Ad van Ties

an Erastothenean Additive Oscillator

— Manual —

v2.4.0

Matthias Sars

‘Ad’ is a module in the ‘van Ties’ plugin for VCV Rack.[1] It’s a sound source based on the concept of additive synthesis, so I figured Ad is a good name.

Fourier analysis is the study of wave signals, by decomposing (‘αναλυσις’ in the true sense of the word) them into sine waves (partials). For example, it turns out that a saw wave (with frequency f) can be written as:

$$\sum_{i=1}^{\infty} i^{-1} \sin(2\pi i f t).$$

Additive synthesis turns this idea around:[2] one creates a sound by adding

[1] https://vcvrack.com/
https://library.vcvrack.com/vanTies

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

many sine waves (partials) together.[3] (‘$\sigma\upsilon\nu\theta\varepsilon\sigma\iota\varsigma$’ in the true sense of the word.)

In the case of Ad: it works by generalizing the above expression for the saw wave as follows:

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

Instead of taking an infinite amount of partials (which is impossible to begin with), we select a certain set of partials, by choosing a lowest partial n, a number of partials N, and $\sigma_i \in \{0,1\}$ which ‘sieve’ certain partials. Furthermore, there is an exponent α, a ‘stretch’ parameter s, and for each partial there is a factor $a_i \in [-1,1]$ that scales that partial’s amplitude. C is just a normalization constant.

We’ll get into all of this and more in detail.

The Front Panel

Let’s have look at the front panel, and first at the knobs and jacks with yellow labels. There are five of these parameter knobs: **pitch**, **number of partials**, **tilt**, **stretch** and **sieve**, with corresponding CV inputs (for the pitch both a **V/octave** as well as an **FM input**), and bipolar attenuverters / a unipolar attenuator for FM.

On the bottom right there’s a corner where the background yellow and the labels are dark blue: here are the three output jacks: one for the **fundamental** wave (the first partial) and two for the **summed** waves (‘**Σ**’), labelled **Left** and **Right**.

On the bottom left there’s a **reset** button with a corresponding input.

On the top, the **spectrogram** is shown.

The knob and jacks with red labels belong to the ‘**partial amplitudes / CV buffer**’ section: there’s a **delay time** knob with corresponding CV input (without attenuator), a **clock** input and a **CV to buffer** input.

Parameters

Before we continue: all of Ad’s parameters act continuously on its output signals, including those parameters that actually only make sense as integers (e.g. the number of partials). In that case there is some fading going on. Exceptions are the **quantization** modes. These can be set in the context menu (right click on the panel to get there).

The **pitch** knob can be quantized in **octaves** or **semitones**. This only affects the knob, not the **V/octave** input. The **FM** jack provides linear through-0 FM. It only affects the main Σ outputs, not the fundamental output, so you can use the fundamental output for self-patching, for example into the FM input.

The ‘**the number of partials**’ parameter has more or less the effect of a lowpass filter.

[3] `https://youtu.be/SCujIf5eJ2w`

The **tilt** knob controls both the lowest partial as well as the exponent α in above expression. It has two zones:

- On the left-hand side it changes the exponent. The lower it is, the more the lower partials are emphasized. In this case the lowest partial is always the fundamental.

- On the right-hand side it changes which partial is the lowest. The exponent is fixed to 0, i.e. the partials all have the same amplitude. It has the effect of a high-pass or band filter.

The **stretch** parameter is the distance between partials in units of the fundamental frequency, so if it is set to 1, we have a harmonic spectrum. If it is greater than 1, the spectrum gets stretched out, if it is between 0 and 1 the spectrum gets squeezed, and if it is 0 the spectrum collapses into a single frequency. If it is negative, you get partials sounding lower than the fundamental, and the spectrum 'folds' around 0. It's easier to understand through experiment: play with the number of partials and the stretch parameter and see what happens on the spectrogram.

Like the pitch, the stretch parameter can be **quantized** via the context menu. This is done in such a way that the second partial is quantized to a consonant with respect to the fundamental. With consonant I mean here a perfect prime, a minor or major third, a perfect fourth or fifth or a minor or major sixth, plus or minus octaves, all in just intonation. See appendix 1 for a detailed list of quantization steps.[4] Unlike the pitch quantization, the stretch parameter is quantized after adding the knob and values together.

The term 'erastosthenean' 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 2 at the right-hand side, all partials that are proper multiples of 2 (i.e. not 2 itself, but 4, 6, 8 etc.) are set to zero. If it's set to 3, all proper multiples of 3 (9, 15, 21 etc.) are sieved out. (Note that 6, 12 etc. are already sieved out in the first step.) If it's set to 5, all proper multiples of 5 are sieved out. (Note that we don't need 4, since those multiples are sieved out in the first step as well.) If it's set fully clockwise, the prime numbers and the fundamental are left over.

A similar thing is going on on the left-hand zone of the knob, except that here it goes in reverse order and the primes themselves are sieved out as well: If you go counter-clockwise from $\times$, first 127 (the largest prime below 128) is sieved out, then 113, etcetera. At some point we get to the prime 61: then also $2 \cdot 61 = 122$ gets sieved out, then 59 and $2 \cdot 59 = 118$ etcetera. If it's set fully counter-clockwise, only the powers of 2, i.e. the octaves, are left.

[4] Being nitpicky: note that a stretch value of -1 means a second partial with 0 frequency, which implies that the quantization steps are infinitely dense around there. That's why between $-\frac{1}{3}$ and $-\frac{5}{3}$ the number of quantization steps is reduced.

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

The 'Partial Amplitudes / CV buffer' Section

As said the knob and input jacks with red labels all belong to the '**partial amplitudes / CV buffer**' section. This only becomes active when the '→ **buffer**' jack is connected. The CV coming in there is then recorded into a 4 second buffer.

There are three modes in the context menu. Let's look at the '**low** → **high**' mode first. Assume the **delay** time is set to a finite value. Then the lowest partial is attenuated[6] by the current CV value (where 10 V corresponds to unity), 1× the delay time later the next partial is attenuated by this value, 2× the delay time later the next partial and so on. In other words: the incoming CV travels from the lowest partial upwards. If the delay is set to 0, all the partials are affected simultaneously, in other words: it works then as an overall VCA. If you turn the knob all the way right to ❄, the buffer freezes and no CV is recorded.

Maybe it's easier to understand it by trying it yourself: send an LFO into the buffer input and set the number of partials to maximum. Experiment then with the rate of your LFO and the delay time and look what happens on the spectrogram. If the LFO is a sine, you can get comb filter-like effects. Or when you send in an envelope, the individual partials each get a delayed envelope.

You can send in an external **clock**. The delay knob then becomes a clock divider. (The knob ranges from a division by 8 next to 0 to a division by 1 next to ❄. Unless it would exceed the buffer length, then the clock gets divided by a power of 2.)

The '**high** → **low**' mode works analogously.

In unclocked **random** mode, the delay for each partial is determined by a uniform random distribution. In clocked random mode, the delay times lay on a grid in time, given by the incoming clock and the division set by the delay knob.

The random values are generated again on a **reset** trigger (via the button or input) or if all amplitudes are 0.

If the 'empty buffer on reset' option is selected in the menu, a reset trigger, indeed, empties the buffer.

Other Features

Ad works with polyphony. (It can get CPU-heavy, though.)

If the **stereo mode** in the menu set to **mono**, both the **left** and the **right** Σ outputs are the same.

If it is set to **hard-panned**, the partials are distributed over the two channels, except for the fundamental, which goes to both channels. This is done in such a way that for any value of the sieve parameter, those two channels are pretty much in balance. See appendix 2 for details. On a **reset** trigger or when all amplitude are 0, the channels are **flipped**. Initially, the channels are also flipped for the odd-numbered polyphony channels.

[6] This is the factor a_i in the wave formula.

The **soft-panned** mode is similar, except that the left partials also appear more quietly in the right channel and vice versa, in such a way that the lower partials are more panned less than the higher ones.

There won't be any panning going on if the right output is disconnected.

The x-axis of the **spectrogram** on the panel represents the frequencies on a linear scale, ranging from 0 on the very left to the Nyquist frequency (half the sample rate) on the very right. The y-axis represents the amplitudes on a logarithmic scale. The left channel is represented with yellow lines and the right channel with red ones.

The phasors inside Ad can drift apart, especially when one plays with the stretch parameter. Usually this doesn't cause any audible difference. The effect of wave shapers like wave folding can be audibly different, though.

Audible drifting can also occur, when the module runs for a while. The reason for that is, in order to save CPU, the 128 sines are not all computed brute force, but recursively from each other. Some artefacts can occur then.

The phasors are reset either of these cases: on a **reset** trigger, if the amplitudes of all the partials are 0 or if the three outputs are disconnected.

This means that the reset input can also be used for oscillator sync.

In many cases, the parameters can be pushed beyond the knob ranges with CV. Experiment with it to find out.

Ad has a huge pitch compass, 9 octaves with the knob only, especially towards the lower side. The idea behind that is, to make it also possible to generate chords, rather than timbres. You can do this by selecting only a few partials by using the tilt (on the right side), number of partials and sieve parameters. It could also 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, generally more efficient algorithm computing the sines. 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.
- Got rid of a few attenuverters.

… in v2.2.1?

- The phasors can't drift any more in quantized stretch mode.

… in v2.3.0?

- The CV buffer section is reworked. The most significant bit is that there is now a large buffer which runs at audio rate. Now one can send snappy envelopes into it, which are replayed without any smoothening.
- More compact front panel,
  - by replacing the auxiliary oscillator section with only the fundamental output,
  - getting rid of the stereo width and balance CV inputs, because I consider those 'set and forget' parameters,
  - getting rid of the amp parameter, since the new CV buffer section can take this role when the delay time is set to 0.
- Got rid of the change of v2.2.1. I didn't like the clicky sound when turning the stretch knob.
- Added the reset button.
- Changed some knob ranges.

… in v2.4.0?

- Added a spectrogram display.
- Still, it was possible to make the front panel got more compact (12HP) by
  - combining the lowest partial and tilt parameters into one parameter,
  - moving settings to the context menu,
  - getting rid of the buffer CV attenuator/amplifier,
  - getting rid of the balance knob (this is replaced by detecting the connection at the buffer CV input),
  - getting rid of the octave knob (this is replaced by the pitch quantization settings),
  - rearranging.
- Changes in the CV buffer section:

  - Added random mode.
  - Added the clock input.
  - Reworked the delay knob. The left-hand side is replaced by the 'high $\rightarrow$ low' mode.
- Reworked the left-hand side of the sieve knob. Basically changed the order.
- Added reset trigger input.
- Reworked the normalization of the amplitudes, such that the output audio is in some cases a bit louder.
- Reworked the stereo section: the lower partials are panned less than the higher ones now. And the continuous width parameter is reduced to three options.
- Added pitch quantization and reset modes.
- Added left/right channel flipping.

To do:

- Optimization.
- Due to the nature of the normalization procedure of the partials, the output audio can still be quite quiet. Would be nice to find a solution for this.
- Maybe find a purpose for the delay knob in case the buffer input is not connected.

Appendix 1: The Stretch Parameter in Quantized Mode

stretch	2nd partial (ratio w.r.t. fund.)	2nd partial (interval)
-2	-1	unison
$-\frac{11}{6}$	$-\frac{5}{6}$	– minor third
$-\frac{9}{5}$	$-\frac{4}{5}$	– major third
$-\frac{7}{4}$	$-\frac{3}{4}$	– perfect fourth
$-\frac{5}{3}$	$-\frac{2}{3}$	– perfect fifth
$-\frac{3}{2}$	$-\frac{1}{2}$	– perfect octave
$-\frac{4}{3}$	$-\frac{1}{3}$	– perfect fifth – octave
$-\frac{5}{4}$	$-\frac{1}{4}$	– 2 octaves
$-\frac{9}{8}$	$-\frac{1}{8}$	– 3 octaves
-1	0	$\times$
$-\frac{7}{8}$	$\frac{1}{8}$	– 3 octaves
$-\frac{3}{4}$	$\frac{1}{4}$	– 2 octaves
$-\frac{2}{3}$	$\frac{1}{3}$	– perfect fifth – octave
$-\frac{1}{2}$	$\frac{1}{2}$	– perfect octave
$-\frac{1}{3}$	$\frac{2}{3}$	– perfect fifth
$-\frac{1}{4}$	$\frac{3}{4}$	– perfect fourth
$-\frac{1}{5}$	$\frac{4}{5}$	– major third
$-\frac{1}{6}$	$\frac{5}{6}$	– minor third
0	1	unison
$\frac{1}{5}$	$\frac{6}{5}$	+ minor third
$\frac{1}{4}$	$\frac{5}{4}$	+ major third
$\frac{1}{3}$	$\frac{4}{3}$	+ perfect fourth
$\frac{1}{2}$	$\frac{3}{2}$	+ perfect fifth
$\frac{3}{5}$	$\frac{8}{5}$	+ minor sixth
$\frac{2}{3}$	$\frac{5}{3}$	+ major sixth
1	2	+ perfect octave
$\frac{7}{5}$	$\frac{12}{5}$	+ minor third + octave
$\frac{3}{2}$	$\frac{5}{2}$	+ major third + octave
$\frac{5}{3}$	$\frac{8}{3}$	+ perfect fourth + octave
2	3	+ perfect fifth + octave

Appendix 2: Stereo Distribution of the Partials

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