FM with Multiple Operators

Review of Simple FM

Synthesis with frequency modulation has been presented elsewhere. Most descriptions thoroughly cover the fundamental feature-- the result of frequency modulation is a composite tone created by adding sidebands centered on the carrier with a spacing determined by the modulator frequency. It is also made clear that the number of sidebands is determined by the strength of the modulator. Less clear, as it is usually buried in the mathematics, it the fact that the strength of modulation must be proportional to the modulation frequency in order to produce sounds that are consistent from one pitch to another.

The amplitude and spacing of sidebands determines the quality of tone. The spacing derives from the relationship of the modulator frequency to the carrier frequency. If we combine this understanding with practical experiments, we can make some general observations about FM sounds.

If the modulator and carrier frequencies are a simple ratio, the result will be a clearly pitched tone. The nature of the tone is further affected by the relationship of the modulator to the carrier:

- If the modulator and carrier frequencies are the same, the result is similar to subtractive synthesis, with the carrier as the fundamental and a tone that brightens as modulation increases. As the modulation is increased, certain harmonics (including the fundamental) will briefly drop out.
- If the modulator is higher in frequency than the carrier, some harmonics will be missing. A 2:1 ratio produces the hollow sound associated with clarinets, a 3:1 ratio will include every third harmonic and so on.
- If the modulator is lower than the carrier, the modulator frequency becomes the fundamental, but the energy of the tone remains near the carrier. This produces a very bright sound similar to a high pass filter.

If the frequency ratio is a fraction, the sound is more or less clangorous, depending on how close the fraction comes to irrationality. A ratio of 1.5 produces paired harmonics with a definite pitch center (an octave low) but some edge, whereas a ratio of 1.237 sounds (with appropriate envelope) like a big bell. If you sweep the ratio of a sounding tone, you hear the characteristic rising and descending pitches within a matrix of complex partials.

It is possible to use a fixed frequency for either the carrier or modulator. If the fixed frequency is within the range of the played pitches, the results no longer follow a scale and adjoining notes may have quite different sounds. If the fixed frequencies are very low the result is some type of vibrato. If the modulator is fixed this vibrato is a pure tone, if the carrier is fixed the tone will be fairly rich with subtle variation.

Complex FM

With envelope control over the carrier and modulator amplitudes, simple FM will produce a wide range of rich sounds, but there are definite limits. For instance, it is not possible to change the frequency ratio without passing through very strange sounding intermediate ratios. More elaborate and more subtle sounds require a more complicated patch. The patch can evolve in two ways-- extra carriers can provide additive combinations of basic FM sounds, and extra modulators can use complex waveforms for modulation.

The general procedure for additive combinations is straightforward. For instance, a carrier-modulator pair adjusted for a bright, non-harmonic tone may be added to the attack of a fairly pure tone. This will provide a percussive punch or the breath at the start of a flute tone.

The effect of a complex wave used as a modulator is harder to describe. The mathematics involved are simple-- sidebands are produced according to the frequency ratios of carrier and modulator, but the calculation must take every partial of the modulator into account. Since the partials of the modulator are not at equal strength, the effective modulation index of each partial is different. We quickly reach a point where prediction of the outcome is difficult. Complex FM is best explored through practical experiment.

The TX Architecture

The most successful implementation of FM is still the original Yamaha architecture as applied to the DX7 and TX81 instruments and imitated in Native Instrument's FM8 softsynth. The conceptual heart of this architecture is the concept of the "operator", a subsystem containing an oscillator, amplifier and envelope generator. All operators are passed pitch and velocity of the note and each is programmed to respond with its own frequency and envelope. The DX7 had 6 operators, the TX instruments had only 4, but their sound palette was extended with non sinusoidal waveforms. The 4 operators of the TX could be arranged into one of 8 algorithms (the DX had 32) as shown in figure 1.

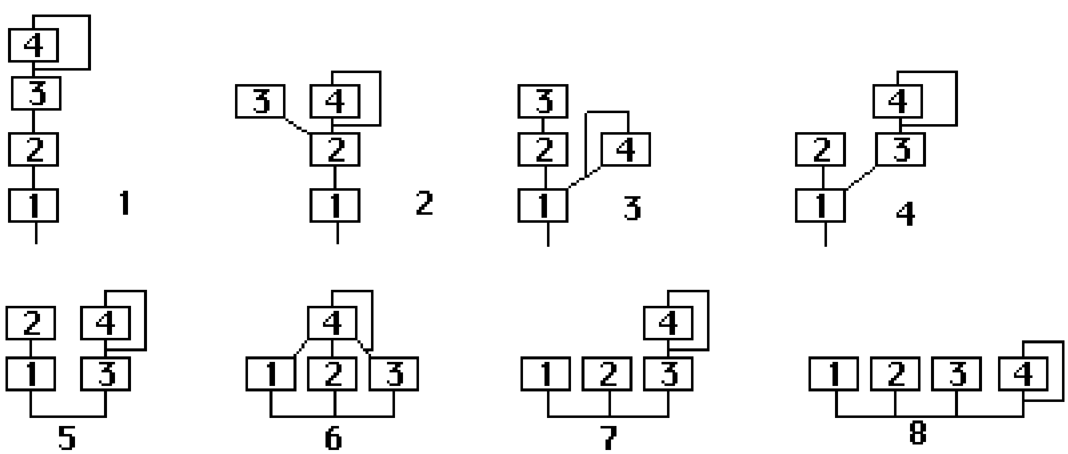


Figure 1. Ways to connect 4 operators.

TX Emulation in MSP

The TX architecture makes a good model for an MSP patch. In this example, we will make an abstraction for each algorithm to load into a poly~ object with the patchname command. The core of the outer patch is shown in figure 2.

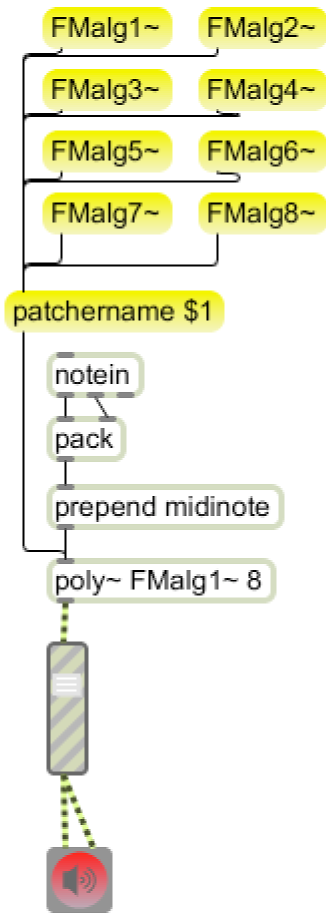


Figure 2. the FM uberpatch

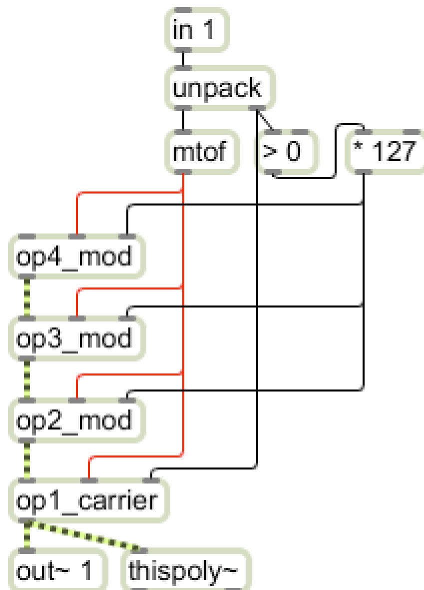


Figure 3. Contents of poly~

This is a standard poly~ setup with the midi notenumber and velocity packed into a last and passed as arguments to midinote.

The abstractions will be named FMalg1~, FMalg2~ and so on. These will correspond to the combinations shown in figure 1. Figure 3 shows Fmalg1~. The operators are also in abstractions with names like op4_mod. There is a difference between the processing required in a modulator and a carrier as we shall see shortly. The number refers to the system that will be used to give independent parameter control to each operator.

When a note arrives in the abstraction, the data is unpacked, with the note number going to each operator and the velocity sent to the carrier(s). In this version, the velocity is converted to 0 or 127 for the modulators, which makes the tone quality independent of velocity.

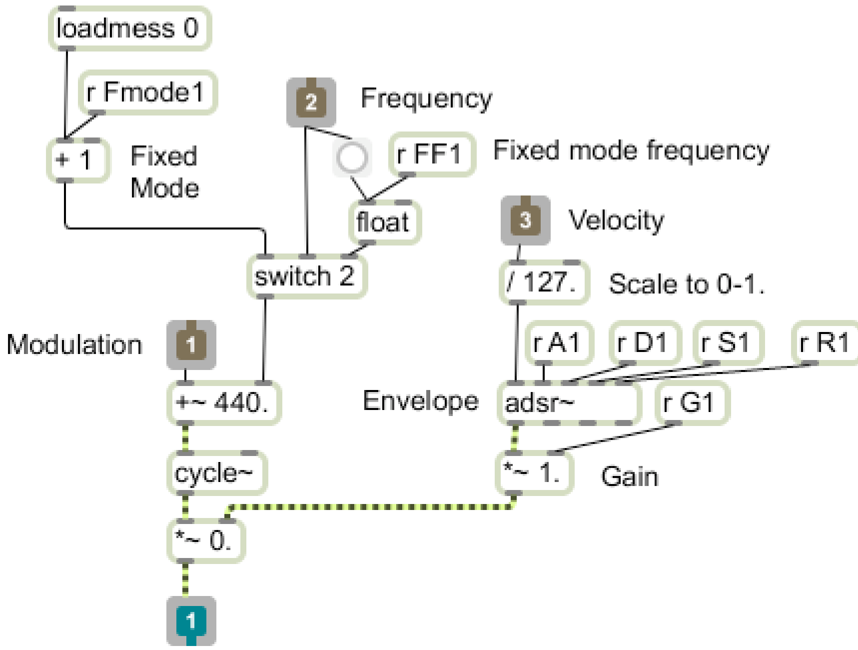


Figure 4.

Figure 4 shows the carrier version of the operator. The frequency input is inlet 2, the velocity input is inlet 3 and the modulation signal is inlet 1. The modulation signal is added to the frequency input to determine the frequency of the output signal, The amplitude of the signal is controlled by the ADSR envelope and a gain multiplier. The switch apparatus sets fixed frequency mode. When the carrier is in fixed frequency mode, the value received from FF1 is substituted for the note frequency. There is a bang and float combination to make sure the fixed frequency is passed through the switch on a new note. Parameters are set via send and receive pairs with names like "A1". The send section is shown in figure 10. The meanings:

- Fmode sets Fixed Frequency mode
- FF sets the Fixed Frequency
- A,D,S,R control the envelope
- G controls overall gain. A 0 mutes the operator, which is useful in voicing the patch.
- Ratio sets the frequency ratio on modulators
- Index sets the modulation index.

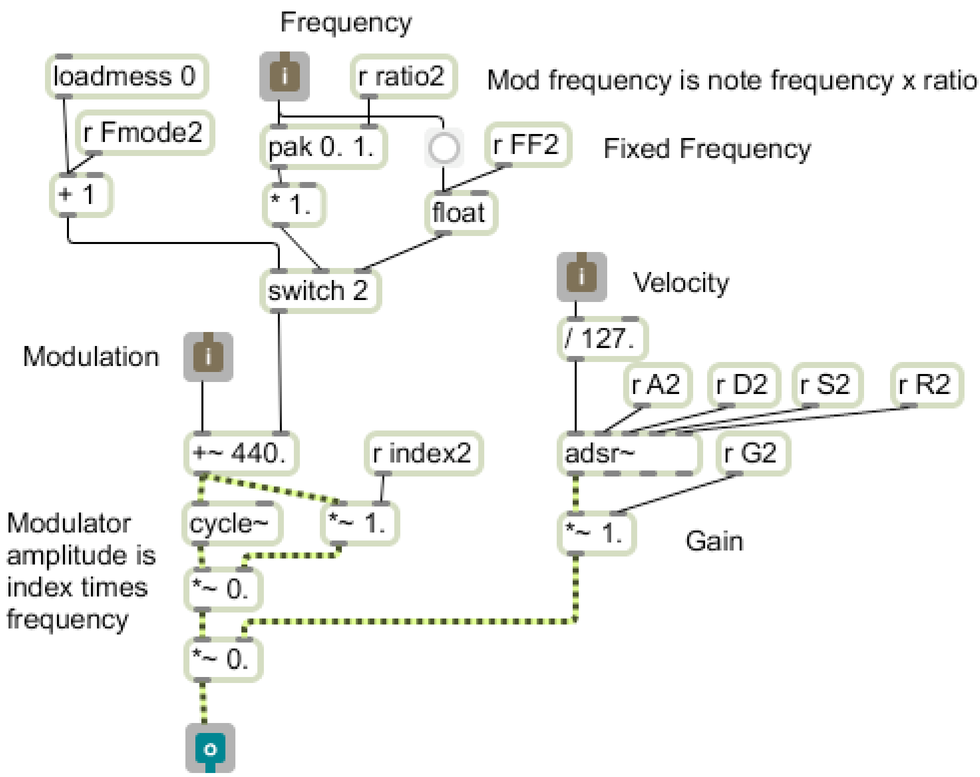
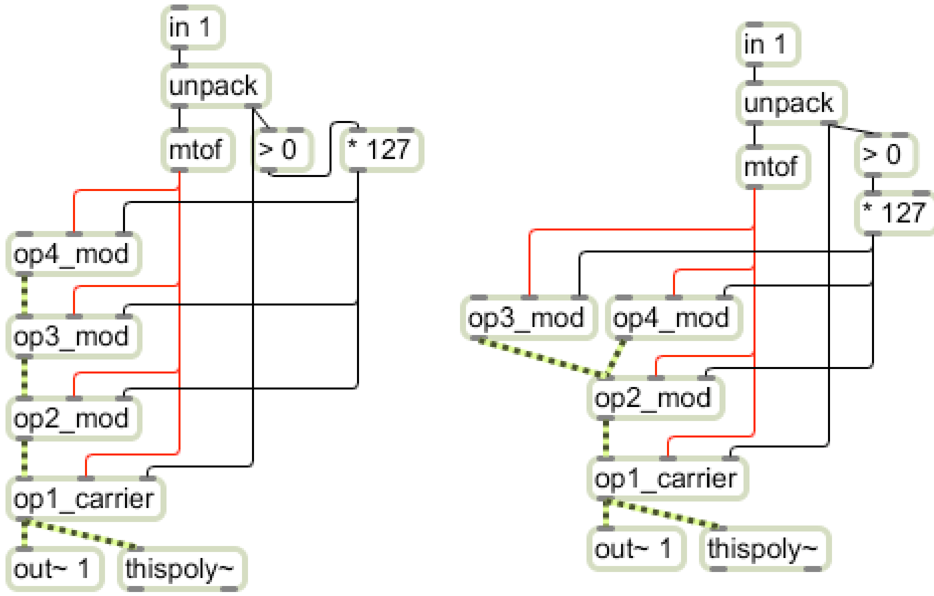


Figure 5.

Figure 5 shows a modulator section. The basic structure is like the carrier, but the frequency must be derived from the note frequency and the operator ratio. The use of a pak to combine the frequency and ratio ensures that a change of either will cause a new value to be calculated. Remember that when a list is applied to a math operator, the operation is performed on the first two members of the list. The amplitude of the modulator is determined by the index and the basic frequency. To be specific, the deviation in Hertz is the product of the modulation frequency and the index. (Remember that this deviation is added to the note frequency in the carrier.) The frequency here is also determined by any modulation applied to the modulator.

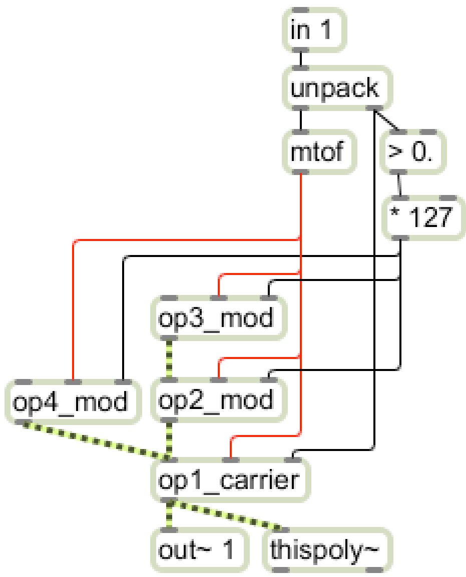


Algorithm 1 Algorithm 2

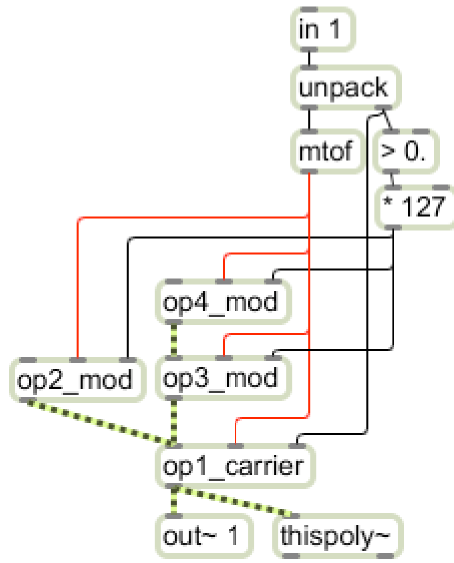
Figure 6.

Figure 6 shows the patch for algorithms 1 - 4 which have various arrangements of 3 modulators working on one carrier. In algorithm 1 the modulators stack up. This amounts to simple modulation with a complex waveform. The envelopes on the modulators define the evolution of the sound. With mod 3 off, simple FM ensues. As mod 3 comes in, the sound quickly becomes complex as the mod 2 waveform develops sidebands. Each sideband in the modulator generates further sidebands in the output. Mod 4 makes the result even more complex, but only during the time mod 3 is active.

In algorithm 2 the top 2 modulators are connected in parallel rather than series. Two modulators should produce the same sound in parallel as they do in series, but the parallel connection allows a transition between two complex sounds during the course of the note.



Algorithm 3



Algorithm 4

Figure 7.

Algorithms 3 and 4 appear to be the same at first. The difference is in the placement of op4_mod. Operator 4 has a feedback function. When an operator modulates itself, the waveform is modified, effectively adding harmonics to the spectrum. Each harmonic modulates the carrier producing another type of rich sound. Algorithm 3 supports two kinds of complex modulation and algorithm 4 supports a combination of complex and simple modulation. These are good with slow envelopes to drive a gradual transition form one tone to another.

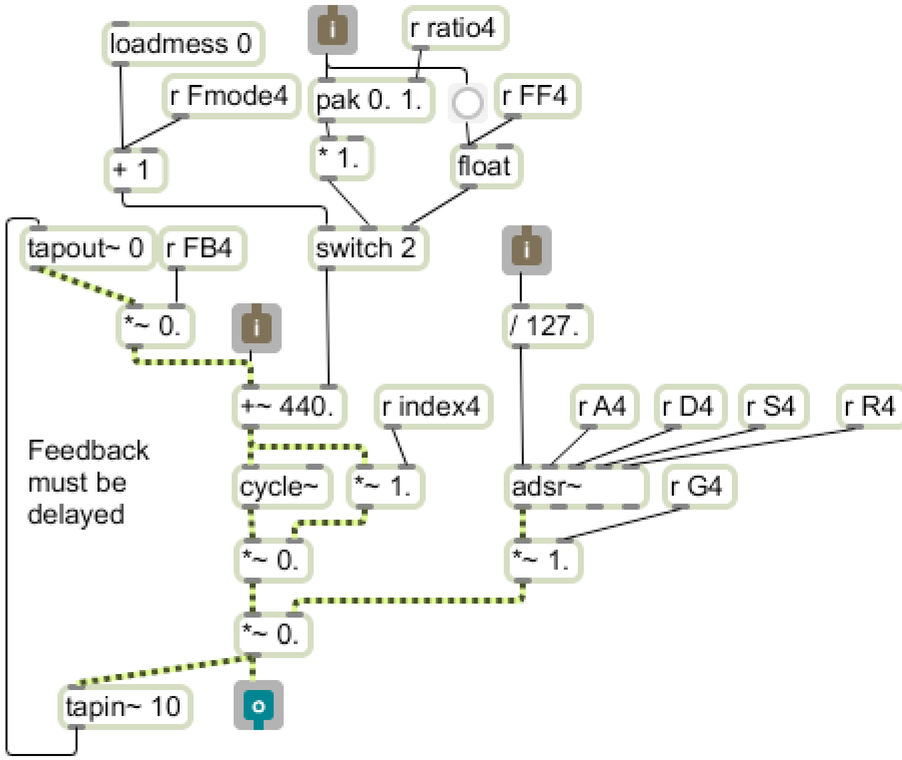
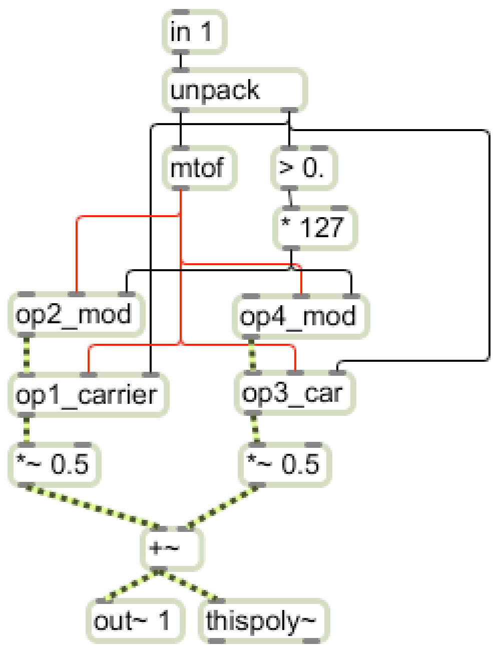
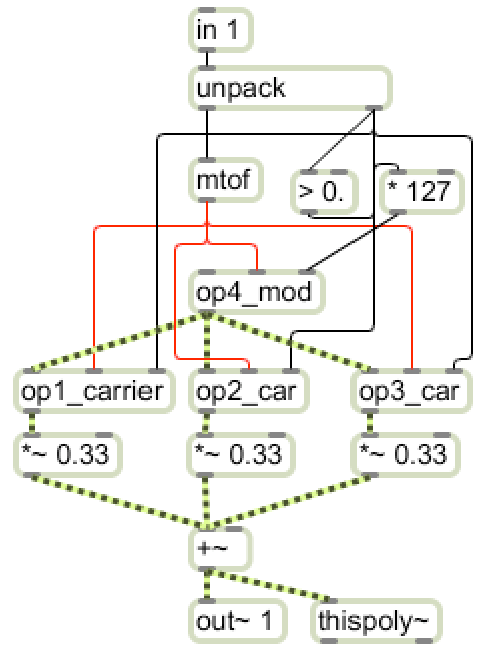


Figure 8. Operator 4.

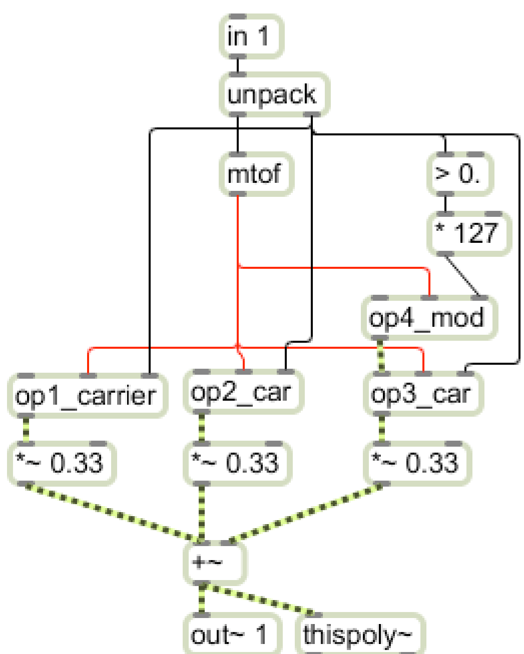
Getting the feedback for operator 4 requires a special trick. The MSP compiler rejects any patch that would produce feedback, because it could lead to an infinite processing loop. Running the fed back signal through a tapin~ tapout~ pair (even with 0 delay) prevents this.



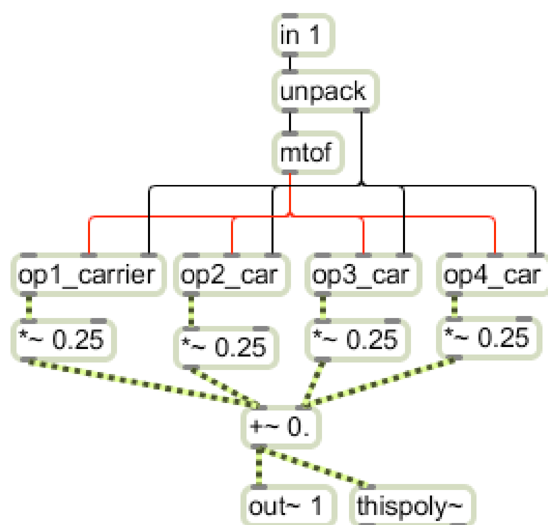
Algorithm 5



Algorithm 6



Algorithm 7



Algorithm 8

Figure 9.
The remaining algorithms provide additive synthesis in various ways. The operators labeled Car are similar to op1_carrier. Note that the outputs are scaled to add up to 1 in all cases. The add~ oblect is not strictly necessary, but it simplifies the wiring of thispoly~.

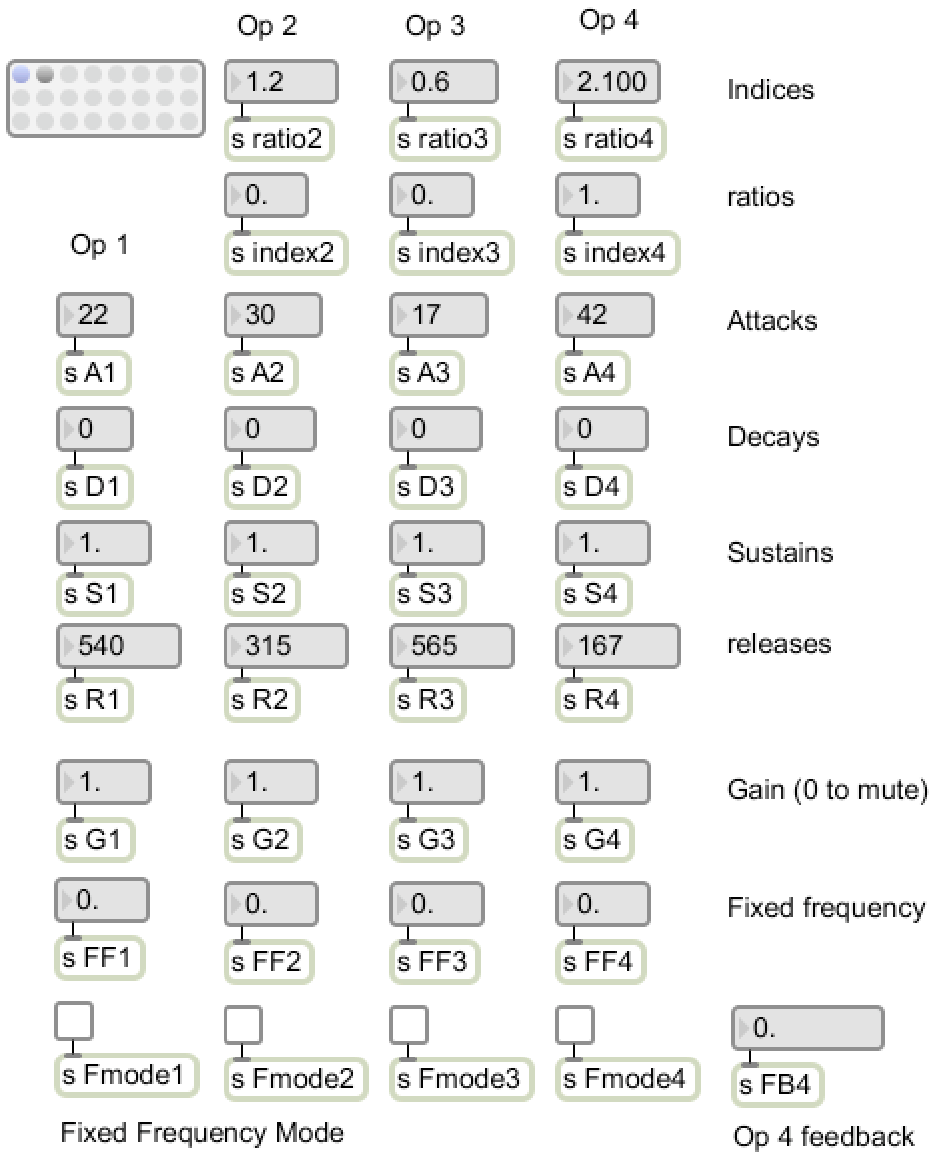


Figure 10. Control panel for the FM player.

Further Work

This patch can be further refined in several ways.

- The only response to velocity is a change of amplitude. The TX81 features velocity scaling, where the gain of each operator responds to velocity indivually. This will create a timbre change on heavily struck notes.
- The TX81 also features key scaling, where the gain of operators is optionally adjusted by note number according to a table.
- The TX81 includes a choice of waveforms for all operators.