1 Concept

The Harmotron is an idea for a dual control synthesizer. One of the controls will be the pitch of the tone generator (a harmonic rich sound) while the other will control the bandpass frequencies. This way, the user will be able play on both the main note (background) and the isolated frequencies for harmonic effect.

1.1 todo

continue to test effect of a specific bandpass fc on static signals

find appropriate serie with shiftable parameter (like exponent exp<->log)

generate sines on user input (midi input ? ? ?).

generate appropriate filter on user input.

implement adsr

2 Source synthesis

The source will be synthesized by adding sinusoids that are integer multiples of the fundamental frequency (user input).
Each harmonic will have an amplitude equal to its term in a serie determined by user input (harmonic, geometric, or something like that).

2.1 Formulaes and results

The following synthesis formulaes have been used and tested :

$$1) synth(f,t) = \sum_{n=1}^{N} a^n cos(2\pi f t n / F_s)$$

$$2) synth(f,t) = \sum_{n=1}^{N} a^n cos(2\pi f t n / F_s) + \sum_{n=1}^{N} a^n cos(2\pi f t n / F_s * \frac{3}{2}) + \sum_{n=1}^{N} a^n cos(2\pi f t n / F_s * 3)$$

It could be interesting to generate different type of spacing for the harmonics. Different series of ratios of other enharmonic (or is it xenharmonic ?) ideas.

3 filtering

The filter will be a bandpass (maybe morphing to a comb filter centered on F_c). F_c will be dynamically determined by user input.

3.1 reflexions on filtering

After a few initial tests, the concept works but needs refining. Some amplitudes were too low, others were a bit dissonant. The present hypothesis is that the filter frequency F_c does not perfectly align with all harmonics, causing loss of amplitude and dissonances from letting pass untargeted frequencies in a greater ratio.
As it is, a single harmonic may be passed at a time. By using parrallel filters, it would be possible to play simultaneous harmonics by computing them all and controlling their amplitudes by user input (i.e. on/off from keyboard input).

4 envelope

4.1 initial gain

Since the signal is the result of coefficients of a geometric series multiplied by the amplitude of the `cos` function, it can be multiplied by its inverse to reduce peaking. The total sum of a geometric series is known to be $\sum_{n=1}^{N} a^n = \frac{1}{1-a}$ if $|a| < 1$. It's iverse is $1 - a$. We estimate the energy of the `cos` function by taking

$\int_{\frac{-\pi}{2}}^{\frac{\pi}{2}} cos(x)\,\mathrm{d}x = 2$ and its inverse is of course $\frac{1}{2}$. Since int the case of the second synthesis formula, the series is applied three times,it seems wise to divide the gain by 3.
So the initial gain will be $\frac{1-a}{6}$