Experimenting & Processing Samples + Modulation/Control Voltage in Automatonism

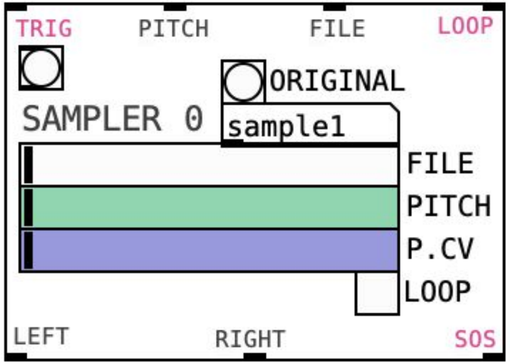
Click to Download Software

A Note:

So as we go through this workshop, I want to say honestly - the sampler module in automatonism is not the BEST sampler ever. However, this is a free tool and you can really start to explore modular in a fun with with samples/sounds you know and love. It can be a great way to understand signal flow and modulation sources and what each module can do for you.

SAMPLER:

The "SAMPLER" module can load 8 different .wav tracks. The files are located in ../samples/ and are named sample1.wav, sample2.wav, sample3.wav etc. To change the samples you have to replace those files with your own files, while keeping the same name structure. This might seem counter-intuitive but the advantage of this system is that your files will alwyays be pointed to the right location and you can move your project main folder anywhere and to different platforms without losing your samples. If you want a sampler that can load files from anywhere on your harddrive I suggest you try the excellent sampler module made by Claude Barker. More info about that at:
https://github.com/megalon/pd-AUTOMATONISM-sampler



Attenuator - A control that can reduce the strength of a signal or voltage going through it.

CONTROLS:

FILE - select between sample 1-8

PITCH - change the pitch of the sample

P.CV - attenuator for cv at the PICTH input

LOOP/1SHOT - toggle between looping mode and 1shot mode. In 1shot mode the sampler needs a trigger signal at the TRIG input to fire the sample.

INPUTS:

TRIG - trigger input to play sample in 1SHOT mode

PITCH - cv input for PITCH

FILE - cv input for modulating what sample is active

LOOP - trigger input to switch between LOOP and 1SHOT mode

OUTPUTS:

LEFT - sample left channel output

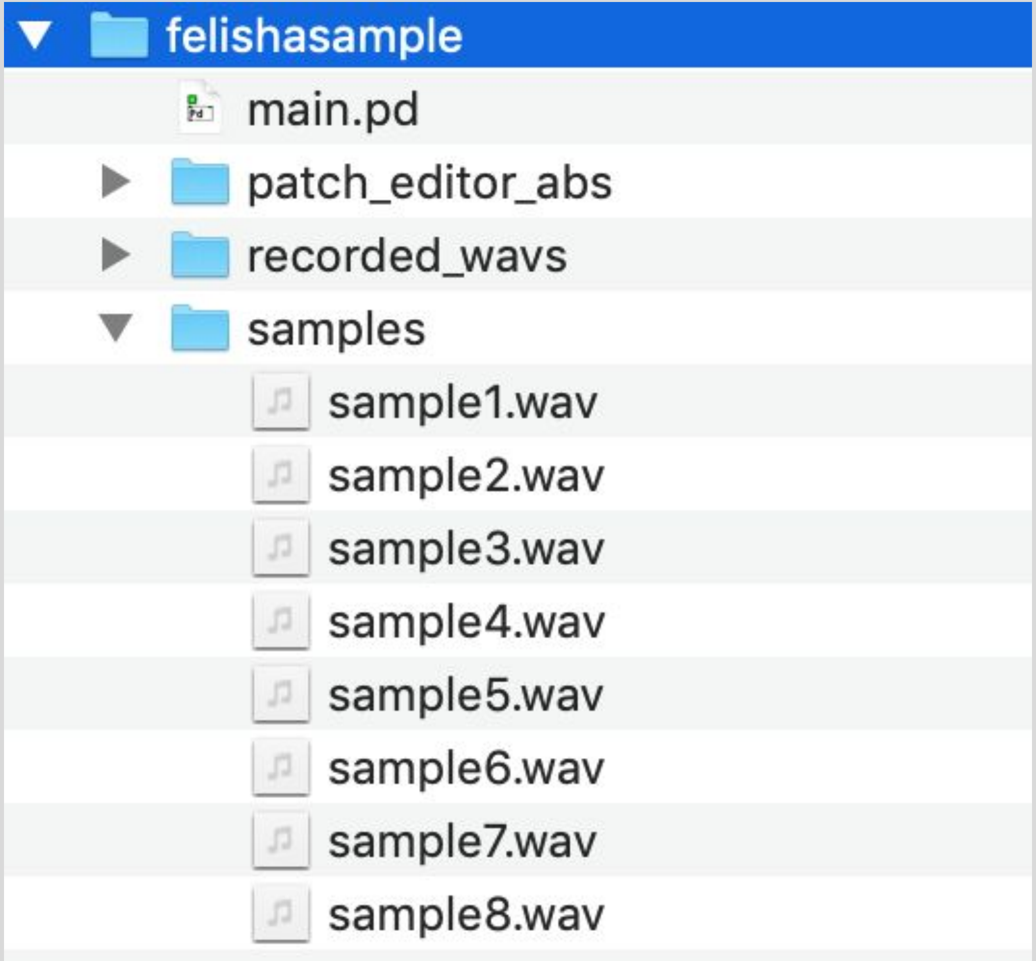
RIGHT - sample right output

SOS - trigger output at the start of each sample

Your file must be .wav

The structure must be the same in the sampler folder

you cannot load more than 8 samples at a time



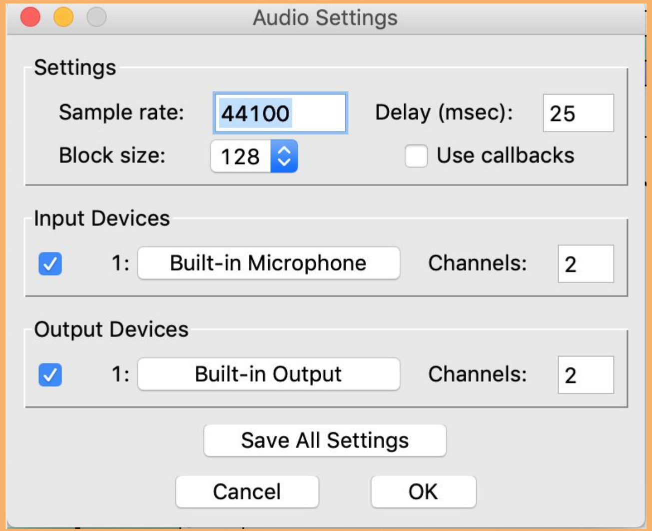
Sample rate

Is the resolution of the digital audio

Think of it like a picture, the higher resolution the more details you can see. A fixed interval of how many sample data points are stored and/or read per second

48kHz sample rate means that there are 48000 data points (called samples) being stored per second.

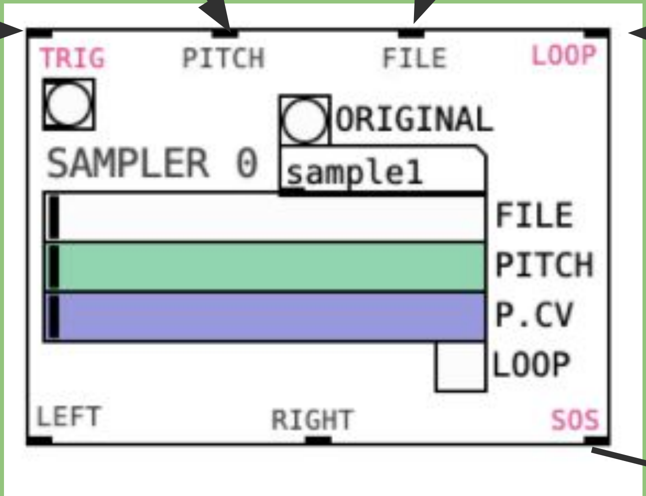
This is just something to keep in mind when you are recording from different sites.. You may get digital artifacts (little glitchy sounds) when the audio becomes to compressed / the sample rate is low.



If you are running Automatonism and it's going slow for your computer - go into audio settings and change block size and delay ms - giving a buffer so I can play in Zoom, but also check here if you are having issues/it is making your computer slow.

Both of these CV inputs should be experimented with. Later in our patch sheets you will find some examples.

Just like all of our pink inlets here you will need a trigger/gate/clock to get started OR you can click loop and it will loop the full sample for you.

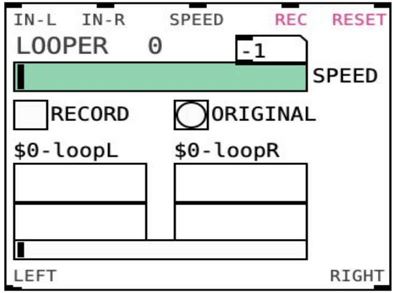


You will still need something to trigger the module to toggle between the two modes loop/1shot. You can try a trigger sequencer here to start.

Theoretically you should be able to trigger the start of another sampler in sync (or another module that calls for a trig in). PLEASE tell me if you get this working HAH

LOOPER:

The "LOOPER" module can record signals from your patch and loop them. It works like a live sampler. Press the RECORD toggle to record and stop recording. The recording can be automated by trigger signals at the REC input. The LOOPER can record CV signals as well as audio. It keeps the last recorded loop in memory when you quit PD, however when you launch the patch again it needs 5-7 seconds to load the recording.



CONTROLS:

SPEED - control the playback speed of the recorded loop

RECODS - toggle recoring on/off

ORIGINAL - press the button to get back to the orginal speed of the recording

INPUTS:

IN-L - left input

IN-R - right input

SPEED - cv input for modulating the SPEED parameter

REC - trigger input to start/stop recording

RESET - trigger input to start the recording from the beginning

OUTPUTS:

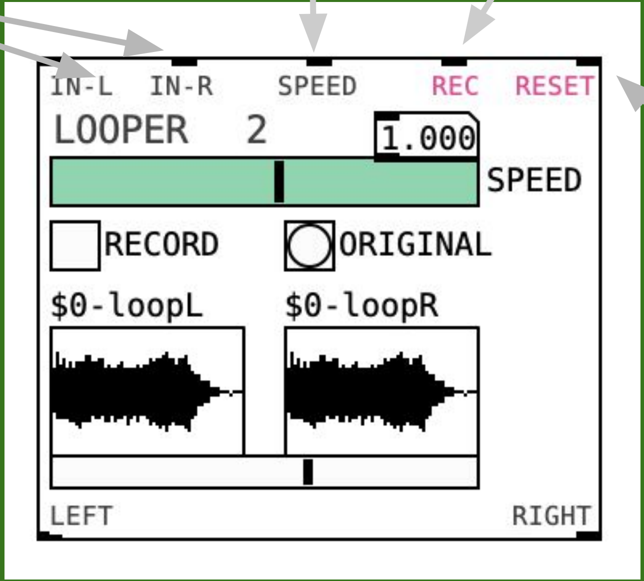
LEFT - sample left channel output

RIGHT - sample right output

Audio or CV Source to record

Modulate the speed of your loop with an LFO or 8step sequencer into a VCA (slower modulation needed must use VCA as your attenuator))

Try with a clock variation and trigger sequencer - it will record when the trigger hits in a timed manner that you choose



Setting up a VC counter or trigger sequencer here could make some solid rhythms as you reset the recording over. But try different trigger inputs to get a result that fits best for you!

Sound Sources Online

archive.org

ubuweb

Audio Routing

Blackhole (macOS)

Jack Audio

VB-Audio VoiceMeeter

FREE DAWS (Digital Audio Workstations)

Reaper

Garageband (mac/iOS)

Audacity

WHAT DO YOU WANT TO CONTROL?

TIME

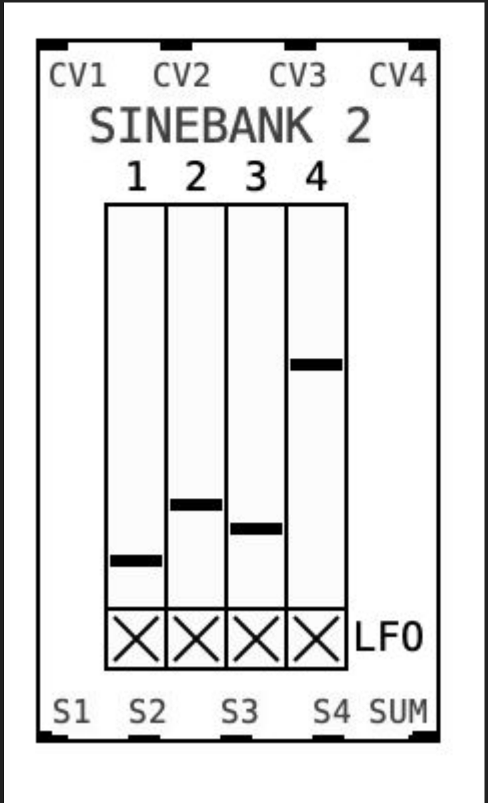
PITCH / FREQUENCY

VOLUME

FEEDBACK

Anything else you want!!

Low Frequency Oscillators



Bipolar - negative and positive voltages (look for a green slider attenuverter)

Unipolar - positive voltages (look for a purple slider attenuator)

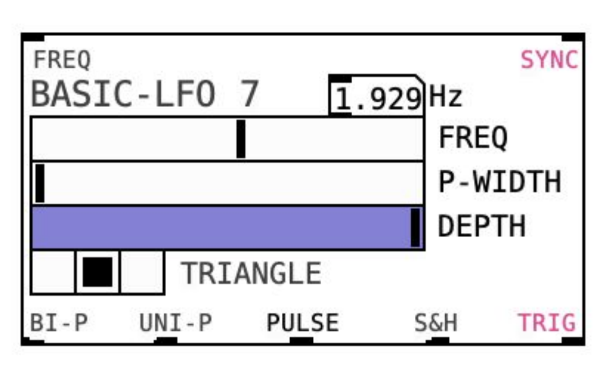
S&H - *sends out a snapshot of held voltage from the LFO*

LFO modulating OSC Pitch for vibrato (use bipolar)

LFO modulating VCA for tremelo (use unipolar)

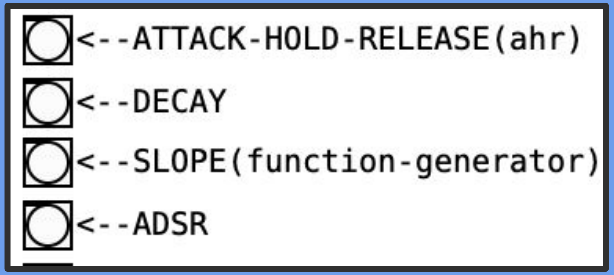
LFO modulating VCF to effect timbre/filter sweeps (use unipolar)

LFO to MIXER PAN for stereo panning (bipolar)



Creative Ways to Use an LFO

ENVELOPES



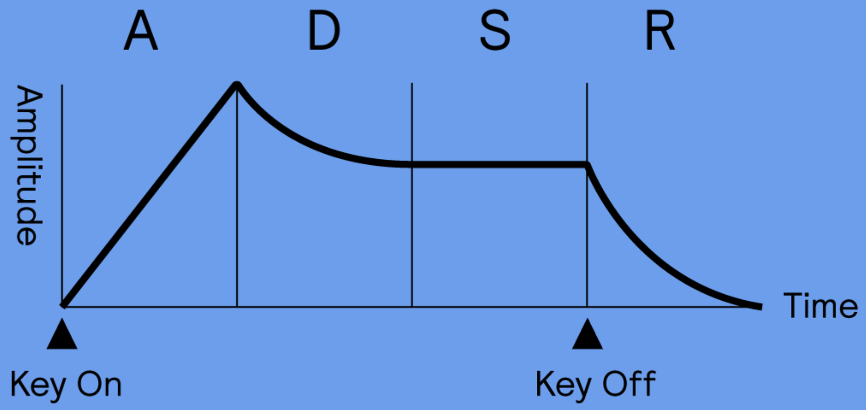
creating emphasis on different aspects of your patch

VCF Cutoff changing brightness of your sound over time

VCO Timbre changes when you modulate with an EG

VCA Voltage control the amplitude (volume of your patch) over time

Modulating Delay Time

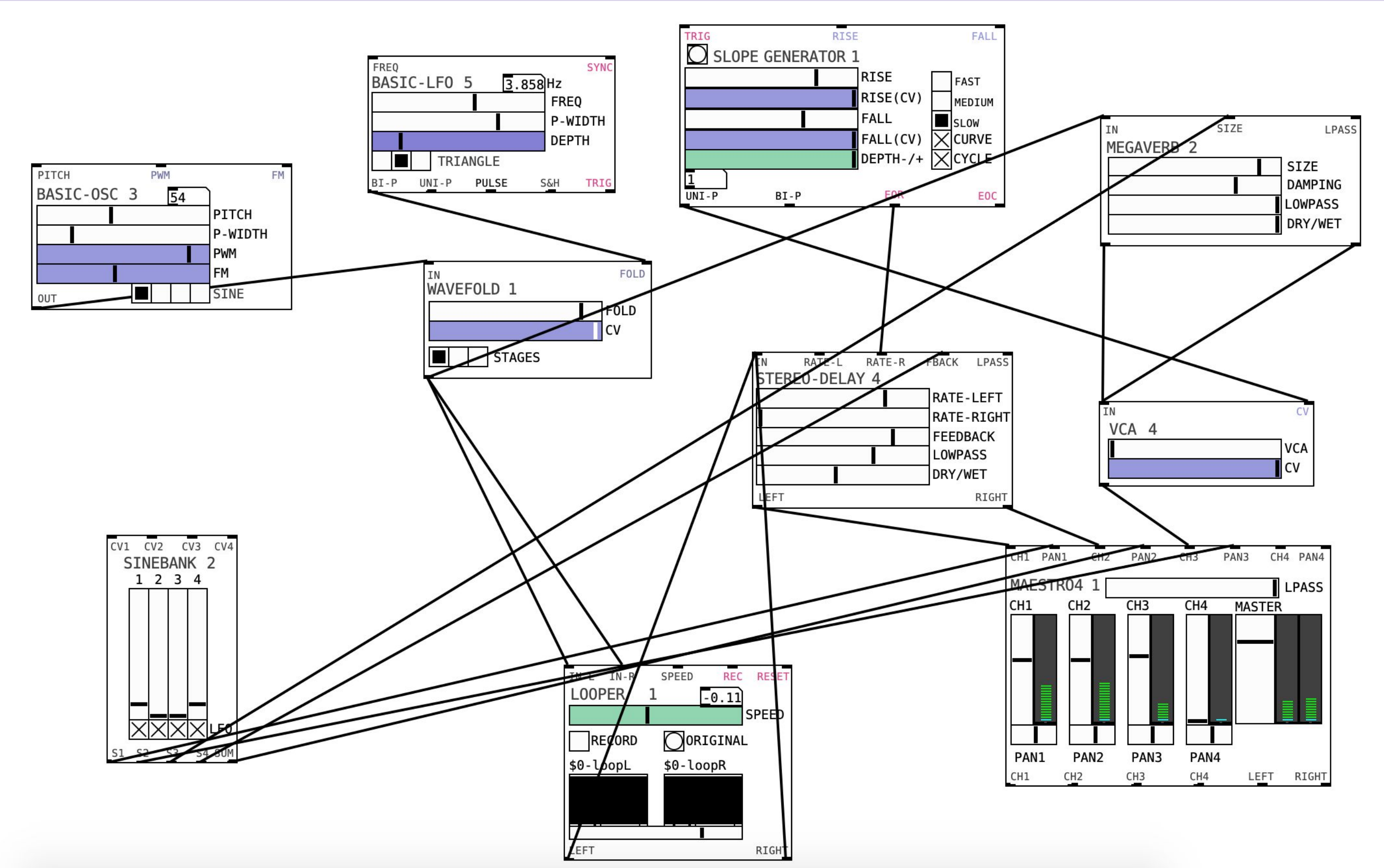


Gates tell destination module when to turn a parameter on or off

In a similar way, the *trigger* sends the destination module an on/off signal but unlike a gate this will only be a short pulse using the rising edge of the signal

Clocks keeping your whole system in time - use clock multipliers/dividers/VC counter/trigger sequencers to keep in time with different rhythmic patterns and speeds

Look for pink inputs. These modules will need a trigger/clock/gate input to start an event.



Patch 1

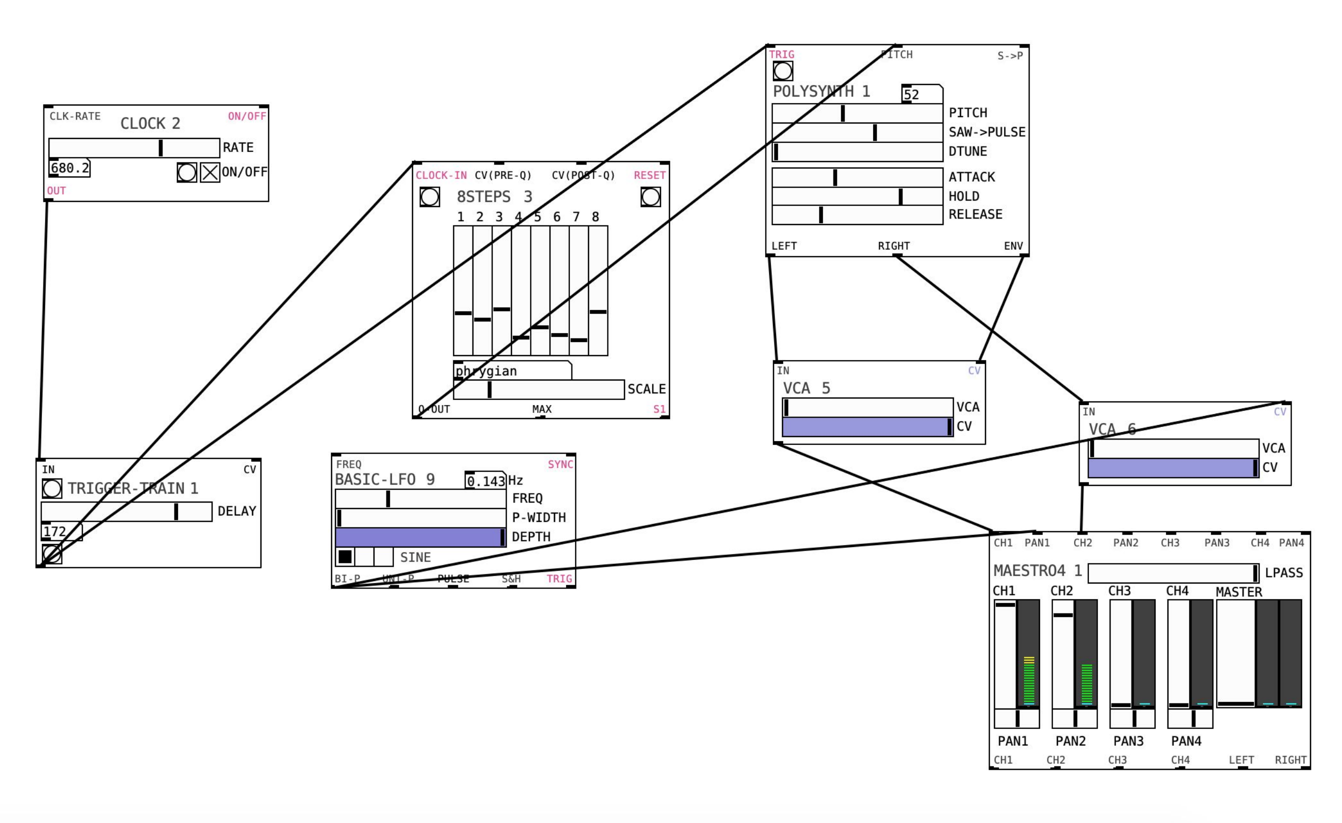
Wave Folding is a form of wave shaping that literally folds of a wave inwards on itself until the wave starts compressing creating distortion.

I sent the oscillator into the wavefolder and looped that. I then took the sinebank in LFO mode and sent it ALL OVER THE PLACE :)

Running the looper into a stereo delay to the mixer

Also running the source sound into a reverb into a vca modulated by an EG.

Patch 2



#2 showcasing a simple patch using the polysynth for chords. To do this you will need to use a clock/trigger and send that to the trigger train - which outputs 4 trigger signals with a delay parameter.

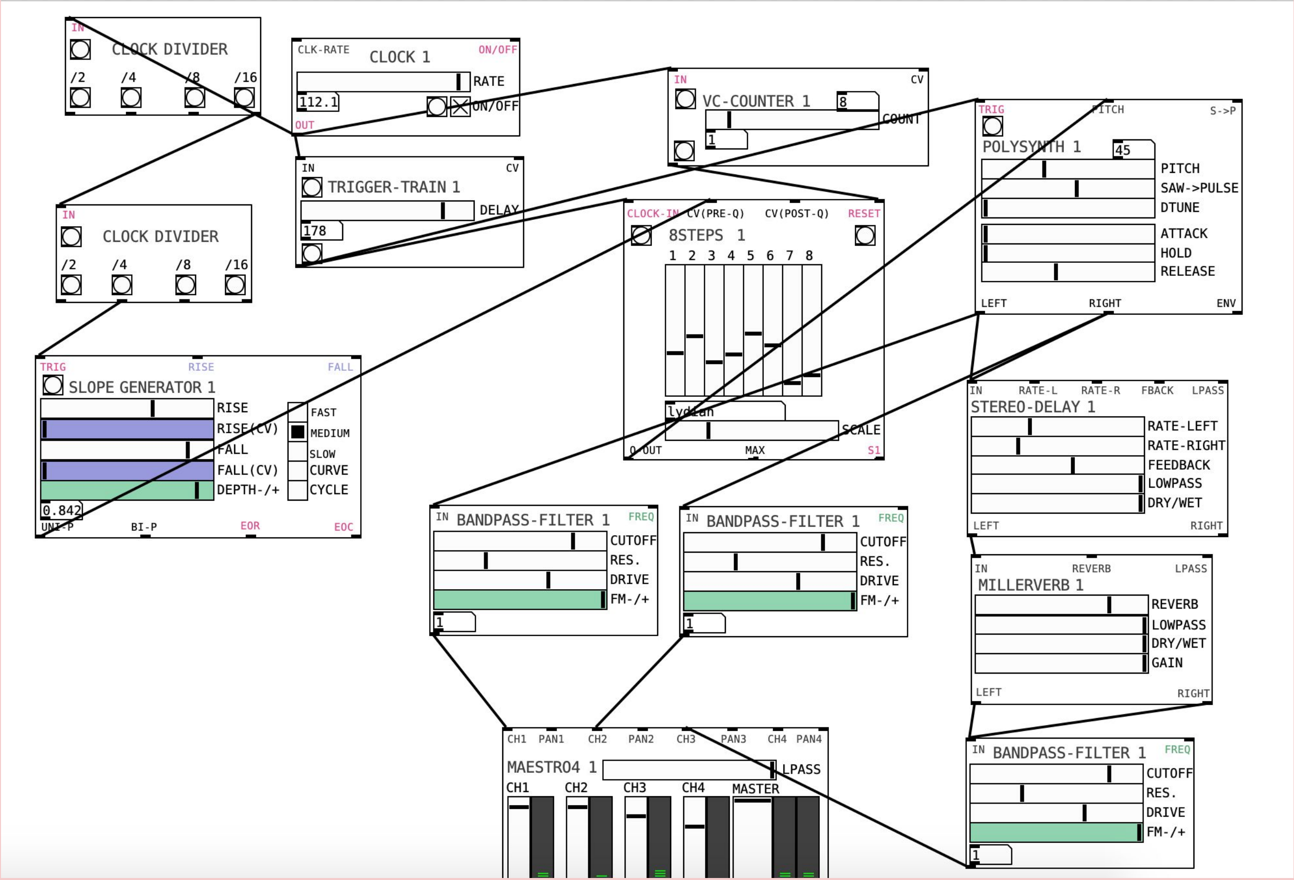
The polysynth's built in envelope is CVing one VCA while an LFO is CVing another as well as panning one of the channels on the mixer.

Patch 3

Here is a slightly more complex version of of a polysynth patch. We are using multiple clocks (dividers/vc counters) and a solid little effects chain.

Channel one and two have l & r out of the polysynth running through bandpass filters and we are also sending l&r through a delay/reverb and another bandpass filter.

We are CVing our pre-quantized sequencer input with our envelope to get some new melodic structures.

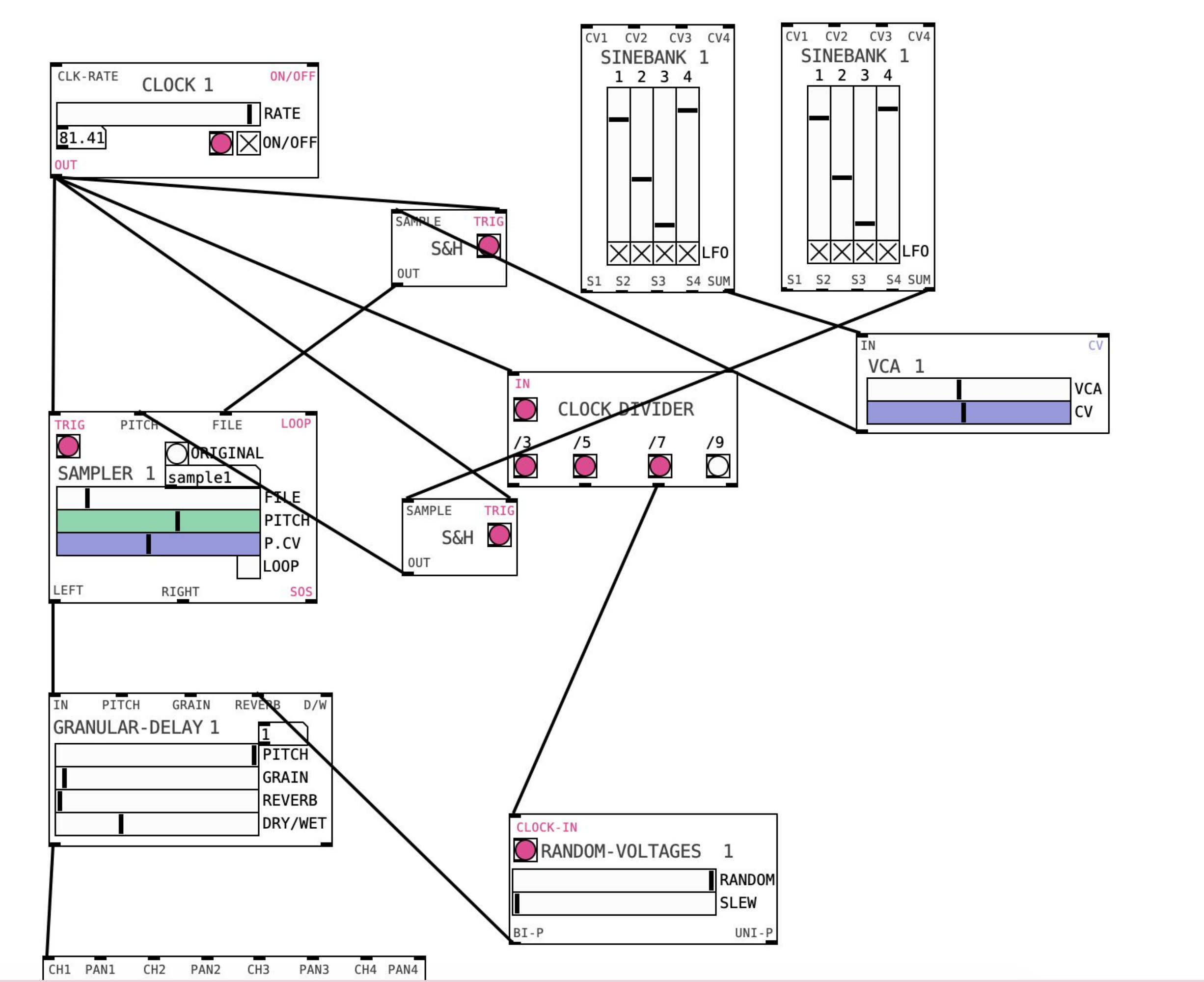


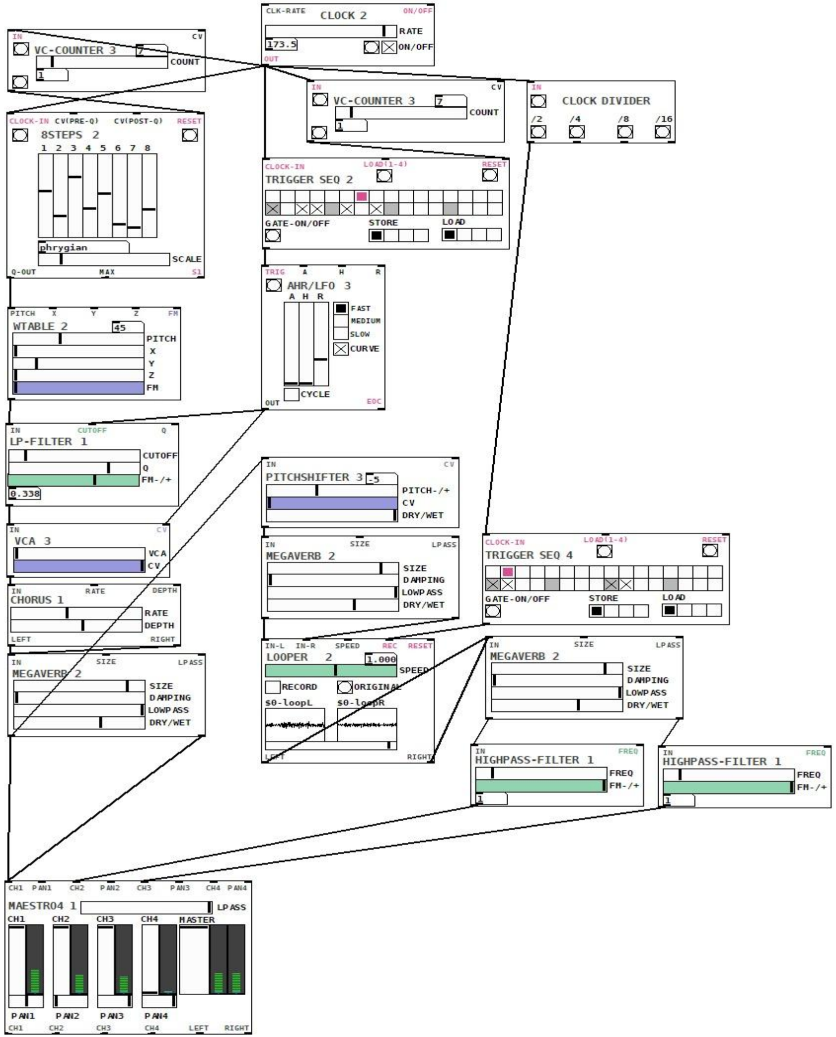
Patch 4

Here we are using an LFO as a signal source for the S&H. A sample & hold takes a static signal every time it is triggered- like a snapshot of CV.

Attenuate Signals (in this case the LFO) with VCA. This is another great way to use VCAs in your patch.

Lots of little control voltage tricks here - including scrolling through our sample files in a rhythmic manner..





Patch 5

Here our patch starts with a wavetable oscillator - a collection single cycle of a waveform samples that you can scan through.

We are building a classic subtractive synth voice with a VCO > VCF > VCA modulating both the filter and amplifier with an EG.

We expand on this simple idea by adding a looper that continually records in short snippets repeating our wavetable voice going through multiple effects.

You will also find a clock that is being divided, providing triggers for all over our patch.

Here's a tip: use a sinebank in LFO mode to change the sound of your wavetable - send into your X Y Z CV inputs.