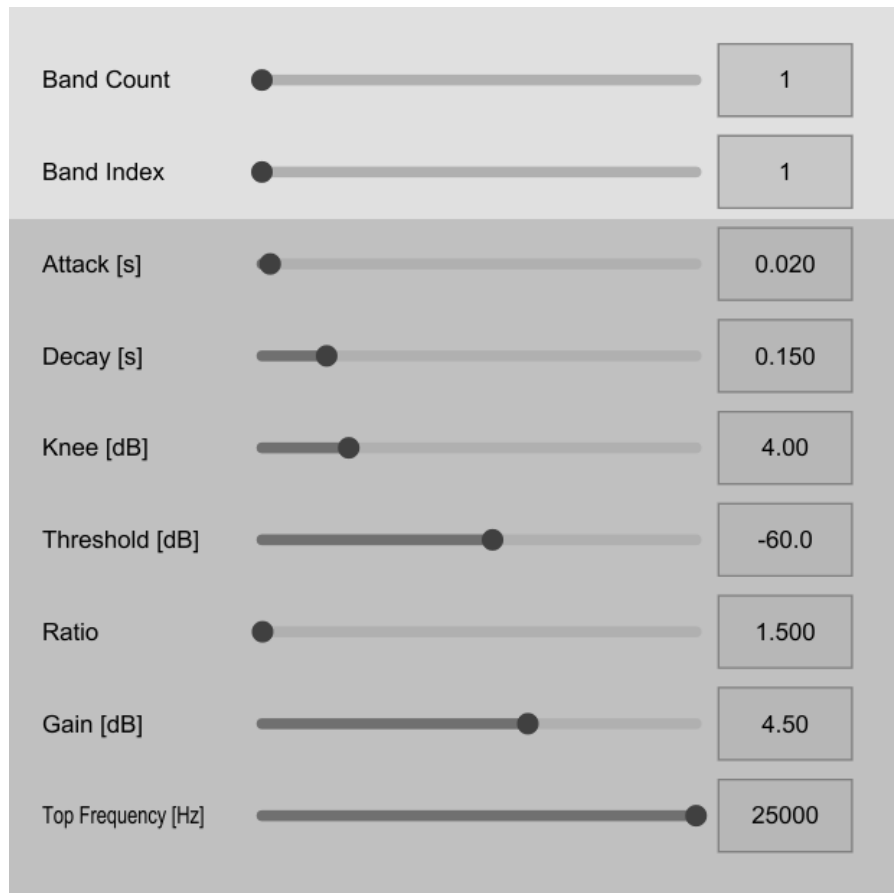


Figure 3.7: Panel for SoX Plugin Compand

Parameter	Description	Unit
Band Count	the count of the bands (here: 1)	—
Band Index	the index of the band to be adapted (here: 1)	—
Attack	the attack time of the compander	s
Decay	the decay time of the compander	s
Knee	the rounding of the corners in the transfer function	dB
Threshold	the threshold of the compander	dBFS
Ratio	the compression factor of the compander	—
Gain	the compander gain before processing	dB
Top Frequency	the compander band top frequency (for all but the last active band)	Hz

This effect implements a compander to compress or expand the dynamic range of the audio. A compander and multiband compander are both vari-

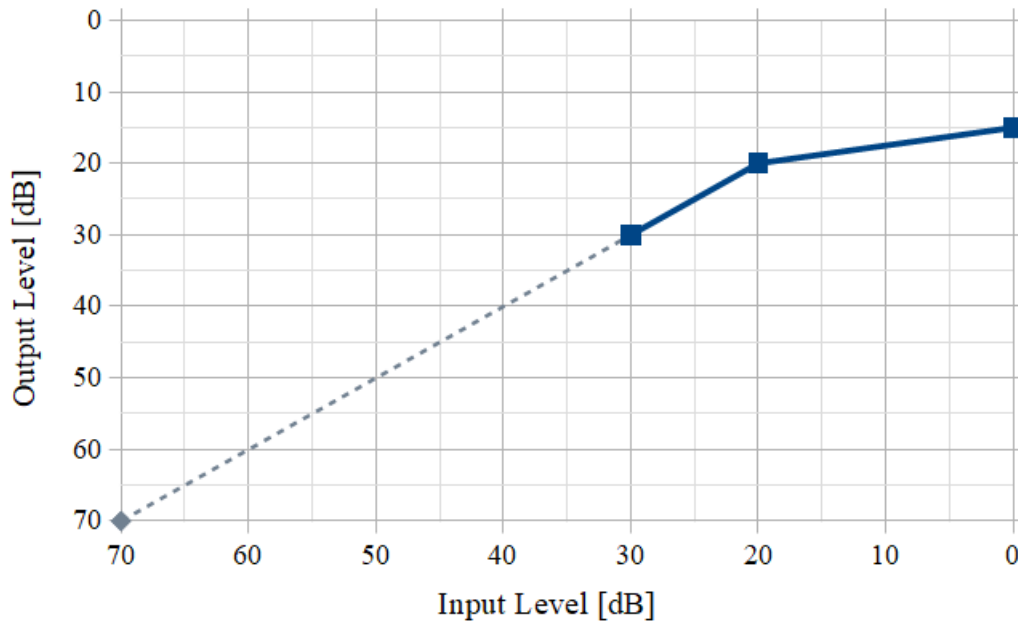


Figure 3.8: Example Transfer Function (Threshold 20dBFS, Ratio 4:1)

ants of the plugin **SoXCompander** where a simple compander is a multiband compander with just one page.

Band Count tells the number of bands in a multiband compander, for a single band compander this must be set to 1. **Band Index** selects the band whose parameters shall be modified, for a single band compander this also must be set to 1, because there is only one active band.

The parameters **Attack** and **Decay** (in seconds) determine the time over which the instantaneous level of the input signal is averaged to determine its volume; attacks refer to increases in volume and decays refer to decreases. For most situations, the attack time (response to the music getting louder) should be shorter than the decay time because the human ear is more sensitive to sudden loud music than sudden soft music. Typical values are 0.3s for attack and 0.8s for decay.

The transfer function of the compander is given by parameters **Threshold**, **Ratio** and **Knee**. The compander leaves the original level unchanged, when it is below threshold and compresses it by ratio beyond this threshold. So e.g. for a threshold of 20dBFS, a knee of 0dB and a ratio of 4:1 the transfer function is a graph shown in figure 3.8. Note that for technical reasons SoX uses a linear lead-in segment with size 10dB below threshold value.

If the parameter **Knee** is greater than 0, the corner points of the transfer function will be rounded by that amount.

The parameter **Gain** is an additional gain in dB to be applied at all points on the transfer function and allows easy adjustment of the overall gain.

As mentioned before the compander is a multiband compander with one band; hence the setting for the **Top Frequency** is not used.

Restriction: Only one overall pair of attack/decay parameters may be specified (where SoX allows one pair per channel). This is in principle supported by the effects engine of SoX-Plugins, but not supported in the current user interface.

Restriction: The original SoX allows an arbitrary multi-segmented transfer function. This is in principle supported by the effects engine of SoX-Plugins, but not supported in the current user interface.

Restriction: There is no delay parameter for delayed compansion.

3.9 SoX Equalizer Filter

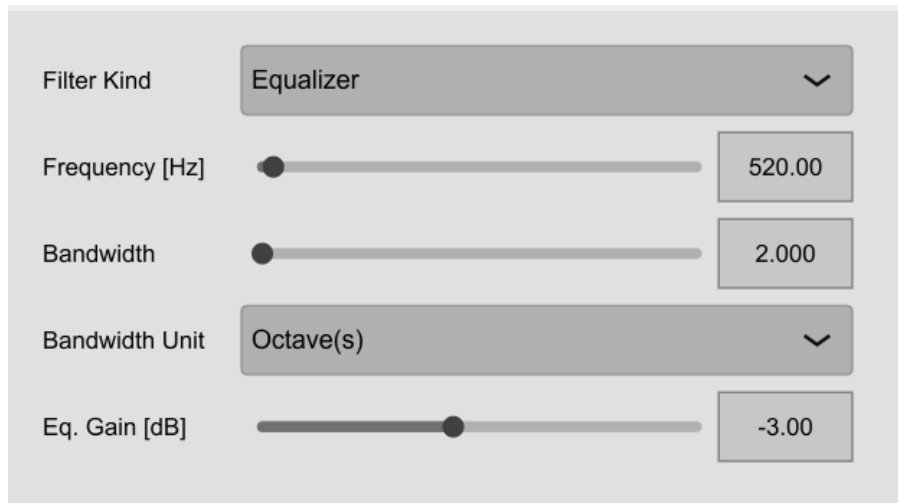


Figure 3.9: Panel for SoX Plugin Equalizer

Parameter	Description	Unit
Filter Kind	the kind of the filter (here: Equalizer)	Allpass / Band / Bass / BandPass / BandReject / Biquad / Equalizer / HighPass / LowPass / Treble
Frequency	the 3dB point frequency of the filter	Hz
Bandwidth	the bandwidth modulus of the filter	—
Bandwidth Unit	the bandwidth unit of the filter	Frequency / Octaves / Quality / Butterworth
Eq. Gain	gain of filter at frequency	dB

This effect is a variant of the plugin `SoxFilter` and implements a two-pole peaking equalisation (EQ) filter. With this filter, the signal-level at and around a selected frequency can be increased or decreased, whilst (unlike band-pass and band-reject filters) that at all other frequencies is unchanged.

The parameter **Frequency** gives the filter’s central frequency in Hz, parameters **Bandwidth** and **Bandwidth Unit** the bandwidth and **Gain** the required amplification or attenuation in decibels. Beware of clipping when using a positive gain.

The filter is described in detail in [RBJFILT].

3.10 SoX Gain



Figure 3.10: Panel for SoX Plugin Gain

Parameter	Description	Unit
Gain	the amplification or attenuation factor	dB

This effect is an amplifier or attenuator for the audio signal with a single **Gain** parameter in decibels. The gain factor applies to all channels identically.

Nothing special, but note that the calculation is exact, hence a gain of -6dB does *not* halve the signal (but a gain of -6.0206dB does quite well).

3.11 SoX Highpass Filter



Figure 3.11: Panel for SoX Plugin Highpass

Parameter	Description	Unit
Filter Kind	the kind of the filter (here: High-pass)	Allpass / Band / Bass / BandPass / BandReject / Biquad / Equalizer / HighPass / LowPass / Treble
Number of Poles	selects between single and double pole filter	single/double
Frequency	the 3dB point frequency of the filter	Hz
Bandwidth	the bandwidth modulus of the filter	—
Bandwidth Unit	the bandwidth unit of the filter	Frequency / Octaves / Quality / Butterworth

This effect is a variant of the plugin `SoxFilter` and implements a high-pass filter with a 3dB point **Frequency**. Depending on **Number of Poles** the filter can be either single-pole or double-pole. The parameters **Bandwidth** and **Bandwidth Unit** apply only to double-pole filters; a Butterworth response is given by butterworth selection or by a q of 0.707. The filters roll off at 6dB per pole per octave (20dB per pole per decade).

The double-pole filters are described in detail in [RBJFILT].

3.12 SoX Lowpass Filter

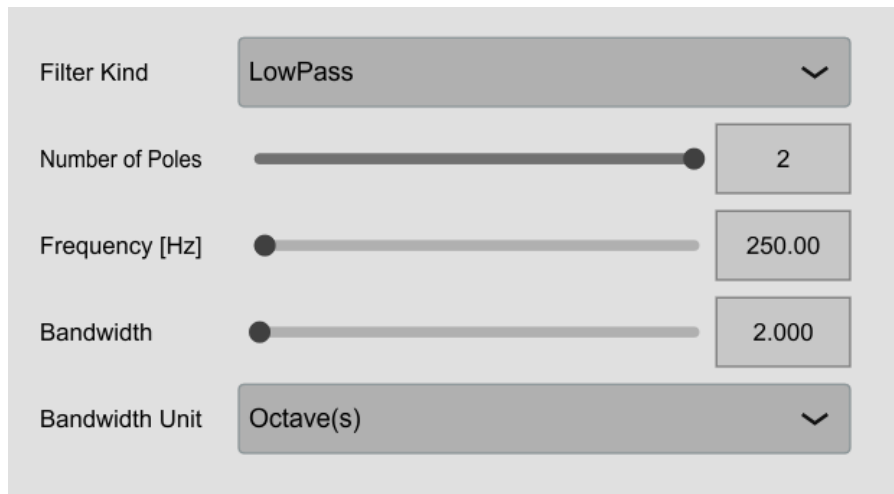


Figure 3.12: Panel for SoX Plugin Lowpass

Parameter	Description	Unit
Filter Kind	the kind of the filter (here: Low-pass)	Allpass / Band / Bass / BandPass / BandReject / Biquad / Equalizer / HighPass / LowPass / Treble
Number of Poles	selects between single and double pole filter	single/double
Frequency	the 3dB point frequency of the filter	Hz
Bandwidth	the bandwidth modulus of the filter	—
Bandwidth Unit	the bandwidth unit of the filter	Frequency / Octaves / Quality / Butterworth

This effect is a variant of the plugin **SoxFilter** and implements a low-pass filter with a 3dB point **Frequency**. Depending on **Number of Poles** the filter can be either single-pole or double-pole. The parameters **Bandwidth** and **Bandwidth Unit** apply only to double-pole filters; a Butterworth response is given by butterworth selection or by a q of 0.707. The filters roll off at 6dB per pole per octave (20dB per pole per decade).

The double-pole filters are described in detail in [RBJFILT].

3.13 SoX MCompand

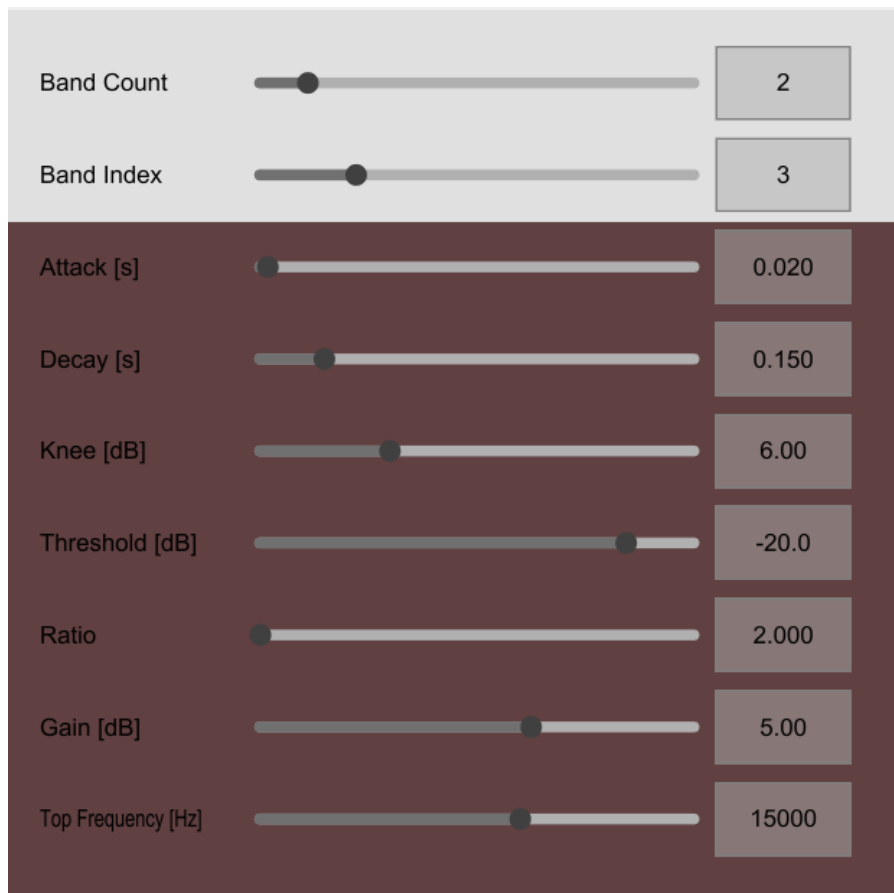


Figure 3.13: Panel for SoX Plugin MCompand

Parameter	Description	Unit
Band Count	the count of the bands	—
Band Index	the index of the band to be adapted	—
for every band		
Attack	the attack time of the compander band	s
Decay	the decay time of the compander band	s
Knee	the rounding of the corners in the transfer function	dB
Threshold	the threshold of the compander band	dBFS
Ratio	the compression factor of the compander band	—
Gain	the compander band gain before processing	dB
Top Frequency	the compander band top frequency (for all but the last active band)	Hz

Compander and multiband compander are both variants of the plugin **SoX-Compander** where a simple compander is a multiband compander with just one page. For a general multi-band compander the audio is first divided into bands using Linkwitz-Riley cross-over filters and later separately specifiable compander run on every band (see the compand effect in 3.8 for the definition of its parameters).

Band Count gives the number of bands in a multiband compander which can be set to the desired number, where the maximum is 10. **Band Index** selects the band whose parameters shall be modified; note that it is possible to modify any band, but the **inactive bands are signified a special reddish background** (see figure 3.13).

Restriction: Only one overall pair of attack/decay parameters may be specified per band (where SoX allows one pair per channel). This is in principle supported by the effects engine of SoX-Plugins, but not supported in the current user interface.

Restriction: The original SoX allows an arbitrary multi-segmented transfer function. This is in principle supported by the effects engine of SoX-Plugins, but not supported in the current user interface.

Restriction: There is no delay parameter for delayed compansion.

Restriction: It is not possible to change the number of bands, but one has to select one of several multi-band companders with two to five bands.

3.14 SoX Overdrive

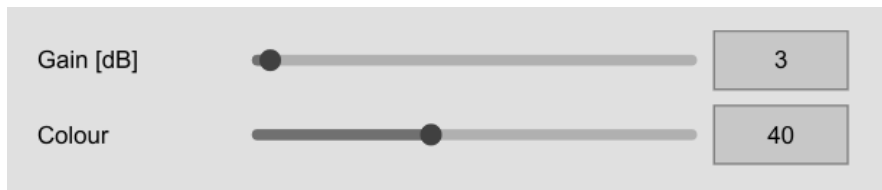


Figure 3.14: Panel for SoX Plugin Overdrive

Parameter	Description	Unit
Gain	the overdrive gain before processing	dB
Colour	percentage for the amount of even harmonic content in output	—

This effect implements an tanh overdrive. **Gain** gives the input gain in decibels, The parameter **Colour** controls the amount of even harmonic content in the over-driven output.

3.15 SoX Phaser

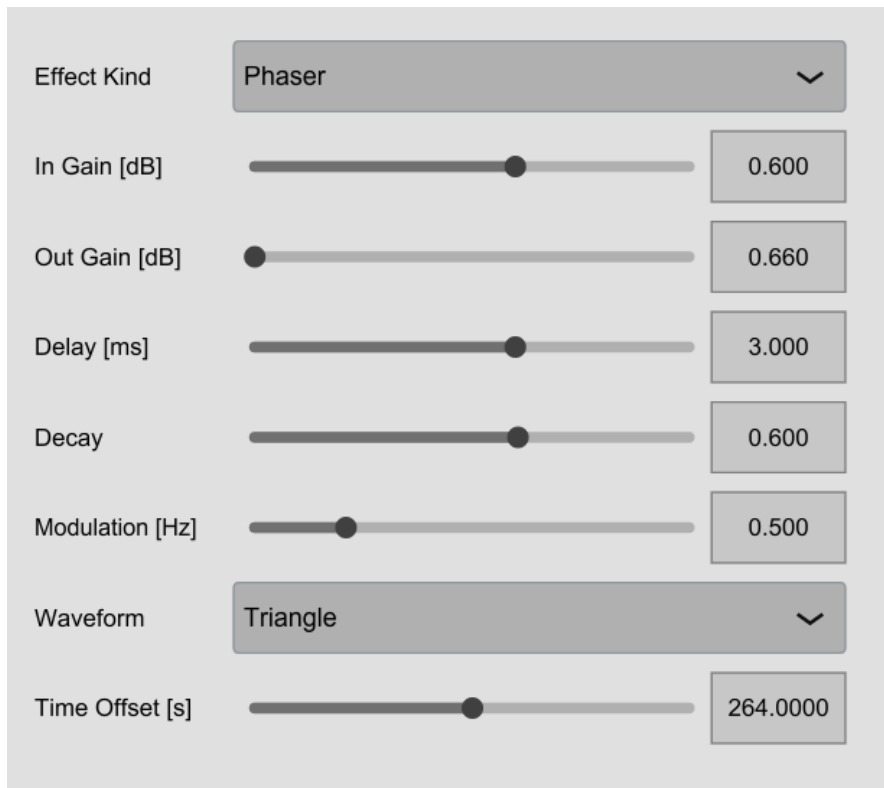


Figure 3.15: Panel for SoX Plugin Phaser

Parameter	Description	Unit
Effect Kind	the kind of the modulation (here: Phaser)	Phaser / Tremolo
In Gain	the gain before processing	dB
Out Gain	the gain applied after processing	dB
Delay	the predelay of the effect	ms
Decay	the decay factor of the phaser	—
Frequency	the phaser modulation frequency	Hz
Waveform	the modulation waveform	Sine / Triangle
Time Offset	the point in project time where modulation is at phase 0° (see 3.19)	s

This effect is a variant of the plugin `SoxPhaserAndTremolo` and implements a phaser effect to the audio.

In Gain is the amplification factor for the input in decibels. **Delay** gives the delay in milliseconds, **Decay** a factor for the decay within the phaser and **Frequency** gives the modulation frequency in Hz. The **Waveform** of the modulation is either sinusoidal — preferable for multiple instruments — or triangular — gives single instruments a sharper phasing effect —. **Out Gain** is the volume of the output.

The decay should be less than 0.5 to avoid feedback, and usually no less than 0.1.

Time Offset shows that this effect is time-locked. For details refer to section 3.19.

3.16 SoX Reverb

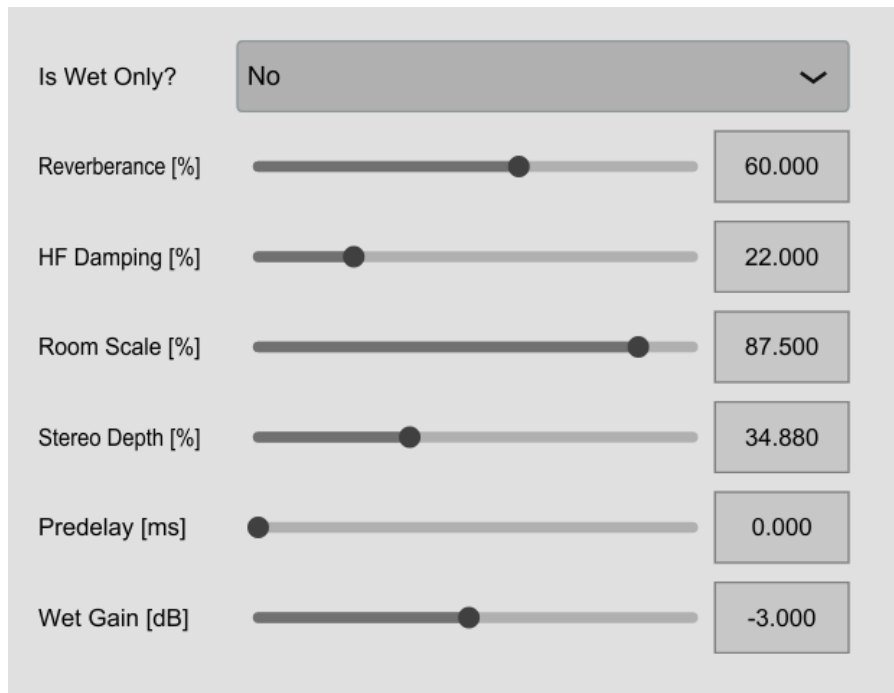


Figure 3.16: Panel for SoX Plugin Reverb

Parameter	Description	Unit
Is Wet Only?	tells whether only the wet signal should be produced	Boolean
Reverberance	percentage for reverb density	—
HF Damping	percentage amount of damping of high frequencies for every reflection relative to low frequencies	—
Room Scale	percentage for size of the room (more precisely the reflectivity of the room)	—
Stereo Depth	percentage amount of stereo effect	—
Predelay	time offset until first reverb occurs	ms
Wet Gain	gain of wet signal relative to dry signal	dB

This effect implements reverberation of audio using the “freeverb” algorithm, which uses eight parallel Schröder-Moorer filtered-feedback comb-filters followed by four Schröder allpasses in series.

Details on this algorithm can be found in [FREEVERB].

3.17 SoX Treble Filter

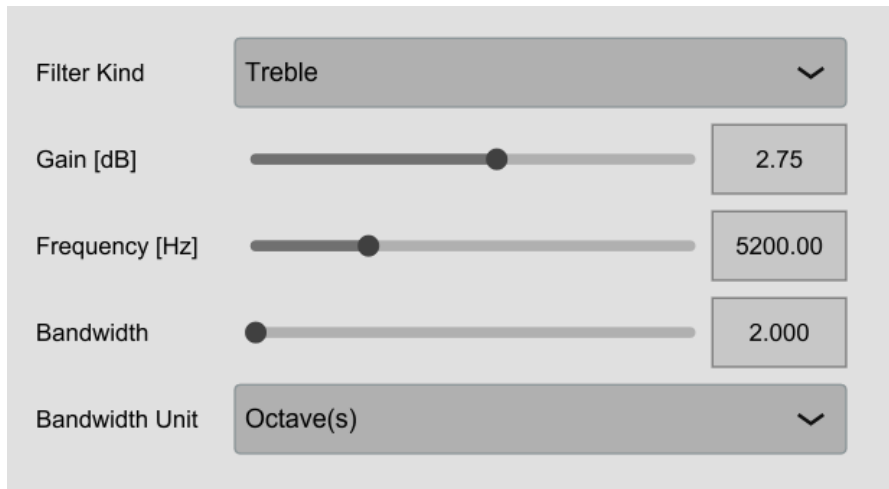


Figure 3.17: Panel for SoX Plugin Treble

Parameter	Description	Unit
Filter Kind	the kind of the filter (here: Treble)	Allpass / Band / Bass / BandPass / BandReject / Biquad / Equalizer / HighPass / LowPass / Treble
Gain	gain of filter at 22kHz	dB
Frequency	the center frequency of the filter	Hz
Bandwidth	the bandwidth modulus of the filter	—
Bandwidth Unit	the bandwidth unit of the filter	Frequency / Octaves / Quality / Butterworth

This effect is a variant of the plugin **SoxFilter**; by selecting the appropriate **Filter Kind**, this effect boosts or cuts the treble (upper) frequencies of the audio using a two-pole shelving filter with a response similar to that of a standard hi-fi's tone-controls. This is also known as shelving equalisation (EQ).

The parameters are as follows:

- **Gain** gives the gain at a frequency whichever is the lower of 22kHz and the Nyquist frequency (for treble). Its useful range is about -20 (for a large cut) to +20 (for a large boost). Beware of clipping when using a positive gain.
- **Frequency** sets the filter's central frequency and can be used to extend or reduce the frequency range to be boosted or cut.

- The band-width given by parameters **Bandwidth** and **Bandwidth Unit** determines how steep is the filter's shelf transition. In addition to the common width specification methods described above, "slope" may be used. The useful range of slope is about 0.3, for a gentle slope, to 1 (the maximum), for a steep slope.

The detailed filter description can be found in [RBJFILT].

3.18 SoX Tremolo

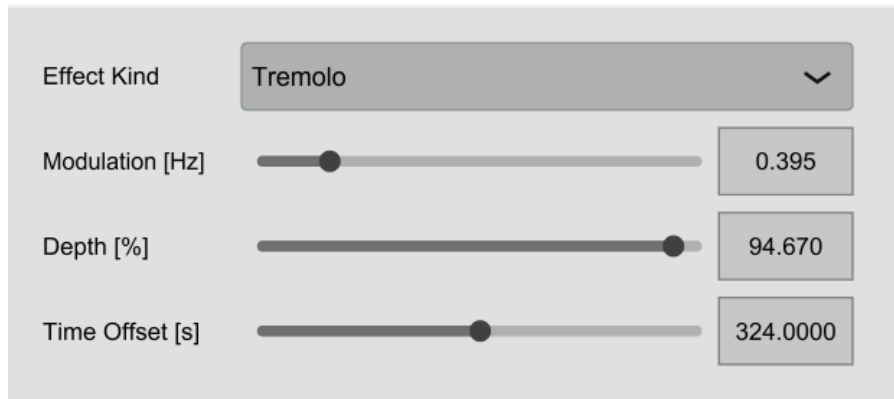


Figure 3.18: Panel for SoX Plugin Tremolo

Parameter	Description	Unit
Effect Kind	the kind of the modulation (here: Tremolo)	Phaser / Tremolo
Frequency	the modulation frequency of the tremolo	Hz
Depth	percentage value for the intensity of modulation	—
Time Offset	the point in project time where modulation is at phase 0° (see 3.19)	s

This effect is a variant of the plugin `SoxPhaserAndTremolo` and implements a tremolo effect. This tremolo is done by signal multiplication; hence it is a low frequency double sideband suppressed carrier modulation. Parameter **Frequency** gives the tremolo frequency, **Depth** gives the intensity as a percentage.

Time Offset shows that this effect is time-locked. For details refer to section 3.19.

3.19 Timelocking

There are effects that behave differently in time, technically they are *time-variant*. A filter does not care *when* a signal arrives, but a modulated effect like e.g. a phaser produces a different sound for different start times because the modulation is normally in another phase.

Hence when looking at the behaviour at a specific point in time, those time-variant effects would behave differently when the effect start time is varied.

For example, assume a phaser with a 0.25Hz modulation (one cycle every 4s): when you start the effect 1s later, its modulation is now off by 90°. This is not helpful when the effect now depends on start time or loop positioning.

To circumvent this problem, all time variant effects from above (phaser and tremolo) are *time-locked* i.e. they check the current play position and always behave the same at some specific point in time regardless of the playback start time.

Additionally those effects have a parameter called **Time Offset**. This parameter tells at what time the effect has a phase of zero in its modulation. The default is 0s, but it may be adapted accordingly.

Take the phaser above and assume you want to make sure that its modulation is exactly at 0° at position 155s within your song¹. Then you just set **Time Offset** to “155”. Because the period of the modulation is 4s it is also okay to use $155 + 4k, k \in \mathbb{Z}$ as offset (e.g. “3”), but the above saves you from some calculation for complicated modulation frequencies.

By this method even time-variant effects can be synced with externally generated audio material.

¹This is a little lie, because the initial phaser modulation phase is 90°, but the argument is still valid.

Chapter 4

Regression Test

To test that the effects of SoX-Plugins really are bit-identical to SoX, a little test suite has been set up for the Reaper DAW.

The suite assumes that command-line SoX is installed in the Windows search path.

If so, a simple batch script sets up raw audio test files and — externally via the command line — applies SoX effects to them producing audio result files. The parameters used are a bit exotic to ensure that algorithmic differences between SoX and SoX-Plugins will show up. The batch script can be found in the `test` subdirectory and is called `makeTestFiles.bat`

Since there are so many DAWs available, it is hard to provide a test suite for each of those. The distribution contains a Reaper project referencing those audio test files and result files in autonomous tracks (see figure 4.1). Adaption to other DAW should be straightforward.

The SoX-Plugins effects are configured with the exactly the same parameters as given in the batch file and are correspondingly applied to the raw audio test files.

When subtracting the rendered audio in Reaper and the externally rendered audio from SoX, they (almost) cancel out. This can be checked by a spectrum analyser in the master channel, which is shown in figure 4.2. It shows a noise floor of typically less than -140dB. This comes from the 24 bit sample depth used in the FLAC files of the test suite; increasing that sample depth would even lead to less residual noise.

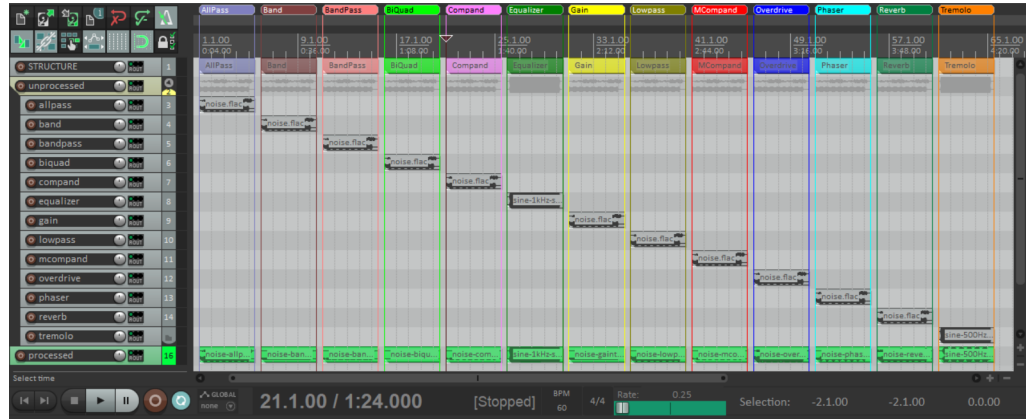


Figure 4.1: Regression Test Setup in Reaper

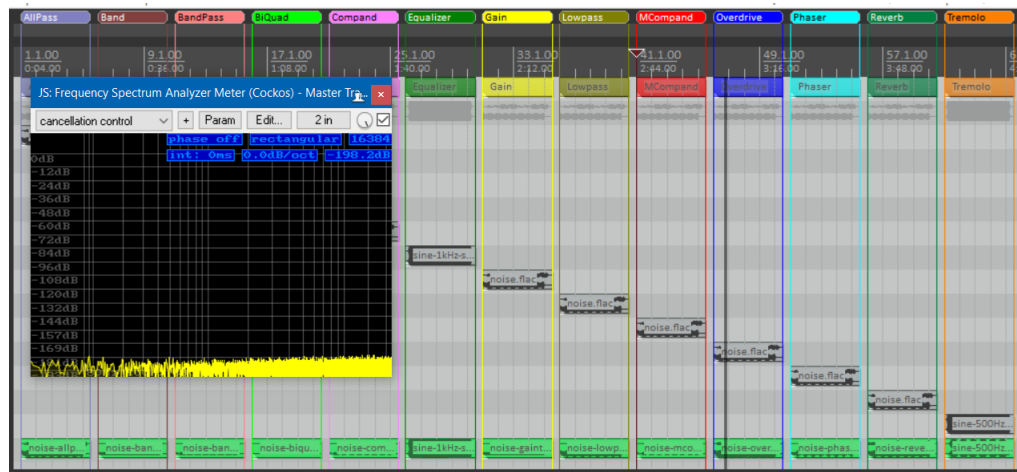


Figure 4.2: Example Noise Floor for Regression Test in Reaper

Chapter 5

Notes on the Implementation

5.1 Overview

The implementation of the SoX-Plugins-plugins is done in C++ and relies on the JUCE library [JUCE]. The algorithms of SoX have not been copied, but significantly refined and reordered in an object-oriented manner. Also redundancies between different modules have (as far as possible) been eliminated, and the processing adapted to double precision floating point numerics where command-line SoX only uses 32bit integer processing for a faster processing.

Those changes have been done with the goal in mind to produce *bit-exact reproductions* of the original algorithms. As shown in section 4 this goal has been achieved.

The complete source code of SoX-Plugins is open-source for easy review and adaptation. Currently there is only a tool chain for Windows 10 (Visual Studio) and VST3 plugins, but in principle the code is easily portable to other plugin formats or platforms.

5.2 Building the Plugins

Preliminaries In the GIT-project of SoX-Plugins (at [SoXVST]) there is a build file for CMAKE to build the plugins for different platforms.

Minimum prerequisites for building are:

- a clone of the GIT-project at <https://github.com/prof-spock/SoX-Plugins>,
- an installation of the audio framework JUCE [JUCE] with version 5 or later,
- some C++ compiler suite for your platform (e.g. Visual Studio, XCode, clang or gcc), and

- an installation of the build automation platform CMAKE [CMAKE] with version 3.10 or later

For documentation generation you can *optionally* install:

- a L^AT_EX installation — like e.g. MikTeX — (for the manual), and
- doxygen [DOXYGEN] and graphviz [GRAPHVIZ] for the internal program documentation

Doing the Build The full build process is started via CMAKE. It is recommended to do a so-called out-of-source-build for the SoX-Plugins, that means, you define some build directory where all build activity is done.

The steps are as follows:

1. Define some build directory (lets say BUILD) and change to it.
2. Find the path of the CMakeList.txt configuration file and configure the build process via

```
cmake -S <pathTo>/CMakeList.txt -B . --config Release
```

3. Build all the plugins via

```
cmake --build . --config Release
```

4. Install the plugins into a architecture-specific subfolder in the _DISTRIBUTION directory and install also the documentation there via

```
cmake --install .
```

5.3 Internal Documentation

In the github repository there is an extensive doxygen documentation available for the inner workings of the plugins at

<https://github.com/prof-spock/SoX-Plugins/internalDocumentation>

with entry point

<https://github.com/prof-spock/SoX-Plugins/internalDocumentation/index.html>.

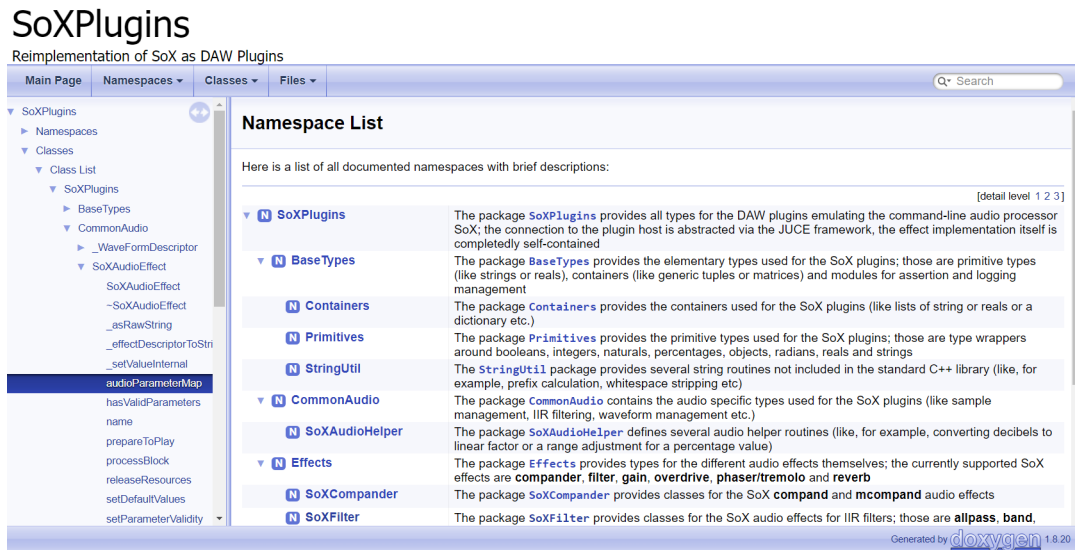


Figure 5.1: Example Namespace Page for Plugins from doxygen

Every public and private feature of all classes and data types is documented and can be analyzed in an HTML browser. Figure 5.1 gives an impression how such an HTML page looks like for the namespaces in SoX-Plugins.

If you want to regenerate this documentation from the code, you need an installation of doxygen [DOXYGEN] on your computer. If you have that available, the generation can be done via the CMAKE chain as target `doxygenDocumentation` in the build directory:

```
cmake --build . --config Release \
      --target internalDocumentation
```

If the command completes, the documentation in the `internalDocumentation` subdirectory of the project is updated.

To trigger regeneration, it suffices to delete the file `internalDocumentation/html/index.html`.

5.4 Debugging

For debugging purposes, every VST can also exist as a debugging version that does an extensive entry-exit-logging into the temp directory. Note that this debugging slows down processing extremely, but it helps to understand problems in case of errors. Figure 5.2 shows how a logging file looks like.

Every non-trivial function is logged there at least twice with timestamps: “»” indicates the entry of that function (possibly with information on the argument values), “«” the exit of that function (possibly with the return

5.4. DEBUGGING

```
START LOGGING -*- coding: utf-8 -*-
>>SoxAudioParameterMap.SoxAudioParameterMap (004227.66)
<<SoxAudioParameterMap.SoxAudioParameterMap (004227.67): SoxAudioParameterMap(_parameterNameList = StringList(),
_parameterNameToValueMap = Dictionary(), _parameterNameToKindMap = Map(), _parameterNameToValueRangeMap = Dictionary(),
_activeParameterNameSet = StringSet())
>>SoxCompander_AudioEffect.SoxCompander_AudioEffect (004227.68)
>>_createEffectDescriptor (004227.69)
>>SoxMultibandCompander.SoxMultibandCompander (004227.70)
<<SoxMultibandCompander.SoxMultibandCompander (004227.71): SoxMultibandCompander(_allocatedBandCount = 0, _effectiveBandCount =
0, _channelCount = 0, _companderBandList = _MCompanderBandList())
<<_createEffectDescriptor (004227.72)
>>_initializeAllParameters (004227.73)
>>SoxAudioParameterMap.setKindInt (004227.74): 0#BANDCOUNT, range = (1, 5, 1)
>>SoxAudioParameterMap.setValue (004227.75): parameterName = 0#BANDCOUNT, value = 1
>>SoxAudioParameterMap.isAllowedValue (004227.76): parameterName = 0#BANDCOUNT, value = 1
>>SoxAudioParameterMap.valueRangeInt (004227.77): 0#BANDCOUNT
>>SoxAudioParameterMap.kind (004227.78): 0#BANDCOUNT
<<SoxAudioParameterMap.kind (004227.79): intKind
>>SoxAudioParameterMap.valueRangeInt (004227.80): (1, 5, 1)
<<SoxAudioParameterMap.isAllowedValue (004227.81): true
>>SoxAudioParameterMap.setValue (004227.82)
<<SoxAudioParameterMap.setKindInt (004227.83)
>>SoxAudioParameterMap.setKindInt (004227.84): -1#BANDINDEX, range = (1, 5, 1)
>>SoxAudioParameterMap.setValue (004227.85): parameterName = -1#BANDINDEX, value = 1
>>SoxAudioParameterMap.isAllowedValue (004227.86): parameterName = -1#BANDINDEX, value = 1
>>SoxAudioParameterMap.valueRangeInt (004227.87): -1#BANDINDEX
>>SoxAudioParameterMap.kind (004227.88): -1#BANDINDEX
<<SoxAudioParameterMap.kind (004227.90): intKind
>>SoxAudioParameterMap.valueRangeInt (004227.91): (1, 5, 1)
<<SoxAudioParameterMap.isAllowedValue (004227.92): true
>>SoxAudioParameterMap.setValue (004227.93)
<<SoxAudioParameterMap.setKindInt (004227.94)
>>SoxAudioParameterMap.pagedParameterName (004227.95): parameterName = Attack [s], page = 1
<<SoxAudioParameterMap.pagedParameterName (004227.97): 1#Attack [s]
>>SoxAudioParameterMap.setKindReal (004227.98): 1#Attack [s], range = (0.001000, 1.000000, 0.001000)
>>SoxAudioParameterMap.setValue (004227.99): parameterName = 1#Attack [s], value = 0.001000
>>SoxAudioParameterMap.isAllowedValue (004228.00): parameterName = 1#Attack [s], value = 0.001000
>>SoxAudioParameterMap.valueRangeReal (004228.01): 1#Attack [s]
>>SoxAudioParameterMap.kind (004228.02): 1#Attack [s]
<<SoxAudioParameterMap.kind (004228.03): realKind
>>SoxAudioParameterMap.valueRangeReal (004228.04): (0.001000, 1.000000, 0.001000)
<<SoxAudioParameterMap.isAllowedValue (004228.05): true
>>SoxAudioParameterMap.setValue (004228.06)
<<SoxAudioParameterMap.setKindReal (004228.07)
>>SoxAudioParameterMap.pagedParameterName (004228.08): parameterName = Decay [s], page = 1
<<SoxAudioParameterMap.pagedParameterName (004228.09): 1#Decay [s]
>>SoxAudioParameterMap.setKindReal (004228.10): 1#Decay [s], range = (0.001000, 1.000000, 0.001000)
>>SoxAudioParameterMap.setValue (004228.11): parameterName = 1#Decay [s], value = 0.001000
>>SoxAudioParameterMap.isAllowedValue (004228.12): parameterName = 1#Decay [s], value = 0.001000
```

Figure 5.2: Example for Logging File

value) and “_” indicates some intermediate information during the function processing. The logging data is hierarchical, hence you can see the function call structure in this file precisely.

All logging files go to the directory specified by the `temp` environment variable.

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