Odin 2
Synthesizer Plugin

Manual
for version 2.2.0

TheWaveWarden

www.thewavewarden.com

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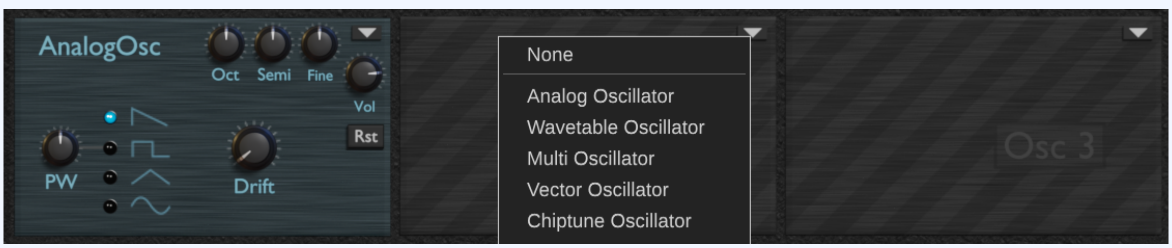
Chapter 2

Oscillators

Three oscillators form the basis of sound generation in Odin 2. You can choose from a wide variety of different modules, which are capable of a wide palette of sounds, even without any further processing. Initially, Odin 2 starts out with an Analog Osc in slot 1 and none in slot 2 & 3.

Osc Type Modulation: ✗ Automation: ✗

You can change the module being used with the small dropdown button on the top-right of the osc-module:



2.1 Common Parameters

There are some controls which are common to all oscillator modules:

Osc Octave

Modulation: ✓ Automation: ✓

Detunes the oscillator in whole octaves.

Osc Semitones

Modulation: ✓ Automation: ✓

Detunes the oscillator in semitones.

Osc Finetune

Modulation: ✓ Automation: ✓

Detunes the oscillator in cents.

Osc Volume

Modulation: ✓ Automation: ✓

Regulates the volume of this oscillator in deciBels. Can be used to shut the oscillator entirely. Modulating this parameter from the modulation matrix with -100 will always shut the sound. Modulating this parameter with $+100$ will raise the sound to 0dB if the current value is smaller than -12dB. If it is bigger than -12dB, it will modulate to +12dB from the current value.

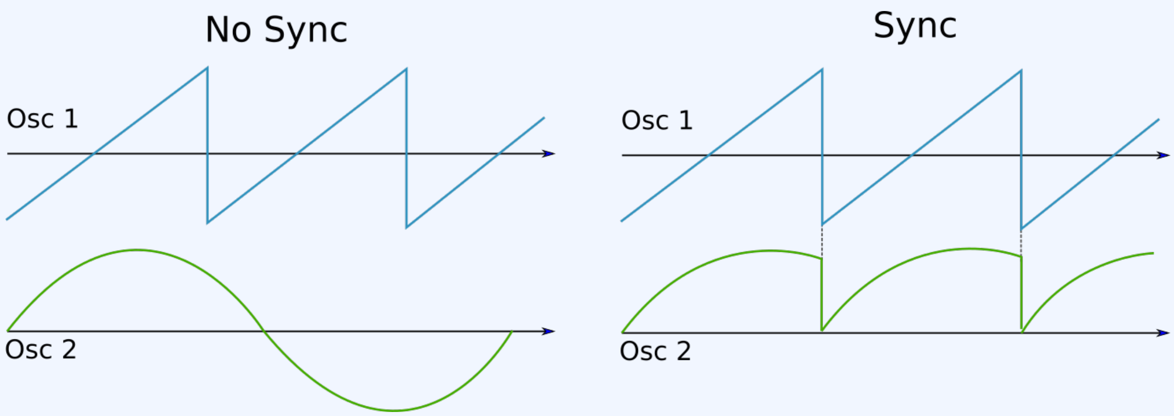
Osc Reset (Rst)

Modulation: ✗ Automation: ✓

Resets the waveform to its initial position each time a key is pressed. This is usefull to get more consistent sounding notes, for example for tight basslines. If this is turned off, the wave will continue where it ended on the last note.

Osc Sync Modulation: ✗ Automation: ✓

This parameter is only available for Osc 2 & 3. Activating sync will sync this osc to Osc 1. That means each time Osc 1 completes a cycle, this osc is reset to its initial position. The pitch of the oscillator is thereby controlled by Osc 1. This can introduce lots of harmonics, even for soft waveforms like the sinewave.



Internally, any osc with activated sync will use 3x oversampling to prevent aliasing on the hard resets. Additionally, any osc with enabled sync uses a DC-blocking filter to remove constant offsets in the wave.

2.2 Analog Osc

The analog osc aims to emulate the sound of classic analog synthesis. The first obvious choice you have is the waveform:

Analog Waveform

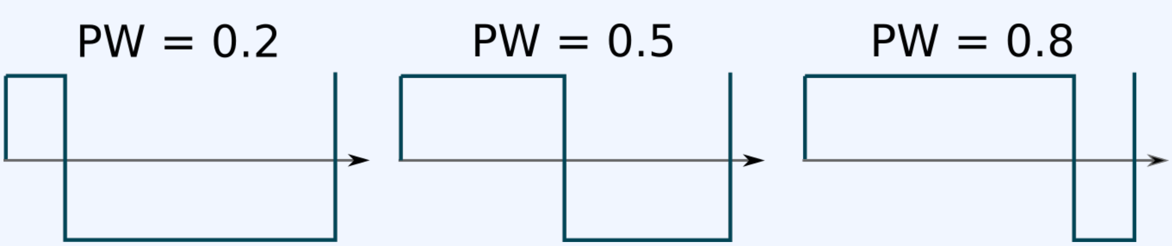
Modulation: ✗ Automation: ✓

Sawtooth:
The classic sawtooth wave. It is very rich in harmonics and forms an excellent starting point for a wide variety of sounds. This particular Sawtooth emulates the way analog syntheizers generate saw-waves. The result is a (phase-corrected) ”fat-saw”. This variant doesn’t rise linearly as the icon would suggest, but in a slight curve, providing a different tonal character.
Pulse Wave:
The pulse wave has a thinner sound than the savetooth, sometimes giving the impression of a ”hollow” sound body being emulated. The pulse still has a lot of harmonics, making it a common alternative to the sawtooth. The width of the pulse can be adjusted, see the next parameter **Pulse Width**.
Triangle:
The triangle wave is much gentler than the saw and pulse waves. It still has a lot of harmonics present though. This wave is well suited for flute like sounds.
Sine:
The purest of all waveforms. The sinewave (by its very definition) has no harmonics at all. The resulting sound is very easy on the ears.

Pulse Width (PW)

Modulation: ✓ Automation: ✓

This parameter has no effect if the waveform selected is not a pulse. It shifst the duty cycle of the pulse wave, making it stay longer in the lower section for higher values.



The pulse width control can not be used to shut the sound completely ($PW = 0$ or $PW = 1$), but that can be achieved when modulating via the modmatrix.

Drift

Modulation: ✗ Automation: ✓

Analog oscillators tend to not be stable in their frequency. Drift emulates this behaviour by randomly shifting the pitch up and down just a little bit over time. For a single osc, the effect is not very apparent, but becomes clear once two oscillators are used.

2.3 Wavetable Osc

The Wavetable Osc allows you to create evolving sounds, which feature more than one waveform. Each of the 35 selectable wavetables consists of four waves itself. You can sweep through these easily by hand or with pre-setup modulation.

Wavetable

Modulation: ✗ Automation: ✗

Selects which wavetable to be used. A wide variety of sounds is available, starting with analog waveforms, human voice like sounds, additive waves, waveforms taken from instruments and many, many more.

Wavetable Position

Modulation: ✓ Automation: ✓

Fades through the four waves in the selected wavetable. A value of 0 will give the first wave, 0.333 the second, 0.666 the third and 1 the last wave.

Modulation

Modulation: ✗ Automation: ✗

Selects a modulation source, which can be used to modulate the Wavetable Position. Modulation Envelope and LFO1 are selectable. Please note that arbitrary modulation sources can be sected when working with the modulation matrix . This slot is merely for a fast and convenient way to set up modulation.

Amount

Modulation: ✗ Automation: ✓

Sets the amount of modulation being used to modify the Wavetable Position. Positive and negative values are possible.

2.4 Multi Osc

The Multi Osc is four oscillators disguised as one. These can be arbitraritly detuned and can even use different waveforms, which results in a thicc, rich sound.

Detune

Modulation: ✓ Automation: ✓

Detunes the four sub-oscillators agains each other. The detune values are calculated to avoid beating (random phase-cancellation).

Wavetable

Modulation: ✗ Automation: ✗

The same as in Wavetable Osc: Selects which wavetable to be used. A wide variety of sounds is available, starting with analog waveforms, human voice like sounds, additive waves, waveforms taken from instruments and many, many more.

Wavetable Position

Modulation: ✓ Automation: ✓

The same as in Wavetable Osc: Fades through the four waves in the selected wavetable. A value of 0 will give the first wave, 0.333 the second, 0.666 the third and 1 the last wave.

Wavetable Spread

Modulation: ✓ Automation: ✓

Spreads the four sub-oscillators over the wavetable: The first sub-osc wavetable position will be shifted to the left, the last will be shifted to the right. These shifts happen around the value chosen by Wavetable Position.

2.5 Vector Osc

The Vector Osc gives even more options for evolving sounds than the Wavetable Osc. Four freely definable waves can be interpolated in a very intuitive graphic way via an XY-pad.

A, B, C & D

Modulation: ✗ Automation: ✗

Select the waves to be used. Each of the four letters mark one cornder of the XY-pad, as the graphic suggests. Virtually any waveform from the entire synthesizer can be chosen for any of the corners. This also includes any of the (see Draw Oscillators).
When selecting Draw Osc 1, 2 & 3, the waves you have drawn in osc slots 1, 2 and & 3 respectively are used.

X & Y

Modulation: ✓ Automation: ✓

Moves the handle over the XY pad. Each of the corners represent the waveform chosen from the A, B, C and D dropdowns. Moving closer to a corner will make the sound closely relate the waveform of that corner. When being in the corner, the resulting waveform is purely the one selected for that corner. Uses bilinear interpolation to fade through the four tables.

2.6 Chiptune Osc

The Chiptune Osc is an easy way to get nostalgic for your childhood. It aims to emulate the sound of yesteryear while emulating the processing capabilities of a 4-Bit soundchip, like it was used in the Nintendo Entertainment System NES or original Nintendo Gameboy. It also features a simple arpeggiator, with two or three steps being selectable. Whilest being able to produce harmonic sounds, it also features a dedicated chiptune noise module.

Waveform

Modulation: ✗ Automation: ✗

Lets you select from a variety of waveforms, like you would typically find on the soundchips of yesteryear. Available are a bunch of pulse waves, a triangle, saw and sine variant. All of these waves are limited to a 4Bit resolution (16 steps) on the Y-axis. On top of these, you can select any of the ChipDraw waves.
To clarify: ChipDraw 1, 2 & 3 refer to the waves you have drawn in osc slots 1, 2 and & 3 respectively. You need to apply changes in the ChipdDraw Oscs for the change to take effekt (see ChipDraw Osc).

Arpeggiator

Modulation: ✗ Automation: ✓

Turns on an internal arpeggiator module, which makes the oscillator jump over predefined semitone values. See the next parameters for specifics.

Arp 1, 2 & 3

Modulation: ✗ Automation: ✓

Select the semitones to be played by the arpeggiator module. For the third step to be used, the next parameter Step 3 needs to be active.

Step 3

Modulation: ✗ Automation: ✓

Enables the third step in the arpeggiator. When Step 3 is not active, the arpeggiator will only loop between the first two steps.

Speed

Modulation: ✓ Automation: ✓

Sets the speed of the arpeggiator in Hz.

Noise

Modulation: ✗ Automation: ✓

Enabling Noise will change stop the output of the selected wavform. Instead, the oscillator will generate a random value to be output each time a cycle is complete. This creates a classic noise effect like it was used on early game consoles. Internally, 3x oversampling is used to remove aliasing on the jumps between values. Note that this noise is dependent on the note being played and has a perceived pitch. It is also possible to use the noise module while the Chiptune arpeggiator is enabled.

2.7 FM Osc

The FM Osc is a convenient way to set up Frequency Modulation, or FM. The basic idea behind FM is that you have two oscillators: The carrier and the modulator. The modulator is solely used as a modulation source for the frequency of the carrier. The carrier is the oscillator you will actually hear. While you can set up FM via the modulation matrix as well, the FM osc is the easy way to do it. The theory behind FM is very well documented in other literature, for example on Wikipedia.
FM will usually produce a metallic, bell-like sound.

Waveform

Modulation: ✗ Automation: ✗

Both carrier and modulator can be assigned a waveform. This will be the actual waveform that the sub-osc is using. Virtually any waveform from the entire synthesizer can be chosen. This also includes any of the (see Draw Oscillators).
When selecting Draw Osc 1, 2 & 3, the waves you have drawn in osc slots 1, 2 and & 3 respectively are used.

Ratio

Modulation: ✓ Automation: ✗

The numbers above/below the waveform describe the base-frequency relation modulator and carrier have to one another. The frequency of the modulator will alwlays be

$$f_{mod} = f_{car} \frac{Ratio_{mod}}{Ratio_{car}} \tag{2.1}$$

So for example using the values $Ratio_{mod} = 2$ and $Ratio_{mod} = 1$ will put the modulator one octave (double the frequency) above the carrier. The base freq of the carrier is the pitch played for the note.
Using fractions which are not reducable to "simple" fractions, like $\frac{11}{7}$ will yield wilder results than "simple" ones like $\frac{1}{2}$.
Note that modulating these values from the modulation matrix , allows for fractions which are non-rational (continuous modulation).

FM

Modulation: ✓ Automation: ✓

This is where the magic happens: The FM amount controls how deep the modulator modulates the frequency of the carrier. A value of zero will show no modulation at all, so the carrier is playing like a normal osc. When increasing the amount, the sound gets more and more metallic. The range of this parameter can be extended over its natural range via the modmatrix.

2.8 PM Osc

PM or Phase Modulation, is closely related to FM or Frequency Modulation. Like the FM Osc, it features a modulator and a carrier oscillator, but this time the modulator modulates the phase of the carrier. When only using sine-waves, the frequency contents generated by FM and PM are indistinguishable.

Waveform

Modulation: ✗ Automation: ✗

Both carrier and modulator can be assigned a waveform. This will be the actual waveform that the sub-osc is using. Virtually any waveform from the entire synthesizer can be chosen. This also includes any of the (see Draw Oscillators).
When selecting Draw Osc 1, 2 & 3, the waves you have drawn in osc slots 1, 2 and & 3 respectively are used.

Ratio

Modulation: ✓ Automation: ✗

The numbers above/below the waveform describe the base-frequency relation modulator and carrier have to one another. The frequency of the modulator will alwlays be

$$f_{mod} = f_{car} \frac{Ratio_{mod}}{Ratio_{car}} \tag{2.2}$$

So for example using the values $Ratio_{mod} = 2$ and $Ratio_{mod} = 1$ will put the modulator one octave (double the frequency) above the carrier. The base freq of the carrier is the pitch played for the note.
Using fractions which are not reducable to ”simple” fractions, like $\frac{11}{7}$ will yield wilder results than ”simple” ones like $\frac{1}{2}$.
Note that modulating these values from the modulation matrix , allows for fractions which are non-rational (continuous modulation).

PM

Modulation: ✓ Automation: ✓

The PM amount controls how deep the modulator modulates the phase of the carrier. A value of zero will show no modulation at all, so the carrier is playing like a normal osc. The range of this parameter can be extended over its natural range via the modmatrix.

2.9 Noise Osc

The Noise Osc provides a source of noise in Odin 2. The initial noise generation produces white noise. The noise can be further preprocessed by the included lowpass and highpass filters.

 **Highpass** Modulation: ✓ Automation: ✓

Sets the cutoff frequency for the included highpass filter. The filter is a first order (6dB / Oct) virtual analog highpass filter.

 **Lowpass** Modulation: ✓ Automation: ✓

Sets the cutoff frequency for the included lowpass filter. The filter is a first order (6dB / Oct) virtual analog lowpass filter.

2.10 WaveDraw Osc

The wavedraw osc lets you experiment with waveforms by letting you draw them yourself.

> The changes you make to the waveform will have no effect until you press the apply button on the bottom-right of the oscillator. If this button is red, then there are still unapplied changes to the waveform.

The drawn waveform is sampled using 200 discrete steps. When you press the apply button, the waveform is processed into the spectral domain to create a usable wavetable.

2.11 ChipDraw Osc

The ChipDraw Osc lets you draw a custom chipdraw waveform. It resembles the capabilities of the "custom waveform" on an Nintendo Entertainment System, NES soundsystem. The waveform consists of 32 steps in horizontal direction, which can be offset to 16 values (4Bit) in vertical direction.

> The changes you make to the waveform will have no effect until you press the apply button on the bottom-right of the oscillator. If this button is red, then there are still unapplied changes to the waveform.

When you press the apply button, the waveform is processed into the spectral domain to create a usable wavetable.

2.12 SpecDraw Osc

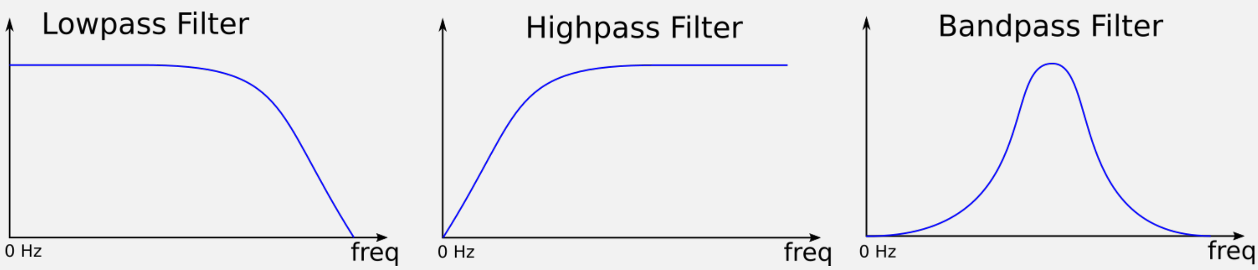
The SpecDraw Osc opens the sonic capabilities with some **additive synthesis**. Unlike subtractive synthesis, where you filter frequencies from harmonically rich waves, additive synthesis lets you build a sound by stacking up individual harmonics. The n-th harmonic is a sinewave which has n-times the frequency of the base note. The Specdraw Osc lets you draw the amplitude of these sinewaves. The left-most bar represents the fundamental. In the initial state, only this bar ist present, resulting in an overall sinewave osc. As you bring more overtones, the sound gets richer. Addiitve synthesis is capable of creating timbres which are not possible with subtractive synthesis.

> The changes you make to the waveform will have no effect until you press the apply button on the bottom-right of the oscillator. If this button is red, then there are still unapplied changes to the waveform.

Chapter 3

Filters

Signal filters are one of the basic tools to shape your sound in subtractive synthesis. While the oscillators in Odin 2 are already capable of a wide array of sounds, unprocessed oscillators usually sound very sharp and not pleasing to the ears. So what is a filter? A filter selectively removes frequencies from the spectrum, usually with some dials for the user to control the rolloff. Filters can be charcaterized by their frequency response, which tells us which frequencies are being attenuated or boosted:



Odin 2 has three slots for filters which can be filled with a extensive selection of modules to shape your sound. A wide array of high quality virtual analog filter emulations is availabe, which emulate various analog filter circuits from synthesizer history.
TODO poly vs stereo.

Filter Type

Modulation: ✗ Automation: ✗

To change the filter module, use the dropdown to the top-right of the filter module:

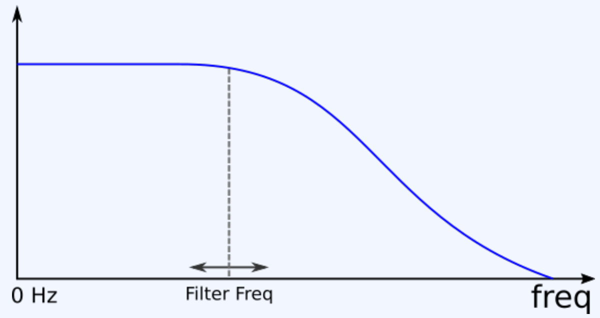
3.1 Common Controls

Some of the controls are shared among most filter modules:

Filter Frequency

Modulation: ✓ Automation: ✓

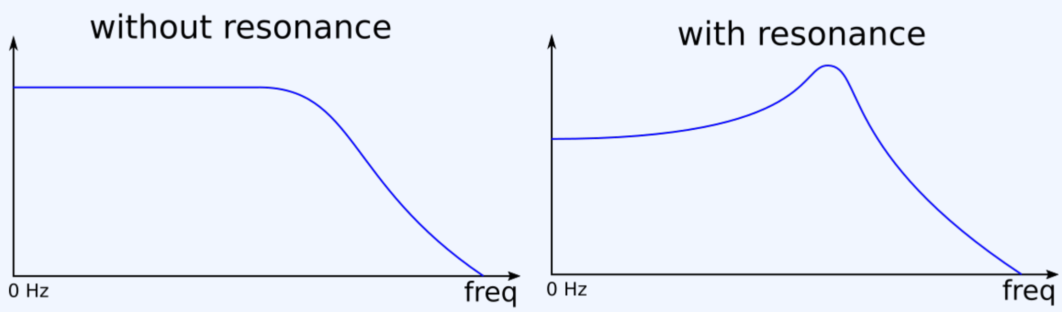
Controls the cutoff point of the filter. The frequency value marks the point where the frequency is attenuated by 3dB.



Filter Resonance

Modulation: ✓ Automation: ✓

Increasing resonance creates a peak in the spectrum at the position of the filter cutoff.



Note also that the frequencies which were previously unaffected by the filter are being attenuated by the resonance parameter.
None of the filters in Odin 2 are capable of self-oscillation for the sake of your ears and speakers.

Filter Velocity (Vel)

Modulation: ✓ Automation: ✓

Adds velocity from MIDI-Notes to the filter frequency. This allows for expressive play, as harder key-hits move the filter freq up. Note that the value is added on top of the current value, so to achieve a similar resulting timbre, you might need to lower the filter frequency accordingly.

Filter Envelope (Env)

Modulation: ✓ Automation: ✓

Controls the amount of Filter Envelope which is applied to the filter frequency. To see how the Filter Envelope itself is operated, see section 6.1.

Filter Keyboard (Kbd)

Modulation: ✓ Automation: ✓

Controls how much the MIDI-note is put ontop of the filter frequency. Increasing this value makes the filter open up more for higher notes. This allows for more consistent notes across the keyboard, since higher notes are filtered differently more by static filters. Note that the value is added on top of the current value, so to achieve a similar resulting timbre, you might need to lower the filter frequency accordingly.

Filter Gain

Modulation: ✓ Automation: ✓

Regulates the volume of this filter in deciBels. Can be used to shut the filter entirely. Modulating this parameter from the modulation matrix with -100 will always shut the sound. Modulating this parameter with $+100$ will raise the sound to 0dB if the current value is smaller than -12dB. If it is bigger than -12dB, it will modulate to +12dB from the current value.

Filter Saturation Modulation: ✓ Automation: ✓

Introduces a slight distortion by shaping the signal with a hyperbolic tangent function. Depending on the filter module being used, the saturation stage is in a different position of the signal loop, yielding different results.

3.2 Lowpass, Bandpass, Highpass

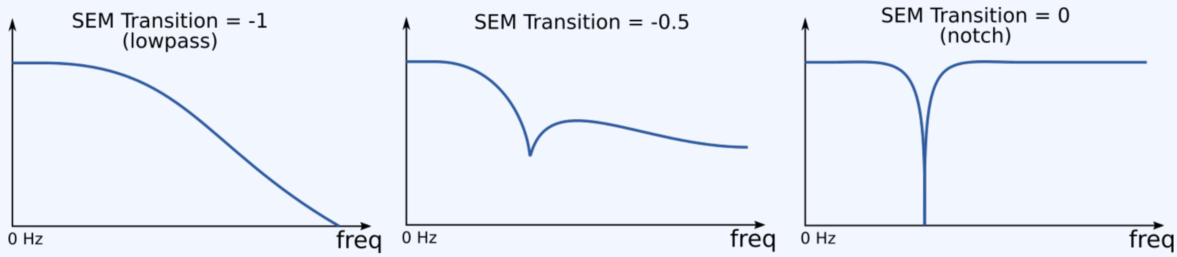
The staples of sound-design in Odin 2. These filters are virtual analog emulations of a certain, famous **ladder filter** which has had a big impact in the history of synthesizers. Each of these filters is available in a 12dB/Oct and a 24dB/Oct variant. These values determine the slope of the filter rolloff. The 24dB/Oct variants filter more frequencies than the 12dB/Oct counterparts.

3.3 SEM-12

TODO SEM BILD.
Another emulation of a classic synthesizer filter. This filter has the speciality of being able to shift between a lowpass and a highpass filter, with a notch-filter in between. The filter slope of this filter is 12dB/Oct.

SEM Transition Modulation: ✓ Automation: ✓

Fades from a lowpass filter over a notch filter to a highpass filter. This allows for special filter variants which still leave some of the frequencies that were filtered before.



3.4 Diode Ladder

The Diode Ladder is a virtual analog emulation of another classic analog syntheiszer filter. Its analog pendant was originally developed to work around a patent on the well established ladder filter. While still being 24dB/Oct, the characteristic of this filter is said to be more aggressive and wild compared to the classic ladder, especially when invoking resonance.

3.5 KO-35

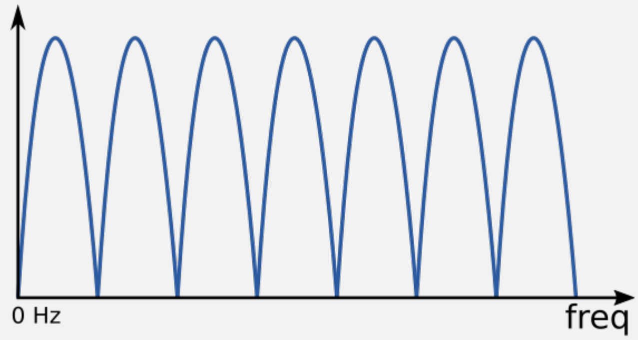
Yet another virtual analog emulation of one of the legendary analog filters of the past. This filter comes in a lowpass and highpass variant. Crancking up the resonance on these filters reveals a dirty, aggressive sound. Note that while the filters are named KO-35, their slope is 12dB/Oct.

3.6 Comb Filter

A comb filter is essentially a tuned delay module. The input signal fed into a delay-line, which echos the sound back after a set amount of time. The delay time is the inverse of the filter frequency:

$$t_{delay} = \frac{1}{f_{freq}} \tag{3.1}$$

The frequency response of this filter usually resembles the shape of a hair-comb, hence the name.



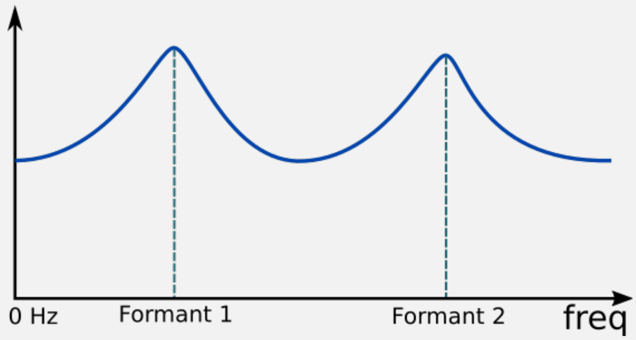
The resonance parameter for the Comb Filter controls how much of the delayed signal is fed back into the delay line again, creating a feedback loop.
Comb filters can sound from subtle to metallic. When automating or modulating the frequency with high resonance values, a psychedelic smearing effect can be produced.

Comb Polarity (+/-) Modulation: ✗ Automation: ✓

Controls whether the insertion of the signal into the delay line is positive (+) or inverted (-). This changes the frequency behaviour. Inverted operation tends to eliminate deep frequencies.

3.7 Formant Filter

The Formant Filter tries to emulate vowels as they are produced in human speech. A tone is perceived as a vowel if two characteristic frequencies are dominant. These are called formants. The Formant Filter emulates this by using a combination of two resonator filters, which increase the frequencies around the two formants.



The Formant Filter allows you to choose two vowels and freely move the formant peaks between the according formant peaks.

Vowel 1 & 2 Modulation: ✗ Automation: ✗

Select the vowels to the left and right of the transition. Selectable vowels are: A, E, I, O, U, Ä, Ö, Ü

Formant Transition Modulation: ✓ Automation: ✓

Transition between the two selected vowels. The transition is not a simple interpolation of the two vowel sounds, but actually moves the resonant formant peaks in the spectrum from one vowel to the next.
The parameters Filter Velocity and Filter Envelope are applied to this parameter for the formant filter.

3.8 Ring Modulator

The ring modulator is an oscillator disguised as a filter. The function of this module is to multiply the input signal with an internal sine-oscillator. This is formerly known as amplitude modulation.

RingMod Freq

Modulation: ✓ Automation: ✓

Controls the frequency of the internal oscillator.

RingMod Amount

Modulation: ✓ Automation: ✓

Controls the amount of ringmod to be applied. This interpolates between the input signal and the processed signal, effectively working like a Dry/Wet control.

Chapter 4

Amplifier & Distortion

The Amplifier and Distortion sections form the only parts in the signal flow in Odin 2 which are both polyphonic and stereo.

4.1 Amplifier

The amplifier section plays an important gain-staging role in odin. The sound can be boosted or attenuated, as well as panned.

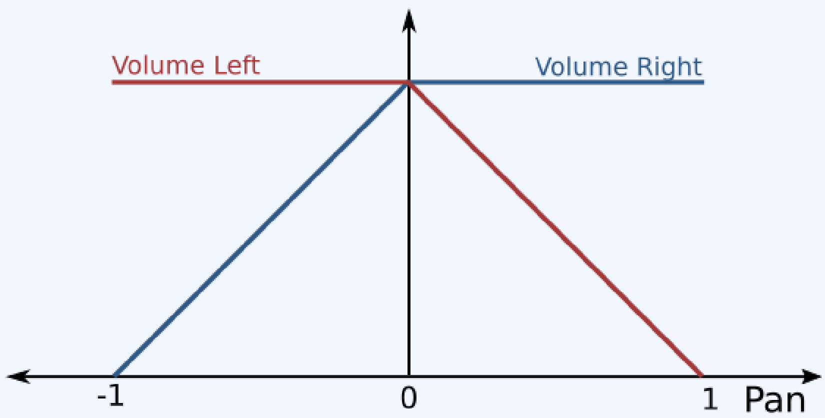
 **Amp Gain** Modulation: ✓ Automation: ✓

Changes the volume in in deciBels. Can be used to shut the sound entirely. Modulating this parameter from the modulation matrix with -100 will always shut the sound. Modulating this parameter with $+100$ will raise the sound to 0dB if the current value is smaller than -12dB. If it is bigger than -12dB, it will modulate to +12dB from the current value.

Amp Pan

Modulation: ✓ Automation: ✓

Pan or Panorama can be used to move the sound over the stereo field. The default value of zero will leave the sound centered. Moving the pan towards -1 will attenuate the right stereo channel, moving towards 1 does the same for the left channel.



Amp Velocity

Modulation: ✗ Automation: ✓

Makes the Amplifier gain sensible to the MIDI-Velocity. This allows for expressive play, where harder notes produce louder sounds. Increasing this value lowers the default gain of the amp, such that a MIDI-note with maximum velocity (127) will bring the level back to its previous level.

Please note that the Amp Envelope is not applied between the Amplifier and Distortion section, like the routing would suggest. The Amp Envelope is applied **after the Distortion section**.

4.2 Distortion

The Distortion module is capable of distorting the sound by various characteristic distortion functions. All the distortion types used in this section are threshold based: Once the wave surpasses a predefined value (in positive or negative direction), the processing will apply. Internally, 3x oversampling is used to prevent aliasing from the sharp cuts made to the waveform.

Distortion On

Modulation: ✗ Automation: ✓

Turns the Distortion module on or off.

Boost

Modulation: ✓ Automation: ✓

Boosts the gain of the incoming wave, making it surpass the internal threshold easier.

Distortion DryWet

Modulation: ✓ Automation: ✓

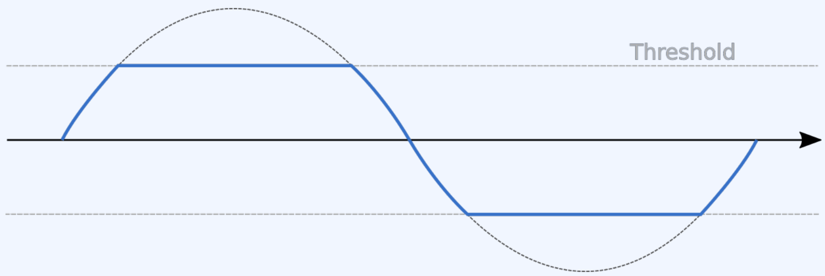
Interpolates the processed and unprocessed signals, thereby controlling the amount of distortion applied to the sound.

Distortion Algorithm

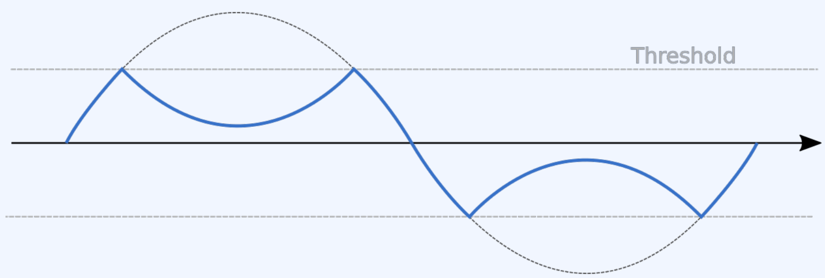
Modulation: ✗ Automation: ✗

Selects which distortion algorithm is used. The options are:

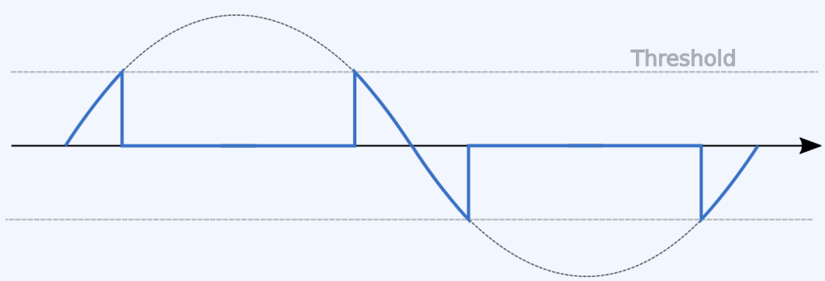
Clamp: Clamps the signal once it surpasses the threshold.



Fold: Folds the wave over once it surpasses the threshold. If the folded wave hits the threshold on the other side, it will be folded again (and so on). Produces more harmonic content than the Clamp algorithm.



Zero: Pulls the wave to zero once it surpasses the threshold. The strongest of the available distortion algorithms.

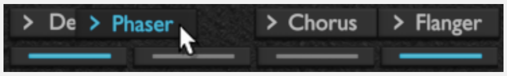


Chapter 5

FX

Odin 2 comes with four internal FX modules: **Delay, Chorus, Phaser** and **Flanger**.

The buttons on top of the FX modules serve multiple purposes:



Clicking the corresponding name of the module reveals the corresponding module. The buttons below the module name are used to turn enable or disable the module. You can also **change the order of the modules**, by drag'n'dropping their name handles to the left or right.

FX On

Modulation: ✗ Automation: ✓

Use the buttons below the module name to turn the FX module on or off. All modules can be used at the same time.

FX Order

Modulation: ✗ Automation: ✗

Drag'n'drop the FX module handles to change the order of the FX. The algorithms are calculated in series from left to right.

5.1 Delay

A delay is a module capable of producing an 'echo' effect: The signal is fed into a delay-line, which outputs the signal again after a set amount of time again. The output of the delay line can also be fed back in, allowing a chain of attenuating echos. By controlling the delay time and feeback parameters, a wide variety of effects can be achieved. The Delay module in Odin 2 goes a step further and offers several additional features.

Delay Time

Modulation: ✓ Automation: ✓

Controls the time the delay line takes to output the sound again. Depending on the parameter "Delay Sync", this is either a dial for continuous values in Hz, or a custom selector to sync the time to the beat. This selector allows for arbitrary fractions of the current host BPM, for example 5/16th notes:

Delay Feedback

Modulation: ✓ Automation: ✓

Controls how much of the output of the delay line is fed back in again. If feedback is zero, only one echo will be audible. If feedback is one, an infinite series of exact copies of sound will be output. Everything inbetween makes for slowly attenuating echos.

Delay Sync

Modulation: ✗ Automation: ✓

Controls whether the Delay Time is set by a knob in Hz or via the sync-time selector, syncing it to the host BPM.

Delay PingPong

Modulation: ✗ Automation: ✓

Enabling PingPong will make the left and right stereo delay lines crossfeed: The output of the left line is fed into the right line and vice versa.
The initial input into the delay lines is mixed down to a mono signal and then fed into the left delay line only. The dry signal remains in the center of the stereo field.

Delay Highpass (HP)

Modulation: ✓ Automation: ✓

The processed signal in the Delay module is filtered through a 6dB/Oct high-pass filter. The Delay Highpass parameter controls the cutoff of this internal filter. This is great for removing the muddiness that deep frequencies can produce in a delay module.
Note that the highpass filter is not applied inside, but after the feedback loop, i.e. consecutive echos do not get filtered further more as they are processed again.

Ducking

Modulation: ✗ Automation: ✓

Ducking attenuates the output of the delayed signal if an input signal is present. This is great for decluttering sections where the delayed signal interferes with the unprocessed signal.

Unlike the other FX modules, the Delay features a separate Dry and Wet control to allow for easier adjustments of processed and unprocessed signals individually.

Delay Dry

Modulation: ✓ Automation: ✓

Controls how much unprocessed signal is output by the Delay.

Delay Wet

Modulation: ✓ Automation: ✓

Controls how much processed signal is output by the Delay.

5.2 Chorus

The Chorus module is a delay based effect capable of thickening sounds. The generated sound resembles that of a slightly detuned ensemble, hence the name chorus. Internally, the Chorus module uses a delay line, which is read from at two diferent positions. The delay times are modulated by an internal Low Frequency Oscillator (LFO). This slightly detunes the result resulting in the Chorus sound. The LFOs for the left and right channel are phase-offset by 90 to spread the sound in the stereo field.

Chorus Rate Modulation: ✓ Automation: ✓

Controls the speed of the internal LFO. Depending on the parameter ”Chorus Sync”, this is either a dial for continuous values in Hz, or a custom selector to sync the time to the beat. This selector allows for arbitrary fractions of the current host BPM, for example 5/16th notes:

Chorus Sync Modulation: ✗ Automation: ✓

Controls whether the Chorus Rate is set by a knob in Hz or via the sync-time selector, syncing it to the host BPM.

Chorus Modulation Modulation: ✓ Automation: ✓

Controls how much the internal LFO modulates the two delay times. Exaggerates the detune effect.

Chorus Feedback

Modulation: ✓ Automation: ✓

Controls how much of the output of the delay line is fed back in again. Pronounces the effect of the Chorus a bit more.

Chorus DryWet

Modulation: ✓ Automation: ✓

Interpolates the processed and unprocessed signals, thereby controlling the strength of the effect.

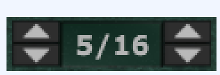
5.3 Phaser

The phaser module introduces movement to the sound by applying a subtle ”windy” character. The internal structure consists of a series of allpass filters: These filters do not alter the amplitude like the filters from Chapter 3, but only shifts the phase of some frequencies. By adding the phase-shifted signal back onto the original signal, some of the frequencies get boosted, attenuated or eliminated entirely via phase-cancellation. The characteristic of the allpass-filters is continuously modulated by an internal Low Frequency Oscillator (LFO), which makes for the movemnet in the sound. The LFOs for the left and right channel are phase-offset by 90 to spread the sound in the stereo field.

Phaser Rate

Modulation: ✓ Automation: ✓

Controls the speed of the internal LFO. Depending on the parameter ”Phaser Sync”, this is either a dial for continuous values in Hz, or a custom selector to sync the time to the beat. This selector allows for arbitrary fractions of the current host BPM, for example 5/16th notes:



Phaser Sync Modulation: ✗ Automation: ✓

Controls whether the Phaser Rate is set by a knob in Hz or via the sync-time selector, syncing it to the host BPM.

Phaser Modulation Modulation: ✓ Automation: ✓

Controls how much the internal LFO modulates the internal allpass filters.

Phaser Feedback Modulation: ✓ Automation: ✓

An extra feedback stage, which feeds the output signal into the input again.

Phaser Freq Modulation: ✓ Automation: ✓

Shifts the base frequency of the internal allpass filters, thereby altering the characteristic of the effect.

Phaser DryWet Modulation: ✓ Automation: ✓

Controls how much of the phase-shifted signal is added to the input signal, thereby controling the strength of the effect.

5.4 Flanger

A Flanger is a modulated comb filter. The signal is fed into a delay line and mixed with the input signal after a small echo. The timing of this effect is modulated by an internal Low Frequency Oscillator (LFO). Additionally, the output of the delay line can be fed in again via the Feedback parameter, to allow for a continuous stream of echoes. The delay times for Comb Filters and Flangers is very short, usually below 50ms.

Flanger Rate

Modulation: ✓ Automation: ✓

Controls the speed of the internal LFO. Depending on the parameter "Flanger Sync", this is either a dial for continuous values in Hz, or a custom selector to sync the time to the beat. This selector allows for arbitrary fractions of the current host BPM, for example 5/16th notes:

Flanger Sync

Modulation: ✗ Automation: ✓

Controls whether the Flanger Rate is set by a knob in Hz or via the sync-time selector, syncing it to the host BPM.

Flanger Modulation

Modulation: ✓ Automation: ✓

Controls how much the internal LFO modulates the delay time.

Flanger Feedback

Modulation: ✓ Automation: ✓

Controls how much of the output of the delay line is fed back in again. Creates a metallic smearing effect for big values. This parameter can be positive or negative allowing for positive and negative comb operation.

Flanger DryWet

Modulation: ✓ Automation: ✓

Interpolates the processed and unprocessed signals, thereby controlling the strength of the effect.

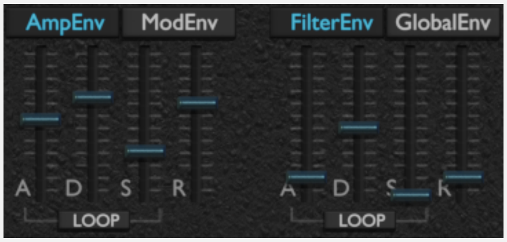
Chapter 6

Modulators

The previous chapters wrote about the audio generators and manipulators. This section will take a look at some usefull modules which are used to modulate parameters within Odin 2.
Some of the modulators have hardwired functionalities in the synthesizer. However, the true potential of these (and the entire synth, really) lies in appling modulation from the modulation matrix .

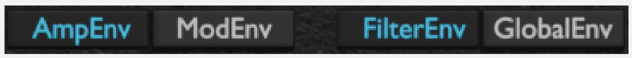
6.1 ADSR Envelopes

An ADSR Envelope provides a handy way of setting up a wide variety of curves like they can be observed in the timbre changes of physical instruments. ADSR Envelopes follow two basic signals: MIDI note-on and MIDI note-off. Depending on these, a curve is produced based on the four parameters **Attack, Decay, Sustain** and **Release**.
Odin 2 provides four of these Envelopes:

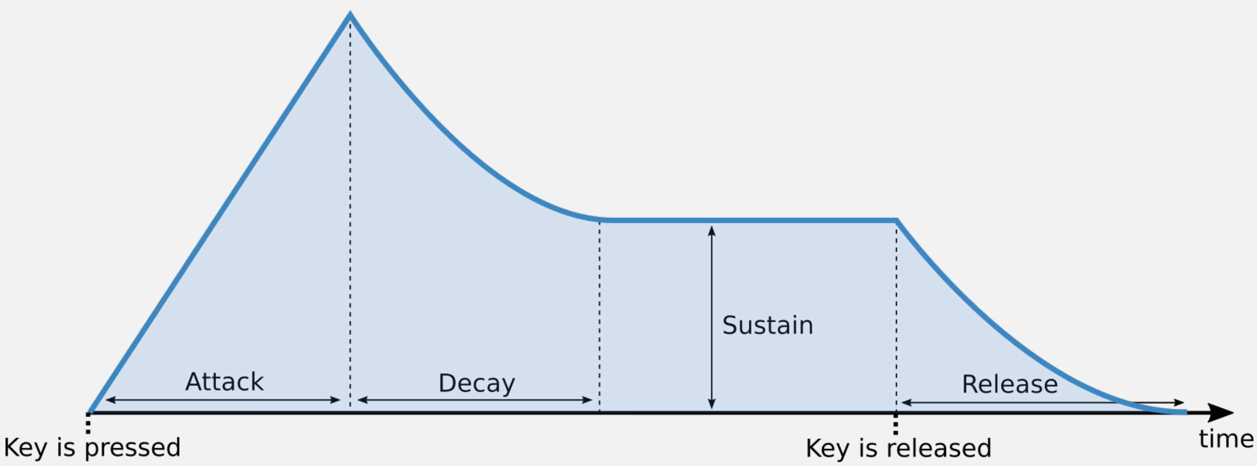


The **Amp Envelope** is hardwired to control the volume curve of the voice. This effect is not applied in the actual Amplifier module, but after the Distortion section (see Section 1.4). Note that the Amp Envelope can still be used as a freely assignable modulation source in the modulation matrix . This Envelope is calculated for each voice independently.
The **Filter Envelope** is hardwired to modulate the frequencies of the various Filter modules in Odin 2. For this to take effect, the parameter "Filter Env" (see Section 3) has to be enabled. The modulation by this envelope can be in either positive or negative direction. This Envelope is calculated for each voice independently.

The **Mod Envelope** is a freely assignable modulation source. This Envelope is calculated for each voice independently.
The **Global Envelope** is different from the other three Envelopes in that it exists only once for all voices. This can come in handy when you want the modulation not to diverge between voices.
To switch between the different Envelope modules, use their handles on top of the section:



Two envelopes are paired to occupy the same space. To acces the currently not visible Envelope, simply click on its name.



Attack

Modulation: ✓ Automation: ✓

The Attack determines the time the Envelope takes from the start to the first peak.
When playing the synth in Legato mode (see Section 9), the attack section will start from the last value the Enveope from the previous voice produced. No matter the start height, the slope of the Attack will always be the same as if it started from zero.
Using really short Attack times can introduce clicks into the sound, so it is advisable to have at least some attack for most situations.
Note that unlike the Decay and Release sections, the Attack follows a linear curvature.

Decay

Modulation: ✓ Automation: ✓

The Decay controls the time the Envelope will take to fall to the sustain value after the Attack reached its highest point. This curve will always start at the internal value one, and will always fall to the value specified by the Sustain parameter.
The Decay section follows an exponential falling curvature.

Sustain

Modulation: ✓ Automation: ✓

The Sustain determines the level that the Evelope will fall to after the Decay section. Note that the Sustain section will be active after Decay finished for as long as the MIDI-key is not released.

Release

Modulation: ✓ Automation: ✓

The Release controls the time the Envelope will take to fall to zero after the MIDI-key was released. No matter what stage is currently active, once the key is released, the Envelope will jump right to the Release section immediately. The starting point for the falling curve will always be the last value that the Envelope produced. A finished Release section of the Amp Envelope gives the synthesizer the signal to end processing for this voice.
The Release section follows an exponential falling curvature.

ADSR Loop

Modulation: ✗ Automation: ✓

The loop parameter gives the option to start the Attack section again after the Decay is finished, thereby creating an LFO-type modulation source.

Chapter 7

ModulationMatrix

Chapter 8

Arpeggiator

Chapter 9

GlobalSettings