Ad van Ties
an Erastothenean Additive Oscillator
Manual
v2.1.0

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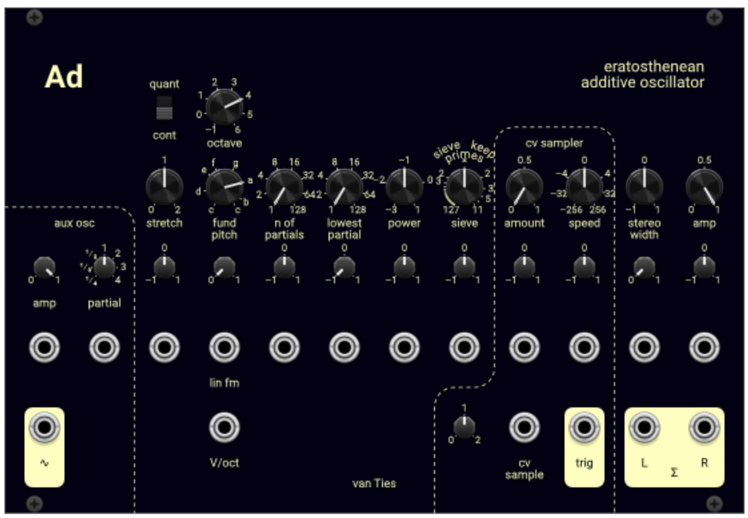

'Ad' in the 'Van Ties' plugin is a module for VCV Rack. It's a sound source based on the concept of additive synthesis, so I figured Ad is a good name. Additive synthesis is based on the idea of Fourier decomposition.[1][2][3][4] Ad works by adding up to 128 sine waves (partials), more or less like as follows:

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

n and N correspond to respectively the '**lowest partial**' and '**number of partials**' knobs on the panel. p is an exponent which is labeled '**power**', 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
[2] https://youtu.be/spUNpyF58BY
[3] https://youtu.be/nmgFG7PUHfo
[4] https://youtu.be/SCujIf5eJ2w

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 **power** 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. This 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.[5] Ad's '**sieve**' parameter is based on this. If it's set to '$\times$', it does nothing. If it's set to clockwise 2, 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 the.

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
4, 8, 12, 16, 20, 24, 28, 32, 36, 40, 44, 48, 52, 56, 60, 64, 68, 72, 76, 80, 84, 88, 92, 96, 100, 104, 108, 112, 116, 120, 124, 128,	6, 10, 14, 18, 22, 26, 30, 34, 38, 42, 46, 50, 54, 58, 62, 66, 70, 74, 78, 82, 86, 90, 94, 98, 102, 106, 110, 114, 118, 122, 126,
15, 27, 39, 51, 63, 75, 87, 99, 111, 123,	9, 21, 33, 45, 57, 69, 81, 93, 105, 117,
35, 65, 95, 125,	25, 55, 85, 115,
77, 119,	49, 91,
	121,
2, 5, 11, 17, 23, 31, 41, 47, 59, 67, 73, 83, 97, 103, 109, 127	3, 7, 13, 19, 29, 37, 43, 53, 61, 71, 79, 89, 101, 107, 113

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

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

There is a **CV sampler**. If the sampling **rate** is set to a nonzero value, the voltage coming in at **CV sample** is recorded in a 128 slot buffer and laid over the frequency spectrum. One can 'draw' so to say in the frequency domain with a voltage in the time domain.

For positive values of the rate, the new samples come in at the set lowest partial and travel upwards, for negative values they come in at the highest partial and travel downward. **Amount** is not a attenuator, rather it is a crossfader between unity and the recorded buffer. The sample input has an attenuator knob, which 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.

Maybe it's easier to understand it by trying: send CV (for example an LFO) into the CV sampler input, with the attenuator left to it open. Set the sampler amount to 1 and the number of partials to maximum. Experiment then with the speed of your CV source and the sampler rate. Monitor the Σ outputs in a spectrum analyzer (for example the Bogaudio Analyzer-XL).

The **trigger** output fires each time the CV input is sampled. This could for example be used to sync the CV sampler and the CV source in some way.

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.

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 phases of all the sines inside Ad can drift apart, especially when one plays with the stretch parameter. Usually it doesn't make an audible difference, but just in case there's a way to reset them: 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 sampler

· the “sieve primes” mode

· bug fix

· some knob mappings