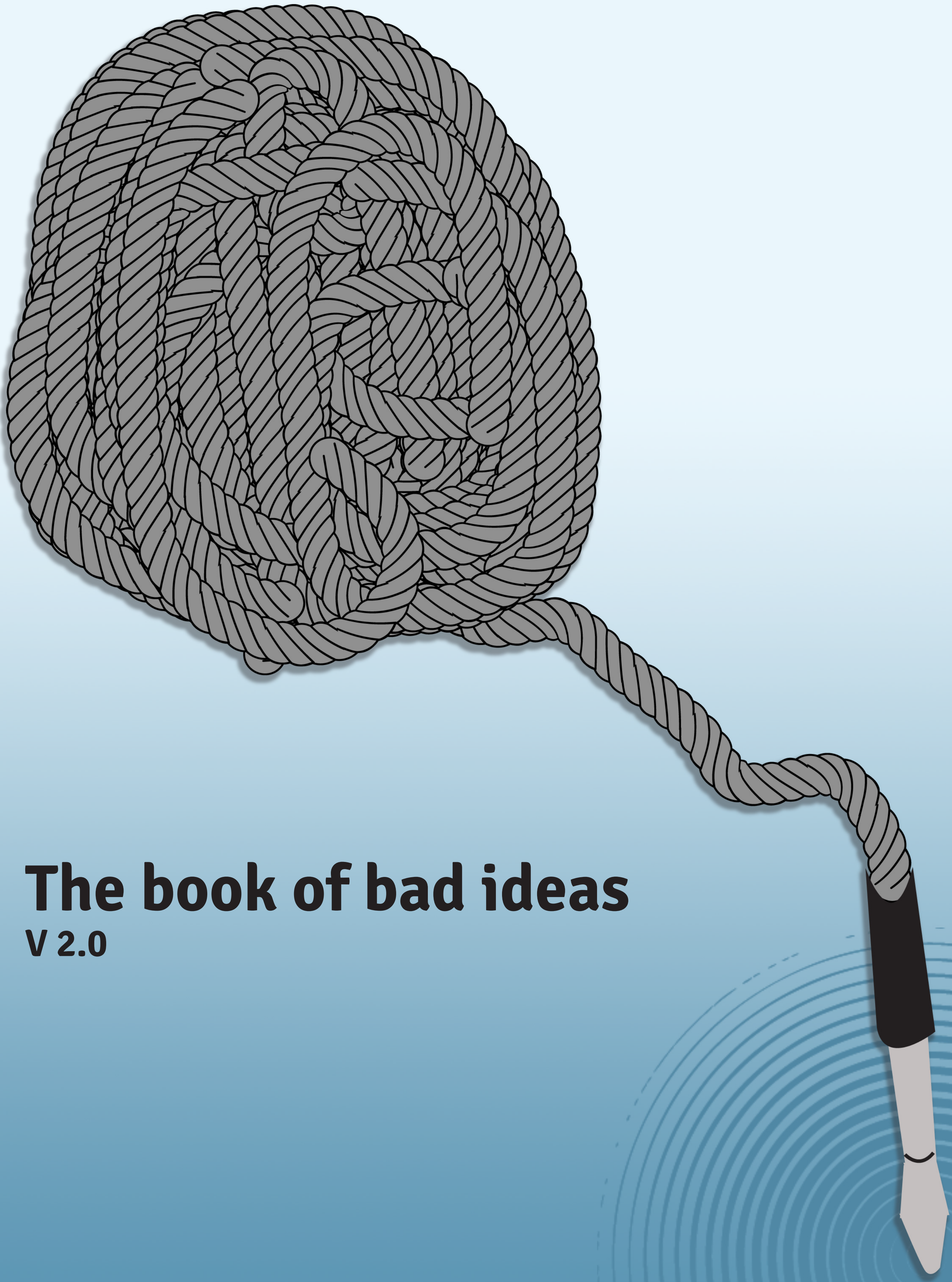

The book of bad ideas

V 2.0

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1.0
Tools & Techniques

1.1
Velocity

The simplest though not perfect patch would be to send the velocity CV to the filter. You can still send the envelope to the filter, but you might need a mixer if you always send key info for filter tracking and you've run out of filter ins.

The most standard patch would be to take an unused VCA and put it before your main VCA, send the velocity cv to it and your amplitude will reflect the velocity. You could also use a dedicated VCA to velocity control any patchable amplitude somewhere else in your patch.

VC envelope generators are game. Usually inverting the velocity CV with an inverter is effective for say a sharp attack on high velocity notes and a slow attacks on low velocity.

1.2
Panning

The essentials to do stereo panning? LFO with inverted wave and 2 vca's?

I suppose I could do this with 2 LPG's and an LFO with Invert also. that can be fun with a vc adsr which can loop. changing the vcas for lpgs can get some very very nice effects. being able to contour your voltage slopes (VC of attack and decay) to both VCAs might be really useful for shaping the character of the pan. triangle DC outs would give you 10v when panned hard on each side and about 5v for both vcas when the image is centered, so the hard pan positions would be noticeably higher in amplitude. having a log countour would give more of an equal power character to the panning.

You have to get everything scaled correctly for the VCAs you have. The first step is to get one VCA working properly so then you can repeat it inverted and fine tune.

VCA's ignore negative voltage inputs when set at zero, so if you don't want half the wavecycle ignored then you have to offset the wave up in voltage so it's whole cycle can modulate. Like if your VCA is half open to begin with and then it gets a negative part of a wave you'll hear a change. At the negative max if you still hear sound then the VCA offset would need adjusting. If it closes before the lowest point is reached then your offset needs adjusting or the amplitude of the LFO needs lowering

I guess the tricky scaling part is a reason why you have modules dedicated to VC panning. You have to get your maximum + voltage to fully open the VCA and minimum - to fully close it. If it's not doing that then your pan won't fully work. You might need a different VCA or deal with amplifying the LFO if it's range is not enough. If your LFO is too hot then it's easy, simply attenuate it.

Once you get your first LFO to open and close the VCA then inverting it (or using an already inverted wave) with everything else duplicated similarly will get you where you want to be.

1.3
Just intonation

tune two VCOs (I used two Doepfer A-110s) to unison in a low register, then hard-sync VCO2 by VCO1. Apply some fixed CV source to the 1V/Oct in of VCO2 (the synced one) and sweep from 0 to 5 V; you will hear VCO2 sweeping through the harmonics/overtones.

I used six steps in a sequencer (Doepfer MAQ16/3 in my case) to fix voltages giving the fifth to tenth harmonics of VCO2. That gives you a pentatonic scale with intervals of ratios 6/5, 7/6, 8/7, 9/8, 10/9, which first sounds strange to the ear used to equal-temperament.

Now VCO1 can function as a drone; I waveshaped (Doepfer A-116) and filtered (Doepfer A-120) it, several parameters controlled by a Malekko/Wiard JAG fed with two slow LFOs. The pentatonic scale of VCO2's triangle out I could "play" by turning the MAQ16/3 to single-step mode and manually dialling through the first six steps, amplitude envelope triggered from a keyboard. A little improvisation in this setup makes up the first half of the track below; I added some phasing by a EHX Small Stone and reverb from a Boss RE-20.
The second half uses the same setup in a more rhythmic manner: VCO1's drone is fed to a low pass filter with cutoff frequencey modulated by a random S/H. The sequencer chooses one of the six harmonics randomly; VCO2's square out runs through a Borg filter. The rhythm is realised with a clock sequencer. Additional delay from the RE-20.

1.4
Offset

Useful for changing a bipolar signal into a unipolar one.

For example changing the centre frequency of an LFO so if oscillates between 0v and 10v instead of +5v and -5v, without changing the shape of the waveform.

A classic east coast example is to change the pitch of several oscillators at once. Offset is your oscillator controller.

Offset can be used to change the response of DC-coupled signal processors. Waveshaping and

distortion can be made asymmetrical and balanced modulators can be unbalanced.
i use it for my filters (self osc sine) and osc to drop them down to lfo rate

You can control a parameter on a group of modules with a single knob, for example:
- Change the A,D,S or R on several VC env generators at once in a polyphonic patch.
- Change the cutoff frequency on a set of stereo filters.

Transpose a sequence by adding it to the sequence cv using a mixer

By summing a sufficient bias voltage (just a constant voltage which "biases" the CV signal towards the positive) to the LFO's voltage, that voltage is lifted into positive territory, and now the VCA will respond to the full range of that voltage, and the resulting modulation will be smooth.

1.5
Inverted audio

Filter Change: Mix original out with filtered out and invert one to turn a lowpass into a high-pass or vice versa. Half-volume-inversion will produce a notch filter. Also, if you have a filter with multiple pole outputs, combine two different pole outputs, invert one, and you get various bandpasses.

You could try mixing different wave outputs of an oscillator. That might not be terribly interesting on its own but sweeping/crossfading between inverted and non-inverted might be. Ditto for LFO to get different waveshapes.

A basic example - two detuned sawtooth oscs sound very different when one of them is inverted. I really like the sound of three sawtooth oscs, with one inverted relative to the others.

It's also very useful to be able to invert one filter relative to another when these filters are doing different things and then mix them together. Adds all sorts of additional sonic possibilities.

1.6
EOR EOC

Something I’ve had fun with is using an end of cycle to trigger a random voltage generator, the output of which is then used to control the rate of the envelope that you're taking the end of cycle from. This way you get a different random envelope length every time you trigger the envelope.

Have it trigger another envelope that controls the cutoff of a filter?

I love mid-function triggers like that on extremely long envelopes/functions. My favorite thing is to have a really looooong ADSR/AD envelope going on an overall filter or VCA, then use the EOR/EOC/other mid-section triggers to modulate filters and other modifiers within that patch. It's great for morphing through sounds and creating movement.

1.7
Delayed modulation

LFO out > VCA > OSC FM in
AR or ADSR with slow attack and full sustain > VCA control in
you could also mult the AR or ADSR to LFO FM if you want the speed to ramp up as well as amplitude.
Use separate env's if needed for more complexity.

1.8
Modular Restyler

1st - you'l need some sort of external processor or envelope follower.. my R35 has got amp level control and slew for getting the pumping and rhythm right. the slew changes the voltage like on the sherman from straight On/Off to a more curved response. i pre-amped the signal so i could get the levels just right.

2nd - you'l need filters.. a band pass before the R35 means i can "home in" rhythmically on elements of my drum loop to process, kick, snare, hat etc.

In addition to this i had a low pass, hi pass and comb filter.. you can use what you want, but youl need a VCA for each for AMing from the R35 or equivalent. (which turns out to sound very cool by the way)

3rd - I put the filtered signal through an RS50 trigger generator, and then through a clock divider to get the rhythm of the loop (again defined by the bandpass filter).. this feeds a sequencer which you can send to anything in the patch (i had it patched to the comb filter, for grainy rhythmical effects).

4th - I sent the level voltage from my R35 to the VCAs for rhythmical pumping of the filter volumes.

5th - the restyler is all very hands on.. By feeding all the VCAs to a signal mixer, i was able to fade in each filter band dynamically with the music. Altering the bandpass changes the emphasis of the whole patch. (and the resulting speed of the triggers) Altering the cuttoff of the other filters is obvious, but in combination with the signal mixer and AM to the VCAs provides a powerful effect.

1.9
Sequencing

patch a complex LFO into sample/hold? (By complex, I mean not your standard tri/square/etc. Lots of wiggle and discontinuities.)

Clock a sequencer or whatever with a LFO, feed the sequencer or whatever's output back into the rate input of said LFO

Example: Sample exactly 8 times per LFO cycle, and you've got an 8-step repeating CV sequence.

I use a combination of envelators and S&H to generate CV's for pitch control. It depends what you put into the S&H; I often route the sound sources through a ring modulator before going to the S&H. With three envelators modulating each other, you can get some very interesting patterns of triggers, and of course the envelopes themselves come in handy as well.

If your LFO has a reset or sync you can hit that every 8 steps(or x steps) I use the third bank of my e350 at LFO rate for this a lot! Changing the x, y, and/or Speed (tuning) works great. I use a quantizer rather then S&H. Clock it and divide that clock down to where you want to restart the pattern. I actually use a uStep for triggering the Quantizer and envelope! And also, also attenuate/offset the E350 before S&H to get it a tighter range.

A cool way to get lots of complex patterns that always repeat exactly, like a sequencer, is to connect two (or more) LFOs to a mixer, then send the mixer to a S/H in, then send the S/H to a quantizer to derive exact notes. Use a clock divider to retrigger both LFOs after every 8 S/H steps. Now just mess around with the various LFO rates and the mixer levels. You'll get all sorts of complex patterns, but each pattern will repeat exactly every eight notes until you change the LFO frequency & mixer level settings. add a noise source (or two) to the lfo's and you'll have a huge range of variations. the trig for the s&h is the rhythmic part of it. if you want tonal and can trigger the quantizer (like with the doepfer) you could skip the s&h as a seperate module.

1.10
Slew

With an LFO, you can get new CV waveforms.

And yes you can tame CV edges if you go thru one, let's say you have a square lfo inot a filter that clicks you can smooth it out

Well, I sometimes use a Slew to lengthen the attack and/or decay of an ADSR, to give me semi-correlated envelopes. Means I take the ADSR to some CV in as well as to the Slew in, and then the Slew out to some other CV in.

A slew limiter/generator/whatever is basically a simple and crude lowpass filter.

if you want AHD:
Gate Signal > Slew > some CV input

The slew will give a slope to the rise and fall, depending on how you have it set. The output from the Slew will 'hold' for the duration of the Gate.
If you want an output that is less than the max Gate voltage level of your system, run the Gate thru a mixer or attenuator of some sort before going to the Slew; that will set the max rise level.

Note, though, that if you have slew generators, you can get correlated adsr outputs from them, with laggy attacks, delays, etc. Sometimes it's nice to have similar but not quite identical envelopes on filters and amps, or fx and filters, or {permute controllable modules set}.

Slew instead of the envelope. Gate envelope EOC with your keyboard gate and feed the result (logical AND) to the trigger input. Then patch envelope to to slew. Send output to your VCO for gated, faded modulation.

Other usages...

- You can create portamento; i.e. a continuous rather than discrete change in pitch, by sending keyboard or step sequencer CV out to slew before the 1V/Oct input of the VCO.

- Do the same thing as above (particularly with the sequencer), except use the output as a complex envelope for stuff.

- You can create something of a gate delay by sending in a gate signal and slewing on the rise. (Obviously this will only work for modules that look for a comparator on the input, not ones that look for a fast-rising edge.)

- If you have a long slew time, and can jump the input voltage from, say, 0 to 5V, you get a continuously rising slope... feed that to a filter's cutoff (or whatever) so the filter slowly opens while your hands are free to play with other knobs.

Slew on audio rate (as a waveshaper) but depending on the rise/fall times you will get a lowpass filter as a result.

Noise --> Slew Limiter with separate controls for up/down. Increase "up" slew, leave "down" slew at 0. You get grainy noise.

1.11
Delayed vibrato

LFO -> VCA -> VCO FM
An envelope slowly brings up the VCA.

I like adding a little low-freq random mod to the lfo rate to add a little "life" to the vibrato. Some AM on the depth might also help...

BTW, try using an LPG with slow vactrols instead of a VCA/ADSR. You can't control the time constants, but it could be handy when you have nothing else available.

Using two VCOs tuned together, input the EG to the FM of one VCO. Adjust the FM amount for a tiny bit of EG mod so the two VCOs beat against each other when the EG is moving. The beating VCOs form the "LFO effect". Depth of modulation sets the rate.

1.12
Feedback

The basic premise is that the output of a module somehow influences the behavior of that same module. Here's a very basic example:

Take one output of an oscillator, and listen to that as your main output.

Now take a second output of the same oscillator and run it through a mixer, vca, or some other kind of attenuator. Take the output of that, and feed it into a CV input on your oscillator.

Now the output of your oscillator is modulating its own cv input. You can vary the amount of feedback with the attenuator. See how the sound changes as you turn up the cv modulation?

The same technique can be used on other modules such as filters, or even complex signal chains. Try splitting your signal into different frequency bands and only feeding back some of it. And so on...

Take one channel of your daw ie. ableton and feed
the output of the cannel >
through a bunch of module ie. filters >
and back into the input of the channel.
then insert a time delay into the channel >
this gives you a digital delay with an analog insert.

then play one tone and keep it up in the air by adjusting the gain in the chain. then you can listen to how the analog devices change the signal progressively until it is totally destroyed.

I would take that second VCO output and ringmod/filter/wavefold/distort/waveshape/etc..
it before returning it into the CV input of the VCO.

Even just gaining the feedback makes for a wider tonal palate.

Another type of feedback loop is cross-modulation, in which you split the output of one oscillator and monitor one side of the split while sending the other into the FM input on a second oscillator. Then take the output of that oscillator and feed it into the FM input on the first oscillator.

You can easily generate very complex, unpredictable, and unstable timbres with only slight variations in the parameters of either or both oscillators.

a few modules which are very helpful or necessary in creating feedback loops:

multiple (as you suggest)
mixer (for combining before and after feedback)
inverter (for inverting voltages)

try multing the output of a delay, one leg goes out, the other goes back into the delay's input via a mixer, but before the mixer feed it something else such as a phaser, filter, ring mod or frequency shifter for evolving delays.

audio feedbacking a filter (with or without inversion, depend son the filter type), and controlling the level of feedback with a vca can give you voltage controlled resonance on filters which don't already have it. again, running the feedback patch through another processor (like a wave folder maybe?) can achieve some very interesting results.

don't forget CV feedback too. there's a great self modulating FM patch in the arp2600 manual where inverted CV's get sent to the carrier pitch and the modulator pitch, results in nice bird noises etc.

1.13
Compression

Babaluma has a really good fb/fw compression patch that is GREAT on snare and/or hihat+snare loops. If you have a borg and a maths or vcs it works great. Basic idea:
Mult the snare out to an envelope follower and a borg (post all filters and vcas). Turn borg to hi pass or band pass. Invert the EF out. Modulate the borg with this signal. This is feedforward i believe. For feedback, mult the out of the borg to the envelope follower instead of the pre borg signal.

The slower the vactrols, the more optocompy it will sound. Its good to use a full featured envelope with EF input like maths or vcs, cus then you can dial in exact attack and decay setting, and also create non linear slopes which are wonderful in this application.

For a traditional feed forward VCA comp:

You need two signals, one is the audio you want to compress, and the other one either a duplicate achieved by a splitter or multiple, or our side chain signal. The audio you want to compress goes straight into the input of the VCA. The other signal is used to create a CV for the VCA:

Run it through an envelope follower. Add a negative offset (threshold). Kill the negative part of the signal so that it only consist of positive voltage. Run it through a slew limiter with separate attack and release. Invert the signal. Attenuate (Ratio). Add +5V or what ever you need to fully open up the VCA. Use signal to control VCA.

1.14
Gates

Inverting a gate can be useful for ducking effects to a VCA. Like during a drone piece when you need a fast transient or different sound. Inverted gates and VCA's can be another rhythmic variation on a sequenced pattern to break up the monotony. A gate in that context could come from a clock with a long triplet or some different bar count.

Most of the time in this form the gate does better if the edges are dampened with a vactrol or even envelope/lpf/slew filtered. Of course then it's really not "just" a gate then either.

My favorite use of gates lies in where they are generated from. A random gates module, S&H or even FM modulated VCO pair's in low frequency square wave mode can cause near-random fun.

Start, stop, reset and shift sequencers. Pulse Vactrol devices for unique percussive sounds. Attenuate and send to CV inputs to save the use of EG's when I jusrt want it to go on and off. Use them for digital logic.

Sometimes they are nice for pinging cutoff or q on filters.

Sent random gates to the control voltage inputs of envelope generators. Attenuate to taste so the articulation of the envelope slowly changes throughout a sequence or whatever.

Instead of sending a clock/gate signal to an envelope, I like sending it to a sample & hold which is being fed noise, and the output of the S&H then goes to the envelope gate input. This results in synchronized but random triggering when the sampled noise surpasses the envelope gate threshold. Using an extra attenuator/amplifier will also allow control of the probability of triggering.

1.15
Drones

iIt's all about a nice slow attack, long decay and long release, on both the VCA and the filter. Tune three or more oscillators at different intervals, add a little modulation to the pitch, filter cutoff ,resonance, pulse width or whatever else you choose. Use envelopes or long LFO's to bring in other elements over time, add a touch of noise and finally season with some nice chorus/phaser/reverb.

the key is to have 3 or 4 LFO's all running at unnoticably different rates, all very slow.

attenuate the cv twice.. so you can make small differences on the second attenuator but over the full range of the knob

an offset generator through a really long slew limiter, and tweak that instead of the actual parameter knob itself, thus stretching/smoothing tweaks out into a much longer timeframe

audio out split in three, one uneffected, one to waveshaper/ringmod, one to bandpass filter/vca. Three different channels with a bit of delay and a big but short reverb (spring emulation) from logic... the rest is just slow hands on knobs.

mix four LFOs with a cv-controllable mixer

but also multiplying and jumbling those LFOs to control their mix levels, yielding one single, unpredictable LFO.

i find that the best drones are borne of phase cancelations and slow evolving frequency modulations and having those modulated waveforms modulate the frequency cutoff... resonance always adds movement.,.. once the core is taken care off, further frequency shifting, ring modulation, reverb and delay and feedback loops

you can flip the polarity of the feedback relative to the input although its more interesting and subtle if you take the feedback out to a VCA and play with the relative levels between the original and phase altered signal

1.16
Poor Man's AM

Take 2 audio sources, plug one into a VCA input and another in the the CV jack. Sounds pretty convincing, though the timber changes significantly when ins are switched.

1.17
VCA Ring Mod

a vca is usually a two quadrant multiplier, and a ring modulator is usually a four quadrant multiplier.

what this means in practice is that with audio rate RM and AM, the RM will output only the sum and difference frequencies, whereas the AM will output the sum, difference AND the original sounds frequencies.

if you wanted a static sound, i guess you could place a tight notch filter after the output of an audio rate AM patch to get rid of the original frequencies.

thinking about it the other way round (and a lot more common), is mixing the original signal back in with a RM signal. this gives you more control of the balance.

1.18
Risset

I would not tune each oscillator an octave apart. try to tune them all the same. Start off trying to have the LFO at a very slow rate, like one cycle every 10 seconds or so.

The thing you are trying to do here is to create an endlessly rising tone. This is an illusion due to the oscillators and amplifiers starting at different times in a manner not unlike a musical "round".

What you will hear is the first oscillator will be silent at the lowest point (if you are making a rising tone), and the first amplifier in the A-135 mixer will increase as the oscillator rises until it reaches midway. At that point the amplifier will decrease in level until the oscillator pitch reaches the highest point.

Now since the A-143-9 is sending out a sine wave at 90 degree quadrature, the next oscillator (#2) and amplifier in the mixer will do the same as the first set of osc/amps. This will be like a delay of the same sound.

Again, the third and fourth oscillator and amplifier sets will follow suit. This will repeat endlessly with the first set taking up after the fourth set, etc..

So as you can see, all four oscillators and amplifiers in the A-135 mixer should be identical in settings. I think this is where you may have run off track.

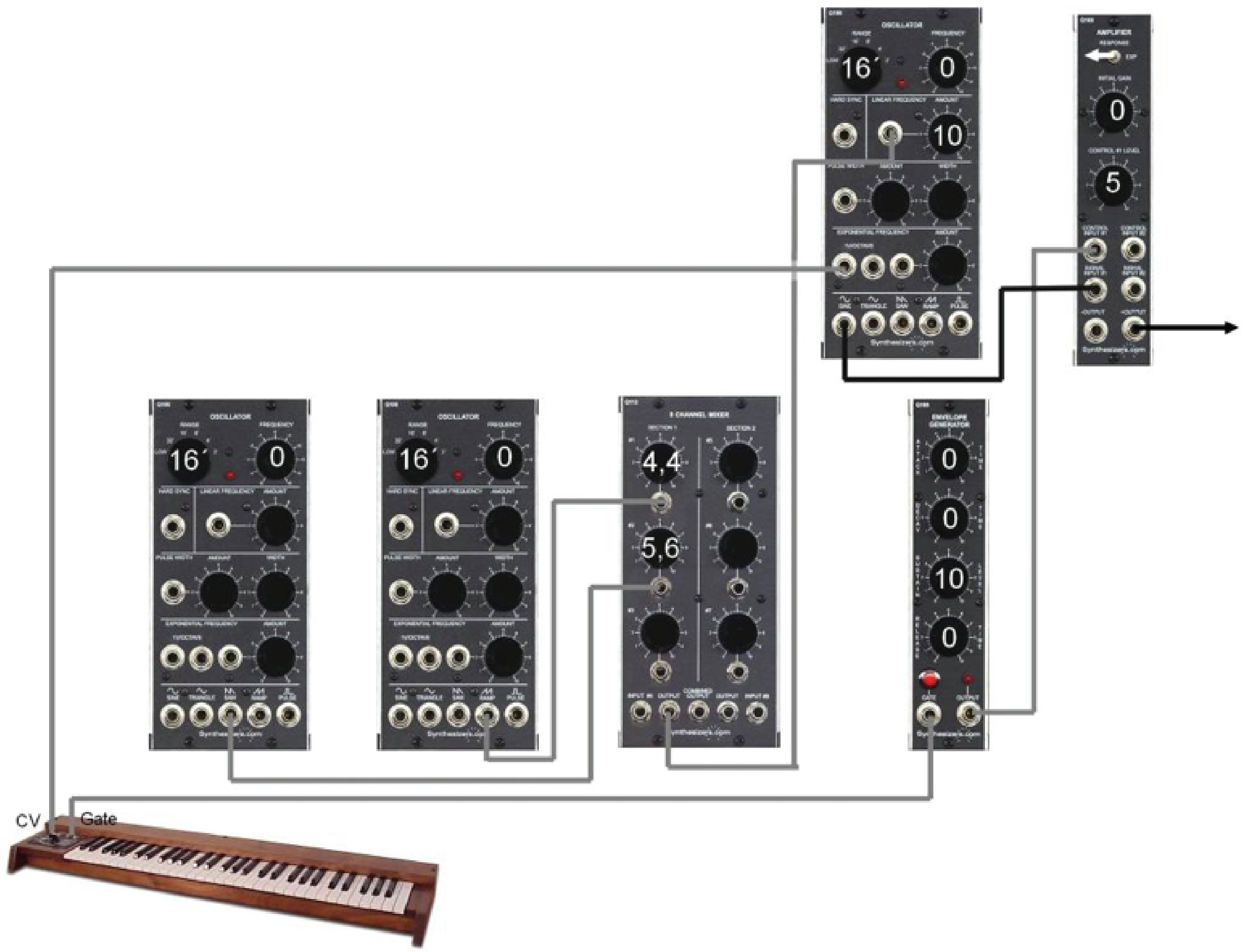
This is really not meant to be a sound that is pitch-centric, but if the speed of the LFO is fast enough then the illusion is faster and you could actually change the pitch of all four oscillators in tandem and the overall pitch could sound like it is changing.

You need not use a sequencer like the SH-01 to control the pitch of the oscillators.

Use the LFO outputs to the oscillators.
LFO output #1 to oscillator #1 and amplifier (mixer cv) #1
LFO output #2 to oscillator #2 and amplifier (mixer cv) #2

and the same for 3 & 4.

You are using the quadrature lfo to generate a set of offset voltages to the oscillators and amplifiers (voltage controlled mixer channels).



1.19
Time machine

an audio delay
a side chain (could be a compressor patch, a filter, whatever really).
a multiple

feed your audio into a delay and a side chain using a mult.
monitor your audio from the delay's output.

now the fun begins, as your side chain will be reacting to the original signal, but you will be hearing the delayed signal. if you patch the output of the side chain (let's say it's an envelope follower/vca/compressor patch) to the delayed audio, it will effectively be reacting to the audio before it has happened...

1.20
New Waves

Mix two saw waves of the same pitch making sure one is out of phase (inverted) let the two wave run out of sync a tiny bit and you will get a lurverly square PWM wave, the detuning sets the PWM speed.

classic fat moog bass sound - saw fundamental, another saw a fifth above that mixed so it's barely audible, and a square wave an octave below the fundamental. Mixed with a STG mixer so you get some nice overdrive, and filtered with your choice of nice filter

A saw and inverted square output, ideally same osc but must be in perfect sync, so rising edge of saw is synced to falling edge of square. Mix the two and you effectively get a saw an octave up. Modulate the mix level of the square and you modulate the mix between a saw at one octave and one an octave above.

1.21
Synced Random Gates

1 VCA'd random signal into the input, and your clock "gate" into the CV input.

another good one is a comparator, great for mixed analogue & digital signals as well, but can be abit fiddly to tune in - great for those massive changes from a single knob turn moments

AND is like multiplication, which is like VCA. Also like MIN.

OR is like addition, which is like mixer. Also like MAX.

The S&H doesn't care if it's sampling a CV, gate or 'silence'. Send your master clock to the CLK-in and your random gate to the sample in.

all you need is a vc switch: steady clock (narrow puls/short trigger)->vc input. weirdo puls-> switch input. switch output gives random pulses but only 'on the beat'

1.22
S&H

feed the gate signal (or the clock) from your keyboard/sequencer or cv-interface into trigger in, an lfo into sample in and take the a148 output to any modulation input in your patch for note-synced variations.

feed a slow lfo into sample in and a fast lfo into trigger in to get staircase patterns at the a148 output.

Feed a triangle or ramp LFO into it and use another faster LFO to trigger. Feed the output into a quantizer. Poor mans arpeggiation.

Send the output of random into the S&H, trigger S&H with a divided version of the clock running the random - you have the noise ring changing values at a quicker rate and you can have longer notes derived from the notes being output by the Noisering.

I like mixing the CV output of a sequencer or keyboard CV with a very narrow voltage range from the S&H, using the sequencer or keyboard gate (via a MIDI convertor) for the clock, for slight tuning variances in a sequence or key depresses, respectively. I think it gives a patch a more organic feel.

You can also think of a S&H as a one bit memory, either to sync transpositions, quantize random gates to your master clock or, in conjunction with a logic/ digital inverter, to patch a flipflop e.g. hit a key and a gate goes high and stays high until you turn it off with the next key command.

take your CV output from your sequencer, prsumably running on some clock, put the CV into the S&H. using my internal clock source on the S&H (or an external source below audio rate) you can get very interesting results at the output. added slew is great as well.
http://www.youtube.com/watch?v=ziFv0OPegJg

Sample and hold will sample its input whenever it gets a trigger (rising edge of a gate) and hold that value at its input until the next trigger. so sending it noise and triggers gets you random values at whatever rate/tempo your trigger is at. personally I like some control so I send LFOs, or better yet waveshaped LFOs mixed with a bit of random

one of my favorite patches is using an ADSR as s&h input, clocked from sequencer, out to a slew limiter, and into filter cutoff or similar waveshaping device. Instant rhythmic goodness.

1.23
Notch Filters

Notch can be used for feedback loops. Try FM'ing the filter with it's own notch output (through an attenuator if the filter doesn't have an FM input control). Similarly try running the notch out back into an audio input of the same filter, again with an attenuator or the filter's input level control if it has one. If you have 2 filters available, try the above only cross-patching instead.

basic usage for a notch filter is a sort of faux phasing effect with the filter being modulated by an lfo/etc. as others have mentioned, the combination of multiple bands or notches and even mixing the notch with other responses from the same filter leads to interesting results and sweeps.

1.24
Tail chasing patch

The following patch is taken from this book and shows how to create very complex permanently changing sound structures by means of the A-149-1 in combination with the voltage controlled LFO A-147 and some additional standard modules (VCO, VCF, VCA, ADSR):

A high magnitude voltage at the N+1 output of the A-149-1 causes a high VCO pitch and simultaneously sets the value of N higher so that the next pitch is taken from a greater range of possibilities. If the N+1 output is low the VCO pitch will be low too and sets the value of N so that the next pitch will have a more restricted range of possibilities. Simultaneously the 2n output controls the frequency of the filter and the A-149 /1 RCV System A - 100 doepfer 6 tempo of the VCLFO A-147. Thus as the range of pitch selection increases the number of possible spectral ranges becomes exponentially (or geometrically) greater.

As the tempo of the VCLFO is controlled by the 2n output too, bright sounds are accompanied by longer events, longer events are accompanied by greater range pitch range possibilities and the number of of range probablities for pitch selection is correlated exponentially. This tail-chasing configuration may last a few hours (to obtain Allen Strange's original patch a voltage inverter A-175 has to be inserted between the 2 n output and the control input of the VCLFO as the CV input of A-147 controls the tempo rather than the period).

More examples with random voltage sources can be found in Allen Strange's book from page 80 (e.g. the "Dream machine" on page 85). Some additional ideas: Use the RND Clock output of an A-117 Digital Noise Generator as clock input for the A-149-1 to increase the randomness of events. Use the Quantizer module A-156 to obtain more restricted pitch voltages (e.g. only notes from major/ minor scale/chords). Combine the A-149-1 with a A-155 sequencer (common clock) to obtain random envelopes (A- 142), timbre (filters), loudness (VCA) or stereo position (VC panning A-134), frequency shifting (A-126)

1.25
Clicks

run a slow (subsonic) LFO square wave through a high pass filter. Each time the LFO transitions from low to high or high to low you get a click. If you don't have a LFO with a square wave, a ramp/sawtooth will also work. That will give one click per cycle instead of two.

A quick AD envelope through a high pass filter will also work.

If you low pass the square wave and then mix that with an inverted copy of the original it will subtract the original and leave you with the spikes from the edges. (basically a brute force high pass)

Run an LFO with super narrow pulse width at low speed and run an LFO or envelope gen to its frequency in to change its speed.

noise to S/H (clocked by a sub-audio oscillator) then HPF, adjust cutoff and resonance to get different click flavor, also adjust clock to control density...if you use a VCA with an AD envelope on it plus get the clock faster you get all kind of noisy and grainy snares, hihats and other percussion sounds

1.26
Flanger

a basic flanger is a delay with feedback and a really short delay time,
something between 1 and 50 milliseconds.
i use bbd-delays quite often for flangings.
you can use the lfo on the delay time to animate the flanging.

To accomplish flanging with modules you would need a triangle wave (or something more interesting) LFO and apply it very subtly to the rate of a delay.

stereo flanging by feeding the delay time inputs with a slow LFO 180 degrees out of phase and panning the outputs hard left and right.

1.27
Clock Multiplication

you might fake it by using a resettable lfo (square) or a cycling envelope with some kind of trigger output (end of cycle/attack) into a VC switch.

run a clock signal through the Audio Divider module and adjust the divisions just right(ie two at mid level and the rest off) you can get some good rhythms

1.28
FM Tips and Tricks

Experiment A
Using Sync in tandem with FM is an easy way to make it sound like the two VCOs are properly in tune, and it can be used to achieve a nice alternative to standard sync sounds.

Experiment B
Cross feedback FM is something I use a lot too. It can add a nice little something special for harmonic FM patches (although it does seem to make things a little more unstable in terms of pitch/tuning) and it can lead to really expansive sounding drones and in-harmonic sounds. Simply take the output (or an output) of your FM'd VCO and feed it back into your modulating VCO. Adjust amounts to taste. Sticking with Linear FM will result in the widest tuning range, but don't ignore exponential FM because it has a totally unique and awesome sound.

Experiment C
FM isn't just for VCOs either. Filter FM, particularly at a harmonic of whatever you're filtering, sounds wicked - and each filter is going to sound different.

Add a VCA+EG/LFO in line with any of your modulation signals in the above and you instantly have an awesome evolving sound. I would definitely suggest patching this towards the end though because you may find that when the modulation signal is full attenuated, the pitch may be slightly shifted compared to the unattenuated modulation signal (doesn't apply to the HD if you're using it's internal modulation VCA).

Experiment D
using 2 vco for FM is great so try chaining 3 or more. Think of a typical digital FM synth and it's multiple operators. Things can get real messy quick but if your pitch scaling and control is in line you can get some very complex evolving sounds with just 3 vco and a couple vca and eg.

My personal favorite is to chain 3 vco serially FMing each other and then send the main vco back to first modulator so it makes a circle. then tap various other outs of the vco and sync them in various ways.

Experiment E
something I have been doing recently is FM a main vco with the ringmod output of 2 other vco or the main+secondary vco.

Experiment F
fixed FM indexes reduce the issue of pitch changes as you increase the FM amount. I do find though that if the 2 vco track the incoming pitch cv very differently as you go up in pitch the high end can get quite noisey as the harmonic relation between the vco strays further away. To illustrate lets assume we are using 2 model 15 vco as they have attenuators on their fm inputs. vco1 is full on with the pitch cv input and the vco2 is set only half way on the attenuator. at the low end they will track fairly close but as you increase the pitch cv the difference between the 2 vco will not probably stay in a harmonic relation to each other so the output gets much more harsh. If both attenuators were set the same this would not occur. It can be used in some instances but you may like the low register sound on the patch but once you start playing higher pitches it will sound very different.

Experiment G
Patch a tiny bit of white/pink/LF noise to an FM input of your osc to make it sound more natural.

Experiment H
Patch an unused (sin/tri/saw) output of your osc back into the linear FM input of the same osc to get a different waveform out of the other outputs

1.29
Attenuators Tips and Tricks

send in a sequence, and feed the other inputs with gates (from a160 clock divider in my case) synced by the same clock (divided by 6 with RCD here) ... then dial in the knobs on the attenuverter for some very fun sequence variations

A slight variation on this is something I do from time to time. Taking a sequence that used somewhere else (as in I don't want to change it) and extracting a rhythm out of it. I usually use the A-166 Logic module, or sometimes an envelope that has an EOA or EOC output. I mult the sequence and run it through the attenuator the attenuated output then goes through the AND input with a steady clock in the Logic module, or the trigger input on the envelope. The attenuator is used to tune the amount of voltage coming from the multed sequence relative to whatever the trigger threshold of the module is. That threshold tends to be tiny, so some fine tuning is necessary, but then what you get is higher voltage notes on the sequencer generate a trigger where as lower notes do not. It's a fun way to generate a clock pattern from some other voltage source.

uses a sawtooth LFO as a repeating 16th-note envelope on an LPG's cutoff. I'm running that positive while also ducking the cutoff with an inverted envelope (A-140) triggered by a uStep - nice rhythmic bassline goodness.

This patch is double-attenuverting fun... if I change the shape control on the LFO to make it a ramp and then 'add' it negatively to the LPG cutoff it has the same nice sharp attack which keeps the hits on the 16th notes. Instead of ducking out with the 140 though, I have to raise the cutoff up to accommodate both negative modulations and so the rests disappear and the rhythm gets subtly more dense.

1.30 VCA Tips and Tricks

As we said before, the main use most VCA's see is to gate notes with an envelope generator. Here are a few other ideas to add to your synth arsenal:

Crossfading and panning
Two VCA's can be connected such that one responds to a control voltage directly and the second VCA responds inversely. In one application of this, the VC panner, a common audio signal is fed into both VCA's and one output is sent to the right stereo channel, the other is sent to the left channel. Varying the control voltage moves the sound in stereo space. A crossfader takes two different sound sources and mixes them to a common output. Here, varying the control voltage changes the mix between the two sounds.

Tremolo
Tremolo is a periodic rapid wavering of the sound's intensity, similar to vibrato. If your VCA has more than one control input, patch a sinewave LFO with a frequency of around 7-15 hz, with a moderate amplitude into one input and your envelope into another. Or just use the LFO and a steady-state voltage. If your VCA has only one control input then you'll need a DC mixer to combine the control voltages. Using other waveforms such as square or sawtooth for the modulation will yield interestingly different effects.

Ring modulation
The VCA and the Ring Modulator are cousins in that a VCA is a two-quadrant signal multiplier and a Ring Modulator is a four-quadrant multiplier. What's that mean? It means that, for a VCA, the output is the instantaneous product of the input signal and the control signal, where the control signal goes to a minimum of zero volts. If you take the control voltage into negative territory it's treated the same as zero. The two quadrants are the top and bottom right quadrants of a Cartesian grid, where positive and negative values are allowed for the audio input but only positive values are allowed for the control input. A Ring Modulator, on the other hand, allows positive and negative values for both control and audio (they're usually called "carrier" and "signal"). When the control goes negative it inverts the audio signal (swaps positive for negative). Just like in math, multiply 1 by -1, get -1, multiply -1 by -1, get 1.

Here's some more math: A ring modulator's output frequency content is the sum and difference of the input frequencies. If you call one input 'A' and the other one 'B' then the output is A+B and A-B the "sum and difference" frequencies are present and the original frequencies are suppressed. A VCA's output will be A+B and A-B and A and B; the original frequencies will be present in the signal along with the sum and difference frequencies.

Ok, what's that mean for your sound? If you want to do ring modulator effects but don't have a ring modulator you can do similar effects with a VCA. Patch audio frequency signals into both audio and control inputs, try different waveforms and different frequencies.

Faking it

You can use other modules in place of the VCA to do the traditional job of gating audio, in case your VCA is busted, needs a rest, or is already in use. A VC Lowpass filter is a great candidate for doing this. Keep in mind that in order for this to work the VCF's initial cutoff frequency should be at its lowest, much lower than your audio signal. It's simplest and cleanest at low resonance levels on audio with a low harmonic content, and more interestingly organic or acoustically imitative with more harmonics.

If you have a VC slew limiter this can be used the same general way as the VC LPF. Keep in mind that a slew limiter is a DC-coupled lowpass filter with a fairly gentle roll-off curve. Set the slew limiter's rate (slew speed) at a minimum to attenuate an input sound, use voltage control to increase the slew rate to increase your sound's loudness.

Another VCA stand-in is the Ring Modulator. On a theoretically perfect Ring Modulator it shouldn't matter which of the inputs you use for audio and which for control, they would be interchangeable. However, many real-world designs optimize signal and carrier inputs, so you might want to experiment with which input is good for audio and which is for control. If the Ring Modulator doesn't have DC coupled inputs then long envelopes will be a problem but you may be able to manage short percussive ones. PAIA's old 4700 series modules didn't include a VCA, they used a Ring Modulator with DC inputs.

Transient Shapes

How flexible is your system? Envelopes can be messed with in different ways and combinations. The mainstay setup of an exponential VCA and linear envelope most accurately imitates a variety of acoustical instruments, but what if you're looking for something weird? Using a linear VCA and envelope yields somewhat "flat" sounds. An exponential VCA and envelope gives strangely sharp or spikey effects.

A linear VCA and log envelope can sound squashed or murky. Most envelope generators don't provide more than one kind of envelope slope, but if it happens to be a voltage controlled envelope generator, feeding the output back into the vc input with some kind of control in the middle (like a pot) will yield exponential envelopes, and, with an inverted signal, provide log envelopes. Experimenting with combinations of VCA types and envelope shapes offers many surprising possibilities.

Interpatch VCA's

VCA's can be used at intermediate stages in a patch to add complexity and flexibility to a sound. As I said earlier, DC coupled linear VCA's are particularly handy for modulating control voltages. Audio VCA's can be used to add brief transients, such as a xylophone's hammer noise or flute 'tonguing', mixed along with the main sound. If you're using multiple VCA's this way keep in mind you'll need a like number of envelopers to drive them and a mixer to combine the various sounds together.

Interpatch VCA's can also be handy in feedback loops, modulating, say, the resonance of a filter or phase shifter.

VC mixing

Those lucky enough to have several VCA's at their disposal can do voltage-controlled mixing to produce especially complex sounds before recording or to automate mix-downs using sequencers, MIDI, or other control sources. Serge offers both stereo and quad multichannel VC mixers, where the customer specifies the number of input channels.

1.31
Various Tips and Tricks

Unstable Saw

Linear-FM a saw with itself.

Self-generating

random CV source (Wogglebug/Noisering) clocked by a clock div (/64) controlling switches... this way you can get dramtic changes that happens a few minutes away.

1) Old Standby - noise to a S&H for random cv. Using this to an LFO to generate odd clocking is helpful too.

2) Soft and Hard Sync feedback. Sending the output of one osc into the hard or soft sync of a second and then returning the output of that one back into the first one. Adding other cv's to each of these and controlling the amount of sync feedback through a vca is useful for changing tonalities. Similar ideas can be done with ring modulators and wavefolders.

3) Gate delay's and random gates. These modules come in handy for breaking up patterns and making things change over longer time periods. I've built and bought some of the Yves Usson designs. I'm planning to build his time divisions modules too.

4) Long clock divisions for sequential stages. Having a way to divide a clock into up to 32 or even 64 divisions and sending these to sequencers to transpose pitches, LFO rates or change filter settings over a long period of time. I currently use a Moon Octal Divider but there are many to choose from. Odd divisions are helpful such as 7 or 13.

5) Long subtle ramps with an inverted (or ramp) from an EG, LFO or function generator applied to something drastic like a wavefolder, ring modulator, filter overdrive or digital delay. Long periods of minutes in which it builds and abruptly ends or changes.

6) Slew generators timed at odd intervals with things like the random gates or sequential switches. Sometimes sending these through a quantizer or S&H to give it glissando rather than portamento.

Even dividers are cool, but I have found odd divisions to be the most rewarding for Noodles.
Slow modulation sources are also important in this context. nrdvrgr mentions the uLFO. I don't own one, but I assume it goes slower than others. I have the Modcan Quad Digital, and that runs down to glacial speeds. Slow speeds are important if your patch is going to evolve. Something as simple as a 5 minute cycle on a VCF's cutoff can produce fairly dynamic results.

Oscillator Self-Modulation

Send an oscillator into its own FM input. Modulating amount of FM can create vibrato. Linear input can be DC or AC with slightly different behaviors. Using Linear FM will provide good pitch tracking while Expo will have a bigger effect. Send oscillator's pulse into itself with PWM to create vibrato + timbre changes.

Harmonically-locked Oscillators

Use a soft-sync-capable oscillator as a sync slave. Master and slave must be as in-tune as possible. Adjust sync and tuning of slave until it produces a clean waveform when playing. Send each into each others' syncs for further precision.

Crackly Noise

Send noise to slew with separate up/down control, slew only up or down portions, not both. Experiment by breaking this rule.

Pitch Snap

Gate the pitch voltage going into the 1v/oct of an oscillator. This creates a nice snappy attack/release.

Feedback / self-influenced clock

the basic concept is a click that is sent to a mult which triggers a sample/hold whose output is sent back into the rate of the click. the sample/hold source is usually white noise but could be anything. the click is sent from the mult to whatever it is clocking (sequencer, clock divider, shift register)

Swing

The simple way to get a swing is to vary a pulse width from a hard sync oscillator and use that as a gate.

PWM Effect

Take a triangle or saw wave
Amplify it right up.
Mix with a slow LFO.
Feed it through something so it's being clipped heavily.

As the LFO rises and falls it will add or subtract an offset to the original wave.
The top and bottom will alternately get bigger and smaller giving a PWM effect.

Chaos

Two VCLFOs cross modulating each other with a slew in between can be tuned to make non repeating (as near as I can tell) signals with chaotic events (bunched up like a creaking door).

Put the slew between the output of one and the input of the other. Skip the slew in the return path, just connect the output of one directly to the input of the other. One slew is enough, and you don't want too many parameters. Monitor either LFO output, or both. Once one is working well, the other will too.

The slew allows the two LFO's to work independently. The amount of slew determines the degree of independence. So you can loosely couple them with enough fiddling.

Experiment A

As an experiment, take 2 sines, get them to track. If you lack a sine a triangle is the next best wave, but stick with sines if you can. One goes through a extra VCA into the linear FM on the other. If the VCA has an manual initial level and input attenuation so much the better. Play something (regular pitch CV into 1v/oct expo input same as usual with pitched CV into 2 VCOs) and send some modulation to the VCA. Adjust things so it ranges from no modulation to however much you think sounds cool.

Now try the same thing in expo and find your results will have more stray out of tune harmonics and you get a different timbres if you play them in unison. So basically you get FM as you expect, but there is little to learn in terms of special situations that make predictable waves. It's not that you can't make crazy sounds with either, it's just that you need linear to do those harmonic rich sounds that track and are reliable and your results will vary from all those books trying to show you FM because you can't duplicate the experiments. It's because the expo tracking that lets you do do pitched CV from a keyboard or converter gives you a different amount of modulation rising compared to falling when you send a wave into expo and that changes with pitch. You need 'symmetrical' changes in the VCO at audio rates to get you can get control of.

Once you master 2 VCO FM you should try a different pitch on the modulating osc, try a different wave, then mess with 3 or more algorithms, though there is a lot to mess with with just 2 vcos

Experiment B

+1 Octave Saw: Ring-modulate a triangle with its own square.

Experiment C

Dual Saw: Mix Saw and phase-aligned Square, invert one, modulate pulse with a triangle or sine, each having its own effect. Triangle causes trill at high speed while sine causes vibrato. for the Dual Saw to work, you need to mix until you hear a +1 octave saw (pulse must be centered and square). This should not produce "trill or vibrato" as described above unless:

Instead of mixing until hear 1+ oct saw, you mix until you only hear the pulse sound disappear.

Basically you will be able to achieve a true (but limited) phase-modulation effect by modulation the PW. Now modulate that pulse width at high freqeuncy using a second oscillator with its pulse output. (Expiriment with pulse width of the second oscillator) to achieve some gnarly effects akin to Bi-N-Tic.Patch the gate signal to the hard sync input of your osc to get a punchier attack (the osc will start at +V instead of a random spot when you open the envelope.)

Experiment D
Patch an oscillator thru a reverb module with the springs damped (foam/cotton/whatever) to add harmonics to the signal. Or, don't damp the springs, if you want it to sound messy.

Experiment E
oscillating filters that don't oscillate by themselves by patching its output back into the input.

Experiment F
Patch a squarewave VCO into a fast VC envelope to soften either or both sides of the waveform, taking the audio from the envelope output.

Experiment G
Patch a fast VCO into a VCA CV input for AM fun, nice when the VCO is being played the same as the main VCO in a patch but at a higher frequency

Experiment H
Patching whatever you're modulating with a VC-LFO back into it for a little bit of chaos.

Experiment I
Another trick would be audio-rate PWM.

Experiment J
Have a square wave LFO triggering one of your envelopes, then modulate the speed of that LFO with a random source.

Put a ringmod in between the cross mod paths for extra fun.

Experiment K
I patched a VCO into my 24dB LPF and the filter into my expo/linear VCA. Both the filter and the VCA are modulated with ADSRs. However, I accidentally patched the LPF into the CV input of the VCA and the envelope into the audio input. It sounded way cooler than the right way round! And, the CV and expo/linear controls gave weird and counter-intuitive results.

the opposite vca connection (the usual way , osc in signal in and envelope in cv in) and it gave the same musical results. when the vca connection are reversed like you describe most of the negative voltage seems to be cut down in the resulting audio signal . That's because the CV of the VCA actually cuts off the negative voltages, unless you have a uVCA (or an equivalent kind of VCA with a bias knob) and can add a positive bias voltage to the CV.

Experiment L
bitcrushing and wavefolding CVs. sweet!

Experiment M
running pitch CV through a delay effect pedal. See what it does to the pitch when you play the keyboard.

Experiment N
inverted VCA audio into vca
envelope generator of your choice into vca cv control

but, instead of the usual having the cv from the envelope generator open the vca, have the vca open already and then invert the eg output so it closes the vca (bipolar VCA needed).

Experiment O
Patch the 1V/Oct pitch CV into the rate input of an LFO (you may need to stick some sort of attenuator in there if there isn't one already). Rout the LFO output to PWM of an oscillator. Makes for very good string sounds because the PWM speed increases as you play up the keyboard . Works even better if you have two or more LFOs at slightly different rates but both being driven by the keyboard CV. If the number and variety of modules permit you may also want to scale the amount of PWM applied so that it's less at the bottom end (to avoid any chance of "seasick" pitch wobble).

Experiment P
running the feedback patch of a delay into a frequency shifter.... or... ANOTHER DELAY!!!!

Experiment Q
I've spent many happy hours fiddling with filters in parallel.

Experiment R
2 LFOs (a and b) into a s/h...
-b is a little bit faster then a.
-pulse of a into trig
-triangle of b into sample
-output into anything
audio-rate PWM

Experiment S
Random Gates/Triggers...here's a poor man's way to get some both sync'd or not.

Gates (sync'd or not) into Signal In of a VCA - preferably at a tempo twice or more as fast as your "master" tempo. LFO (NOT sync'd, but being Rate CV'd) set to a much slower speed to randomly open the VCA - lets gates through only when open.

Experiment T
audiorate modulation of the controls of a wave folder. Stick a VCA in-between the mod VCO & CV input, even better! Mix the audio, into the input, with a slewed CV

Experiment U
If you have a delay with CV time, set the delay time very fast and modulate it with an LFO. You can create chorus and flange effects this way. You can get a lot weirder from here if you get creative with CV and feedback.

Experiment V
use a offset function to 'play' a quantizer manually.

Experiment W
You can use your VCA to compress a signal if you have an envelope follower. Just invert the follower output and use it as the CV for the VCA. Obviously you will want to do some tweaking but the idea is simple. If you don't invert the CV, you would have an expander.

CV Shaping

This is very simple but maybe overlooked. If you have an LFO with CV, try splitting the output and feeding one of the splits into the freq CV. For example if you do this with triangle, the resulting form is a kind of exponential triangle, so that you get a short little peak and a wide valley. Depending on the intensity of the CV, you vary the difference between lowest and highest slope. Useful for creating a sequence of short pulses by feeding through VCA but with continuous changes so the VCA doesn't click, i.e. it does what a short AD cycle on and envelope would normally do. A warning though: changing the intensity also changes the overall frequency. You can do this along with voltage inversion and other waveforms to other effects. For example this can change square to rectangle.

good for envelopes also

Flip-Flop

possibly the most practical and easy patch with the Flip Flop is as a /2 and /4 clock divider that you can directly influence.

If you put a clock signal into the top clock in and then put a sequence into either of the Set inputs. You now have a clock divider that will suddenly jump to a positive state based on the sequencer input. Fun for making a simple beat more complex.

Freq Shifter

got a delay with feedback out?

put your frequency shifter into the feedback out/return but make the shift REALLY small, set your delay ammount to 95-100.... instant barber pole effect. Have nothing going into the bbds input.

and in feedback loops, FM paths, processing drums, etc...

Take, for instance, the hi-pass out into the shifter, then shifter out into a mixer. Take the lo-pass out into the mixer as well. Vary the shift amount and the slope to move harmonics down into the lower range. Anything from odd but delightful harmonic enhancement to total manglement.
Of course, the same goes for the lo to hi, or the bp. I like to use the variable slope for controlling how strongly the shifted harmonics come in/how quickly. This can be a very useful modulation point for adding animation to the sound.

drums -> HP -> frequency shifter = yum!

Any multiple output filter plus a frequency shifter plus a mixer gives huge range for sound mangling. You could probably achieve some pretty interesting effects if you inverted the input coming in from one or the other sources and then shifted so as to take advantage of some almost comb filter effects.

The "talking chaos" patch, for example, has two 300s cross-modulating each other with sign waves, both set to approximately the same basic frequency. The sync buss is connected, and both are set to soft sync. That creates a complex interaction where the modules get stuck and unstuck. Wiggling one a tiny bit (with a low lagged sample&hold random voltage for example) will prevent them from getting locked into a pattern.

As the questioner mentions, The trick of modulating VCO pulse width with a high frequency sine for example adds a great "fret buzz" sort of timbre to bass sounds. For timbral variation within patches, try audio-frequency modulating anything with something else. Sine waves often make the best sources for audio rate modulation, because there's already plenty of overtones resulting, so I frequently put a filter into full resonance for that purpose.

If you have two LFOs with FM input, try cross-modulating them for complex interactions. If you put that output into the VC-in on an oscillator, you'll open up a wide range of bleebly sounds, birds, wiggly bits, etc.

Split the output of a filter and process one half with other filters and VCAs before putting that signal back into an input on the same filter (either audio input or the FM input). The feedback loop will affect the resonance, and dynamic processing will make the interaction very complex.

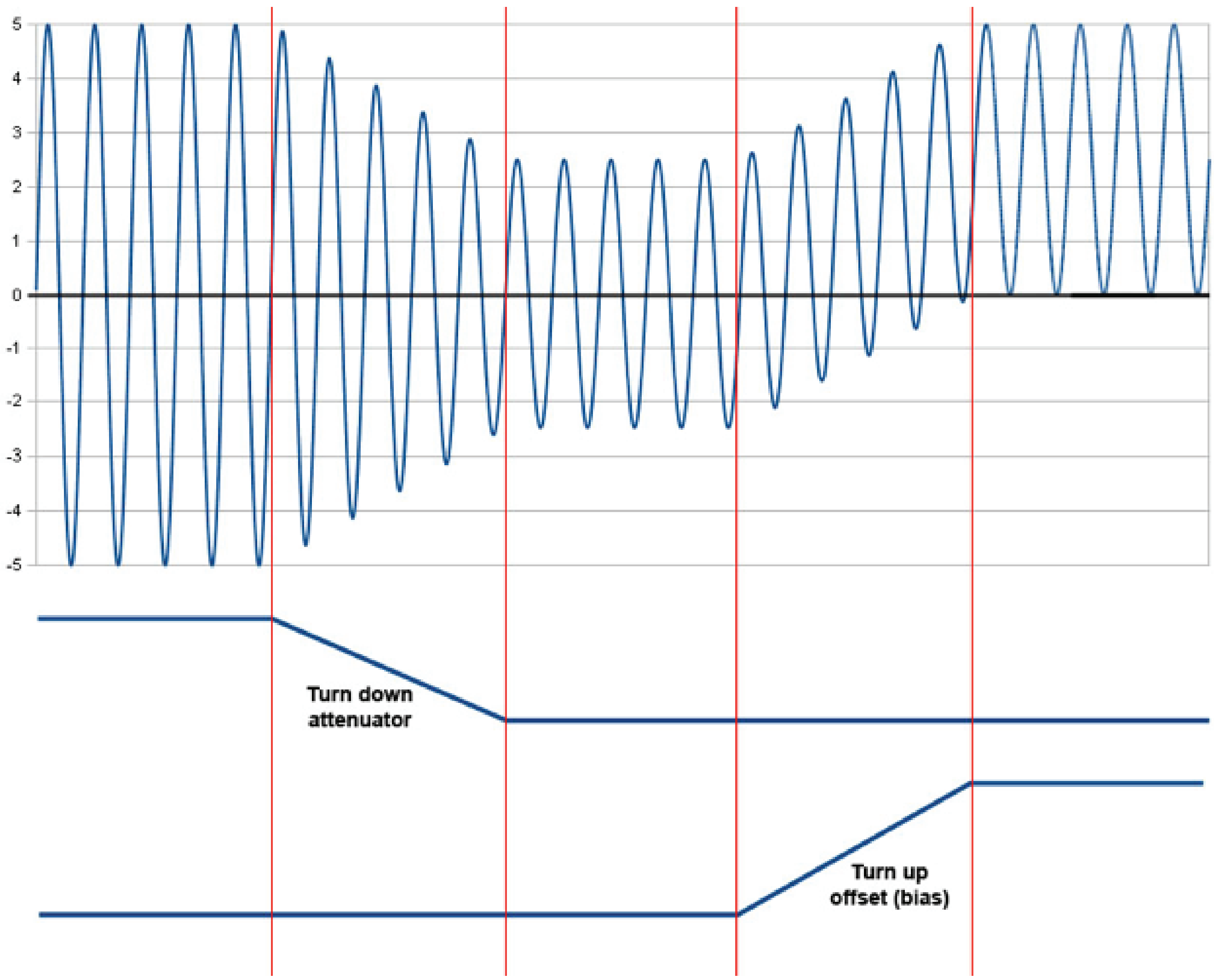
For the above idea and for any other patch, add a time-domain digital effect - like echo, reverb, chorus, flanging, etc - to the inside of a patch. Use outboard effects as if they are modules in the system. This opens up an entire realm of options for feedback-type patches.

Rethink the standard use for a module. Did you know that a lag processor is a low pass filter? An envelope generator is a lag processor? An envelope can even be a waveshaper for low audio frequencies at its fastest settings. Likewise the 320 LFO makes a good audio-rate oscillator for bass sounds with very cool waveshaping features.

Two oscillators at their highest frequencies - above hearing - can cross-modulate each other to create difference tones you can hear. That's how a Theremin works, and radio. You can do it on MOTM oscillators. Try modulating a super-sonic VCO with an external audio signal. It's very odd. You can do the same with resonating filters.

Speaking of resonating filters, you can "ping" them when they are almost ringing by putting a sharp envelope blip into their audio input. It makes a very ghostly gamelan-like sound.

The idea behind all of these tips is to break the established paradigms and rethink the possible role of each module. Don't assume a patch has to go VCO-VCF-VCA.



2.0
Physical Modelling

2.1
Birds

osc into ring mod or pitch shifter, take the higher harmonics and round them out with a soft attack EG for "tweet." slow LFO can then take you to "two-wheat."

Remember the reverb - that's the glue that puts it all together.

For birds that trill a lot you can use chained envelope generators running VCAs with LFOs going threw them modulating the primary VCO. Then stage the EGs to open in the order you want. An A-143-1 or A-143-2 with a VCAM work great for this. Of course the VCO you modulate should in most cases be a sine or triangle, unless you are trying for crows or sea gulls...

wavefolding just the beginning stage of a square wave with a really thin pulse width (the typical recipe for an "ahh" sound) and then dropping the pitch off exponentially, but not too abruptly.

Start with a rather narrow pulse on your VCO. Control pitch with a decaying EG. Run the VCO through a resonant BPF or LPF.

To make the Krr sound, use a fast LFO (saw) that you gate with a VCA or analog switch. Control the VCA/switch with an EG or a gate processor that gives you one short pulse. Use the LFO to control the VCA of your ordinary signal patch.

2.2
Strings

Patch the EG1 into VCA1 CV input, and EG2 into VCA2 CV input.

Patch the input of VCA1 to a white noise source.

Patch the output of VCA1 into the linear CV (modulation) input of the VCO, and be ready to adjust the VCO's external linear CV input as desired.

Set EG1 for a really short attack and slightly longer decay. This will give you the scratchy "attack" of the bow.

The rest of the patch is fairly straight forward. Just adjust EG2 / VCA2 for the sustain profile you're looking for.

The filter is optional at the output of the VCO, but I'm sure you'll find the right combination. Note - a fixed filter bank can do wonders with tone quality.

I've had very pleasing results with complex waveforms and envelopes.

Use a wavefolder or audio-rate PWM, a clean filter, one envelope to FM the VCO for plucked pitch bend and another with an extremely exponential decay for a natural 'tail'.

If you want nylon string sounds, try some dynamic linear FM with an even C:M ratio, like 2:1 etc.

2.3
Snares

White noise + good filter (lpg works) + snappy envelope

white noise and sine/triangle oscillator through a mixer / x-fader
mixer output through a multimode filter with LPF and BPF outputs
run filter outputs through another mixer or x-fader
output of the final mixer through a VCA modulated by a snappy env
VCA output through a reverb

The first mixer/x-fader lets you set the balance between the snares (noise) and the drum tone (oscillator). The oscillator is usually tuned low to the key of the song, say between 110-220 Hz. I often use a filter with dual inputs to save an actual mixer.

The second mixer lets you set the frequency balance and bass content of the snare. Some filters like the Borg already have this built-in. You can also use just the LPF or BPF if you don't have a mixer at hand.

A variation of this patch involves running the white noise and oscillator through separate VCAs controlled by separate envelopes, before going into the filter (in this case, you probably won't need the final VCA). This lets you set a longer decay for the "tone" layer, and a shorter one for the "snap" (white noise) layer. You can also try to amplitude-modulate the white noise with a fast pulse LFO to emulate claps.

Also, you can perform fake compression by running the envelope first through a clipper to give it a short sustain before it begins to decay.

Vc decay is a must. Nothing like accenting a snare with more decay.

White noise to PWM can result in some crunchy snares.

Modulate pitch and VCA with a log rather than exp env decay for some 80's touch.
A slight FM using pink noise will add some organic feel.
Filter to taste.

Try patching the 'tone' portion of the signal (like a triangle osc, but not the noise generator) through a frequency shifter that is set to a low ratio, just enough to alter the sound a bit. This can be used to alter the relationships of the harmonics in ways that mimic some harmonic behaviors of acoustic drums. I've used this trick with great results when synthesizing things like tympani and large acoustic bass drums.

It can also help to send the noise generator through some kind of mild distortion device before sending it to the filter.

Noisering Noise Out > Optomix Ch 1 Signal In
RCD > Optomix Ch 1 Strike
Optomix Control Attenuator Full CCW

1) I bandpass the noise component - I like my Synthasystem VCF for that task usually. (Frequensteiner would be essentially the same thing.) Variant: set resonance near oscillation, then FM the cutoff as in #3 below.

2) Wavefolder on the 'tone' component, CV'd by your envelope. That adds a little 'hash' to the transient strike. I'll do this on kicks too, if I want to get fancy.

3) Try some AM on the final VCA you're using - I like to use an audio-rate pulse wave. Subtle is good here, and it works better on some snare patches than others.

2.4
Crackling electicity

Noise ---> Attenuator/Amplifier ---> Sample&Hold ---> Envelope Gate

Set the S&H to a fast rate. Set the envelope with zero attack, zero sustain, and very short decay & release. You can use the output of the envelope itself (it should produce audible clicks), or use it to modulate anything through a VCA or VCF (not an LPG, since they will impart their own decay and get rid of the clicks).

The gain of the noise attenuator becomes the "probability" with which the clicks are produced. It is also possible to sync them with a clock (use the clock as the S&H gate).

Run the output through a phaser and reverb and you're good.

use an osc and attenuverter
wave out of osc>attenuvert>cv in on osc
take audio from a different wave

I like tri for feedback and sine fro audio out
if you dont have an attenuverter, a normal atten will work too

One approach could be to send audio-rate sampled & held random CV through a HPF, and use the same random CV to modulate the cutoff frequency. Adjustments I made were to use blue-shifted noise, some resonance on the HPF and a really short envelope.

2.5
HiHats

i usually just start with white noise, HPF out all the low frequencies, and then shape it with an envelope and a VCA.

things to tweak include the filter frequency and resonance (adding a lot of resonance to get the filter to self oscillate adds a sine wave which can sometimes add to the "metallic-ness" of the sound"), envelope controls and VCA type (experiment with log/lin/expo shapes if your envelopes and VCAs allow it).

maybe also mix in a pair of carefully ring modded sines for a more metallic edge?

I like to take white noise along with a bunch of very high pitched oscillators playing squarewaves going through a high pass filter. I like to mix the oscillators a bit quieter than the noise but loud enough to hear them, when you have a handful of high pitched oscillators it can sound a lot like a tr-808 cymbal or hihat.

Nice to tune two prominent oscillators in the mix to a 5th

2 VCO set to tri, one into the FM input of the other, output from the one being FMed into a hpf, experiment with cutoff freq and freq of both VCO, can get snares, claps and hats using that method.

2.6
Brushes and Marcas

Tuned noise, Mixed with the feedback amount you get some very realistic brushes and maracas.

2.7
Flute

i would try some linear fm patch with an vca on the modulation-index.

controll this vca with an envelope and you get the breathing.
the total amount of fm accords to how hard the flute is blown in.

I would use a VCF and have it open more in the higher registers. A delayed warble/vibrato is nice for flute emulation. Take your triangle wave and experiment! Adding a little white noise to certain notes at the start of notes may help, etc. Depending on how you are using it (solo baroque or in a heavy mix... live or studio) the complexity/detail may not matter.

2.8
Kicks

Just one piece of general advice: Making a kick that feels good can often be a matter of VERY minute adjustements. Take your time finding the sweet spot. Even on a dedicated drum module such as my DRM1 mkIII it is easy to dismis the kick as "too this or too that", but in fact it can be very good when the sweet spot is found.

More specificly: If you have two envelopes you can really widen the scope. In some cases you need a vca to dampen the release. In other cases you can get away with just sweeping the frq down to sub audio in the release phase.

How about some slightly-resonant High Pass filtering and distortion? Then compress that stuff. Yes, I typed "High Pass" Filter. You can get some 'bump' at a desired frequency without the kick becoming too flabby.

Also, try putting some subtle sweep to the filter on each decay. Or not. How about parallel processing?

How about some slightly-resonant High Pass filtering and distortion? Then compress that stuff. Yes, I typed "High Pass" Filter. You can get some 'bump' at a desired frequency without the kick becoming too flabby.

Also, try putting some subtle sweep to the filter on each decay. Or not. How about parallel processing?

are you putting a relatively quick envelope on the pitch at the beginning of the kick? that is pretty key for the majority of kicks. makes it more percussive.

Parallel processing from the same gate/trigger is a good idea.
Mult. and send one to your resonating filter,
and the other to gate in on an LFO tuned to just the right frequency.
Mix and control shape with envelope to taste

* You can add in a noise source on another channel of the same mixer to achieve a larger pallet of other drum sounds.

* You may want to use multiple envelopes to tailor each part of the sound from the same trigger, but one EG works fine for me.
* EGs & other modules respond differently to gates/tiggers YMMV.
* A dedicated bass drum module will do a better job than this patch, but this isn't bad in a pinch (if you don't mind using up a handful of modules).
* these techniques can be even more useful in addition to dedicated drum modules.
* it gets even more fun when you add a ringmod, and some other processors.

it is often a good idea to layer sounds to achieve different results. For example, finding a sound which gives you the percussive thud you're looking for may lack the bass presence you want. To resolve this, use multiple envelopes/sound sources/VCAs as LDT suggested and use the second set to build a sound which provides the bass presence to round out the percussive properties. Layer it into the original sound and you end up with the best of both worlds - percussive attack and a powerful boom.

More versitile kicks can be made by modulating a sine (or if you like gabber kicks pulses) with an adsr (asr on 0, d and modulation depth is critical and need to be in balance) and a second adsr for the vca (a on 0, d don't matter, s max and r to taste). This patch doesn't need a compressor, thats what the adsr for the vca does.

3.0
Explained

Linear and Exponential Envelopes

Expo curves are where you can hear clicks when used with audio, particularly with exponential response vca's.

It is important to know that VCAs only act on positive control voltages. Negative control voltages cause them to shut off (this is how a 2-quadrant multiplier works, and that is what a VCA is).

Hence, if one feeds a typical bipolar LFO voltage to the CV input of a VCA and tries to get vibrato (by modulating pitch at the FM input of a VCO) or tremolo (by modulating the VCA directly somewhere within the signal chain), then unless one "lifts" the LFO voltage completely into positive territory, one will get a strange sort of modulation which only works for the positive half of the LFO's cycle, and leaves things unaffected during the negative half.

Exponential FM Vs. Linear FM

Pretty much every modular VCO has a 1v/oct CV input as standard. You typically connect it to your keyboard/sequencer/midi to CV converter to get the VCO to play in pitch. I guess there must be a couple more DYI type "drone box" devices and LFOs that don't as well as the Korg MS-20 that are exceptions. Then there are some units other than 1.0 volts per octave that for most intents behave the same way, like EMS, EML, Buchla.

Filters too usually have a dedicated 1v/oct input, which will often but not always mean they can go into self oscillation and track the voltage. The main practical purpose though is to have the cutoff track the keyboard/sequencer. This means the timbre stays the same brightness with different pitches. (lots of newbies don't do that and wonder why simple monosynths sound better playing filtered pitched notes, anyway...)

VCOs and VCFs typically have another similar input with a pot to attenuate it. You would use it for adjusting the amount of modulation, say from an LFO. That's frequency modulation. To be specific, a 1v/oct input or an unmarked input for modulation with a pot is exponential FM when you hit it with a frequency. "1v/oct" just means it's ready to track.

Then on some VCOs but no VCFs I can think of you have still another FM input, it's just about always marked "linear FM". Just pointing that out because people tend to ask and the answer is almost always the input is expo unless it says so. If the input just says "FM" without the "linear" then the module maker is just out to be trendy of confuse. Every frequency CV input does expo FM, it doesn't need to be called an "FM' input

What happens with FM is when you CV modulate an oscillator at an audio rate you create more frequencies that vary with the amplitude of the modulation. The greater the amplitude of the modulation the more new frequencies get generated above and below the original pitch and they will vary with the frequency and shape of the wave doing the modulation.

What happens with a linear FM input is the modulating wave will lower the pitch of the VCO being modulated the same frequency change as it raises it. when this modulation is sped up to audio rates, the "average" frequency ends up right where it was with no modulation, meaning your ear will perceive the same pitch as before, but with a more complex waveform/harmonics.
If the same thing is done with V/Oct FM, the modulating wave would, for example, pull the intial frequency say 100 Hz down, and 200 Hz up (because each octave is a doubling (multiplication) of the intial frequency and not an additon of frequency.) during each cycle of the modulating waveform. Thus the perceived pitch tends to be dragged up (sharp) as the depth of modulation is increased. So like as an experiment, if you slowed it down to an LFO doing the modulating you could hear the carrier swing say up an octave and then down an octave. When you do that at audio rates you can really work with creating new waveforms that potentially track the keyboard because you can potentially achieve the same timbre when both carrier and modulator track.

When you hit an expo input with audio rate waves the new frequencies being generated are different above and below the original fundamental pitch. You can keep tweaking and find something cool but then if you change the pitch of both carrier and modulator in unison you get something quite different.

If you hit either linear or expo with enough amplitude of a random frequency you will get a crazy result regardless. The interest in linear is you can get control if adjusted right. The cost is extra circuitry in the VCO and a VCA to get the levels right
FM as a form of synthesis works using the amplitude of the modulating wave. Depending on the carrier's frequency, your higher amplitude modulator wave can modulate the carrier down to a stop at 0 cycles a second and want to go further. That's trouble because it's not going to generate the frequencies it would have. So the through zero capability as I understand it flips the phase and keeps on modulating.
frequency modulation (FM) and phase modulation are completely unrelated, and don't sound anything alike when done at low modulation frequencies.

When done at low modulation frequencies you can't hear phase modulation at all unless you mix the modulated signal with another signal of similar frequency. That's not true of frequency modulation. You can hear that directly.

When modulated with frequencies that are close to the oscillator's output frequency then they sound a bit more similar. That's because phase changes at that speed can create portions of a cycle at higher or lower frequencies that can be a large percentage of the main frequency.

VCAs:

Any input on any module, whether it's audio or CV in, can have a VCA connected to that input and use another signal to modulate the signal going in to the input. The more modules you have, the more inputs you have, the more places you can stick a VCA. Hence "you can never have enough VCAs".

You can modulate audio with a CV. You can modulate CV with audio. You can modulate audio with audio. You can modulate CV with CV.

You can even plug a VCA into a VCA. Either in the signal or the control input.

For example, an oscillator typically puts out a constant tone. So it sounds like:
AAAAAAAAAAAAAAAAAAAAAAAAAAA
Feed it into a VCA and an LFO can force the oscillator be silent at regular intervals, like:
AAAA----AAAA----AAAA----AAAA----AAAA
Feed that into another VCA and another LFO can then inject another patern of silence:
AAAA------AA------------A-------AAA-

Of course if you have an analog AND module (like the MIN part of a Doepfer MIN/MAX) you can combine the two CVs first and feed them to a single VCA. And there are lots of other ways to combine LFOs to make complex modulation signals. This is just one example.

With a VCA connected to the CV input of another VCA you can not only modulate a signal with a second signal, but you can use a third signal to modulate how strong the first modulation is.

Of course a VCA has two inputs, so you need enough signals to feed into them. So you need more modulation sources. Then you can use more VCAs. Then you need more modulation sources.

VCAs are good for overdrive distortion and doing what you said above which is boosting quite or low amplitude signals. Also if you work with non-euro equipment (e.g. pedals) in conjunction with a euro rig you'll either amplify on the way out or coming back in.

Linear VCAs are way more useful and versatile than expo VCAs. In most situations where you are trying to control the amplitude of an audio source you are driving the VCA with an envelope that already has an exponential shape. If you use an exponential envelope with an exponential VCA you get a super exponential curve that is not very useful.
The Linix is suited well to both audio and CV.

The optomix, Obviously it's incredible as a frequency and dynamics modifier, and it even works wonders when sending an fm modulator through it, striking it, and sending it to a carrier osc. But, obviously, it's rubbish for the vast majority of vca duties due to the response of the vactrols (the extended "ring"), lack of linear cv response (due in large part to those vactrols),and the coloration of all signals passed, whereas for the vast majority of typical vca duties you want a vca that is clean, controllable, and as fast as you need it to be. This is not at all to deride the optomix in any way, in fact, as an lpg it's damn near perfect and I use it all the time, but it's no substitute for actual vca's, especially for things like cv'd depth control of a cv modulator or gating a signal sent through a typical vcf (where you might want to preserve the sound coloration imparted by the filter more than an lpg would allow). I think the fact the vactrols make the response sluggish makes it pretty unusable for cv control and processing, unless you want something like a slew limiter…

exp is better (sometimes) at producing more natural sounding percussive shapes when controlled by an envelope

i tend to pretty much always use exp. for envelopes, and lin for lfos (on an oakley classic vca)

I only use exp. For mixing and CV.
Lin for regular triggering the VCA
It depends on what your envelope is capable of as well,
If you have an exp. curve on your envelope I usually find exp. VCA unnecessary
I use exp for wave shape/overdrive sounds.
Punchy, fast, plucky, percussive is exp to me.
Slow, fading, bowed, gentle is lin

Slew

a slew will will continue to slew until the peak volatge is reached however if the input drops before the max is reached it switches to the fall phase of the slew. You really need to send it gates for a poroper eg type result as a trigger is a fraction of a second which is why you are not getting anything. Also remember that a slew will hit the same maximum level of what you feed it. If you give it a 10v gate it will rise to 10v - 5v gate goes to 5v.

Coupling

AC coupling is where the signal goes through a capacitor to remove the DC part of the signal. This happens in anything designed specifically to accept audio signals as opposed to control voltages. Since audio does not go below about 20Hz, the DC part of a signal is high pass filtered out and only frequencies above around 20Hz are allowed through.

So, for example, the inputs on an audio mixer will be AC coupled. You could feed a sub-sonic square or ramp wave to an external audio mixer and then send that back to the modular and the mixer would remove the subsonic frequencies but pass the clicks.

Logic Explained

The truth table for an OR is:

A B OUT
0 0 0
0 1 1
1 0 1
1 1 1

These logic gates(NAND,AND,NOT etc.) are really great for combining say a trigger to control other gear. AND is:
A B OUT
0 0 0
0 1 0
1 0 0
1 1 1

XOR
A B OUT
0 0 0
0 1 1
1 0 1
1 1 0

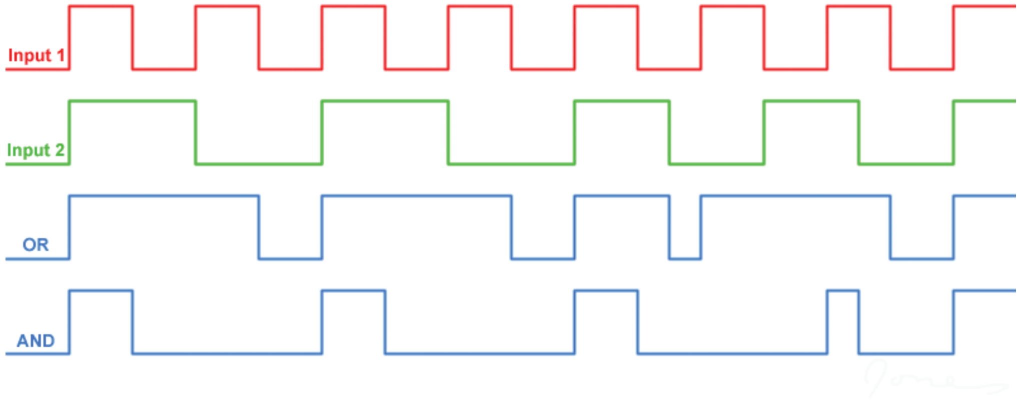
OR = max
AND = min
NOR = negative or inverted OR
NAND = negative or inverted AND

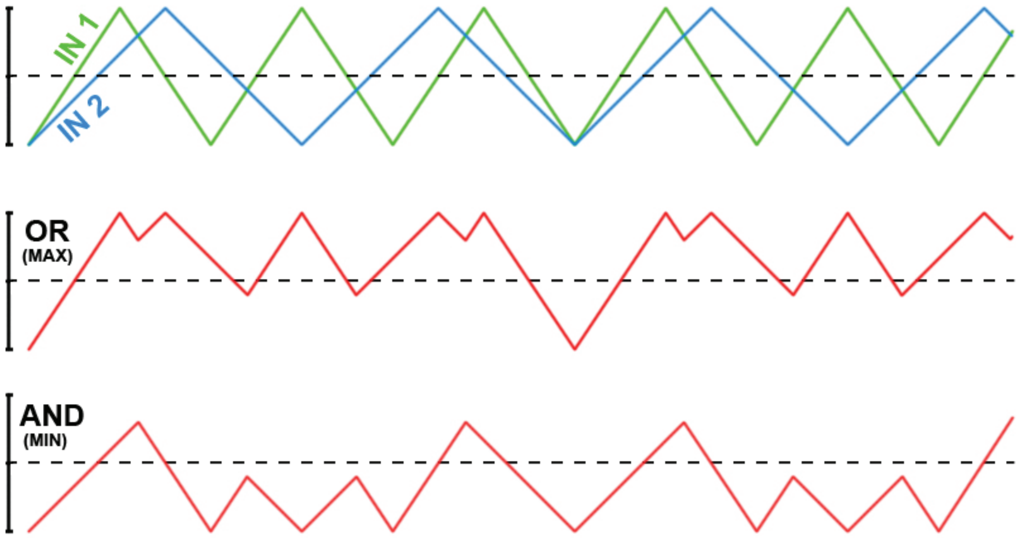
(NAND and NOR are simply an AND and OR run into an inverter)

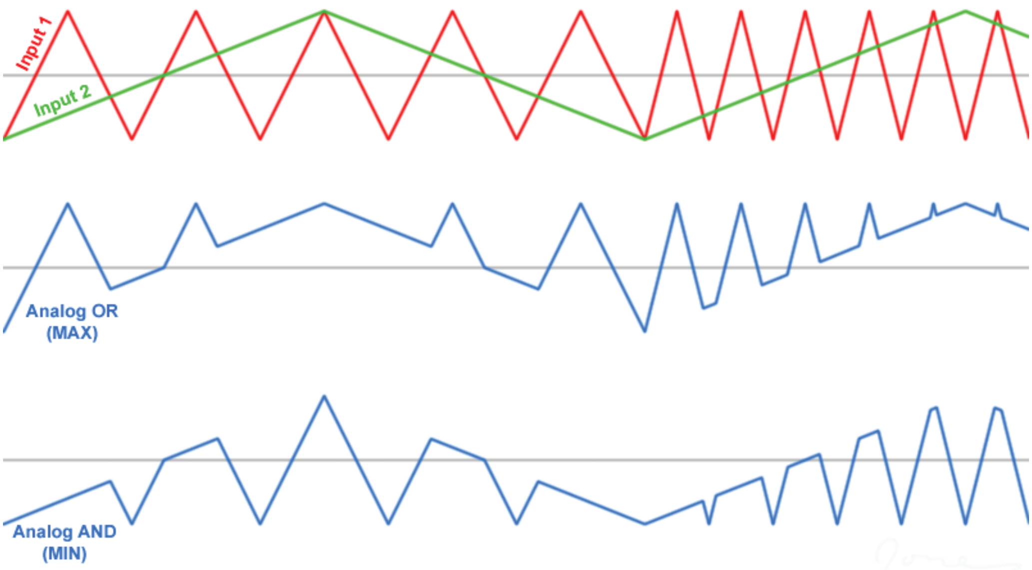
you can think of XOR as a gate controlled invertor as well. Pulses on A will go thru unchanged so long as B is low, but, if B is held high the pulses at A will be flipped - when A goes low the out goes high

In analogue logic OR would be a peak detector (the highest input takes priority) whereas XOR would be the absolute value of the difference between two inputs-so if one input was +2V and the other was -5V the XOR value would be +7V.

I think a 4 quadrant multiplier (VC Polarizer) with an inverter on one of the inputs would be the closest to an analog XOR. With one input low the other input passes through normally. With one input high the other input passes through inverted. And in between gives lots of fun.







There are two types of "logic" found in modular synths. Digital or binary logic, where signals in and out are expected to be in one of two states. A low voltage (often zero volts) and a high voltage (typically +5v, +8v, or +10v). Signals are combined using Boolean rules and are often used for creating complex gates.

The other type of logic in modulars is "analog logic". This isn't the same as the classic digital logic but attempts to apply the same Boolean rules to combining a pair of analog signals, with an analog result.

"Sum" is not a logic or Boolean function. It is a mathematical function. In general, math functions are applied to analog signals (if applied to digital signals they are simply treated as analog waveforms).

Boolean functions are things like AND, OR, XOR, and NOT.

AND = input 1 and input 2 must both exist in order for the output to exist. In digital logic this means that the output goes high only if both inputs are high. Otherwise the output is low. In analog logic the AND function is also called a MIN (minimum) function. The output voltage reaches a given voltage if input 1 and input 2 are at that voltage. Otherwise the output is the lower of the two voltages.

OR = If either input 1 or input 2 exist, the output exists. In digital logic this means that if either of the inputs is high, the output is high. If both are high the output is high. So the output is only low if neither of the inputs is high. In analog logic the OR function is also called the MAX (maximum) function. The output voltage is the voltage on input 1 or input 2, whichever is higher.

XOR = "exclusive or". Similar to the OR function except the output is high if one input or the other input is high, but not if both are high. This can also be thought of as a way to see that the inputs are different. If the inputs are the same the output is low but if they are different (one high and one low) then the output is high. There are a couple of analog interpretations of how to do this and they are not as common as an analog AND or analog OR because it is a more complex way to combine signals. Basically as one of the inputs goes up the output goes up, but as the second input also goes up it causes the output to reverse and start going down.

NOT = the NOT function is simply an inversion of the signal. With digital signals if the input is high the output is low. If the input is low the output is high. This is also called an inverter. With analog you simply turn the signal upside down so positive is negative and negative is positive.

Mathematical functions work on analog signals:

Sum = (addition) Mixing two (or more) signals together without attenuating the input signals. So a +2v signal on one input and a +3v signal on the other input puts out a +5v signal. A +2v signal on one input and a -3v signal on the other input puts out a -1v signal.

Two +5 inputs will output +10v. The output voltages are limited by the power supplies and the mixer circuitry, so summing four +5v signals can not reach +20v and simply gets clipped by the circuitry typically somewhere around +/-10v or +/-11v.

Average = Basically the same as a sum except that you turn down the mixture based on how many inputs you have so that the sum of all inputs never goes higher than full scale. For example if you feed two signals into a mixer and set both inputs to 1/2 gain then the mix will stay within the same voltage range as the originals.

Biasing = a special case of summing or addition where only one signal is fed in and it is added to the voltage on a pot. The pot then adds or subtracts a fixed amount of voltage transposing the input up or down by a certain amount without changing the size (scale) of the input signal.

Multiply = The output voltage is one input's voltage multiplied by the other input's voltage. There are two common types of multipliers in modular systems. A VCA is a "2-quadrant" multiplier. Typically the signal input can be positive or negative while the control input is usually only positive. Many VCAs only have unity gain so the output matches the input when the control is all the way up (perhaps +5v). So for mathematical purposes you can think of the control input multiplying the signal input by a value between zero and one as the control goes through it's range (perhaps 0v to +5v). So if the CV is at 2.5v it is multiplying the input signal by 0.5, making the output put out a signal half the size of what is going in.

The other type of multiplier found in modulars is a "4-quadrant" multiplier. Also called a balanced modulator or possibly a bipolar VCA. This is similar to a VCA except as the control signal becomes a negative voltage the output becomes the negative version of the input signal. These also often have unity gain at full control voltage, so are like multiplying the input by 1.0 when the CV is fully positive and like multiplying the input by -1.0 when the CV is fully negative. So when the CV starts at -5v the output is the negative of the input signal. As the CV rises towards zero volts the output gets smaller and smaller. When the CV reaches zero volts the output has no signal and is just zero volts. As the CV continues to rise and becomes a positive voltage the output is now a positive version of the input signal and gets stronger as the CV rises.

Subtraction = Just like in normal mathematics, you can subtract one signal from another by inverting that signal (making it a negative version of the original) and adding it to the second signal.

Pulses Explained

Pulse is a very vague term. It doesn't define voltage, duration, polarity or repetition rate. All that "pulse" defines is a signal that changes from one state to another and back again.

It can definitely have duration. A pulse can be 1 nanosecond wide, or it could be 10 seconds wide. It could repeat immediately, or it could be a one time event. It can be positive or it can be negative.

Many people use the term pulse to describe something that happens for a short time and repeats occasionally. But technically that's only one type of pulse.

The positive portion of any square wave is a pulse. So is the negative portion. But the term usually only refers to one or the other portion.

PWM (pulse width modulation) usually refers to adjusting the duty cycle of a square wave where you make the positive portion wider as the negative gets smaller (and vice versa) maintaining a constant frequency.

Since pulse can describe so many different things, the context it is used in can help narrow down what type of pulse is being talked about. For example, usually when a circuit has a "pulse" input it is expecting a trigger signal that has a short duration relative to it's repetition rate.

What are commonly referred to as "gate" and trigger" in audio synths are both typically pulses, but "trigger" usually implies a much shorter duration. Usually the trigger is a short pulse at the start of an event, while the gate is a pulse that lasts for the full duration of the event.

Attenuation/Scaling Explained

Scaling is making a signal larger or smaller. It does not have to do with musical scales. It has to do with the scale (size) of the signal. An attenuator allows you to manually reduce the scale of a signal. With it all the way down the output is zero. As you turn it up the signal gets larger until it is at full scale (output matches input) when the knob is all the way up. To get an output that is larger than the input you need amplification (gain). Most attenuators only reduce the signal, but some of them might have additional gain allowing you to boost the signal (scale it up).

An "attenuverter" or bipolar attenuator is one where the output is at zero when the knob is in the center. As the knob goes up the signal gets stronger. If the knob is turned down below center then the output is the negative (inverted) version of the input. The further down you turn the knob from center the stronger the negative signal gets.

If you don't have an attenuator that has additional gain, or any other dedicated device to add gain, then you can feed the same signal into two inputs of a mixer. This creates the sum of the two inputs. Since they are the same signal, if both knobs are higher than half way then the output is larger than the input signal. When both knobs are up all the way the output is typically twice the input signal, or a scale of 2.

Just think of an attenuator as an offset for a control voltage. Balance the CV output level and the attenuator to get you to the 'mean' starting point, where the sound is 'good' or desired, and then you can mess with it as you wish by raising or lowering the modulation or control via the attenuator. This obviously means that you need to have the attenuator set to somewhere around the middle of its travel. This attenuator can be multed to any combination of destinations.

Flip-Flop Explained

You can think about it like a sample and hold; it will sample a stream of logic and capture 1 bit when you stop sampling. This is to say its a switch that doesn't simply alternate when you activate it, but instead 'listens' to a second logic line and captures the last state before you deactivate it. (note this is only one possible configuration of a j/k flip flop).

Basically a flip flop lets you build a state machine. If you can think of two 'states' for your synthesis or more, this is very powerful.

'State' can mean whatever you want, which of course is beauty of it. But it could be something like gating an FM signal (a vca would work here). Or it could be an attenuated voltage going to the transpose of a quantitizer, which would have you shifting keys in one state. With multiple flip flops, you can now have multiple states. There's a lot of possibilities here, you could even slew the signals you are transitioning between to make it smooth.

Comparator Explained

A comparator will compare two inputs and put out a digital signal that is either high or low depending on whether input 1 is higher or lower than input 2. It is often used to convert an analog waveform into a digital waveform (square wave) by comparing an analog waveform to a constant voltage. For example if one of the inputs is at zero volts then the output goes high or low as the other input crosses zero volts.

4.0
Other Resource

Physical Modelling
http://www.frodebeats.com/physical-modelling

Numerous Application Notes of interest in the areas of analog and digital music synthesis, audio, and general signal processing
http://electronotes.netfirms.com/

A sample of a complete tutorial about the technique of Analog Modular Synthesis by Voltage Control
http://www.angelfire.com/music2/theanalogcottage/patch.htm

Advanced programming Techniques of modular synthesis
http://www.cim.mcgill.ca/~clark/nordmodularbook/nm_book_toc.html

Doepfer Patch Book
http://www.modular-planet.de/modular-planet-survey.html

Detailed analysis of the components of synthesis
http://rhordijk.home.xs4all.nl/G2Pages/index.htm

Module specific patches
http://navsmodularlab.blogspot.com/

Getting Started in the Synth DIY World
www.synthtech.com/tutor/tutor1.html

Growing your system beyong a Dark Energy
http://www.amazona.de/workshop-doepfer-dark-energy-minicase-teil-1/
http://www.amazona.de/workshop-doepfer-dark-energy-minicase-teil-2/
http://www.amazona.de/workshop-doepfer-dark-energy-minicase-teil-3/

Sound On Sound Synth secrets
http://www.soundonsound.com/sos/allsynthsecrets.htm

An Introduction to Digital Logic - Signals and Gates
http://www.facstaff.bucknell.edu/mastascu/elessonshtml/Logic/Logic1.html

The Facts on Linear vs. Exponential
http://www.synthmuseum.com/magazine/linexpo.html

Workshops based on the Nord Modular
http://nm-archives.electro-music.com/010_NordModular/015_Workshops/

Serge gold book:
http://www.carbon111.com/sergebook.zip

The arp 2600 manual online for free:
http://www.guitarfool.com/ARP2600.html

analogueak videos
simple analog modular patch - How to - part 1 -
http://www.youtube.com/watch?v=rcWllXKMfo0&feature=c4-overview-vl&list=PL4AB15111DCE4EAFD
Simple analog LFO modulation - How to - part 2 -
http://www.youtube.com/watch?v=XR9Rxazye0E&feature=c4-overview-vl&list=PL4AB15111DCE4EAFD
Simple analog envelope modulation - How to - part 3 -
http://www.youtube.com/watch?v=yVn3YRoAxas&feature=c4-overview-vl&list=PL4AB15111DCE4EAFD

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Over 700 Copies of V1 were downloaded. If you each just donated $5, it would help Muffs out hugely. It's been said better than I ever could:

“YES WE CAT!

donate bitches. Muff wouldn't ask to donate.. instead he sells his desert island bits of gear.. so.. help a brutha out and whatever few dollars you can spare i'm sure would go a long way...

if not then just send him happy thoughts and hugs. the universe will hopefully turn that into good days for muff and his fam.”

https://www.paypal.com/ca/cgi-bin/webscr?cmd=_flow&SESSION=8P75OhREFSEFRxD0dThhXm8R2j_sLDd6sfPdIsDBgLCXzW43zoh438niiXq&dispatch=5885d80a13c0db1f8e263663d3faee8d0038486cd0d9a2f30f3a21df7b0d0cee

Notes

I know this may make a few people :waah: but i've decided to not pursue any module specific content. The initial content, like the entire initial effort, was done for my own selfish enlightenment. So the module specific content related to modules I own. Even initially, I was on the fence about including it.

Now having started in the eurorack sub-forum I realize there is simply too much module specific content. We'd land up with huge portions of BOBI with no relevance to the majority (and of no interest to me :oops: , i'm just too self involved...).

Anyway, as I get deeper into this I've come to feel it has to be a bit more focused.

If you wanna get hold of me for whatever reason, mail me hello.its.me.again@gmail.com. Till V3, happy wiggling!