

Vulpus Labs

Chebz

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Introduction

Chebz is a toolbox for building waveforms out of stacked sine waves at different frequencies.



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KNOW MANY THINGS

Chebyshev polynomials are used to efficiently generate the sequence of harmonics of a fundamental sine wave up to the 9th harmonic. The blend of harmonics in the output signal can then be controlled and modulated along with the overall balance of odd and even harmonics.

Modulating signals can be smoothed with three levels of smoothing (none, medium, high) to reduce clicking. The blended output signal can be optionally wave-folded, and the shape of the resulting wave form is displayed in a handy scope.

Finally, each harmonic has its own dedicated output, which can receive either the unmodulated sine wave at that frequency or the modulated harmonic (switchable via a modulation bypass switch).

Usage



The basic controls of Chebz are what you would expect for an oscillator: a large dial controlling the central frequency (centred on 65.41Hz or C2), a **V/OCT** input pitching the frequency up or down by one octave per unit of voltage, and an **FM** input (exponential scale) further modulating the frequency by up to 6 semitones in either direction.

These control the *fundamental frequency* to which Chebz then adds 9 additional frequencies, up to the 9th harmonic:

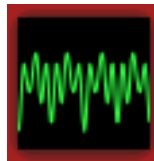
Harmonic	Frequency (Hz) when fundamental is at C2	Note
0th (fundamental)	65.41	C2
1st	130.82	C3
2nd	196.23	G3 + 1.96 cents
3rd	261.64	C4
4th	327.05	E4 - 13.69 cents
5th	392.46	G4 + 1.96 cents
6th	457.87	B \flat 4 -31.17 cents
7th	523.28	C5
8th	588.69	D5 + 3.91 cents
9th	654.1	E5 - 13.69 cents

Slightly confusingly, harmonics 1, 3, 5, 7 and 9 are described as the *even* harmonics, as their frequencies are even multiples of the fundamental frequency, and harmonics 0, 2, 4, 6 and 8 are described as *odd* harmonics, as their frequencies are odd multiples of the fundamental frequency.

Chebz has three main outputs, to which it sends the sum of the **ODD** harmonics (each amplified by a scaling factor), the sum of the **EVEN** harmonics (ditto), and the sum of **ALL** the harmonics. The **FOLDING** switch controls whether wave-folding is applied to sample values outside the range -5v to +5v - each individual harmonic will stay within this range, but some stacks of harmonics will stray outside it. Wave-folding folds values back inside this range, but will create some distortion in the process.



The shape of waveform created by summing all of the harmonics is shown in the **scope**:



Each harmonic has its own dedicated volume control, with a volume modulation input (on the left) and an output (on the right) for just that harmonic. You can cross-patch one harmonic's output into the modulation input of another.

Modulating the volume of a harmonic at audio rate by a signal with very fast transitions, such as a square wave, is likely to produce clicks. These can be smoothed out by setting the **SMOOTHING** for the volume controls to "medium" or "high". If however you want to modulate a harmonic at audio rate by a smoother input such as a sine wave, then set **SMOOTHING** to "none" (or the higher frequencies of the input signal will be eliminated).



By default, the output for each harmonic receives that harmonic *after* the volume control/modulation stage. However, you can choose to send the unmodulated harmonics, at full volume, to the outputs by flicking the **MOD BYPASS** switch.



Finally, there are controls for setting and modulating the balance of odd and even harmonics sent to the **ALL** output, for applying 8x oversampling to the oscillator (there *might* be some point if you're heavily modulating harmonics at audio rate), and for setting the level of smoothing to be applied to the input signal controlling the odd/even balance modulation.

Credits and Acknowledgements

Chebz was written by Dominic Fox in February 2023.

This module was inspired by Squinky Labs's excellent *Chebyshev* module for a rival product.

The oversampling button employs Chris Neuberger's IIR resampling filter, generously provided as open source by the redoubtable R_Ware.

Thanks to the developers at Cherry Audio for their great products, especially Voltage Modular.