Ad van Ties
an Erastothenean Additive Oscillator
Manual
v2.2.0

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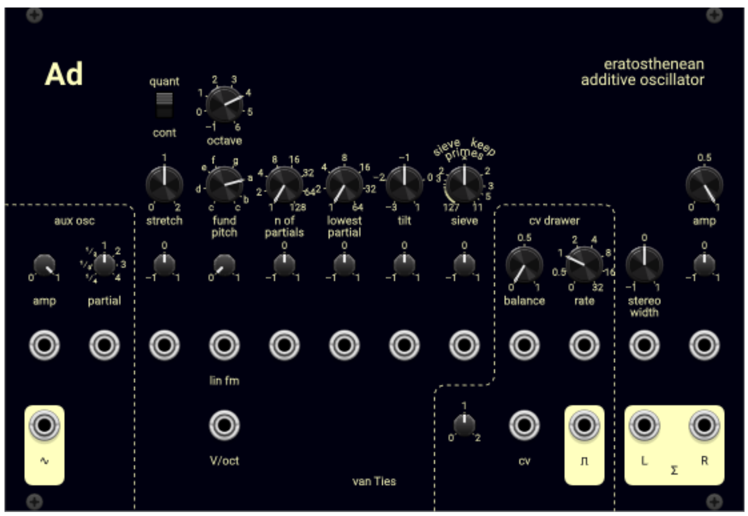

'Ad' in the 'Van Ties' plugin is a module for VCV Rack. It's a sound source based on the concept of additive synthesis, so I figured Ad is a good name.

Additive synthesis is based on the idea of Fourier decomposition.[1] Ad works by adding up to 128 sine waves (partials), more or less like as follows:

$$\sum_{i=n}^{n+N-1} i^{\alpha} \sin\Big(2\pi\big(1+(i-1)s\big)ft\Big).$$

n and N correspond to respectively the '**lowest partial**' and '**number of partials**' knobs on the panel. α is an exponent which is labeled '**tilt**', s the '**stretch**' parameter, f the fundamental frequency, which can be set with the '**fundamental pitch**' and the '**octave**' knobs.

[1] https://en.wikipedia.org/wiki/Fourier_series
https://youtu.be/spUNpyF58BY
https://youtu.be/nmgFG7PUHfo
https://youtu.be/SCujIf5eJ2w

Before we continue: almost all of Ad's parameters act continuously on the output wave. That is, there is a cross fading between parameters that actually only make sense as integers: the number of partials, the lowest partial and sieve. (Exceptions are the partial parameter for the auxiliary sine, the octave knob and, when in quantized mode, the stretch parameter.)

The '**lowest partial**' and '**the number of partials**' parameters have the effect of respectively a high-/band-pass and a low-pass filter.

For negative values of the **tilt** parameter the lower partials are emphasized, for positive values the higher ones. In the latter case the amplitudes diverge. That's why inside Ad the amplitudes are normalized, such that the output voltages never exceed ± 5 virtual volts.

If **stretch** is set to 1, we have an harmonic spectrum. The stretch parameter can be **quantized** to consonant intervals with respect to the fundamental: the perfect prime, the minor and major third, the perfect fourth and fifth and the minor and major sixth, all in just intonation.

There are a few more ideas ideas present in Ad:

The term 'erastosthanean' refers to the sieve of Eratosthenes in mathematics, an algorithm for finding prime numbers.[2] Ad's '**sieve**' parameter is based on this. If it's set to '$\times$', it does nothing. If it's set to 2 at the righthand side, all partials that are proper multiples of 2 (i.e. not 2 itself, but 4, 6, 8, ...) are sieved out, if it's set to 3 all proper multiples of 3 are filtered out, etcetera. If it's set fully clockwise, the primes and the fundamental are left over.

A similar thing is going on counterclockwise, except that in this case the primes themselves are sieved out as well. If it's set fully counterclockwise, only the fundamental is left. This means that (assuming that the other parameters are set to their initial values) we get a continuous transition between a saw, a square and a sine wave when we sweep the sieve parameter from $\times$, via 2 to 127 at the left side.

There are two sum ('**Σ**') output jacks, labeled **L** and **R**. If the **stereo width** parameter is set to 0, both outputs are the same. If the parameter is set to 1, the partials are distributed over both channels, except for the fundamental, which is present in both channels. This is done in such a way that for any value of the sieve parameter, those channels are pretty much in balance:

	left	right
mult. of 2:	4, 8, 12, 16, 20, 24, 28, ...	6, 10, 14, 18, 22, 26, 30, ...
mult. of 3:	15, 27, 39, 51, 63, 75, 87, ...	9, 21, 33, 45, 57, 69, 81, ...
mult. of 5:	35, 65, 95, 125	25, 55, 85, 115
mult. of 7:	77, 119	49, 91
mult. of 11:		121
primes:	2, 5, 11, 17, 23, 31, 41, ...	3, 7, 13, 19, 29, 37, 43, ...

If the width parameter is set to -1, the distribution is flipped. If there's more than one polyphony channel, there's one more flip for the odd channels.

[2] `https://en.wikipedia.org/wiki/Sieve_of_Eratosthenes`

There's a **linear FM** input, which respects negative voltages (i.e. through-0 FM). Also the **amp** CV input respects negative voltages, so it can be used for ring modulation.

The so-called **CV drawer** works as follows: assume the rate is set to a nonzero value and balance and the attenuator left to the CV input are set to 1. There is an internal 'play head' running over the partials, from 1 to 128. The values coming in at the **CV input** are recorded into a little 128 slot buffer. Once the play head hits 128, it jumps back to 1 and the amplitudes of the partials are scaled to the values in the buffer. (There is also some smoothing going on there.) The **rate** parameter determines how fast this happens.

Maybe it's easier to understand it by trying: send CV (for example an LFO) into the CV input and set the number of partials to maximum. Experiment then with the rate of your CV source and the drawer rate. Monitor the Σ outputs in a spectrum analyzer (for example the Bogaudio Analyzer-XL). If the LFO is a sine, one can get comb filter-like effects.

One can say that one can 'draw' into the frequency domain with a voltage that lives in the time domain.

The **balance** parameter works as a crossfader between unity and the recorded buffer. The attenuator knob can also boost the signal up to a factor 2. (This is done with signals with a maximum value of 5 V in mind: this way these can be amplified to max 10 V.) If the rate is set to 0, the buffer freezes.

The **trigger** output fires each time play head resets. This can be used to sync the CV source to the CV drawer, and this is useful for getting a stable spectrum.

On the left side there's a little **auxiliary sine oscillator**. Its pitch is given by the main oscillator's pitch, where a **partial** can be chosen. The stretch parameter of the main oscillator is respected, but not the FM input. Also generalized sub-partials can be chosen. Their frequency is given by $\frac{1}{1+(s-1)i}f$, instead of $(1+(s-1)i)f$ for an ordinary partial.

This little helper can be used for for example FM'ing the main oscillator, mixing it with the main output, triggering an oscilloscope etcetera.

The phasors inside Ad can drift apart, especially when one plays with the stretch parameter. Usually this doesn't cause an audible difference (as long as the output doesn't go through a wave shaper). Drifting can also occur when the module runs for a while, which can make an audible difference. There's a way to **reset** the phases: set both amp parameters to 0 or disconnect all three audio outputs.

Many of the different parameters can be pushed beyond the knob ranges with CV.

Ad works with polyphony.

Ad has a huge pitch compass, 9 octaves with the knobs only, especially towards the lower side. The idea behind that is that is to make it possible to also generate chords, instead of timbres. It could be interesting to play with this transition zone of harmony and timbre.

Demos can be found here:
`https://youtube.com/playlist?list=PLTg6VAqMki3XDdgPw0jcTMevmnDNcnVxR`.

What's new…

… in v2.1.0?

· the CV drawer

· the "sieve primes" mode

· bug fix

· some knob mappings

… in v2.2.0?

· a different, in general more efficient algorithm computing the sum of the partials. This algorithm can cause some phase shifting after running a while, though.

· The CV drawer works differently.

· Some parameters are renamed.

· Some of the knob mappings have changed.

To do:

· a nicer front panel

· including a little screen showing the spectrum

· sync in for the CV drawer

· prevent the phasors from drifting. This can be done in quantized stretch mode.