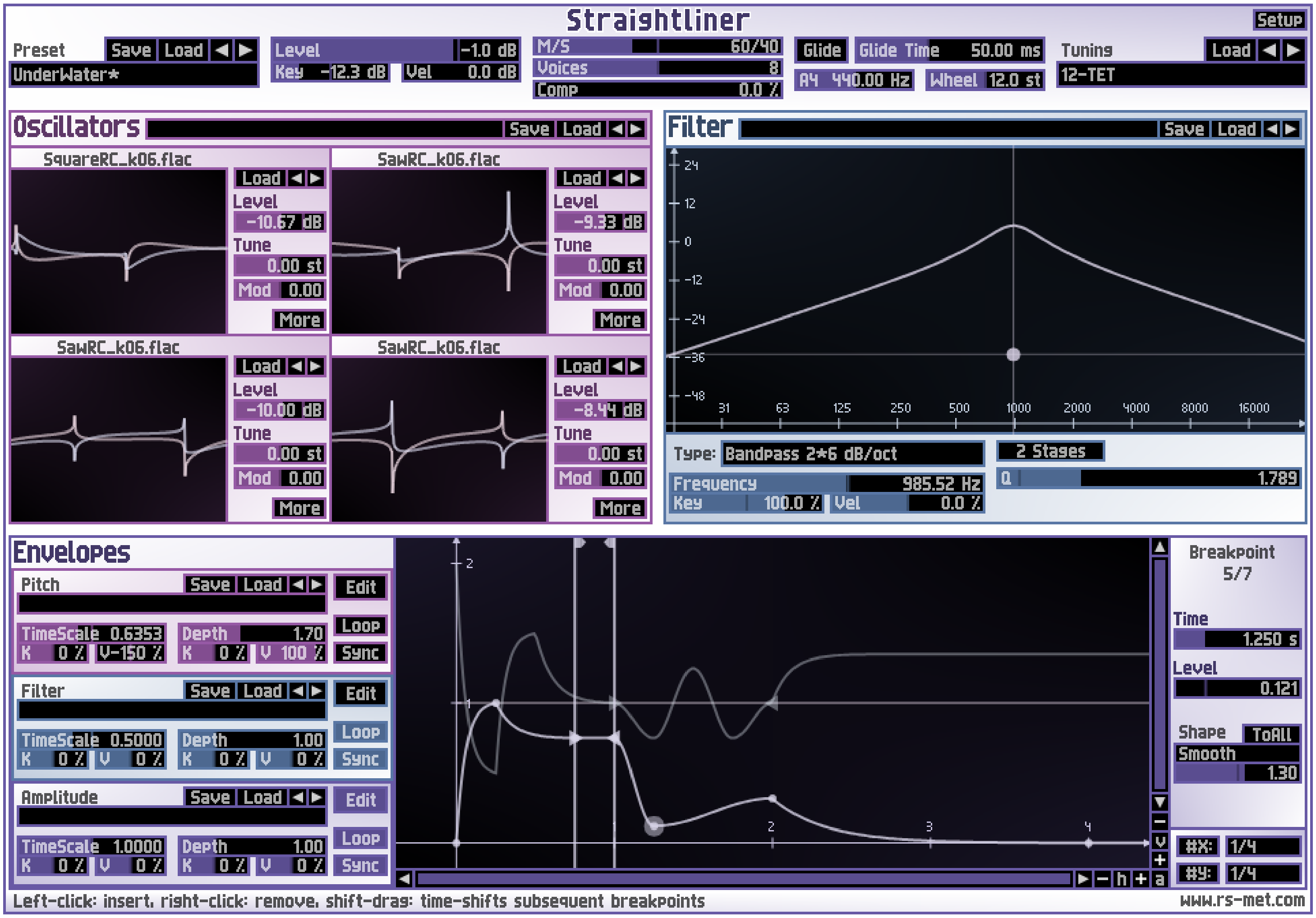
Straightliner - User Manual



What is Straightliner?

Straightliner is a straight-lined subtractive synthesizer in a modern guise. If you like the easy and intuitive feel of subtractive synthesis but are not excited anymore by reverb-drenched virtual-analog appreggios, then Straightliner is the synth for you. You will find the well known building blocks of classic analog synthesizers (4 oscillators, a filter, 3 envelope generators) patched together in the traditional way. However, in Straightliner, these building blocks offer much more flexibility than commonly seen in virtual analog synthesizers: Instead of relying on a set of fixed waveforms, the oscillators support loading of arbitrary (single cycle) audio files which may be manipulated in various ways. The waveforms may even be in stereo and if they are not, they can internally be stereoized by some of the waveform-manipulation facilities. In turn, the whole signal path is laid out in stereo which allows for wide and enveloping sounds even without application of any effects. The standard ADSR model for envelopes is replaced by a flexible modulation generator which allows to define an arbitrary number of breakpoints - by defining a sustain-loop, these modulators can also take over the role of LFOs. The heart of subtractive synthesizers is the filter, of course. In this department, Straightliner offers a multimode filter with 15 modes, among them a sophisticated Moog model. The user interface of the synth focuses on visual editors optimized for editing on screen with a mouse instead of the commonly seen hardware oriented interfaces. These visual editors don't just 'sketch' the resulting curves but exactly represent what is being generated in a 'what you see is what you get' manner.

The Global Parameters

In the upper part of the user interface, you will find the global parameters and the preset management. You can load and save presets with the respective buttons (where the save-button actually works as a 'save as' button). The arrow-buttons allow for fast switching to the next and previous preset in the currently chosen directory (presets are traversed in alphabetical order with wraparound). The sub-editors for the different building blocks (oscillator-section, filter, envelope-generators) have similar functionality to manage their own states as 'module-presets'.

- **Level:** controls the overall output level in dB and the **Key** and **Vel** slider below control the key- and velocity-dependence thereof. The unit of the latter could be explained as 'boost at velocity = 127' - that is: a note played at velocity 127 will be this value louder than a note played at velocity = 64 (which is taken to be neutral velocity). Conversely, a note played at velocity = 1 will be this value quieter. For the key-dependence, it is entirely analogous.
- **Voices:** controls how many voices can play simultaneously, this is also known as polyphony.
- **Comp:** This is an automatic scaling of the overall output level by the number of currently playing voices (and their current amplitudes) in order to compensate for the loudness gain when several voices play simultaneously. When at 100%, the output level (measured in an RMS sense) should stay approximately independent from the number of playing voices.

The Tuning Parameters

Next to the global/preset parameters you find the parameters related to the tuning of the synth. Straightliner supports microtuning by means of importing tuning files in the .tun format and in a proprietary xml-based format. Similarly to the presets, you can load and skip through tuning files. To learn more about tuning files, go here:
http://www.xs4all.nl/ huygensf/scala. Next to the tuning file import, you find:

- **Glide:** Glide is a feature commonly seen in monophonic synthesizers - there, it switches into a mode where the note pitch glides to the new note without re-triggering the envelopes when a new note is pressed without releasing the old one before. When Voices = 1, it works just this way in Straightliner as well, but here the idea is also generalized to the polyphonic case (i call that '**Polyglide**'): when Voices is higher than 1, but you play more notes then there are available voices, an event known as 'voice-stealing' occurs: the most recently pressed note steals a voice from one of the other playing voices (in Straightliner, the oldest voice will be stolen). Now, when glide is active, the stolen voice will not re-trigger it's envelopes but instead glide to the new note. In the monophonic case (Voices = 1), this reduces to the well known standard behavior. The button switches the glide mode on or off and the slider adjusts the glide time.

- **Tuning:** This is the master tuning frequency for the note A4 - most commonly, $440Hz$ is used here. When you are working with a scale different from the standard equal tempered scale, all the frequencies in your scale will be multiplied by a factor of $\frac{\text{Tuning}}{440}$.
- **Wheel:** This is the range for the pitch wheel in semitones.

The Setup

The button on the top-right opens a setup where you can set up your global preferences - currently this is only the colorscheme of the GUI. This should be fairly self-explanatory or easy to explore by experiment, so I won't go into the details here.

The Oscillator Section

The oscillator section comprises 4 oscillators each of which can be equipped with a waveform from a single cycle audio file (accepted file formats are .flac and .wav, stereo or mono). The load- and arrow-buttons work similar as those for the (module-) preset management, a click on the waveform display switches an oscillator on or off (when switched off, CPU cycles can be saved). Level (in dB), Tuning (in semitones) and amount of modulation by the pitch-envelope can be directly dialed in on the GUI's main page, the **More** button opens a context menu with further settings. These are:

Amplitude related settings:

- **Level:** controls the overall output level of the oscillator in dB, with Key and Velocity dependence settings
- **Mid/Side:** adjusts the relative amplitude of the mid and side signal for stereo(ized) waveforms
- **Pan:** panorama position of the oscillator output

Tuning related settings:

- **Tune:** (de)tuning of the oscillator in semitones (with respect to the incoming MIDI note)
- **Detune Hz:** detunes the oscillator by an absolute frequency difference in Hz - this is useful for patches with detuned oscillators where the beating frequency should not depend on the pitch
- **Stereo Detune:** detunes the left against the right stereo channel - half of this value will be applied to the right channel and the other half of the value will be applied with opposite sign to the left channel
- **Stereo Detune Hz:** additional stereo detuning by an absolute frequency difference

Time related settings:

- **Start Phase:** the point inside the waveform at which the waveform starts when the oscillator is re-triggered, expressed in degrees

- **Comb Harmonic:** the waveform can be overlaid with a scaled and time-shifted copy of itself in order to create a comb-filtering effect. With this parameter, you control the harmonic to which this comb will be tuned (thus, it indirectly controls the time-shift of the copy - and it does not need to be an integer value, so it may also be tuned to somewhere in between the actual harmonics)
- **Comb Amount:** This parameter controls the scale-factor of the time-shifted copy, thereby controlling the amount of comb filtering. Positive values will cause an attenuation at integer multiples of the 'Comb Harmonic' and boosts in between, for negative values, it is vice versa.
- **Full Wave Warp:** time warping of the wavetable's readout position. This distorts the time-axis such that early portions of the waveform are read at higher speed and late portions at lower speed or vice versa.
- **Half Wave Warp:** time warping of the wavetable's readout position applied to first and second half-wave separately
- **Reverse:** time reverses the waveform
- **Invert:** inverts the polarity of the waveform

Settings related to the magnitude spectrum:

- **Contrast:** raises the magnitudes of the harmonics to a power - values lower than one even out the differences in the magnitudes, making the spectrum more flat (reduce contrast), values higher than one emphasize the differences in the spectral magnitudes (increase contrast)
- **Slope:** applies a spectral slope/rolloff in dB/oct to the oscillator wave
- **Highest Harmonic:** applies a brickwall lowpass, limiting the highest harmonic that is present in the waveform
- **Lowest Harmonic:** applies a brickwall highpass, limiting the lowest harmonic that is present in the waveform
- **Even/Odd:** Ratio between even and odd harmonics

Settings related to the phase spectrum:

- **Scale:** Scales the phases of all harmonics by a factor.
- **Shift:** Shifts the phases of all harmonics by an offset (in degrees).
- **Even/Odd Shift:** Shifts the phases of the even harmonics against those of the odd harmonics, half of this value will be applied to the even harmonics and the other half will be applied with opposite sign to the odd harmonics
- **Stereo Shift:** Shifts the phases in the right channel against those in the left channel, half of this value will be applied to the right channel and the other half will be applied with opposite sign to the left channel. A value of 90 degrees (plus or minus) turns one channel into the Hilbert-Transform of the other which establishes a zero cross-correlation between the two channels - this is useful for creating stereo-width.

- **Even/Odd Stereo Shift:** Applies an additional phase-shifting of even against odd harmonics half of which which is applied (as is) to even harmonics in the right channel and odd harmonics in the left channel and the other half of which is applied with opposite sign to even harmonics in the left channel and odd harmonics in the right channel

Note that many of these parameters require a re-rendering of the wavetable (actually, a whole mip-map of wavetables) and that requires a lot of computation. They are not supposed to be tweaked/automated in realtime, they are suitable only to set up the waveform statically. On slower machines or under heavy computational load, some of these controls may feel a bit unresponsive/unsmooth.

The Filter

At the heart of a subtractive synthesizer, there is the filter. Straightliner offers a multimode filter with various filter types. These types are chosen with the **Type** combo-box and your choices are:

- **Bypass:** switches the filter off - this saves CPU-power, when no filter is needed.
- **Moogish Lowpass:** a lowpass filter inspired by the famous Moog ladder structure
- **Lowpass 6 dB/oct:** a simple first order lowpass filter with a slope of 6 dB/oct, resembling an analog RC lowpass filter. Lowpass filters let frequencies below their cutoff frequency pass more or less unchanged and (attempt to) block frequencies above the cutoff frequency.
- **Lowpass 12 dB/oct:** a second order (biquad) lowpass filter with a slope of 12 dB/oct and Q control - Q introduces a resonant peak in the vicinity of the cutoff frequency.
- **Highpass 6 dB/oct:** a simple first order highpass filter with a slope of 6 dB/oct, resembling an analog RC highpass filter. Highpass filters let frequencies above their cutoff frequency pass more or less unchanged and (attempt to) block frequencies below the cutoff frequency.
- **Highpass 12 dB/oct:** a second order (biquad) highpass filter with a slope of 12 dB/oct and Q control.
- **Bandpass 2*6 dB/oct:** a second order (biquad) bandpass filter with a slope of 6 dB/oct on both sides and Q control. Bandpass filters let frequencies within their passband pass more or less unchanged and (attempt to) block frequencies above and below their passband. Here, the Q parameter is mainly associated with the bandwidth - high Q values make for narrow passbands.
- **Bandstop 2*6 dB/oct:** a second order (biquad) bandstop (aka notch) filter with Q control. Bandreject filters completely block their center frequency and attenuate frequencies around it. Here, the Q parameter controls the attenuation of these nearby frequencies - high Q values make for narrow notches which means that the attenuation in the vicinity of the notch is restricted to a narrower frequency range. When the notch is wide enough (low Q), the slopes on both sides will asymptotically approach 6 dB/oct.
- **Peak/Dip:** this is a filter which resembles one single band of a parametric equalizer with the familiar controls of center frequency, gain and Q.

- **Low Shelv 1st order:** a first order low shelving filter - it boosts or attenuates frequencies below its characteristic frequency.
- **Low Shelv 2nd order:** a second order low shelving filter - the additional Q controls the steepness of the transition between the affected and unaffected frequencies. With values higher than $\sqrt{2} = 0.707...$, overshoot and undershoot of the desired boost/attenuation will occur in the vicinity of the characteristic frequency.
- **High Shelv 1st order:** a first order high shelving filter - it boosts or attenuates frequencies above its characteristic frequency.
- **High Shelv 2nd order:** a second order high shelving filter - Q works analogous to the second order low shelv type.
- **Allpass 1st order:** a first order allpass filter - allpass filters let all frequencies pass without any boost or attenuation but introduce a frequency dependent phase shift. The human hearing is not particularly sensitive to such phase shifts when these are static. However, when modulated, they become audible as a phase modulation confined to a certain frequency range. In musical terms this means, that it will impose a kind of vibrato that acts only on certain partials/harmonics (assuming that the modulation is LFO-like).
- **Allpass 2nd order:** a second order allpass filter - the additional Q controls the slope of the phase response in the vicinity of the characteristic frequency (which is here the frequency at which the filter's phase response goes through -180 degrees). When we again consider an LFO-modulation of the frequency, higher Q-values (and thus higher slopes) confine the vibrato to a narrower frequency range.
- **Morph Low/Peak/High:** a filter type that allows morphing between a lowpass and and a highpass response, going through a peaking response in between.

Many filter types provide a **TwoStages** button which lets you use two filters of the chosen type in series. The main effect of this is to steepen the filter's slope to twice its value (for example from 12 to 24 dB/oct). Other effects like doubling of the gain at the resonant frequency are compensated for internally such that the user is not plagued with such inconveniences. Among the filter parameters are:

- **Frequency:** characteristic frequency of the filter. Depending on the selected mode, this may be a cutoff-, center- or some other relevant frequency. The attached **'K'** and **'V'** slider control the key-velocity-dependence of this frequency.
- **Resonance/Q:** resonance or Q usually emphasizes frequencies in the vicinity of the cutoff frequency
- **Gain:** controls the boost or attenuation of peak- and shelving types
- **Morph:** controls the overall shape of morphable types

Depending on the chosen type, not all of the listed parameters may be available.

Parameters specific to the 'Moogish Lowpass' type: The Moogish Lowpass type is a digital implementation of one of the most famous filters in the history of subtractive synthesizers: the 4 pole ladder structure consisting of 4 first order lowpass filters in series with negative feedback, as pioneered by Robert ('Bob') Moog. It has some special parameters which require further explanation:

- **Allpass:** Before the actual Moog model, there is a first order allpass filter which can be used to pre-shape the input of the filter - in conjunction with the nonlinear saturation stages inside the filter, this will result in different 'colorations'.
- **Order:** the structure of 4 first order filters series allows for picking up the output signal after any one of the 4 stages or even before the first. Here you choose the signal pickup point - and this in turn determines the order of the filter (at least, when not considering feedback).
- **Drive:** this filter type is nonlinear - it saturates/distorts when the input level is sufficiently high. The 'Drive' slider controls (boosts or attenuates) the input to the filter in order to control the amount of distortion.
- **MakeUp:** When turning up the resonance, we observe a loss in the low frequency range with this filter - this can compensated for by an overall gain factor. When this parameter is set to 100%, the losses are compensated fully.

The Modulation Generators

There are 3 modulation generators - one for the oscillator pitches, one for the filter's characteristic frequency and one for the overall amplitude. These modulation generators are set up in terms of an arbitrary number of breakpoints, which the user can insert (via left mouse-click), remove (via right mouse-click) and drag around (via grabbing some existing node with a left mouse-click) in the envelope-plot. In order to select one of the 3 modulation generators for editing, click on the **Edit** button of the respective envelope (in the left section of the envelope-editor). In the bottom right corner of the envelope plot, there are some widgets for scrolling and zooming into and out of the plot. The mouse-wheel can also be used for horizontal zoom, and a click on the mouse-wheel (middle mouse button) reverts to the maximally zoomed out view. The modulation generators work multiplicatively, that is, their modulation target is multiplied with the instantaneous output value of the envelope. In the case of the pitch-envelope where the modulation amount is adjustable (for each oscillator, via the respective 'Mod' slider in the osc-section), the oscillator's frequency will be multiplied by the value of the envelope raised to the power that is determined by the 'Mod' parameter. When a breakpoint is selected, it's parameters are shown to the right of the plot. Each breakpoint has the following parameters:

- **Time:** the time stamp of the breakpoint either in units of seconds or in units of beats (when 'sync' is active)
- **Level:** the level - this is the factor by which the amplitude or filter frequency will be multiplied with (at the time instant of that breakpoint)
- **Shape:** the shape of the envelope segment which approaches the selected breakpoint. Directly below is a slider which controls the amount of the shape - the **Shapiness** - so to speak. When the button 'ToAll' is active, the shape-settings will affect all breakpoints at once.

Below the area where the breakpoint-parameters are displayed, are two buttons with associated comboboxes which serve for toggling a horizontal and/or vertical grid in the plot on/off and adjusting the resolution of this grid. This is useful when creating rhythmic patterns with the modulation generators. To the left of the plot, you can set up some global parameters for each of the 3 envelopes - global in the sense that they affect the envelope as a whole (as opposed to affecting only a particular breakpoint). These are:

- **Time Scale:** scales (multiplies) the overall time duration of the envelope by some value. The attached **'K'** and **'V'** sliders control the key- velocity-dependence of the time scaling. The combobox above the slider is there to quick select some common values.
- **Depth:** Controls the overall effect of the envelope generator. Because the envelopes are acting multiplicatively, this depth parameter acts as an exponent to the multiplication factor. The attached **'K'** and **'V'** sliders are once again for key- velocity-dependence.
- **Loop:** You can activate a loop here - when this is turned on, two loop locators will appear in the plot which can be dragged to any breakpoint. With this loop, you can realize a generalized sustain stage, which reduces to the standard sustain, when there is no breakpoint in between the start- and end-locator and both breakpoints at start and end are on the same level.
- **Sync:** The modulation patterns can be synchronized to the host tempo by switching this button on. In this case, the time units displayed in the plot will be beats (instead of seconds, in non-sync mode)