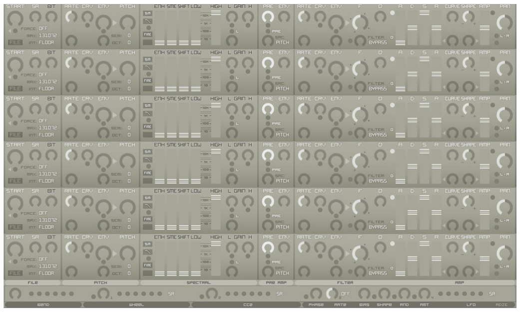
ADZE vst Manual
http://www.xoxos.net





Overview

Adze resynthesizes percussion samples by performing a single frame FFT analysis for the length of the sample, availing the following spectral features:

Enhance: emphasises the preexistent spectral content (2 modes)
Smear: changes the time (phase) information, for resynthesis or timestretching
Frequency Shift
Spectral Inversion
FFT low and high shelf filtering
Stereoisation / Restereoisation
Monophonic Bass

Adze signal chain is richly accoutred with conventional fare:

Downsampling to 32x
6 interpolation modes
Bit reduction: 12, 8-1
Preamp section with envelope assign
54 filter modes + bypass, 8 filter algorithms (4 zero delay feedback), lp, bp, hp, br, pk, punch, add harmonics

Modulation:

Morphing LFO (bias, shape, 2 rnd modes, host sync)
Curve adjustible pitch and filter envelopes
Curve and shape ADSR amplifier
-5 to +7 possible octave range
Start Position/Reverse
Wide Parameterisation
Panning
Channel swap
Signal inversion

The key to using Adze is to remain aware of the cpu resourcing for features you wish to use. If resynthesis parameters (marked in inverted text on the gui) are not changed between triggers, cpu use is the same as wav playback. Cpu can be capped by setting a maximum sample length. Filter and interpolation choices can also result in significant cpu use.

Stoooner is a free/demo version of Adze, limited to two samples instead of six. Adze owners may wish to run Stoooner alongside Adze as the smaller gui will result in faster song file load times.

## Paradigm

The six samples are triggered by each group of six keys across the keyboard beginning at C.

xoxos VST implementation places velocity trimmers underneath modulated parameters. Labeling is deferred due to this standardisation.

The gui for Adze has a number of circular toggle buttons which are unmarked for space considerations. These indicate an option for the parameter they are associated with, such as inverting modulation, or increasing the range of amplification.

## Spectral Features


### ENH Enhance

Enhance raises frequency content present in the file, it will not add frequencies that aren't already there. The effect can warm or add body, or increase the background noise and the presence of artifacts and may vary widely between samples. The SM smooth button options the transformation.

### SME Smear

Smear "spreads" the time information of the sample across the file. If there is an amount of silence at the end of the sample, this produces timestretching as the signal energy is distributed across the window. the FORCE parameter can be used to extend the original sample (remember that long files increase cpu).

If the end of the file is reached, the smear wraps around the filelength. Small amounts gradually disperse the attack, increasing amounts produce a diffused, temporally decorrelated block of sound which can be enveloped to make new samples with the same spectrum.

The outcome of smearing may not necessarily zero teminate, so a TAIL function (button with ramp picture) is included to fade the end of the file.

Smearing becomes especially interesting using IMAGINARY modes, as each frequency has its own stereo image at each setting.

> Hint: an unforeseen interaction in development has been left in - if using a FORCE setting longer then the original sample and a MAX setting shorter than the original sample, more periodic textures are produced with higher smearing as the window is partly processed at the shorter rate.

### SHIFT

FFT frequency shifting is accomplished by offsetting the bin index, so that a 100% shift will return each bin to their original location. Frequency shifting is optioned by the FINE button. The unshifted signal is then located in the middle of the slider range so that slight adjustments in either direction may be made.

### INVERT

The unmarked circular button inverts the spectrum.

### IMAGINARY

The icon indicates the use of complex numbers in the FFT. This button "piggybacks" the second audio channel onto the imaginary spectrum of the first channel (which would normally mirror the real spectrum) and pairs the two signals in one transformation.

The effect of this is to turn mono samples into stereo, and to redefine stereo samples. Adjusting the smear parameter influences the outcome.

IMAGINARY has two settings - the second uses the left channel for LOW frequencies to produce monaural bass (the word MONO appears under the LOW slider).

### LOW, HIGH and GAIN

The spectrum below and above these points can be reamplified, producing a "perfect" shelf eq at the original sample pitch (and near perfect at other pitches with hermite or sinc interpolation).

The GAIN parameters range from 0 to 1x the original volume, or can be optioned to range from 1 to 2x. There is a single velocity response trimmer which can be assigned to either band.

General Features

I. File Panel


START, REVERSE, FILE

8, 16 and 24 bit wavs of 1 or 2 channels.

SR

The sample rate of the file can be reduced by a factor of 1 to 32, most effectively in combination with FLOOR interpolation.

BIT

The bitdepth of the sample can be reduced (original, 12, 8 to 1). For economisation, bit reduction is performed with the FFT resynthesis.

FORCE

Normally the sample is analysed using a window length the power of two greater than the file length (and FORCEing to the same length has no effect). This function is intended to be used in combination with SMEAR to produce timestretching. Higher settings use an immense amount of cpu.

MAX

A limit can be placed on the number of samples to process when using long samples (or browsing samples). The FORCE setting supercedes it.

INT Interpolation

Six interpolation algorithms are included for their timbral variance. FLOOR, or no interpolation, is preferred when emulating early electronics. The difference between algorithms is more significant with extreme repitching, and entirely insignificant if using the original pitch.

Generally the algorithms increase in quality through the series. The second mode, S, uses an s-curve to fade between known samples. The third is LINEAR.

Of the more intensive processes, HERMITE can be relied upon the most to reduce aliasing. CUBIC interpolation is more comparable to LINEAR at some settings.

The most expensive setting uses a 64 point SINC interpolation, the quality of which is dependent on host samplerate (64 samples is a smaller portion of the spectrum at 96k than 44.1k).

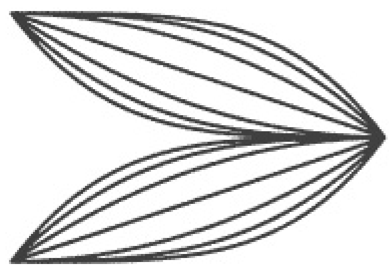
Using the HIGH and LOW settings with a frequency analyser and repitching will exhibit the performance of each algorithm (note that aliasing will vary with pitch).


II. Pitch Panel

PITCH, SEMI, OCT

Range from 0 to 1 octave, 0 to 12 semitones, and -5 to +5 octaves respectively. The pitch knob has a nonlinear response so fine adjustments can be made.

The pitch and filter envelopes are identical. The single envelope stage can be curved in either direction and inverted.




III. Pre Amp Panel

PRE, ENV

Several of the filter modes have internal clipping and nonlinearities which can be overdriven. Adding an envelope to this allows for eg. the attack portion to be emphasised by generating more harmonics. Any of the three envelopes can be tapped for this modulation.

The button under PRE boosts both PRE and AMP. Like the GAIN section, one assignable velocity control is available.


## IV. Filter

F, Q

The filter algorithms vary in performance and tone. The algorithm designations are BI, BQ, SV, RS, MG, KL, NT, and MS.

BI Bilinear Integrator

Zero delay feedback 6dB low and highpass

BQ Cookbook Biquad

Robert Bristow-Johnson's widely used 12dB filters

SV State Variable

Andrew Simper's (Cytomic) linear trapezoidally integrated state variable 12dB/octave filter

RS Robin Schmidt

Topology-preserving transform zero delay feedback state variable. 12dB ;)

LADDER FILTERS

Four pole ladder filters can be tapped at any pole for 6, 12, 18 or 24dB lowpass. Devising other output modes can sometimes be accomplished by combining the poles in ways that may not have been intended. Ladder filters are often quieter than other modes because of nonlinearities that limit the signal.

MG Stilson/Smith Moog Ladder

A widely used open source algorithm

KL Karlsen Fast Ladder

Peak and lowpass modes are included, some with nonlinearities

NT

An open source zero delay ladder

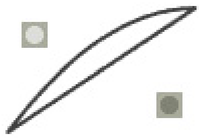
MS Mystran Ladder

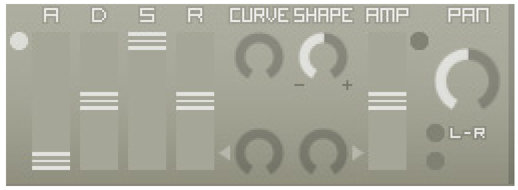
A high performance open source zero delay ladder with nonlinearities, limited to two samples instead of six. Adze owners may wish to run Stoooner alongside Adze as the smaller gui will result in faster song file load times.

IVb. Filter Selections

Some filter modes with nonlinearities include additional overdrive. The gui selector displays these filter algorithm designations in reverse (black text outlined with white).

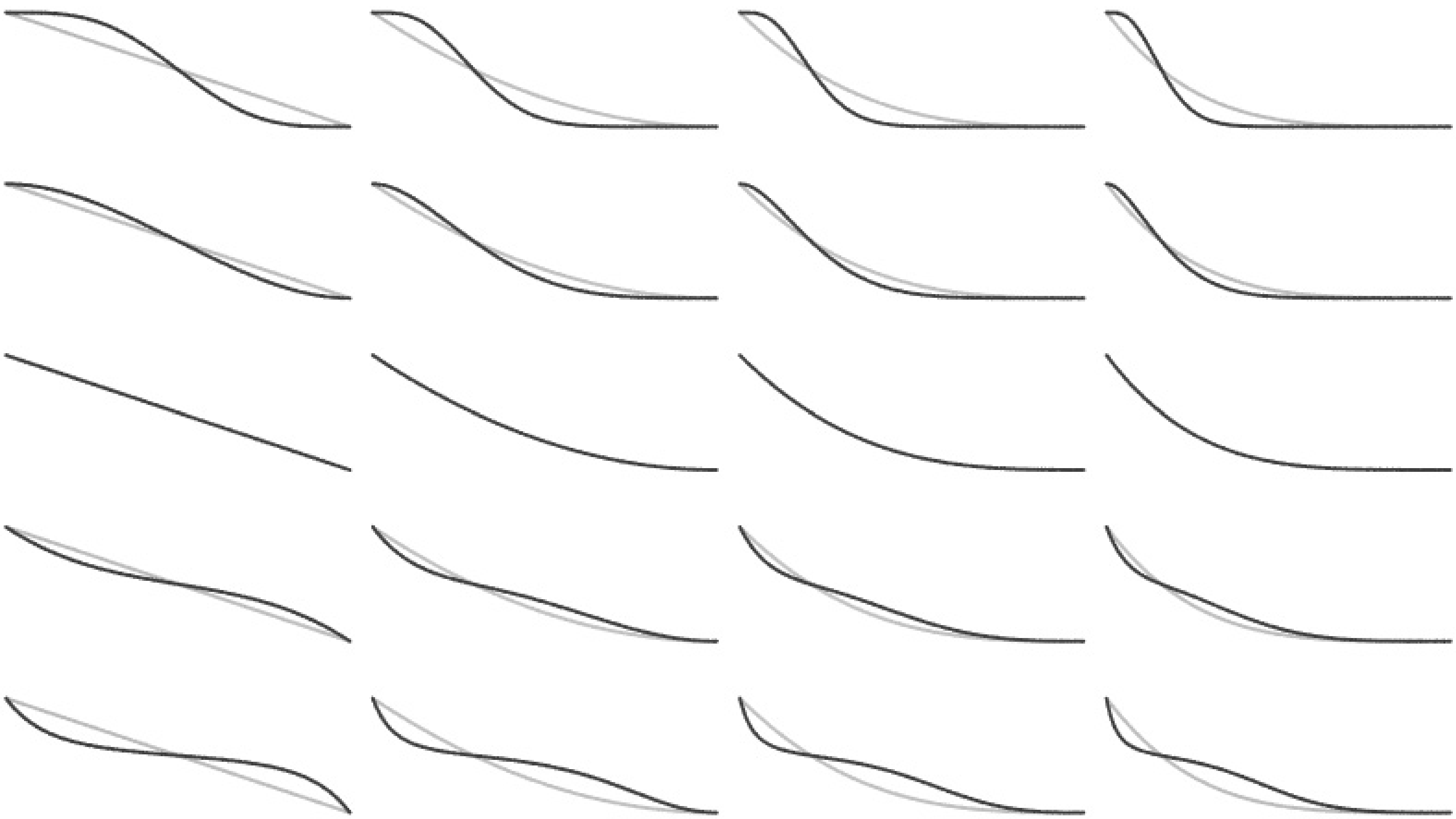
| | | | |
|---|---|---|---|
| 0 | bypass | | |
| 1 | BI | 6dB | lowpass |
| 2 | NT | | |
| 3 | BQ | 12dB | |
| 4 | SV | | |
| 5 | RS | | |
| 6 | MG | | |
| 7 | KL | | |
| 8 | NT | | |
| 9 | MS | | |
| 10 | MG | 18dB | |
| 11 | KL | | |
| 12 | NT | | |
| 13 | MS | | |
| 14 | MG | 24dB | |
| 15 | KL | | |
| 16 | KL-nl | | |
| 17 | NT | | |
| 18 | NT-nl | | |
| 19 | MS | | |
| 20 | NT | 6dB | bandpass |
| 21 | BQ | 12dB | |
| 22 | SV | | |
| 23 | RS | | |
| 24 | MG | | |
| 25 | NT | | |
| 26 | MS | | |
| 27 | BI | 6dB | highpass |
| 28 | NT | | |
| 29 | MS | | |
| 30 | BQ | 12dB | |
| 31 | SV | | |
| 32 | RS | | |
| 33 | MG | | |
| 34 | NT | >12dB | |
| 35 | MS | | |
| 36 | BQ | 12dB | band reject |
| 37 | SV | | |
| 38 | RS | | |
| 39 | MG | | |
| 40 | NT | | |
| 41 | MS | | |
| 42 | BQ | 12dB | peak |
| 43 | SV | | |
| 44 | RS | | |
| 45 | MG | | |
| 46 | KL | | |
| 47 | KL-nl | | |
| 48 | NT | | |
| 49 | MS | | |
| 50 | punch 1 | | |
| 51 | punch 2 | | |
| 52 | punch 3 | | |
| 53 | add harmonics 1 | | |
| 54 | add harmonics 2 | | |






V. Amplifier

The attack of the ADSR can be toggled between linear and exponential contours. The decay and release portions have two contour parameters: CURVE fades from linear to quartic (eg. steep curve), SHAPE applies an S-shaped transform in either direction.



horizontal: CURVE     vertical: SHAPE

The SHAPE parameter effectually changes the balance of the early and late decay/release contour. It was originally intended to function in one direction to achieve a droplet shaped contour. Applying the S-curve in the "other" direction proved to be useful for raising the volume of the tail while preserving a defined attack.

The amplifier section also includes a PAN control, and switches to swap the left and right channels and to invert the output (eg. for when combined signals are routed to a peak compressor or limiter).

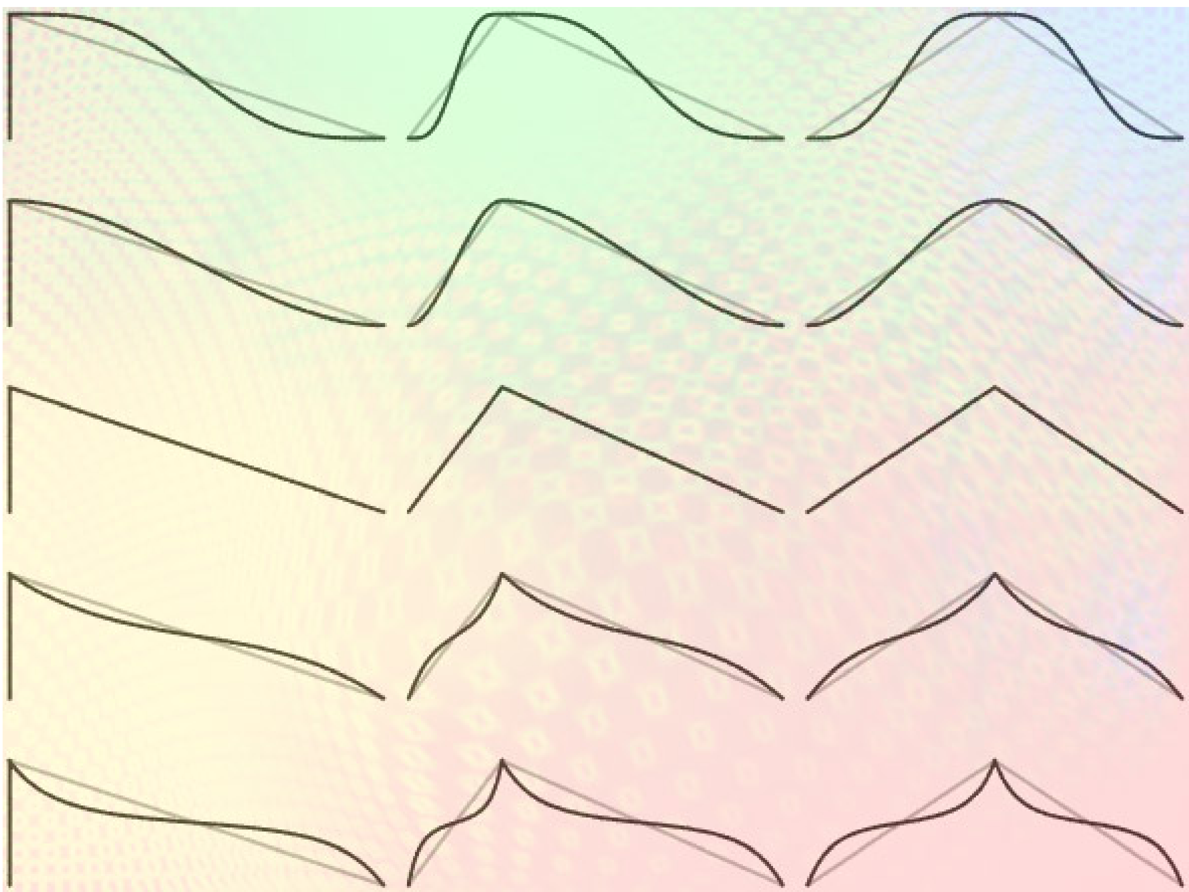
PAN and SHAPE are both reset to their central position to have no effect. The center "detent" spans about a thirteenth of the range.


Modulation

Modulations have a single amplifier and can be routed to any selected voices. Pitch bend, mod wheel (cc1) and cc2 are assignable on the gui. MIDI cc3 and 4 are routed to the RATE and AMOUNT of the lfo.

The lfo uses a single flexible algorithm instead of waveform selections. BIAS moves a peak between saw (downwards) and triangle shapes. The signal may be inverted to provide a ramp (upwards) contour.

The SHAPE parameter works the same way here as on the amplifier envelope. At about the 1 o'clock position the output is close to a sine, and at the full amount a rounded square. The opposite polarity has sharp peaks and dips and tends towards the median.



RANDOMISATION has two modes. The first gradually crossfades the amplitude extremes to random values, availing a variety of textures across its range, from gentle variance to creating dynamic tempo synced random patterns. The second mode also applies variation to the lfo rate.

Randomisation is seeded by the rate and phase parameters whenever the lfo is reset. If different patterns are needed at the same tempo, holding [ctrl] can be used to make slight adjustments to the phase parameter (or any other) in order to reseed the pattern without overtly affecting the contour.

The lfo can be reset to PHASE by the button next to this parameter. To facilitate the automation of this button, it is named _lfo reset for alphabetic precedence.

Postface

Filter algorithms should be considered to be adapted by the developer as the variables of implementation (such as oversampling, floating point, coefficient scaling, et c.) may modify the tone and experiential qualities of the process to some degree.

Some of the filter algorithm designers also develop vst:

Andrew Simper
http://www.cytomic.com

Robin Schmidt
http://www.rs-met.com

Ove Karlsen
http://sourceforge.net/projects/pxu/files/

Teemu Voipio (Mystran)
http://www.signaldust.com

Continued gratitude is extended to those who do not for the many ideas their open source filters have allowed me to realise, eg. Tim Stilson, Julius Smith, and Robert Bristow-Johnson and others.

There is no expression of guarantee, please demo before making purchases. Adze VST was created with the SynthEdit SDK - www.SynthEdit.com
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