Akai Rhythm Wolf Mods with schematics

Source : http://wiki.openmusiclabs.com/wiki/RhythmWolf.

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Original English Version

Overview

The Rhythm Wolf by Akai was one of the most universally panned synths when it came out in 2014. It creates some interesting sounds, but these are often limited to a small range of knob settings. Other issues like excessive noise and the unfixable bass synth oscillator drift did not help the situation. But, the negative reviews are probably equally fueled by disappointment. Here was a combo, analog drum machine and bass synth with sequencer, that looked perfect on paper, but fell short when people actually listened to it. They were expecting an 808 and 303 in a single box, and instead got something else, something less. Now that you can get a cloned 808 and 303 for cheap, maybe there is value in something that sounds different. Due to the market failure of the Rhythm Wolf (RW), it can be had for under $150, and perhaps be modded into something that sounds good in its own way. This project is an attempt to see if that is possible.

There is also a Muffwiggler thread with more discussion on the topic: RW Muffwiggler thread

Schematics

The first step in turning the RW into something else, is to understand what it currently is. To this end, i have reverese engineered the schematics and analyzed them to see where things went wrong (and some things went right). My apologies for the hand drawn, and somewhat hard to read schematics. It was way faster to do it this way than to CAD everything up. Perhaps someone will remedy this in the future. They are seperated into sections, with each section linking to schematics and a discussion (Note: there are no values for ceramic capacitors, as they are hard to measure in-circuit):

1. Bass synth VCO and VCF
2. Bass synth Envelope
3. Bass synth VCA
4. Hi-Hats
5. Percussion
6. Snare Drum
7. Kick Drum
8. Mixer Section
9. Power supply Section
10. All Schematics (zipped)

NOTE: there might be some mistakes in the schematics (wrong designator numbers, values, etc.), so if you can't find a part, it might be an error in the schematics.

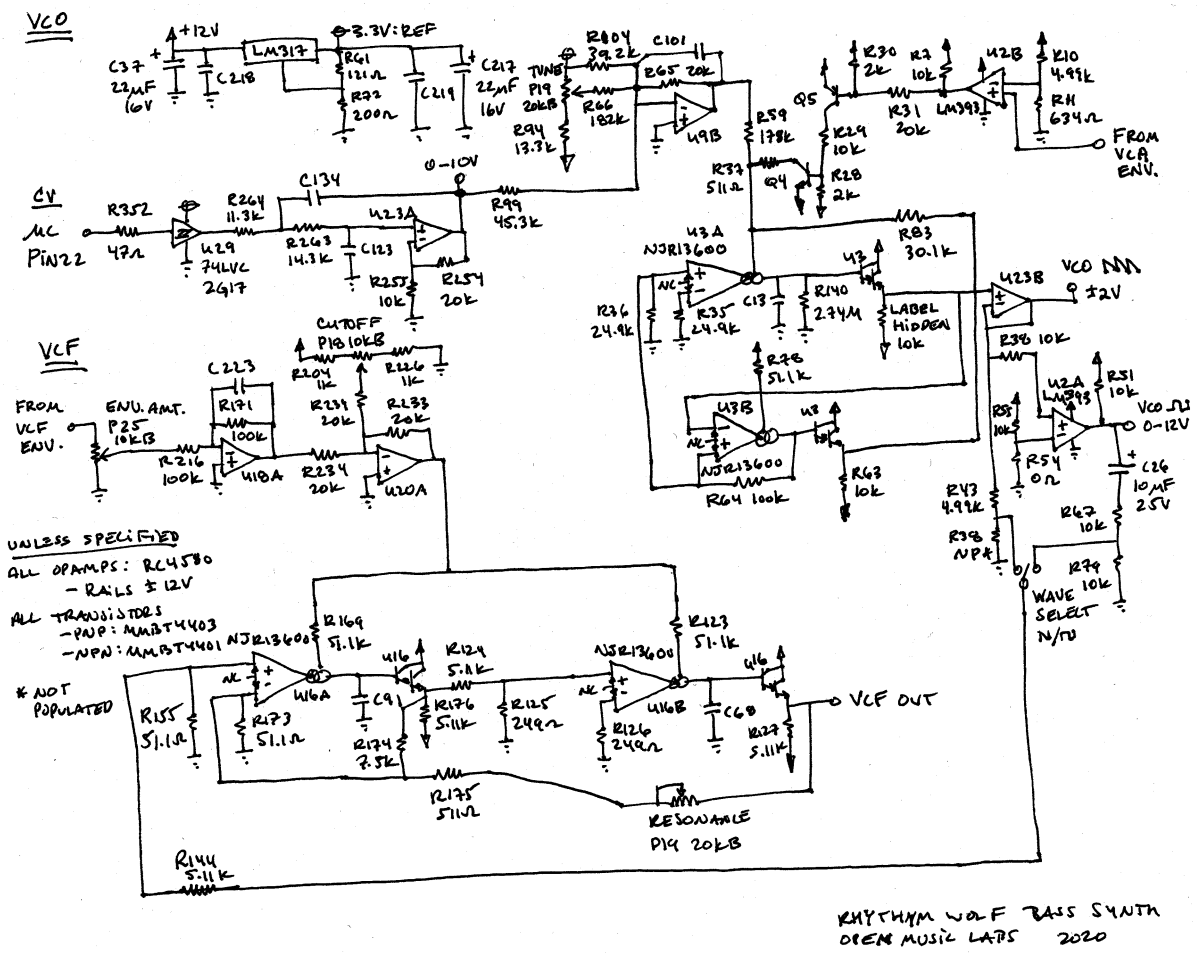
Videos

i wanted to document the before/after sounds to give folks an idea of what the mods did. They are recorded with the direct out in the left ear, and the discussion in the right ear. right now i only have the before videos ready:

1. Stock sound discussion - part 1
2. Stock sound discussion - part 2

Bass synth VCO & VCF

[Original Schematic]



VCO

The VCO is a linear design, with an exponential control voltage (V/Hz), which is a reasonable design decision considering how small of a range the VCO needs to operate over (3.5 octaves - 1:10 ratio of low:high note). This means any drifts "should be" linear, and can be compensated for with a simple tune knob turn. They buffer the PWM signal with a reference voltage, and send it to a mixing stage, which then goes to the OTA for waveform generation. Unfortunately there are a lot of nonlinear drifts in the remaining stages, which can not be accounted for with a simple tune knob. For example, the current into the OTA is not linearly related to the CV, as the OTA input is 2 Vbe off the negative rail, and those Vbes vary with current and temperature. Replacing R65 with a transistor would fix this, but it would also invert the relationship of CV to frequency, and i'm not sure if the RW tuning software could overcome this. The next major issue is the existence of R140. this causes a nonlinear drain on current, which is amplitude dependent, and the amplitude at this point varies with the output buffer Vbe drops. the only reason i can think of for this resistor, is to discharge the cap when the VCO is shut off. The only problem with this is that the Vbe drops in the output buffer make it so that the VCO output is not at zero when this happens (more on this later).

The PWM signal is at 731Hz (a sub-multiple of 48MHz) which is quite low. There is a lot of ripple after the filter. The filter has a 2ms transition time, with a few milliseconds of ripple afterwards. Any portamento

happens on the digital side, but a pot and capacitor could be added after the filter to have manual control. In order to get 1cent resolution over a 1:10 ratio of frequencies, 14b resolution is required on the DAC, which means the main PWM clock is probably running at 12MHz, and the low frequency output is just to achieve this level of accuracy. a dual PWM scheme could have gotten the same accuracy with a higher frequency (and the buffer they use is a dual unit already!).

The VCO is a pretty standard dual OTA setup, which could be easily converted to a triangle wave, if so desired (remove R63). This would drop all notes down one octave. For both this and the VCF, the 13600 is an odd choice over the 13700. The 13600 send a copy of the bias current to the output buffer, which helps match the buffer impedance to the OTA output impedance. Unfortunately this makes the buffer input current higher than it could otherwise be. Swapping them out for 13700s would help. i'm a bit confused as to why the buffer was used at all in the VCO, when the signal goes straight to an opamp buffer in the very next stage. The amplitude is set by R64,36. This is +/-2V, which is a fair bit of overdrive for the transistors at the input of the OTAs. This isnt a big deal on the one used as a comparator, but the other one will have leakage currents associated with this, which will effect timing. Putting a resistor between R64 and R36 would fix this, dropping the overdrive to 0.6V. R63 should be removed, as it provides a leakage path for the frequency setting current, and is not required for the reset cycle. Speaking of which, the reset cycle is quite long (100us), which adds to the nonlinearity. There really isnt much that can be done about this, other than going to a different reset format (JFET across the integraction capacitor). The square wave is made with the comparator U2A. PWM could be added here by removing R54 and adding a pot in its place.

One other oddity of the VCO, is that there is a comparator on the VCA envelope, to turn off the VCO when the envelope gets below a certain level. The problem here, is that the VCA is not shut off at the same time, and just continues to amplify the DC voltage the VCO spits out when it shut off. So the output of the VCA has this long DC tail on it, which adds to the noise of the synth. Also, the LM393 could have been tied to -Vcc, and the internal transistor used for this purpose, and Q4,5 and all those resistors could go away.

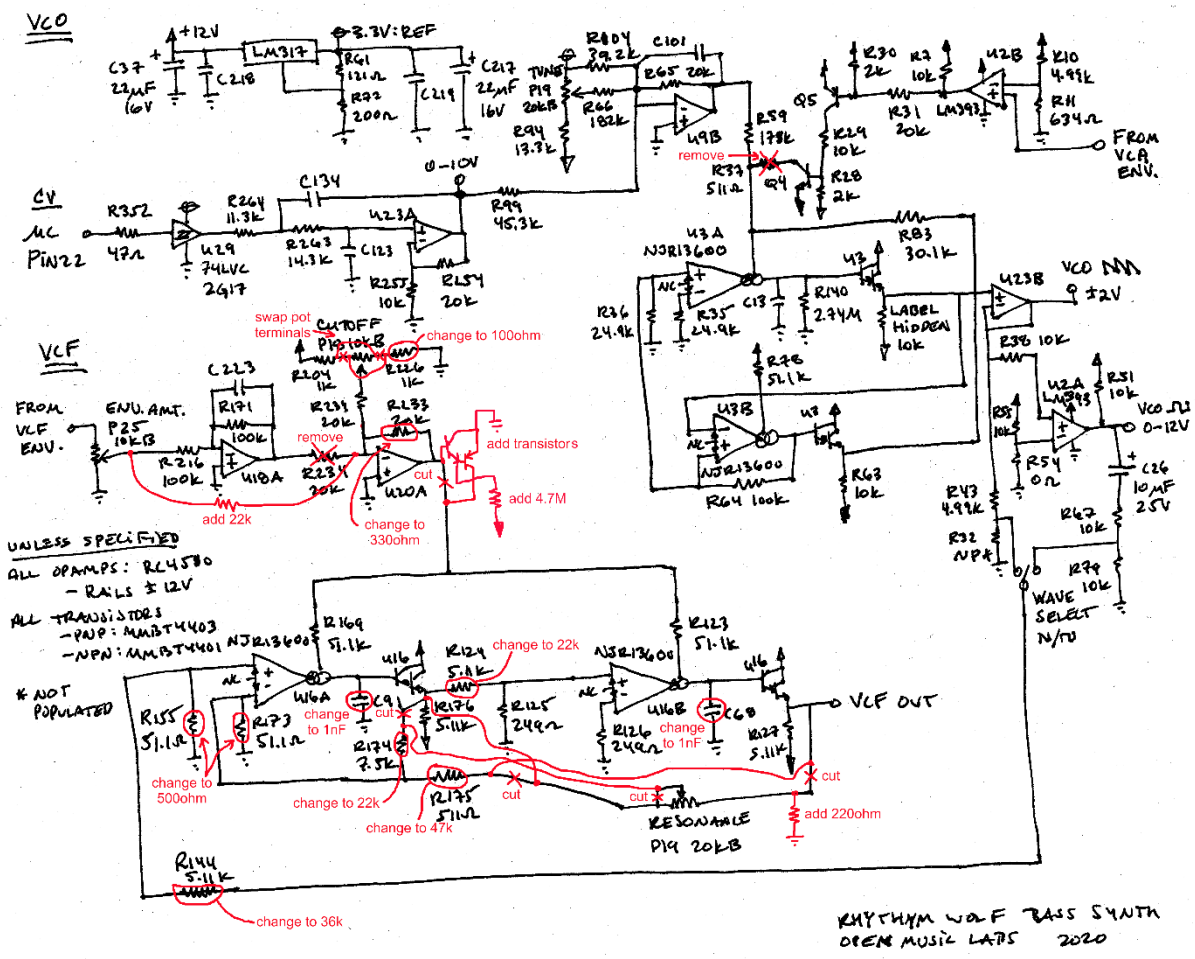
The tune knob has its bottom leg tied to -Vcc through a resistor. This completely defeats the point of having the reference voltage on the other side of it. Any powersupply drift now ends up in your VCO frequency. That should have been connected to ground.

VCF

This really needs exponential control, but other than that, its a pretty straightforward SVF design. Resonance is taken from the final stage, rather than the usual middle stage, and this reduces the low frequency amplitude with resonance, and shifts the cutoff frequency with resonance setting. The gain between the two stages is not identical, which leads to poor SNR as one signal is kept very small so that the other doesn't become too large.

Mods

[Modified Schematic]



VCO

i mostly left this alone, as there is no easy way to fix the major issues with it and not mess up the synth's factory tuning. You can replace R65 with a PNP transistor to make U9B a current source, which would improve the situation, but then you still have the issue of leakage in the darlington buffers, so i decided to not bother with this section. i only removed R59 so the VCO would decay out naturally and not get muted in an awkward way.

VCF

This section required a major rework, and the SVF was converted to a more normal layout. This made the gain increase with resonance, and now there is severe VCF overdrive at high resonance settings. To mitigate this, R144 was increased to strike a balance between decent volume at low resonance, and too much distortion at high resonance. The VCA was modded to boost the gain after the VCF to accomodate these fluctations. I originally was going to leave it with the linear cutoff control, but it was really hard to play, so a simple exponential converter was added, which made a world of difference.

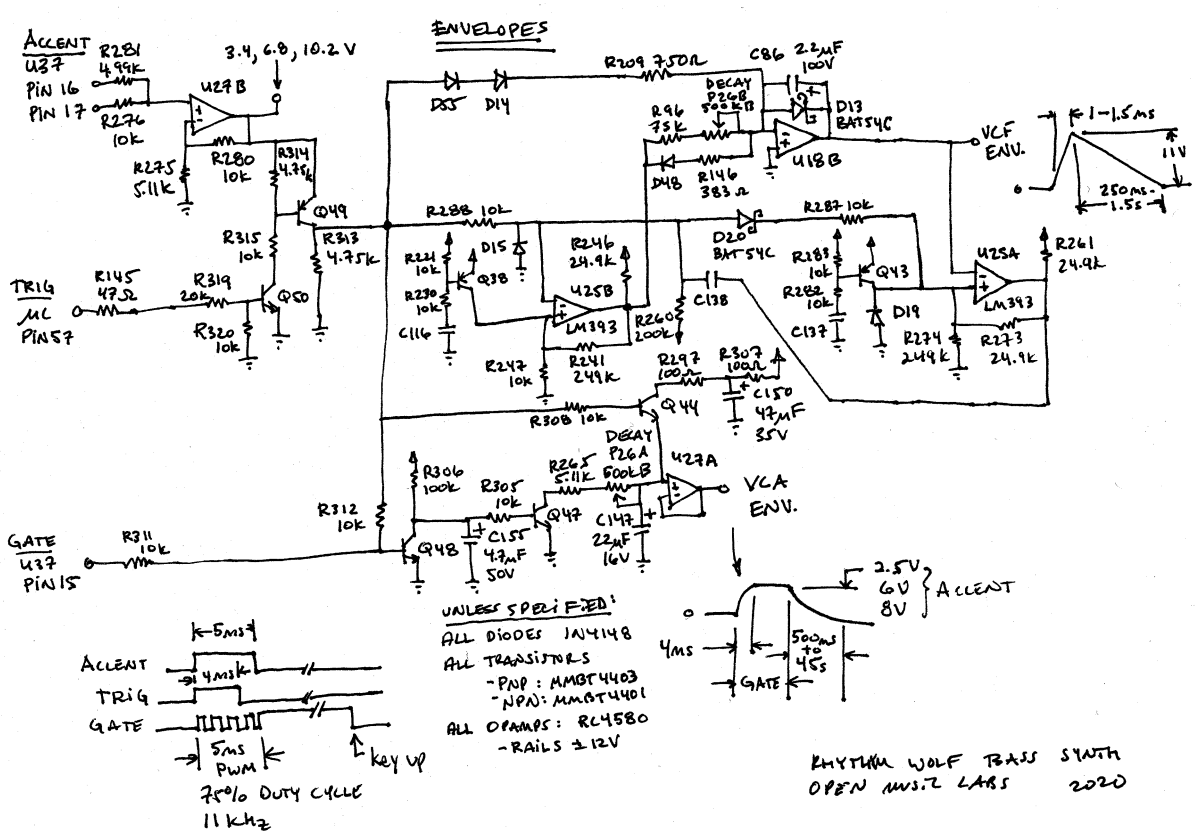
1. R233 reduced to 330ohms to reduce gain on U20A output.
2. R226 changed to 100ohms to reduce lower limit on cutoff frequency.
3. U20A polarity reversed to work as exponential controller. R234 removed and the ENV MOD knob directly couple with a 22k resistor to U20A. Cutoff potentiometer reversed.

4. U20A disconnected from OTA inputs, and two transistors and a resistor added to make a simple exponential current generator that is contrlled by U20A and feeds the OTAs.
5. C9 and C68 changed to 1nF to set the cutoff frequency in the right range.
6. R155 and R173 changed to 500ohms and R121 changed to 22k to set the two OTA stages to roughly the same gain, and to have the signals be of an appropriate amplitude at their inputs.
7. Resonance pot moved to first stage and second stage directly coupled via R175 which is changed to 47k to set gain.
8. 220ohm resistor added at bottom of resonance pot to set max resonance and R174 rerouted to resonance pot and changed to 22k to set min resonance.

Bass Synth Envelope

Overview

[Original Schematic]



It seems like the intent here was to make both lin and exp envlopes, with the exp going to the VCA, and the lin for the VCF. Since there is no exponential converter on the VCF, the lin envelope makes for a really weak filter sweep. It spends a relatively smaller amount of time in the lower range, near the cutoff frequency. There are 3 accent levels (the 4th accent level is "off"). These are evenly spaced between 0 and 10V. The accent and trigger signals arrive at the same time, with the trigger ending first. i'm not sure why there are 2 signals here, but its handy; if you wanted to add a manual accent knob, you would drive U27B with your accent voltage. The gate signal is even more bizarre, in that it has PWM on the first 5ms. im not sure if this is intentional, or just some artifact of the way the signals are multiplexed. i'm assuming the former, as it gets multiplexed the same way as the other signals. This PWM signal has no effect on the remainder of the circuit, as its over-ridden by the accent/trigger signal where they come together at R312. But, if it were PWM'd after

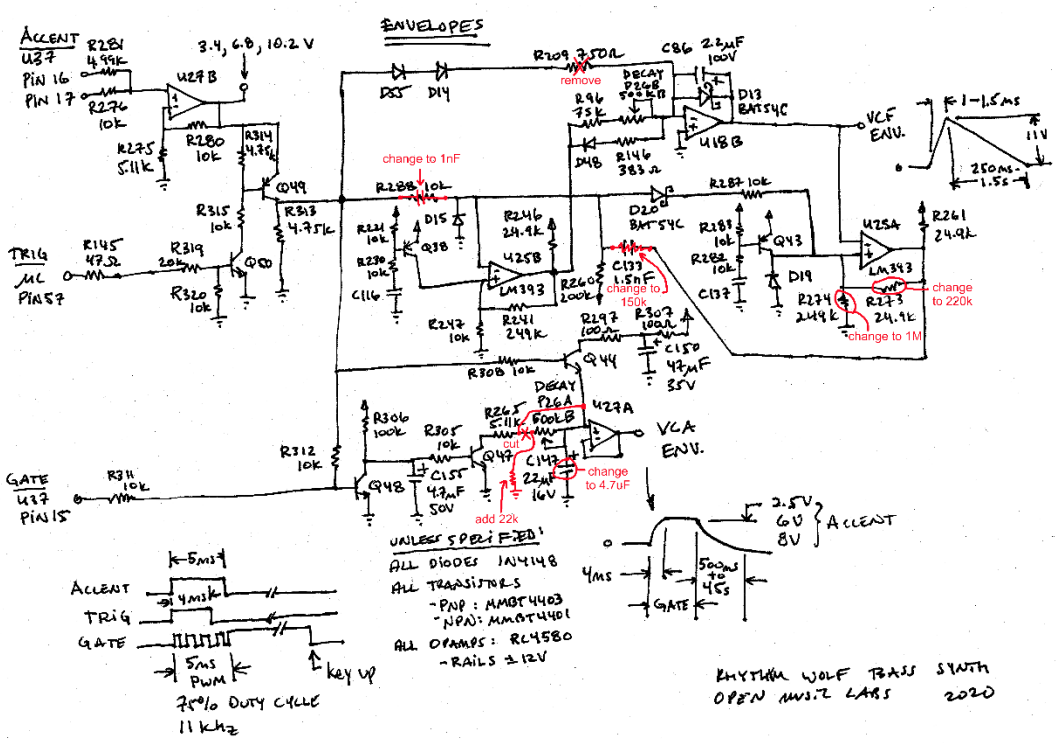
key-up, then you could have digital control of the decay time. The existance and size of C155 suggests that pwm was intended here. If you dont want your envelopes to be gated, you can just remove R311, and instead put a 10k to ground there. Q44 follows the accent voltage to charge C147, while at the same time Q47 is turned off to disengage the decay pot. This sets the min time before decay to 4ms (the trigger time).

The decay time on the VCA envelope is crazy long: 45s! Reducing C147 would fix this. Another odd thing here is that its a linear pot. Typically log pots are used for time and frequency control. There are mostly linear pots in the whole synth, but a number of different values, so getting a log wouldn't have been any extra cost. In this particular case, its a dual pot, so its an odd one out to begin with. Again, an interesting choice to have 2 different styles of envelopes, as it requires a dual pot.

The VCF envelope is less straightforward. First off, it has 2 comparators, each with startup pulses (Q38,43) on them to ensure a fixed state at startup. i'm not fully convinced these are necessary, and a quick pulse on the trigger would have done the same thing anyways. Both of these comparators merely follow the trigger signal, so its not too relevant. what it looks like is supposed to happen, is that U25B detects the trigger, and starts charging the integration capacitor, and U25A checks the envelope voltage, and shuts off U25B when it hits 10V. but, this doesnt happen, as the signal is not strong enough. there are a few different way to fix this. But, the upshot is U18B saturates at 11V, and there is a delay before decay begins as the integration capacitor needs to discharge quite a bit before U18B goes back into linear mode. R146 sets the attack time, and D48 ensures its only active during attack. D55 and D14 ensure that the gate signal does not discharge the capacitor. But, i'm not sure what this pathway is for anyways. It slightly adjusts the attack time, from 1ms to 1.5ms at higher accents. i'm not sure why you would want slower attacks for accented notes. Personally, i would re-route this signal to the comparison voltage on U25A, giving higher peaks on accented notes.

Mods

Modified Schematic



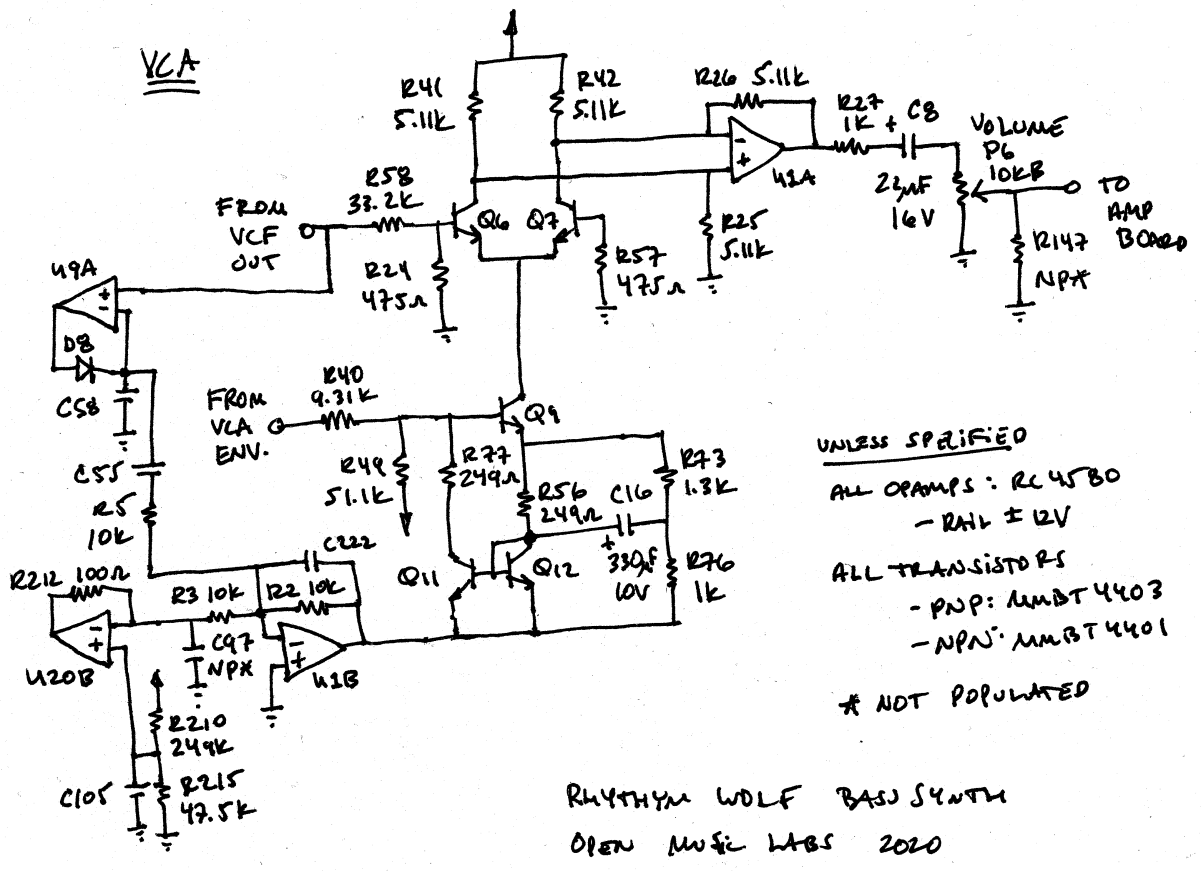
The two envelopes were mostly modded to fix the min/max decay times to have the knobs be in a more useful range. Although a linear envelope is a bit odd for the VCF, this was kept in order for the VCA and VCF to have different decay times. The VCF envelope could have been converted to exponential as well, but the way they change with settings is different, with the VCA extending much longer and it makes for some interseting sounds at long decay times. The most important fix was to change how the VCA envelope triggers, with it closing fully at the end of each note, rather than decaying out. This gives a difference between slid and normal notes that otherwise would not exist.

1. Change C147 to 4.7uF to reduce max VCA decay time.
2. Reroute R265 to C147 to shut off envelople immediately with note off.
3. Add a 22k resistor to the VCA decay pot to set min decay time.
4. Remove R209 to get rid of inverted accent control of VCF attack time.
5. change R288 to a 1nF capacitor and C133 to a 150k resistor to make the envelope peak detection circuit actually function.
6. Change R273 to 220k and R274 to 1M to set the envelope peak value to 10V.

Bass Synth VCA

Overview

Original Schematic



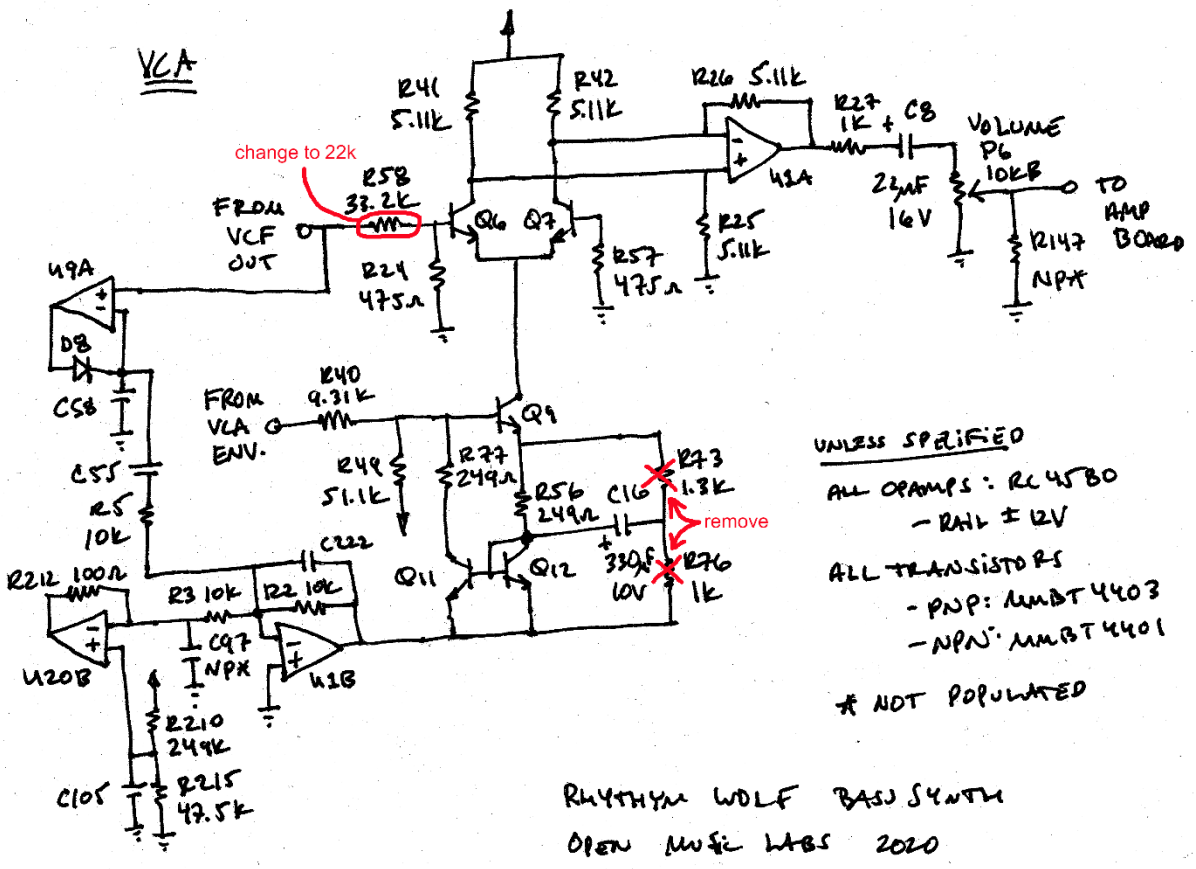
The VCA has compression on it. The main gain cell is a standard diffpair, but the tail current generator is unique. it has a current mirror that tracks the tail current, and then brings down the CV input as the gain goes up. My main issue with this, is that it completely defeats the purpose of having exp envelopes for the CV, as this compression completely undoes the exp behavior. This compression starts when Q12 turns on,

which happens when the current through R73,76 cause a voltage greater than a volt or so (Vbe + R56 drop). C16 puts a time constant on this compression, which is slightly backwards from what you might expect: fast moving increases in CV increase compression, and fast moving decreases in CV decrease compression.

The real oddities in this circuit are the inputs to U1B. U1B sets the lower rail that serves the tail current. U20B is a just a fixed voltage generator, but with its signal taken from the inverting pin, rather than the output. C97 is not populated, as this would make the opamp unstable by delaying the feedback signal. ultimately, this opamp isn't required at all, as a resistor from Vcc to the summing junction of U1B would have done the same thing. U9A is even more curious. This takes the peak of the VCF signal, and tries to adjust the lower rail accordingly. This could act as a sort of compression on the audio signal, as opposed to just the CV. This could be pretty useful for dealing with resonance related volume changes, or if there was a distortion circuit before this. But, since there is no resistor across C58, it just charges up and stays charged forever, so it can't respond to changing ampitudes. its just the peak value it has ever seen. i'm not sure what C55 is about, as it would only allow for a temporary compression. The oddest thing about this, is that the feedback is backwards, such that larger signals actually increase the gain of the VCA. Reversing D8 would fix this. But, utlimately the CV compression circuit mostly over-rides any effect this has.

Mods

[Modified Schematic]



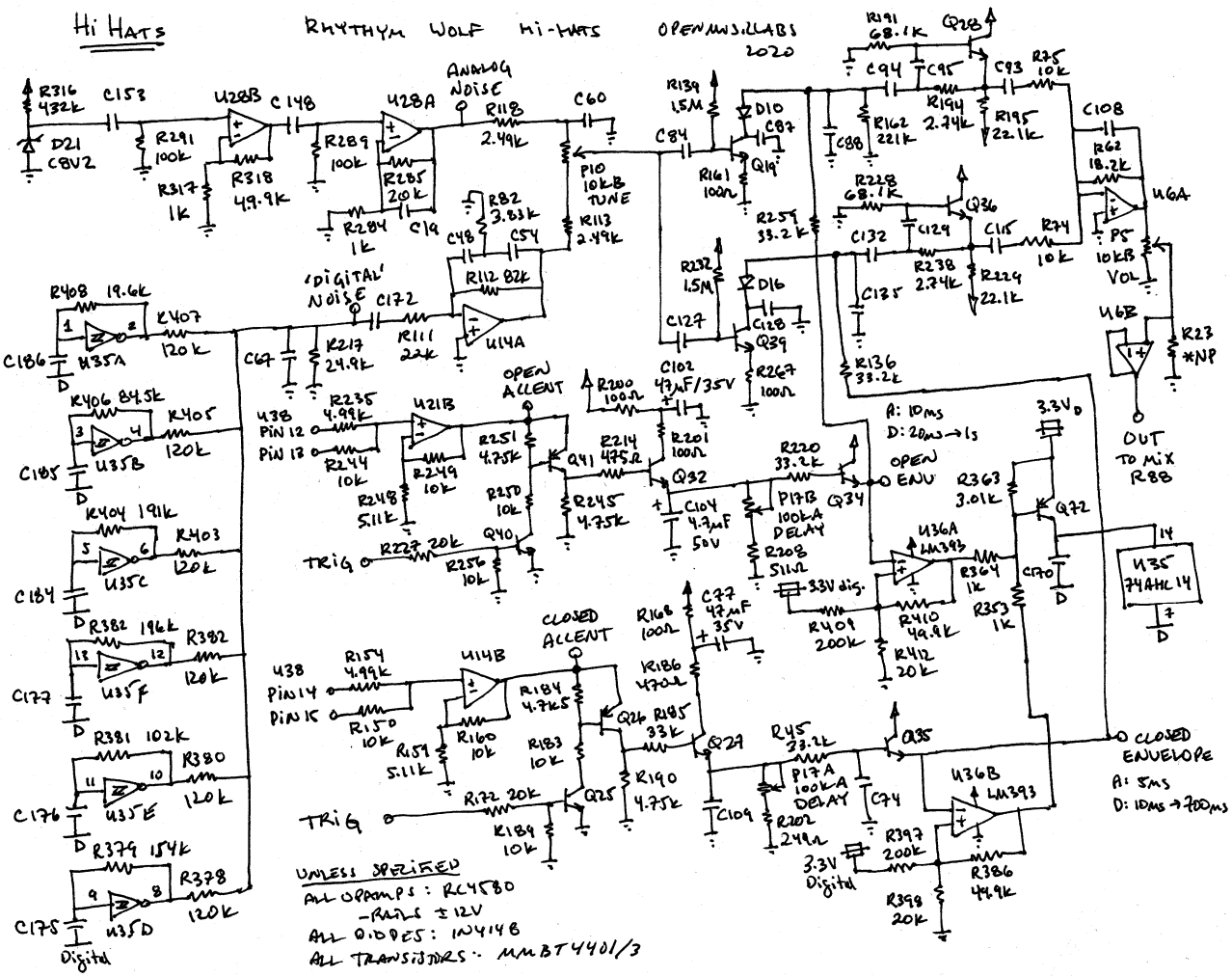
This has been converted to a standard linear VCA. Since the compression circuitry was non-functional, i just left it in place, and R73 and R76 were removed to make the current mirror function properly. R58 was

decreased to 22k, but this wasn't strictly necessary, it just brought the signal up to be a bit more in line with the other voices on the synth.

Hi-Hats

Overview

[Original schematic]



The RW utilizes both an analog and digital noise generator, and the tune doesnt actually tune anything, and rather sets the mix of these two. R118,113 limit the extent of this mix, and the relative volume of each is quite different. The digital noise is around 5Vpp max, whereas the analog is 500mVpp or so. There are two lowpass filters on the analog noise (C19,60) and the digital noise has a resonant lowpass and a regular RC lowpass (C48,54,67). This combined noise source goes to two pretty standard drum 'VCAs' (Q19,39 - one each for open/closed). The envelopes for these are generated with the same gate/accent setup as the bass synth. The envelope's peak value tracks the accent voltage. The decay time is half for the closed hat. The envelopes also go to comparators to shut off the digital noise source. The output of the VCAs go to a sallen-key highpass filter, using a single transistor. The output then goes to a mixer stage and the volume pot.

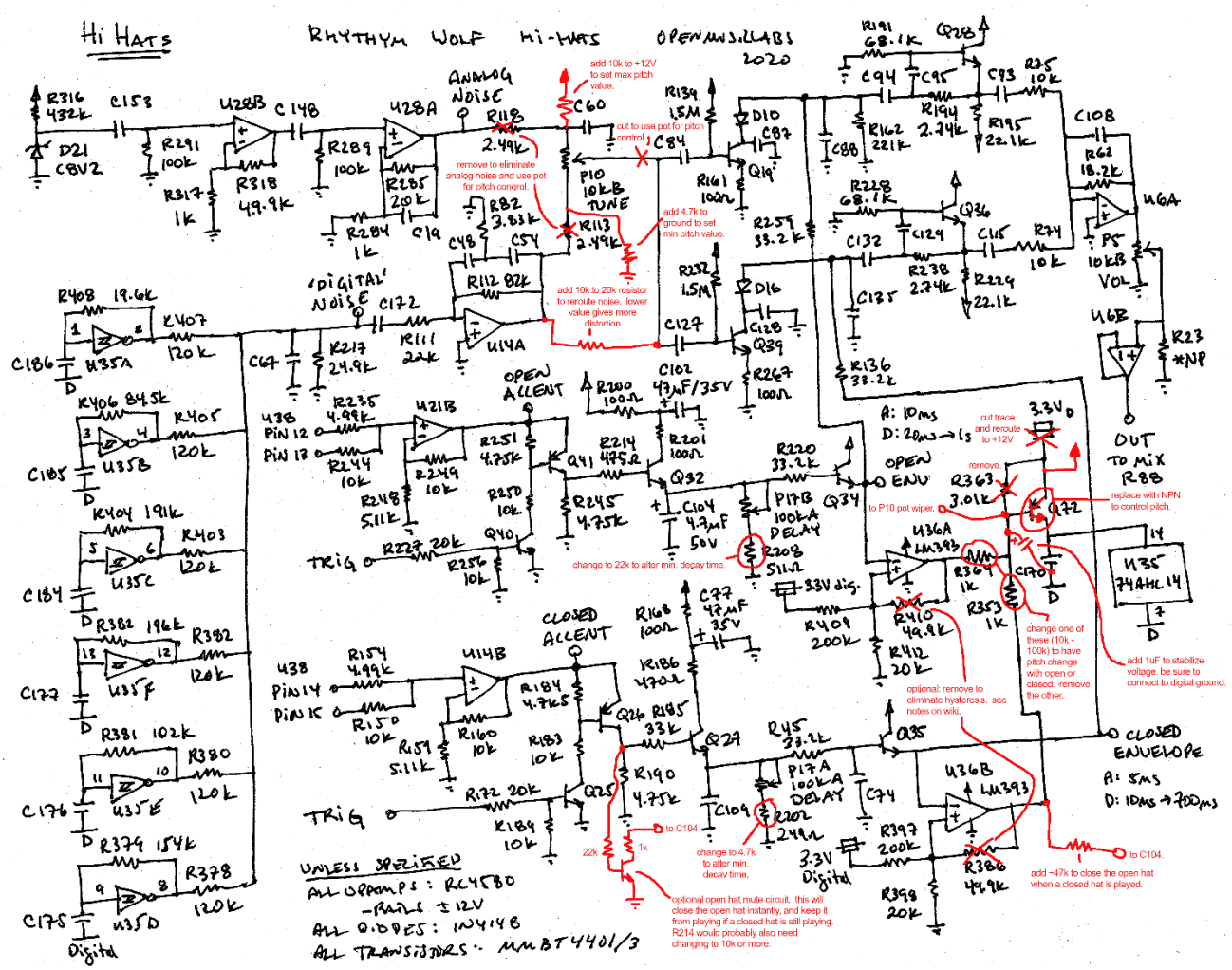
I find it odd that the digital noise source is connected to digital powersupplies. This means any noise in the digital side will be coupled into the hihat outputs. The amount of current drawn by the HC14 is really low, so i'm not sure it would have been splattering noise onto the analog supplies if it was connected to them. The

other thing is that the comparators do not work as intended here. Since there are diodes and capacitors in the VCAs which hold their charge after the envelope is off, the emitters of the envelope buffers sometimes do not get down below the threshold to shut off the comparator. A pulldown resistor is needed at the emitters to ground. i'm not sure this matters much, though. This section could be modified to apply a sort of 'mute' to the voices, so they end more abruptly. This might be good for the closed hat. Another oddity is that as the decay knob is turned down, it loads down the charging circuit, and the envelope actually gets completely eliminated, so the voice pretty much stops sounding at really low settings. R208,202 could be increased to basically set a more reasonable limit on the decay time.

All in all, i like the sound of this, and fixing the decay times is a pretty simple mod. i would like to be able to alter the pitch in some way, and i think the easiest way would be to alter the supply voltage to the HC14. These can run from 2V to 5.5V, and the pitch of the oscillators change with the supply voltage, as the hysteresis increases with increasing supply voltage. A little modulation on it might be nice as well. If i felt that the mute function was needed, then this voltage would be applied to the emitter of Q72, otherwise it would be easier to just remove Q72 and apply it directly to the supply pin of the HC14.

Mods

[Modified schematic]



i tried a number of different mods, and ended up using a small set of them. The ones i kept were as follows:

1. Minimum decay time - change R208 to 10k and R202 to 4.7k. This is very much to your taste, so i would suggest trying a few different values to see how they sound.
2. Tune knob rework - i eliminated the analog noise, as it didnt sound as good as the digital noise, and repurposed the tune knob to tune the digital noise. This is accomplished by varying the voltage to the 74AHC14 IC that creates the digital noise. To do this, remove Q72 and replace with an NPN transistor. The trace the 3.3V rail does not need to be cut if that pad is no longer used. Instead, run a trace to the +12V supply. i connected up at the comparator on the backside of the board, but any +12V point would do. Remove R363 and place a 1uF capacitor between the base of the transistor and digital ground. Digital ground can be found closest at C177. If you are not planning on doing pitch bending, i would suggest a 10uF capacitor instead, as it will be more stable. Run a wire to the wiper of P10.
3. To rework the pot, cut the wiper trace and wire it to the transistor on the bottom of the board. Remove R118 and replace it with a 10k to +12V. Remove R113 and replace it with a 4.7k to ground. You can use a smaller value for the 4.7k, but it might not work. i installed a 3.9k, and it sounded great for a while, but then the digital noise started to cut out at low pot settings. The 74AHC14 is only rated down to 2V, and a 3.9k supplied 1.5V or so. i'm not sure if it was the low voltage or something else i had done that messed it up. A 3.6k is the lowest that will work at all. This will allow a wide frequency sweep. You can reduce the 10k value to get higher frequencies, but it seemed high enough to me at 10k.
4. Reconnect the noise source to the VCAs with a 10k resistor from the output of U14A (pin1) to the input of the VCAs (C84,127). This value can be made smaller or larger to give varying amounts of clipping in the VCA output. Larger values will give a softer sound, but at some point will make the volume very quiet. Smaller values will give a harsher tone.
5. Pitch shift - you can make the closed hats a different pitch than the open hats by adjusting R364 and R353. When one of these resistors is engaged, it pulls down the voltage on the 74AHC14, lowering the pitch. So, if you want the closed hat to be a lower pitch, put a resistor in R353 and not in R364. To make the open hat lower pitched, do the reverse. The value of the resistor determines the pitch depth, with values from 10k to 100k being reasonable. A 47k would be a good starting point. If you use a larger value than 1uF for the transistor base capacitor in (1), the pitch shift will have a slew to it. its interesting, but not something i wanted as a fixed option. See discussion below for tweaks to this mod.
6. Open hat closure - to close the open hat when the closed hat triggers, run a 22k resistor from the output of U36B (pin7) to C104. This will discharge the envelope of the open hat whenever the closed hat is on, very similar to how the 606 or 808 works. Smaller values of this resistor will close the hat faster, but will also attenuate the hat more on subsequent retriggers when the closed is still sounding. An alternate version of this involves a transistor and a few resistors off the trigger signal. This latter method is far more abrupt, and mutes the open hat whenever the closed is activated. i didn't like it in comparison to the former version, so i didnt explore it completely. R214 would need to increase in order to keep Q32 from over-powering the pulldown resistor on the mute. In theory, it should be >10x the value of the pulldown resistor.

Other mod options

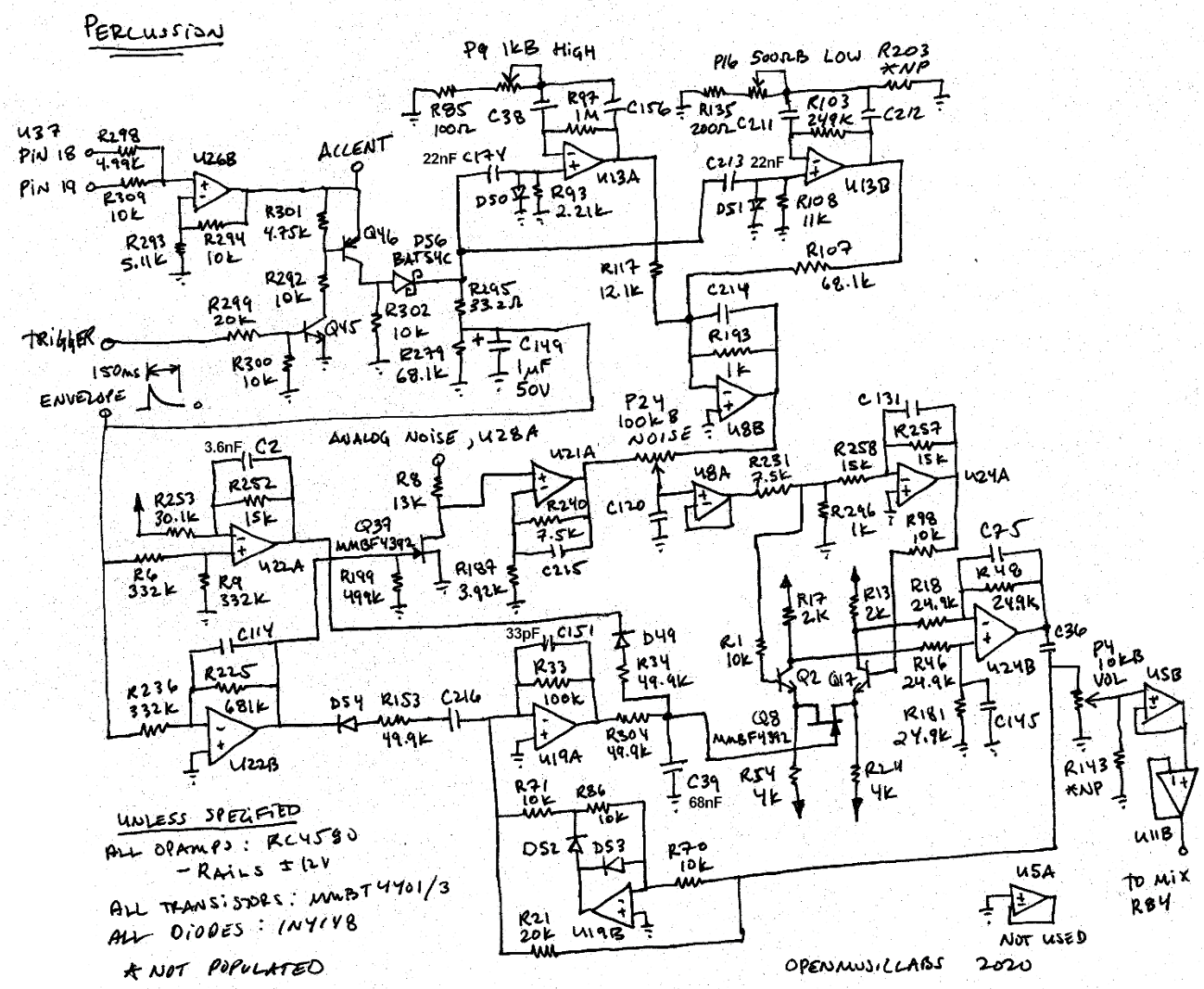
There is a lot of design space here, and i did not explore it fully. i tried adjusting one of the oscillators on the 74AHC14, as there were two set to 1kHz, but after changing it, it didn't sound as good, so i put it back. The main area that could use some work, is the settings on the comparators. i removed R386 and R410, which eliminated the hysteresis. This raised the threshold for detecting note off, which worked well in keeping the pitch from always hanging low. For pitch shifting purposes, it depends upon which note is being shifted

down. I found that triggering off the closed hat made the most sense, as it was the shorter of the two envelopes. But, this sends some voltage to the open hat envelope if you have the open-hat closure mod. Instead, cut the signal from the open hat to U36A, and run the output of U36B to the U36A input. This basically gives you an inverter with a buffered output that can be patched to the pitch mod. As it also inverts which hat is pitched down, you can also send the signal to the non-inverting input of U36A, remove R412, and tie the inverting input to something around 1V. i found that the pitch shift had a bit of slew, and also turned off halfway through note at times, which was rather cheesey, so i didn't use this mod. For open hat muting, you might not want to hold the mute too long, so a higher threshold would be preferred, which can be achieved by increasing R398. If the mute comes off too early, it won't matter as much as the envelope will have probably decayed to a very low level by that point anyways.

Percussion

Overview

[Original schematic]



This has three main sections. The first is a standard pinged resonant filter (U13A,B) for the low and high percussion sounds. The second is a noise source gated by a transistor (Q37), and the third is a compressor built around a few opamps (U19,24) and transistors (Q2,17,8). The pinged filters work as they should and

sound fine, but are limited by D50/51 such that accent does change the volume much. The gated noise source sort of works. It turns the noise on butonly moderately follows the envelope, and it doesnt really shut off. The JFET has a nominal on resistance of 60ohms, so with the 13k drain resistor, that should give -45dB attenuation for "off", which it might do, but that's not really enough. You can definitely still hear it at the ouput, even when the voice is not triggered. This is compounded by the fact that it goes through a compressor, which amplifies it when its "off". If you use the "howl" function, this noise is completely unreasonable. If you have the percussion voice turned up at all, the noise is constantly at the output, and its distorted, so it splatters all across the audible range.

The compressor is the most sonically interesting part of the circuit. It removes the usual exponential decay of the envelope, and instead keeps the amplitude constant for the majority of the range. This also has the effect of extending the apparent length of the notes, increasing the gain as the signal decays out. This makes a more "hollow" sound as compared to the typical "ping" of the resonant filter.

The compressor is driven by three signals: feedback from the output, a short attack spike, and the envelope. The envelope is a simple RC decay set by C149 and R279, and this is intended to serve as both an envelope, and a gate to turn the compressor off when no notes are played (U22A). Whether due to JFET variations, or just poor design, it doesn't do either on the unit i tested, as the level is too low to turn the compressor off. The attack spike (U22B) also has a failing, in that the blocking capacitor C216 has no discharge path, so it charges once on the first gate, and then never passes the attack signal again. The compressor's main input is from the fullwave rectifier (U19B) which is fed from the output. This has a gain of five, and a lowpass filter on the output. The heart of the compressor is a VCA composed of Q2,17,8, where Q8 is JFET used as a variable resistor to vary the gain of the VCA.

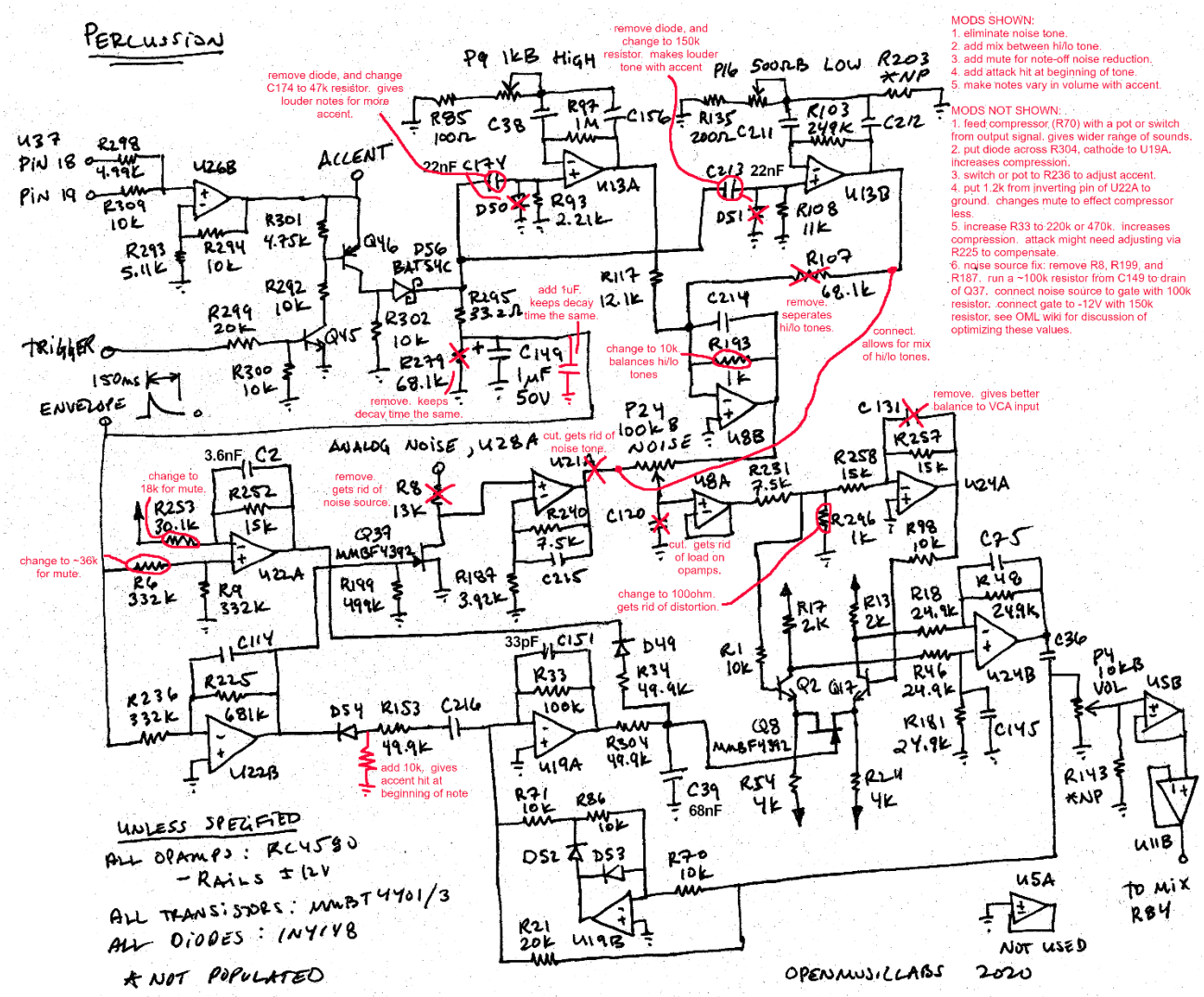
The inclusion of JFETs is a bit strange as they could have used a standard diffpair with a current source. The JFET in the noise gate also doesn't really do its job, and they would have been better off using the gating scheme from the hi-hats. If those introduced too much distortion, another diffpair could be used. It suprises me to see a six voice analog synthesizer with no trimmers inside. I have to imagine a bunch of things are just out of spec.

And now on to the electrical oddities:

1. There is a cap on the mixing pot wiper. Either end of the pot is driven by an opamp, so when its at the extremes it puts a hefty capacitive load on the opamps. This makes them less stable. A series resistor on the wiper would have fixed this.
2. The gate of Q39 is driven directly by an opamp output. This means that any excursion above 0.6V will destroy the JFET. Normal operation precludes this, but who knows what happens at startup. A series resistor would have fixed this, and reduced capacitive feedthrough on the attack phase.
3. D54 discharges C126, but there is no return current path, so it only happens once on the first trigger, and then never effects the circuit again. A current drain resistor is need on D54. This pathway is intended to gate the compressor open for the attack phase, but this is already done with the D49 pathway, and could have been replicated with a capacitor in parallel with D49.
4. U11B is completely redundant, and off in a corner of the board above the digital section. U5A is not used. i'm guessing they originally ran the signal all the way over to the corner and then back to the mixing section, but later decided this was not worth it, and added another opamp (U5), but then did not unroute the old opamp. So you have half an opamp going unused and a signal being double buffered.

Mods

[Modified schematic]



i made the decision for my box to completely cut the noise tone. It didn't sound good, was a lot of work to get sounding ok, and duplicated the same sounds i got from other voices. Instead, i used the mix pot to mix between the hi/lo tones. This gives a much wider range of sounds. This involved disabling the noise source so it wasn't splattering all over the circuit, and cutting the trace to the mix pot. On mine, i cut both ends of the pot, and wired directly from the outputs of U13. But, its easier to just cut one side and adjust the volume on U8B. C120 is removed to keep the opamps from having to drive a large capacitive load when the pot is at the extremes. R296 needed to be reduced as the levels were much hotter now. 100ohms was a good value to put it just at the edge of distortion across the whole range. You can go a bit smaller if you want it perfectly clean, but the SNR gets worse.

The next thing i did was mod the hi/lo tones so they are louder with more accent. i think it was originally done as a fixed volume because the mute circuit also did envelope control. But, once i got the mute function going, i decided i liked the variable volume effect anyways. This is done by removing the diodes at the inputs of U13, and replacing the capacitors with resistors. i picked the values to give a bit of distortion at the highest level. the resistors can be increased by 20% or so to eliminate this, or reduced by 20% to increase this distortion. Due to these capacitors being replaced with resistors, the decay time was made shorter. Its ok, but the notes lost a bit of that hollow sound, so i removed R279 and added another 1uF cap to bring the decay time back to where it was. Using an SMT capacitor, i was able to just replace R279 with this cap.

Next up was the accent circuit, which just needed a 10k drain resistor for C216 to discharge through. This gives a click at the beginning of the notes, which is louder with more accent. i'm not sure i fully like it, but it can be easily switched out by disconnecting the drain resistor (or any other method of blocking the signal).

Finally, the mute circuit, which probably worked on their bench but not in production due to the wide spread of JFET parameters. i measured the JFET active range for my VCA compressor, and it was -4V for full off, and -3V for 1/2 gain. Full gain was up at -0.5V. i modded R253 so the output was around -8.5V with no signal. This, mixed with the 0V output of the fullwave rectifier, gives -4.2V, which ensures that the VCA is off when there is no signal. This greatly reduces the background hiss of this voice. Next, i decreased R6 to 36k to increase the envelope. This allowed the compressor to open up, giving longer notes that were louder. This is very similar to the original sound, as the envelope was not working at all before modding R253. If you put a 1.2k from the inverting pin of U22A to ground, the gain increases and gets rid of the envelope effect entirely, just gating the VCA.

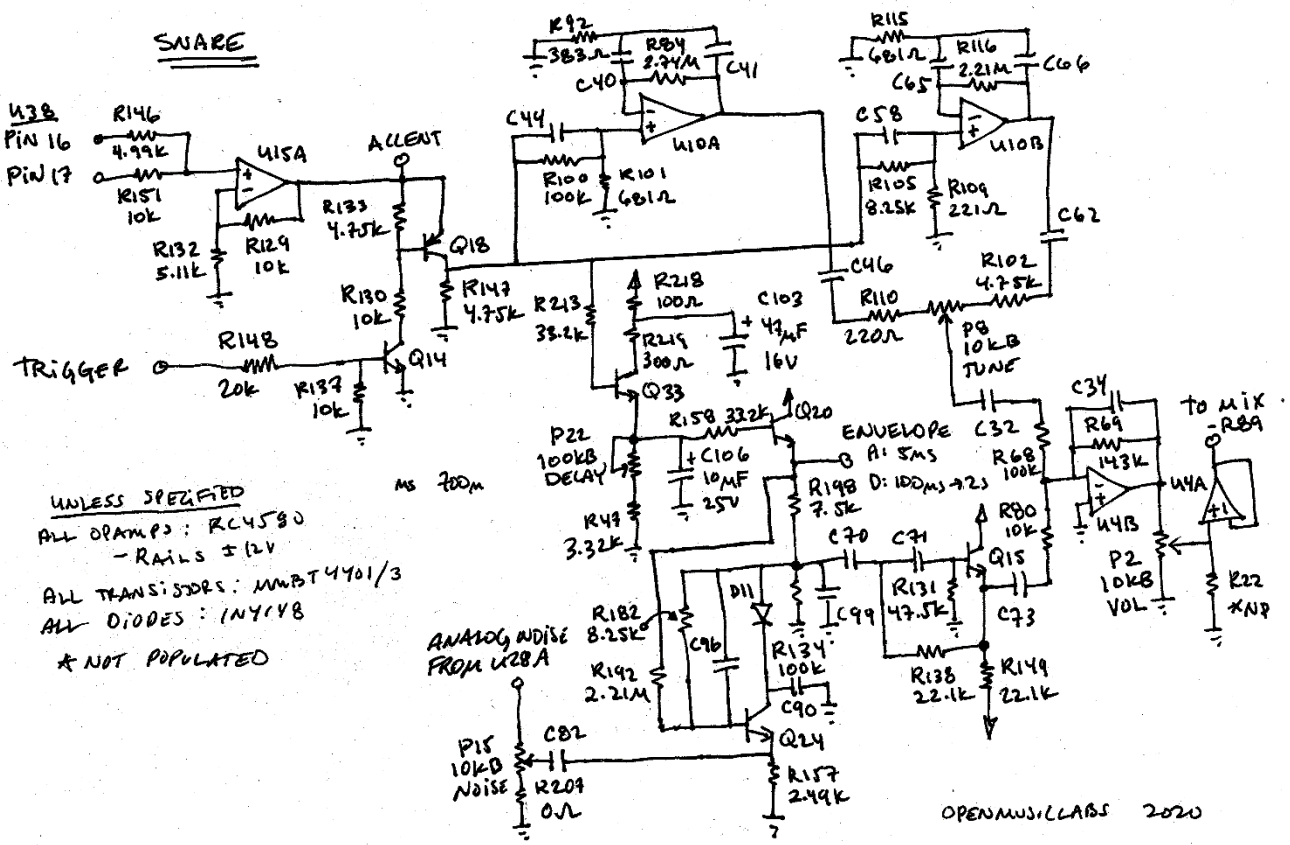
Other things i tried, but didnt want to add switches/pots for at this time:

1. Put a pot or switch on the signal going from the output to the compressor. A pot here is great. For testing, i just wired from the volume pot wiper, and was tempted to leave it like this. This allows for a really wide range of sounds from the voice. Without the compressor, they are very much just resonant ringdowns, very sharp and short. With the compressor, they get a nice hollow sound to them.
2. Put a diode across R304. This makes the compressor turn on instantly, which sounds pretty good when there is no accent added. It also increases the compression effect, but distorts the beginning of notes some.
3. i tried higher compression gain settings (increased R33), and they were fun, but ultimately i decided it was too much for a fixed setting, and they distorted a lot. A pot here could allow for some fun variations.
4. Fixing the noise source. This is possible, but complicated. Basically, the JFET needs to be wired differently. Currently, the envelope is sent the gate, and noise is at the drain. But, due to the nonlinear nature of the gate input, there is only a small window where the JFET is active, and the envelope just gates the signal as it doesn't spend much time in this regoin. The noise source should go to the gate, and the the envelope should go to the drain. Since JFETs vary a lot, you will need to test values to get the response you like.
5. First, start by removing R8 and R199 and connect a 100k resistor from the envelope at C149 to the drain. With a 100k drain resistor to C149, the envelope will shorten by ten percent or more, so the capacitance might need to increase by 220nF or more. If the attack time is too long a bit at this point, short out R295. Next, cut the trace between the the drain and U21A and put in a 0.1uF capacitor and 10k resistor to ground. This will act as a highpass filter and lessen the DC thump. Using a larger capacitor or resistor here will pass more low frequencies, but will also pass more DC thump. Finally, take a 10k resistor and tie one end to the noise source and the other to the gate. Then connect the gate with a 100k pot to -12V. Adjust the pot until the noise source is sounding right. Check the value of the pot and solder that value in place of the pot.
6. Second, check the output volume. It will most likely be distorting or too loud. If too loud, remove R187. If it is still too loud or distorting, decrease the value of the 10k resistor in the highpass filter on the drain. You will most likely need to increase the capacitor value accordingly. To further shape the noise, a capacitor can be placed between the gate and ground to cut out high frequencies.

Snare Drum

Overview

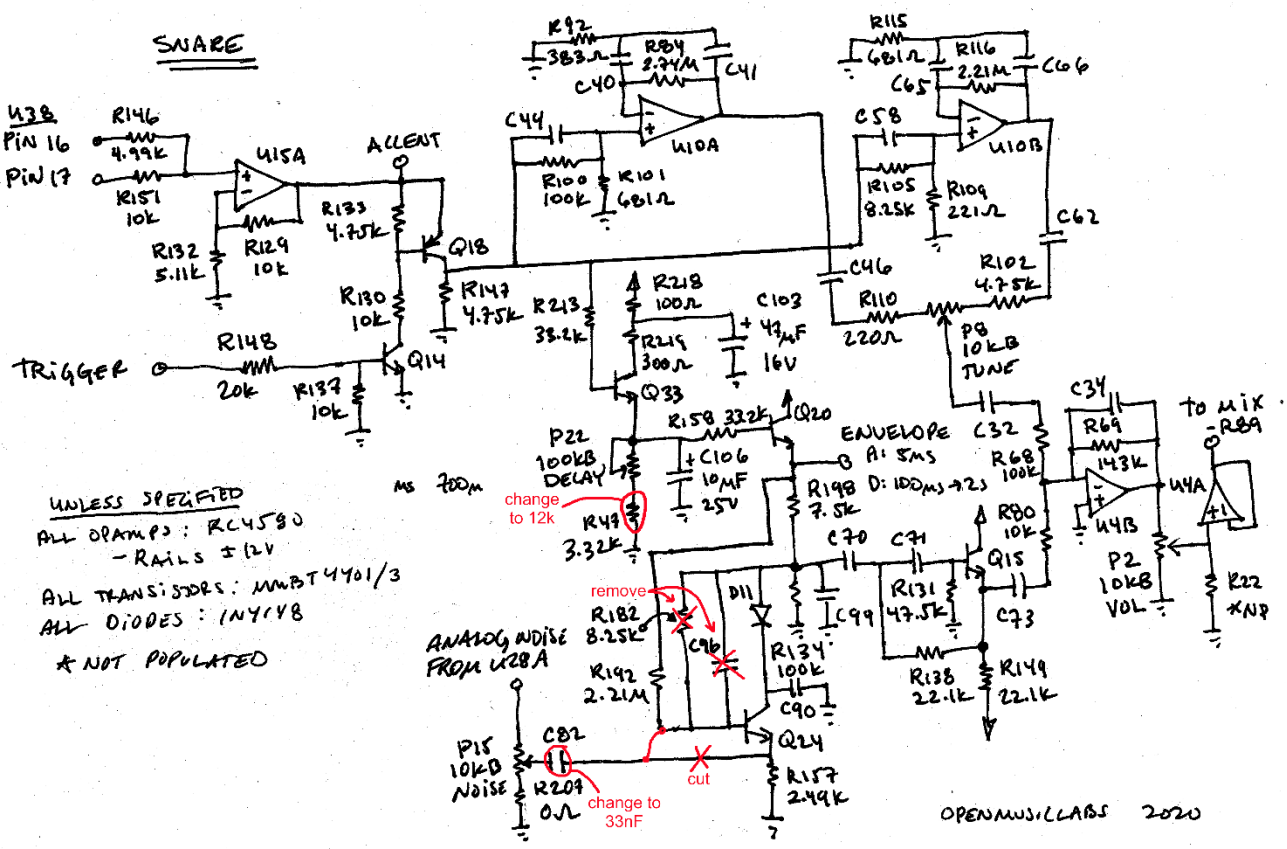
[Original Schematic]



This is very similar to the 808 snare (can't compare the pinged filters too much without the capacitor values), except the noise VCA is a common base amplifier stage, which doesn't work at all. The envelope pretty much acts as a gate for 90% of the time, and then does decay for the last 10%. Also, this topology has a higher voltage to operate since the emitter is higher (they could have biased it to -12V to eliminate this), so the envelope cuts out sooner for lower accent levels. Its close to twice as long between low and high accents. It would be pretty straightforward to convert this back to an 808 style VCA.

Mods

[Modified Schematic]



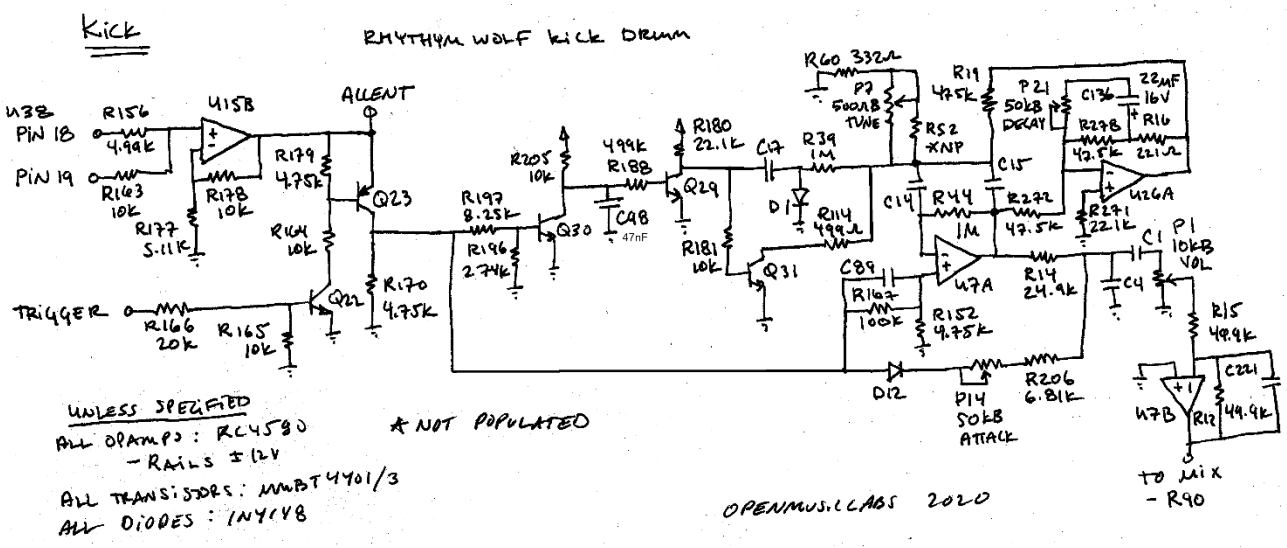
I didn't mod this part too heavily, mostly because i was running out of steam on this project. I set it back to mostly an 808 style snare.

1. Removed C82 and jumpered from the noise source side pad to the Q24 base with a 33nF capacitor.
2. Removed R182 and C96, which makes it the standard VCA.
3. Changed R157 to 100ohms.
4. Changed R47 (typo on schematic - shown as R147) to 12k to make the decay not shut the noise off completely.

Kick Drum

Overview

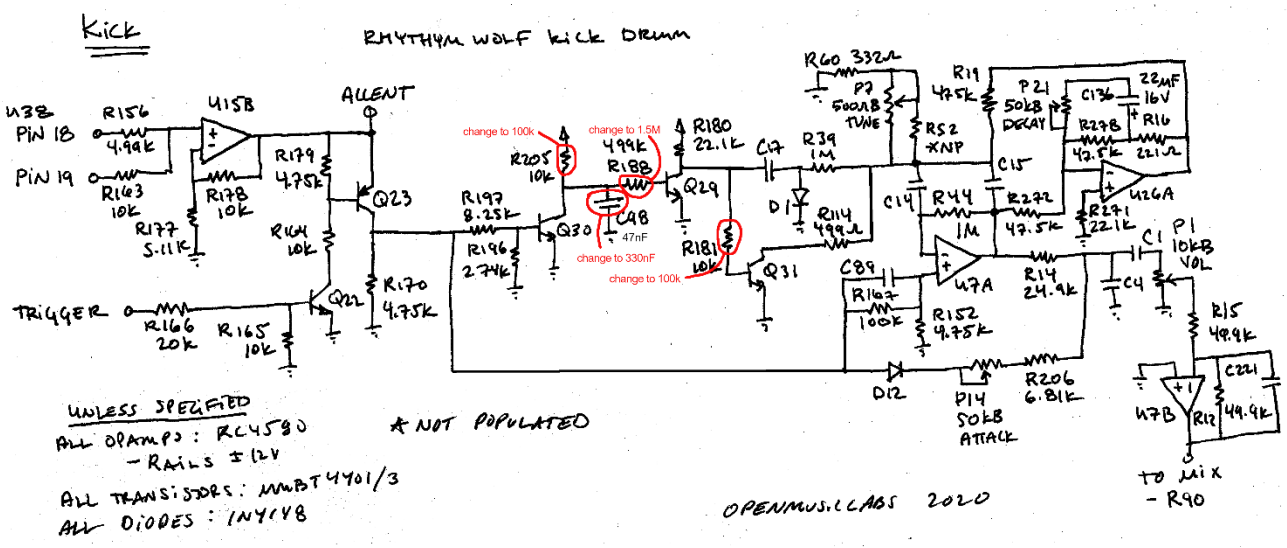
[Original schematic]



Its interesting to see the level they went to copy the 808: on the regeneration amp (U26A) there is a resistor from the noninverting input to ground. This is typically done to counter the offset voltage generatred by bias currents on the opamps, and this done throught the 808. But, this is the only opamp in the rhythym wolf that uses this technique. Also, the output pot goes to an inverting amplifier stage rather than a buffer, which is the only instance of this as well; not sure why. There is the addition of the "attack", which is a small pulse right at the beginning of the drum hit (through D12). The frequency of the drum voice does not shift down throughout the note, as it does on the 808 due to R205 and R188 being apparently swapped, making the decay too quick to be noticeable. Overall, this is the best designed part of the RW.

Mods

[Modified schematic]



i liked how the kick sounded, so i didnt make many mods, only adjusting the tone sweep. After doing the mod, i decided i liked it without the tone sweep for a lot of the range, but that the low end sounded better with the tone sweep, so i did a compromise (100nF C98). i would reccomend trying a few different values, as the sound changes pretty drastically depending upon depth and decay time. Ideally, you could put a pot on this parameter. i was thinking of repurposing the attack knob for this, but i wanted that as well, so maybe later i'll put one in, or at least a switch.

A word of caution with this mod: this part of the circuit is entirely transistor beta dependent, which means it will change with temperature and between synths. For this reason, i've picked some conservative, midrange values, so that when beta changes, the circuit is still in a reasonable operating point. This is particularly important if you pick large value resistors, as beta could shift and the transistors won't turn on anymore.

1. Change R181 to 100k. This adjusts the pitch bend amount. The low and high tones are 40Hz and 62Hz, respectively. With the pitch bend these can go up to 64Hz/83Hz at the beginning of a note. 10k is the lowest i would use here, and that gives the full bend and a sharp transition between the hi/lo end of the note. 47k also gives the full bend, but has a smoother transition. 100k only pitches up to 58Hz on the low end, and gives a longer transition. 220k goes up to 55Hz and gives the longest transition. going above 220k makes the effect pretty subtle. i decided to go with 100k as the really high sweeps seemed a little comical.
2. Change R188 to 1.5M. anything between 500k and 3.3M would probably be fine. The larger the value, the longer the sweep becomes for a given capacitor value. i went with a middle range value to ensure operation across temperature.
3. Change R205 to 100k. This can be anything from 10k to 1M, with larger values giving longer decay times. But, when R205 becomes greater than half the value of R188, the max sweep goes down, so there are diminishing returns. 100k to 500k is probably ideal. This can be replaced with a pot to make the sweep time adjustable. The 50k pot could be fitted here, with a 2.2k resistor in series to limit the current at max value. Ideally, a 500k pot would be used, as the current is still pretty high with the 50k/2.2k.
4. Change C98 to 330nF. This changes the sweep time with R205. for a 100k R205, values of 100nF to 2.2uF are reasonable. 330nF this will give a 40ms sweep time. The way i measure sweep time, is the time it takes for the collector of Q29 to go from 10V to ground. Around 5ms is when the effect is no longer noticeable, at 10ms it is subtle, and 50ms - 200ms are very noticeable. After that, its hard to hear the effect anymore, as the tone has decayed out. This voltage drops linearly, by increasing R188 it can be made more exponential.

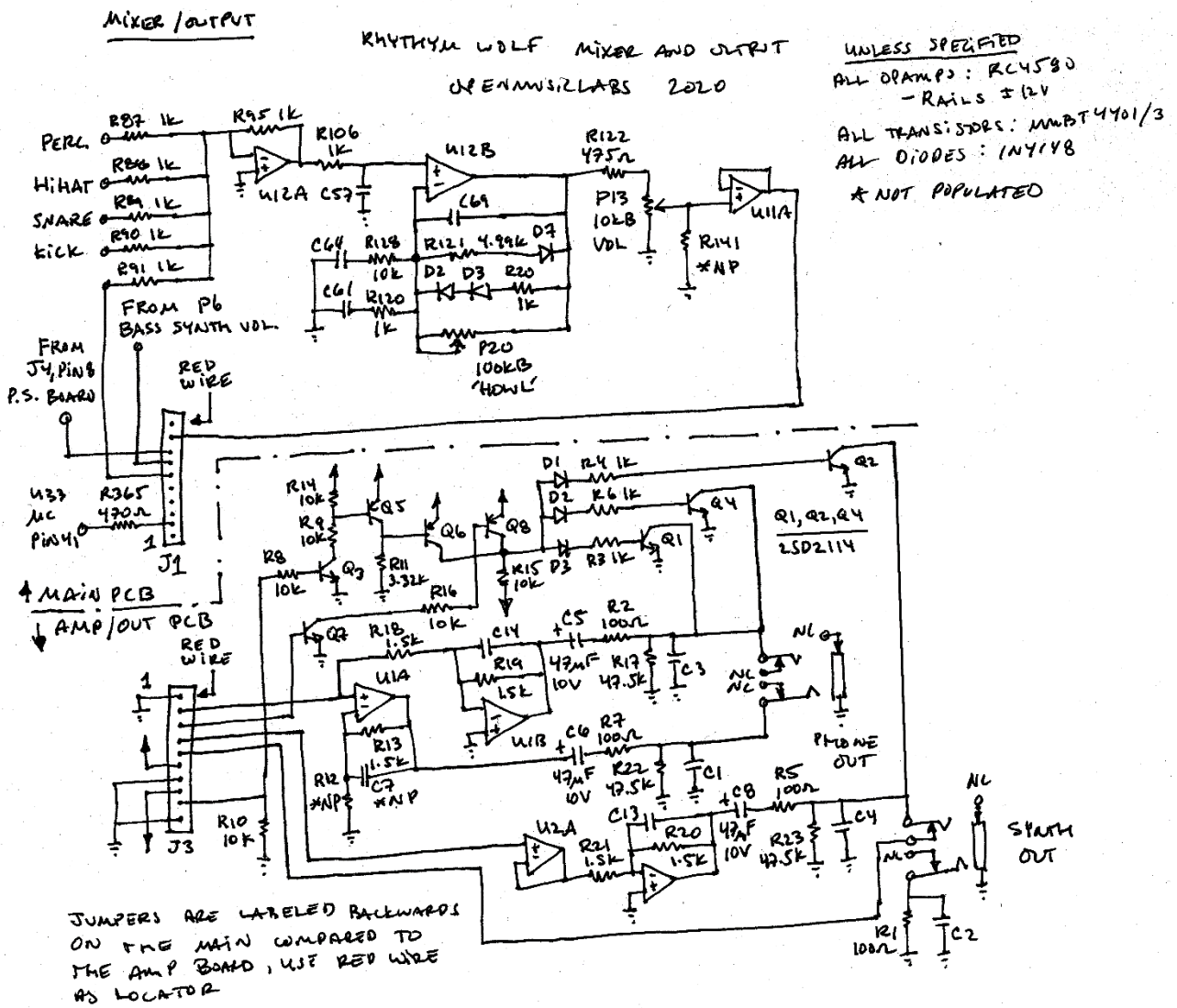
Comparison to the 808

With the level of detail they paid in trying to replicate the 808, its interesting to note that they picked a value of the TUNE pot to be 1/100 that of the 808. This has some severe consequences for the behavior, and if your goal is to replicate the 808, the only way to do so is to change P7, and R60,114 and C14,15. Due to the low impedance at this juncture of these components, the voltage level never gets large enough to pass enough current through R39 to pitch the tone of the drum. D1 never turns on. This pitch shifting can be accomodated for with the above mods, setting the decay time to 150ms. But, the pitch shifting will be slightly different, as the 808 mostly shifts pitch when the equivalent of D1 turns on, which only happens on positive excursions, so on the half cycle. So there are two different mechanisms for pitch shift, one at the beginning of the tone for ~10ms, and one over the duration of the tone, which is a much more subtle effect.

Mixer Section

Overview

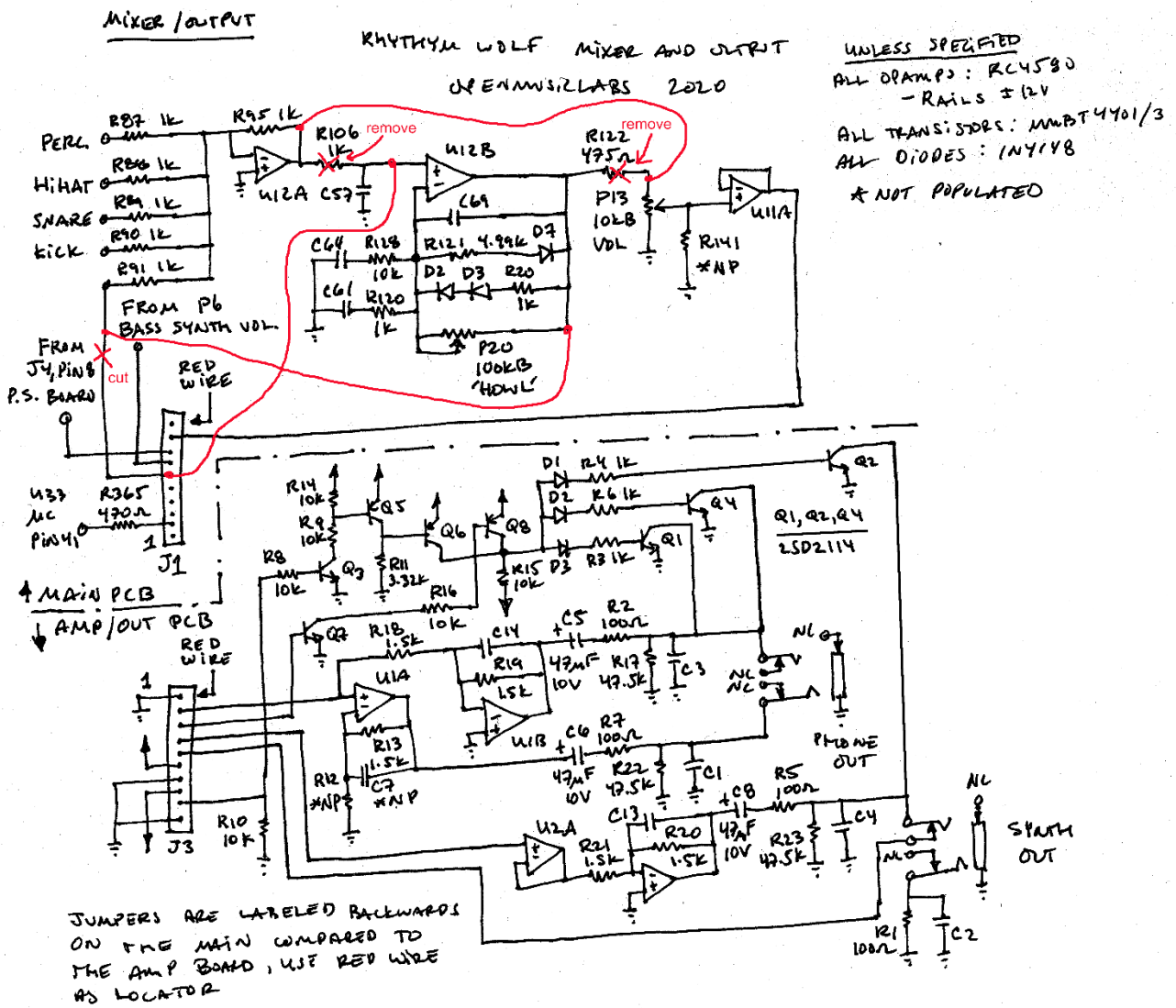
Original Schematic



The mixer section is a pretty straightforward summing amplifier followed by a differential output driver for the headphone output. The synth output is sent to a seperate jack before the mix, and is only included in the mix if no plug is inserted. There is a mute circuit on the outputs which is controlled by both the powersupply board and the microcontroller. This seems a bit redundant, as the microcontroller also controls the the analog rail generation, and i cant see why you would want to un-mute the output before turning on the analog power supply. Between the mixer and output driver is the "howl" distortion circuit. This has asymmetric clipping, a low-pass filter, and a less than 6dB/octave slope high-pass filter.

Mods

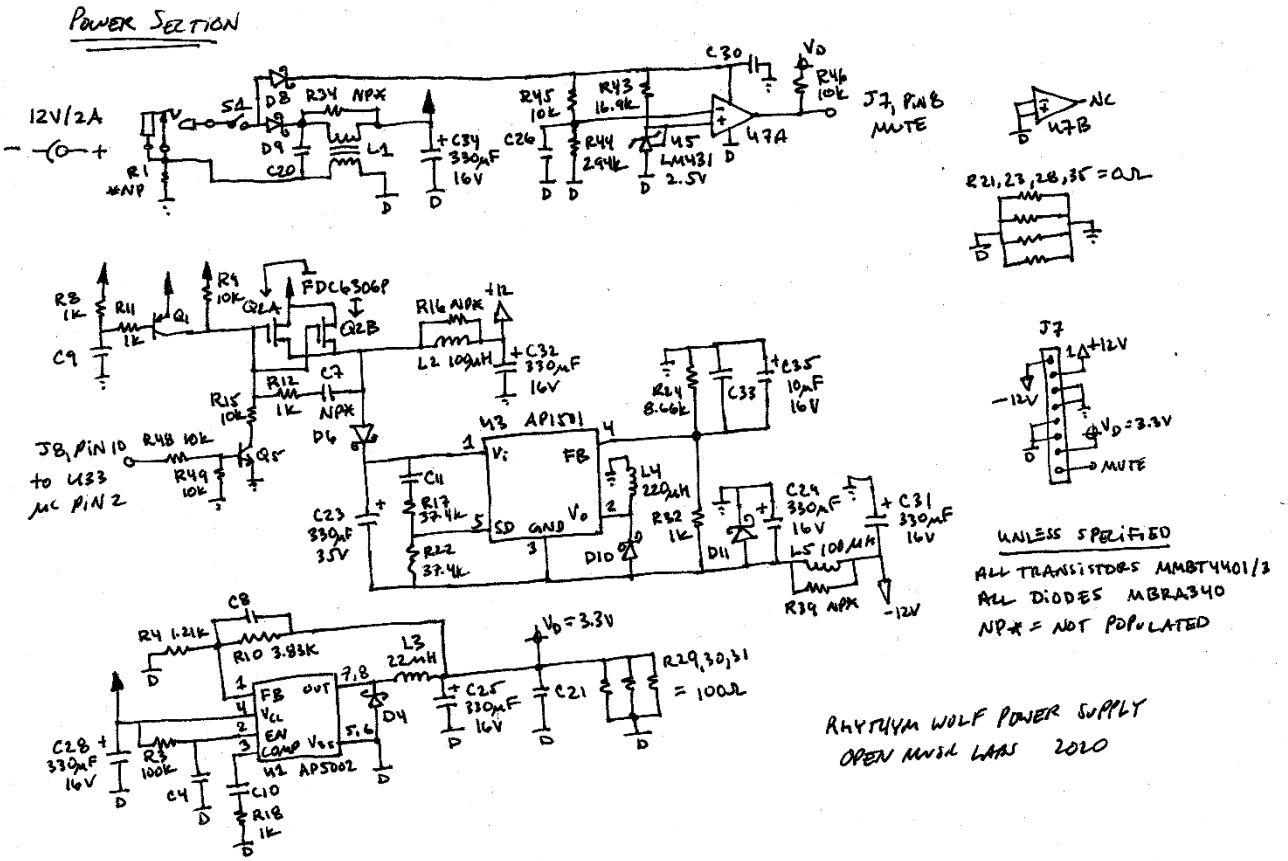
[Modified Schematic]



The howl function being applied to all voices makes no sense, and i would either use it on the kick or the bass. It was a tough decision, but eventually decided to put it on the bass synth. This was a simple matter of cutting the trace from the bass pot, removing R106 and R122, and then wiring the bass to U12B and back again to its normal input in the mixer. The output from the mixer then is wired to bypass U12B and go straight to U11A and the output. A similar thing could be done for the kick drum.

Power Supply Section

Schematic



The RW draws 0.5A from the 12V wallwart (its rated for 2A). This 12V switching supply wallwart runs through an LC filter for the iternal +12V analog supply line. It also supplies a switching regulator to make the -12V analog supply and the 3.3V digital supply. There is a comparator used to mute the audio output for a bit on powerup, and another signal back from the microcontroller to delay the start of the analog rails until the digital line is set.

French translated Version

Vue d'ensemble

Le Rhythm Wolf d'Akai a été l'un des synthétiseurs les plus universellement critiqués lors de sa sortie en 2014. Il crée des sonorités intéressantes, mais celles-ci sont souvent limitées à une petite gamme de réglages de boutons. D'autres problèmes tels que le bruit excessif et le décalage inamovible de l'oscillateur de la basse synthé n'ont pas contribué à améliorer la situation. Cependant, les critiques négatives sont probablement tout autant alimentées par la déception. Voici un combo, une boîte à rythmes analogique et un synthé basse avec séquenceur, qui semblait parfait sur le papier, mais qui a déçu lorsque les gens l'ont réellement écouté. Ils s'attendaient à avoir à la fois un 808 et un 303 dans une seule boîte, mais ont finalement obtenu quelque chose d'autre, quelque chose de moins. Maintenant que l'on peut obtenir un clone du 808 et du 303 à moindre coût, il se peut qu'il y ait de la valeur dans quelque chose qui sonne différemment. En raison de l'échec commercial du Rhythm Wolf (RW), on peut l'obtenir pour moins de 150 $ et peut-être le modifier pour obtenir un son agréable à sa manière. Ce projet est une tentative de voir si c'est possible.

Il existe également un fil de discussion sur Muffwiggler avec plus de discussions sur le sujet : RW Muffwiggler thread

Schémas

La première étape pour transformer le Rhythm Wolf en quelque chose d'autre consiste à comprendre ce qu'il est actuellement. À cette fin, j'ai réalisé une ingénierie inverse des schémas et les ai analysés pour comprendre où les choses ont mal tourné (et où certaines choses ont bien fonctionné). Mes excuses pour les schémas dessinés à la main, qui peuvent être quelque peu difficiles à lire. C'était beaucoup plus rapide de le faire de cette manière que de tout concevoir en CAO. Peut-être que quelqu'un remédiera à cela à l'avenir. Les schémas sont séparés en sections, et chaque section est liée à des schémas et à une discussion (Remarque : il n'y a pas de valeurs pour les condensateurs céramiques, car ils sont difficiles à mesurer en circuit):

11. Bass synth VCO et VCF
12. Bass synth Enveloppe
13. Bass synth VCA
14. Hi-Hats
15. Percussion
16. Snare Drum
17. Kick Drum
18. Mixer Section
19. Power supply Section
20. All Schematics (zipped)

NOTE: Il pourrait y avoir des erreurs dans les schémas (numéros de désignation incorrects, valeurs, etc.), donc si vous ne trouvez pas une pièce, cela pourrait être une erreur dans les schémas.

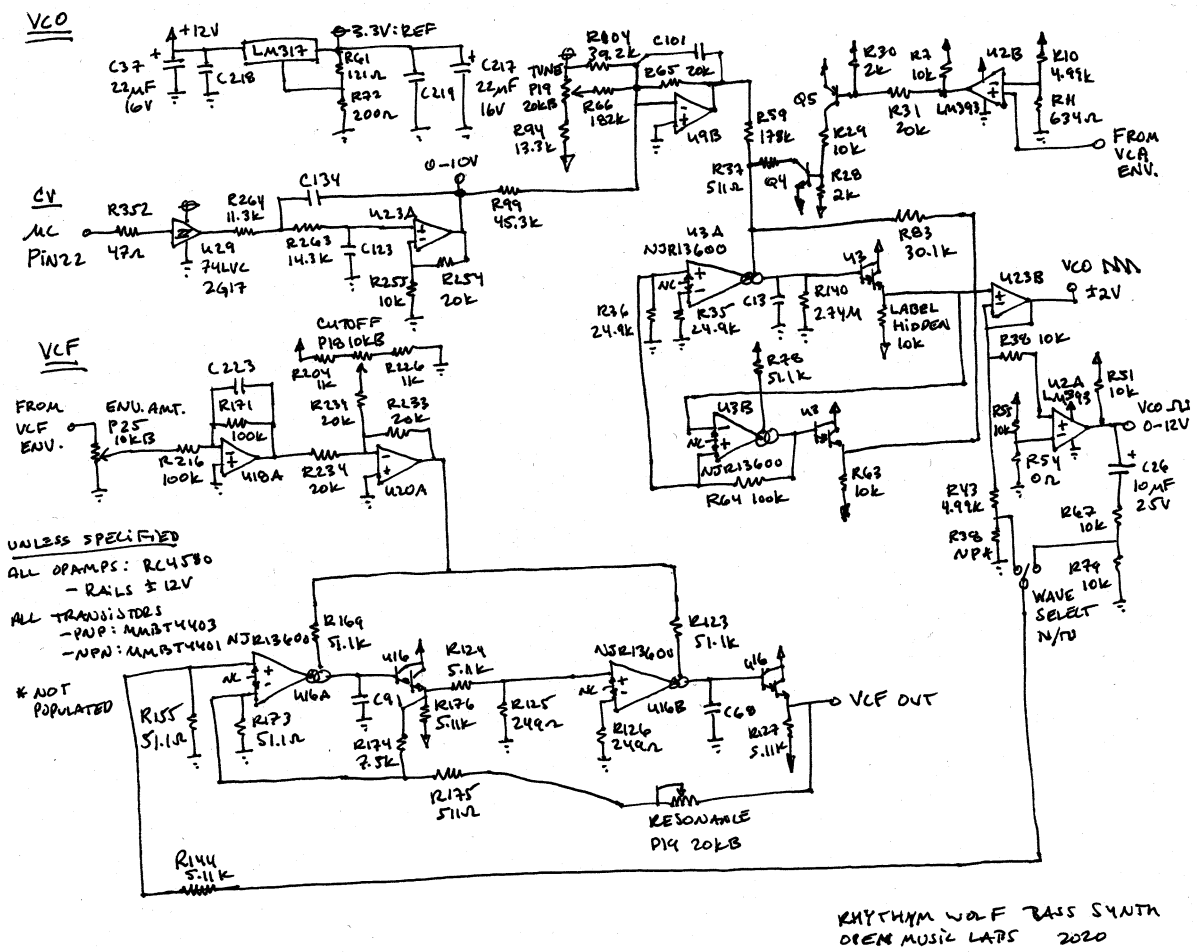
Videos

Je voulais documenter les sons avant/après pour donner aux gens une idée de ce que les modifications ont fait. Ils sont enregistrés avec la sortie directe dans l'oreille gauche et la discussion dans l'oreille droite. Pour le moment, seuls les vidéos "avant" sont prêts:

3. Stock sound discussion - part 1
4. Stock sound discussion - part 2

Oscillation & Filtre (VCO & VCF) du synthétiseur de basse

[Schéma d'origine]



VCO

Le VCO (Voltage-Controlled Oscillator) est conçu de manière linéaire, avec une tension de commande exponentielle (V/Hz), ce qui est une décision de conception raisonnable compte tenu de la petite plage sur laquelle le VCO doit fonctionner (3,5 octaves - rapport de 1:10 entre les notes basses et hautes). Cela signifie que tout décalage "devrait être" linéaire et peut être compensé par un simple réglage du bouton de syntonisation. Ils tamponnent le signal PWM avec une tension de référence et l'envoient à un étage de mélange, qui va ensuite à l'OTA (Operational Transconductance Amplifier) pour la génération d'ondes. Malheureusement, il y a beaucoup de dérives non linéaires dans les étapes restantes, qui ne peuvent pas être compensées par un simple réglage du bouton de syntonisation. Par exemple, le courant dans l'OTA n'est pas linéairement lié à la CV, car l'entrée de l'OTA est à 2 Vbe du rail négatif, et ces Vbes varient avec le courant et la température. Remplacer R65 par un transistor résoudrait ce problème, mais cela inverserait également la relation de la CV à la fréquence, et je ne suis pas sûr que le logiciel de syntonisation du RW puisse surmonter cela. Le problème majeur suivant est l'existence de R140, qui provoque une décharge non linéaire du courant, dépendant de l'amplitude, et l'amplitude à ce stade varie avec les chutes de Vbe du tampon de sortie. La seule raison que je puisse imaginer pour cette résistance est de décharger le condensateur lorsque le VCO est éteint. Le seul problème avec cela est que les chutes de Vbe dans le tampon de sortie font en sorte que la sortie du VCO n'est pas à zéro lorsque cela se produit (plus à ce sujet plus tard).

Le signal PWM est à 731 Hz (un sous-multiple de 48 MHz), ce qui est assez bas. Il y a beaucoup de fluctuations après le filtre. Le filtre a un temps de transition de 2 ms, avec quelques millisecondes de fluctuations ensuite. Tout portamento se produit du côté numérique, mais un potentiomètre et un condensateur pourraient être ajoutés après le filtre pour un contrôle manuel. Afin d'obtenir une résolution de 1 centième sur un rapport de 1:10 des fréquences, une résolution de 14 bits est nécessaire sur le DAC, ce qui signifie que l'horloge principale PWM fonctionne probablement à 12 MHz, et la sortie basse fréquence est simplement là pour atteindre ce niveau de précision. Un schéma PWM double aurait pu obtenir la même précision avec une fréquence plus élevée (et le tampon utilisé est déjà une unité double !).

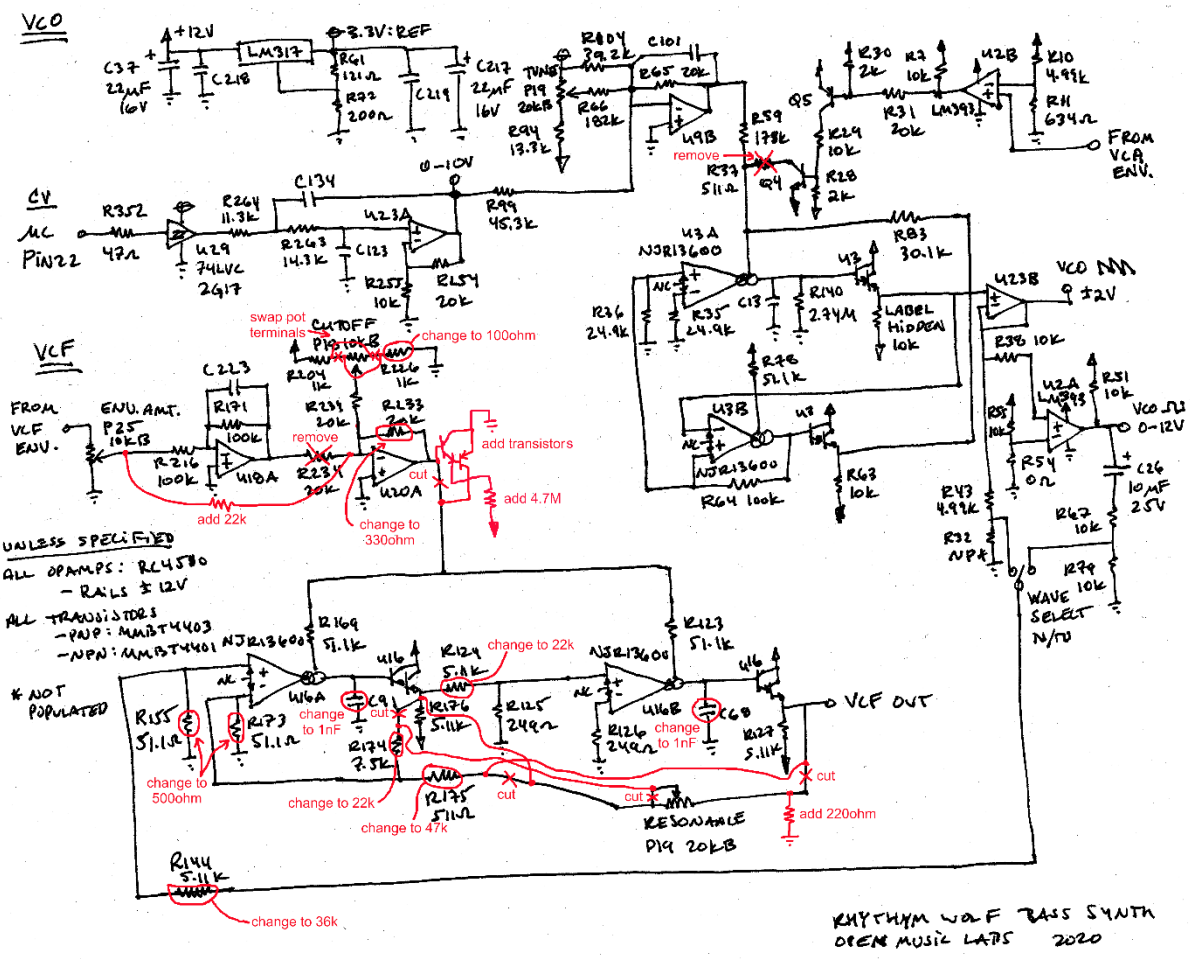
Le VCO est une configuration double OTA assez standard, qui pourrait être facilement convertie en une onde triangulaire si on le souhaite (supprimer R63). Cela ferait descendre toutes les notes d'une octave. Pour le VCO et le VCF, le choix du 13600 au lieu du 13700 est étrange. Le 13600 envoie une copie du courant de polarisation vers le tampon de sortie, ce qui aide à assortir l'impédance du tampon à l'impédance de sortie de l'OTA. Malheureusement, cela augmente le courant d'entrée du tampon par rapport à ce qu'il pourrait être autrement. Les remplacer par des 13700 aiderait. Je suis un peu confus sur la raison pour laquelle le tampon a été utilisé du tout dans le VCO, lorsque le signal va directement à un tampon d'amplificateur opérationnel dans l'étage suivant. L'amplitude est réglée par R64,36. Cela donne +/-2 V, ce qui est assez de surcharge pour les transistors à l'entrée des OTAs. Ce n'est pas très grave pour celui utilisé comme comparateur, mais l'autre aura des courants de fuite associés à cela, ce qui affectera le timing. Mettre une résistance entre R64 et R36 résoudrait cela, réduisant la surcharge à 0,6 V. R63 devrait être retiré, car il offre un chemin de fuite pour le courant de réglage de la fréquence et n'est pas nécessaire pour le cycle de réinitialisation. En parlant de cela, le cycle de réinitialisation est assez long (100 µs), ce qui ajoute à la non-linéarité. Il n'y a vraiment pas grand-chose à faire à ce sujet, sauf à passer à un format de réinitialisation différent (JFET à travers le condensateur d'intégration). L'onde carrée est générée avec le comparateur U2A. La modulation PWM pourrait être ajoutée ici en supprimant R54 et en ajoutant un pot à sa place. Une autre bizarrerie du VCO est qu'il y a un comparateur sur l'enveloppe du VCA pour éteindre le VCO lorsque l'enveloppe descend en dessous d'un certain niveau. Le problème ici est que le VCA n'est pas éteint en même temps et continue simplement d'amplifier la tension continue que le VCO crache lorsqu'il est éteint. Ainsi, la sortie du VCA a cette longue traînée continue, ce qui ajoute au bruit du synthé. De plus, le LM393 aurait pu être lié à -Vcc, et le transistor interne utilisé à cette fin, et Q4,5 et toutes ces résistances pourraient disparaître. La résistance de réglage a sa patte inférieure reliée à -Vcc par une résistance. Cela annule complètement l'intérêt d'avoir la tension de référence de l'autre côté. Tout décalage de l'alimentation se retrouve maintenant dans la fréquence du VCO. Cela aurait dû être connecté à la masse.

VCF

Cela a vraiment besoin d'un contrôle exponentiel, mais à part cela, c'est une conception de filtre à variable d'état (SVF) assez simple. La résonance est prélevée du dernier étage, plutôt que du milieu comme d'habitude, ce qui réduit l'amplitude des basses fréquences avec la résonance et déplace la fréquence de coupure en fonction du réglage de la résonance. Le gain entre les deux étages n'est pas identique, ce qui entraîne un faible rapport signal/bruit, car un signal est maintenu très petit pour éviter que l'autre ne devienne trop grand.

Modifications

[Schéma modifié]



VCO

J'ai principalement laissé cela tel quel, car il n'y a pas de moyen facile de résoudre les problèmes majeurs sans perturber le réglage d'origine du synthé. Vous pouvez remplacer R65 par un transistor PNP pour transformer U9B en source de courant, ce qui améliorerait la situation, mais vous auriez toujours le problème de fuite dans les tampons Darlington. J'ai donc décidé de ne pas m'embêter avec cette section. J'ai simplement retiré R59 pour que le VCO se décline naturellement sans être coupé de manière gênante.

VCF

Cette section nécessitait une refonte majeure, et le SVF a été converti en une disposition plus normale. Cela a fait augmenter le gain avec la résonance, et maintenant il y a une saturation importante du filtre passe-bas (VCF) avec des réglages de résonance élevés. Pour atténuer cela, R144 a été augmenté pour trouver un équilibre entre un volume décent à faible résonance et trop de distorsion à haute résonance. Le VCA a été modifié pour augmenter le gain après le VCF afin de gérer ces fluctuations. À l'origine, je pensais laisser le contrôle de la fréquence de coupure de manière linéaire, mais c'était vraiment difficile à jouer, donc un simple convertisseur exponentiel a été ajouté, ce qui a fait toute la différence.

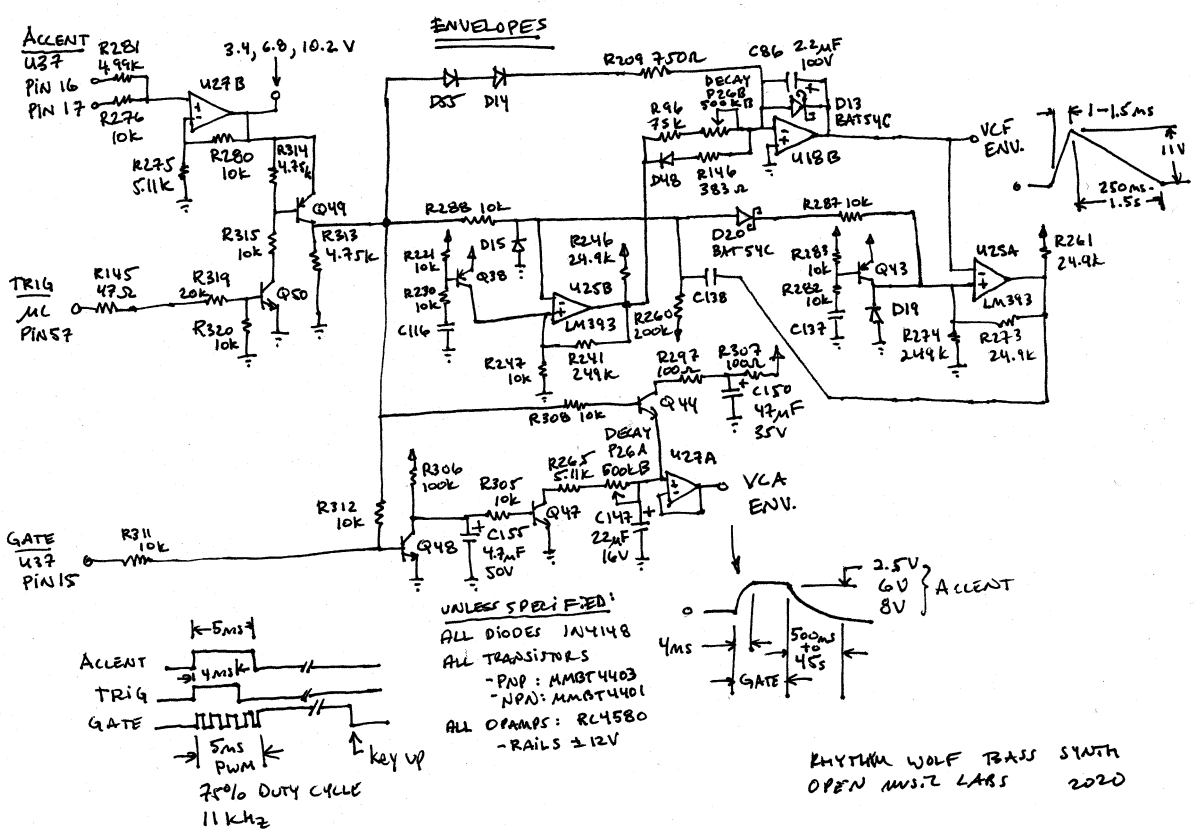
1. R233 a été réduit à 330 ohms pour diminuer le gain à la sortie de U20A.
2. R226 a été changé à 100 ohms pour réduire la limite inférieure de la fréquence de coupure.

3. La polarité de U20A a été inversée pour fonctionner comme un contrôleur exponentiel. R234 a été retiré et le bouton ENV MOD est directement couplé avec une résistance de 22k à U20A. Le potentiomètre de coupure a été inversé.
4. U20A a été déconnecté des entrées OTA, et deux transistors et une résistance ont été ajoutés pour créer un simple générateur de courant exponentiel contrôlé par U20A et alimentant les OTAs.
5. C9 et C68 ont été changés à 1nF pour ajuster la fréquence de coupure dans la bonne plage.
6. R155 et R173 ont été changés à 500 ohms et R121 a été changé à 22k pour ajuster les gains des deux étages OTA de manière à ce qu'ils soient approximativement égaux, et pour avoir des signaux d'amplitude appropriée à leurs entrées.
7. Le potentiomètre de résonance a été déplacé vers le premier étage et le deuxième étage a été directement couplé via R175, qui a été changé à 47k pour ajuster le gain.
8. Une résistance de 220 ohms a été ajoutée en bas du potentiomètre de résonance pour définir la résonance maximale, et R174 a été rerouté vers le potentiomètre de résonance et changé à 22k pour définir la résonance minimale.

Enveloppe du synthétiseur de basse

Vue d’ensemble

[Schéma d’origine]



It seems like the intent here was to make both lin and exp envlopes, with the exp going to the VCA, and the lin for the VCF. Since there is no exponential converter on the VCF, the lin envelope makes for a really weak filter sweep. It spends a relatively smaller amount of time in the lower range, near the cutoff frequency. There are 3 accent levels (the 4th accent level is "off"). These are evenly spaced between 0 and 10V. The accent and trigger signals arrive at the same time, with the trigger ending first. i'm not sure why there are 2

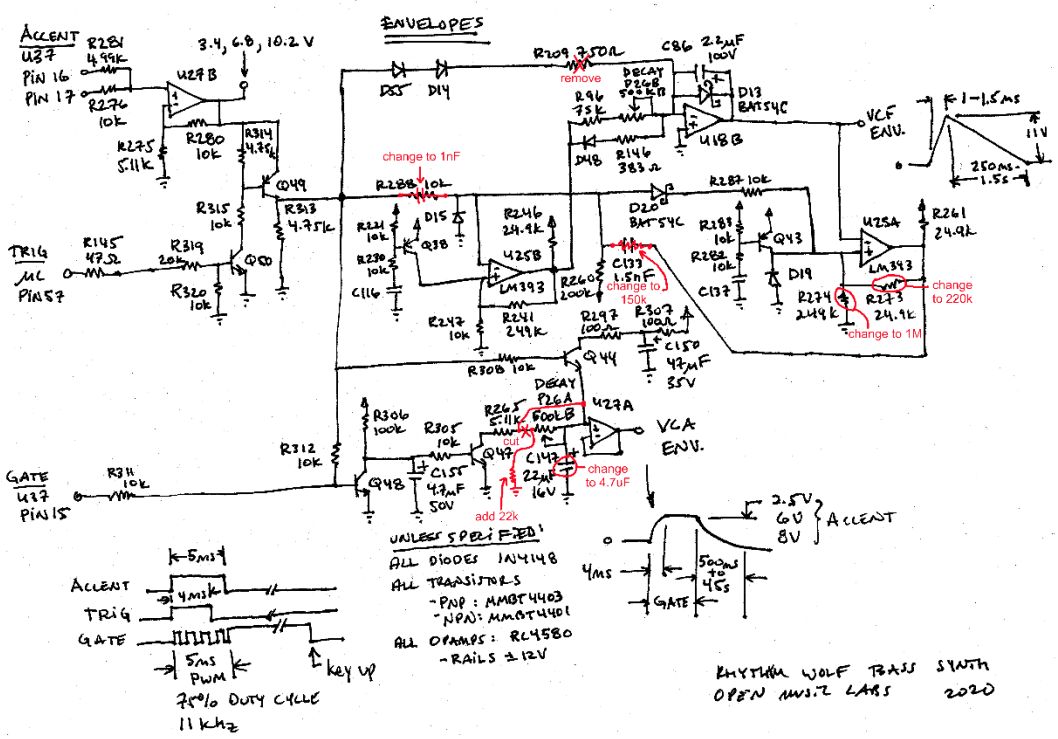
signals here, but its handy; if you wanted to add a manual accent knob, you would drive U27B with your accent voltage. The gate signal is even more bizarre, in that it has PWM on the first 5ms. im not sure if this is intentional, or just some artifact of the way the signals are multiplexed. i'm assuming the former, as it gets multiplexed the same way as the other signals. This PWM signal has no effect on the remainder of the circuit, as its over-ridden by the accent/trigger signal where they come together at R312. But, if it were PWM'd after key-up, then you could have digital control of the decay time. The existance and size of C155 suggests that pwm was intended here. If you dont want your envelopes to be gated, you can just remove R311, and instead put a 10k to ground there. Q44 follows the accent voltage to charge C147, while at the same time Q47 is turned off to disengage the decay pot. This sets the min time before decay to 4ms (the trigger time).

The decay time on the VCA envelope is crazy long: 45s! Reducing C147 would fix this. Another odd thing here is that its a linear pot. Typically log pots are used for time and frequency control. There are mostly linear pots in the whole synth, but a number of different values, so getting a log wouldn't have been any extra cost. In this particular case, its a dual pot, so its an odd one out to begin with. Again, an interesting choice to have 2 different styles of envelopes, as it requires a dual pot.

The VCF envelope is less straightforward. First off, it has 2 comparators, each with startup pulses (Q38,43) on them to ensure a fixed state at startup. i'm not fully convinced these are necessary, and a quick pulse on the trigger would have done the same thing anyways. Both of these comparators merely follow the trigger signal, so its not too relevant. what it looks like is supposed to happen, is that U25B detects the trigger, and starts charging the integration capacitor, and U25A checks the envelope voltage, and shuts off U25B when it hits 10V. but, this doesnt happen, as the signal is not strong enough. there are a few different way to fix this. But, the upshot is U18B saturates at 11V, and there is a delay before decay begins as the integration capacitor needs to discharge quite a bit before U18B goes back into linear mode. R146 sets the attack time, and D48 ensures its only active during attack. D55 and D14 ensure that the gate signal does not discharge the capacitor. But, i'm not sure what this pathway is for anyways. It slightly adjusts the attack time, from 1ms to 1.5ms at higher accents. i'm not sure why you would want slower attacks for accented notes. Personally, i would re-route this signal to the comparison voltage on U25A, giving higher peaks on accented notes.

Mods

Schema modifié



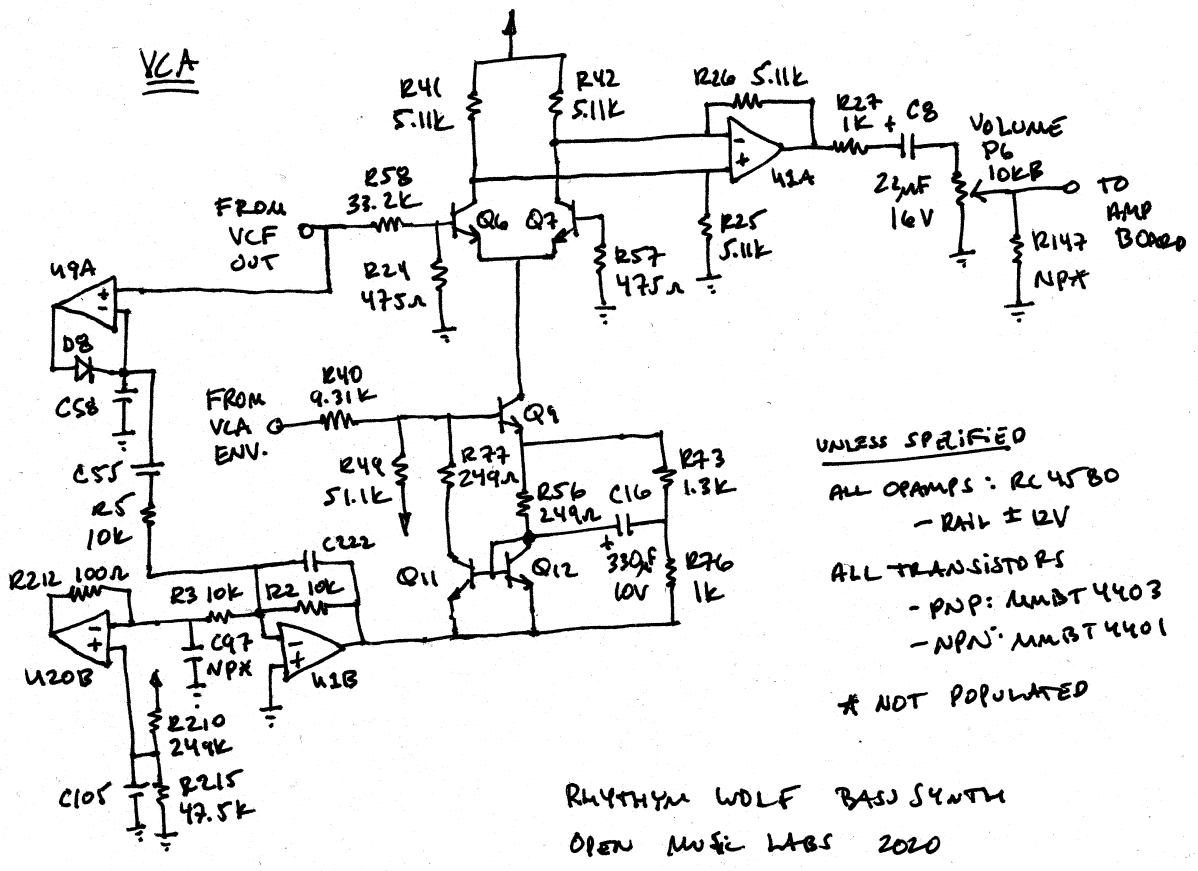
Les deux enveloppes ont été principalement modifiées pour ajuster les temps de déclin min/max de manière à ce que les boutons soient dans une plage plus utile. Bien qu'une enveloppe linéaire soit un peu inhabituelle pour le VCF, cela a été conservé afin que le VCA et le VCF aient des temps de déclin différents. L'enveloppe du VCF aurait également pu être convertie en exponentielle, mais la façon dont elles changent avec les réglages est différente, avec le VCA s'étendant beaucoup plus longtemps, ce qui crée des sons intéressants avec des temps de déclin longs. La correction la plus importante a été de modifier la façon dont l'enveloppe du VCA est déclenchée, en la fermant complètement à la fin de chaque note plutôt qu'en déclinant. Cela crée une différence entre les notes glissées et les notes normales qui n'existerait pas autrement.

1. Changer C147 en 4.7uF pour réduire le temps de déclin maximal du VCA.
2. Rerouter R265 vers C147 pour éteindre immédiatement l'enveloppe avec la note off.
3. Ajouter une résistance de 22k au potentiomètre de déclin du VCA pour définir le temps de déclin minimum.
4. Retirer R209 pour éliminer le contrôle d'accentuation inversé du temps d'attaque du VCF.
5. Changer R288 en un condensateur de 1nF et C133 en une résistance de 150k pour permettre au circuit de détection du pic de l'enveloppe de fonctionner réellement.
6. Changer R273 en 220k et R274 en 1M pour régler la valeur de crête de l'enveloppe à 10V.

VCA du synthétiseur de basse

Vue d’ensemble

[Schémas d’origine]

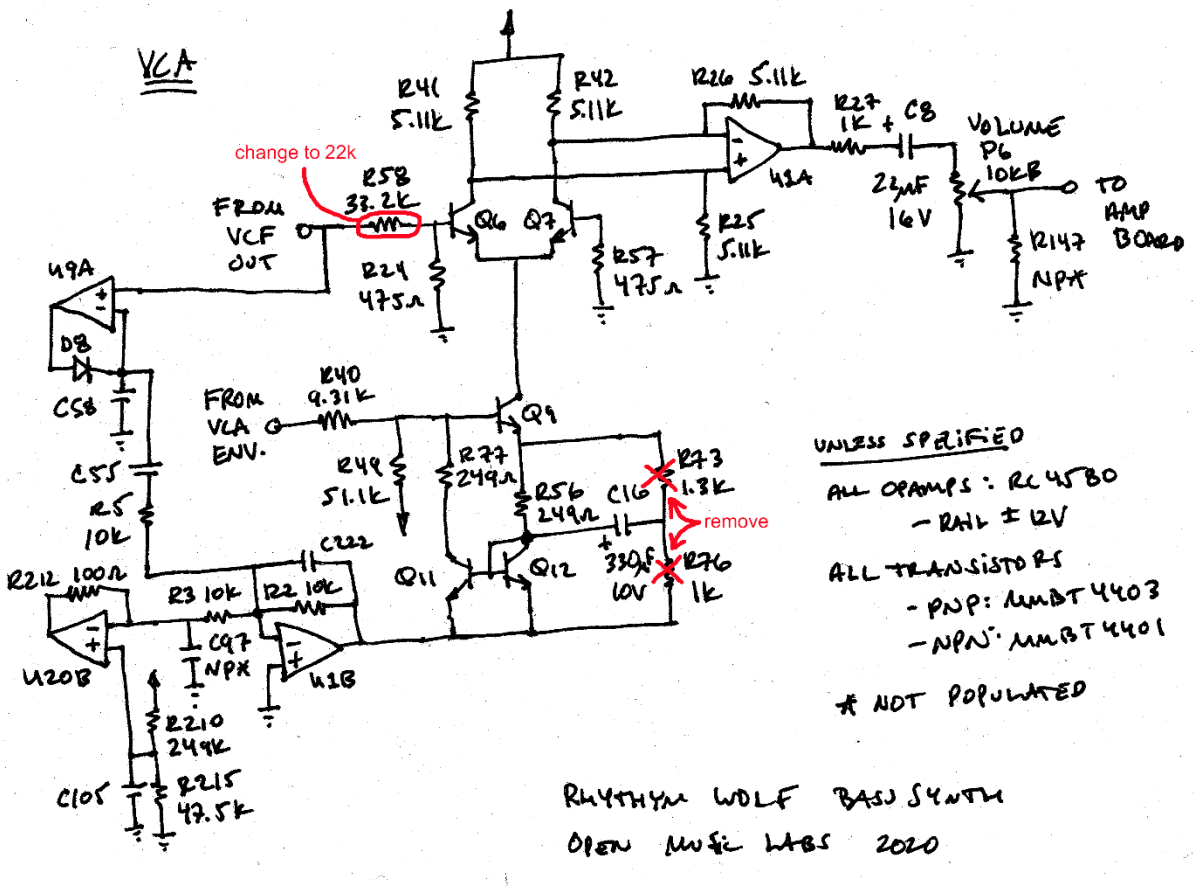


Le VCA a une compression. La cellule principale de gain est une paire différentielle standard, mais le générateur de courant de queue est unique. Il a un miroir de courant qui suit le courant de queue, puis abaisse l'entrée CV à mesure que le gain augmente. Mon principal problème avec cela est que cela annule complètement l'intérêt d'avoir des enveloppes exponentielles pour le CV, car cette compression annule complètement le comportement exponentiel. Cette compression commence lorsque Q12 s'allume, ce qui se produit lorsque le courant à travers R73,76 crée une tension supérieure à un volt ou quelque chose comme ça (Vbe + chute de R56). C16 impose une constante de temps sur cette compression, qui est légèrement à l'envers de ce à quoi on pourrait s'attendre : des augmentations rapides du CV augmentent la compression, et des diminutions rapides du CV diminuent la compression.

Les bizarreries réelles de ce circuit sont les entrées de U1B. U1B fixe la référence basse qui sert au courant de queue. U20B est simplement un générateur de tension fixe, mais avec son signal pris à partir de la broche d'inversion, plutôt que de la sortie. C97 n'est pas peuplé, car cela rendrait l'op-amp instable en retardant le signal de rétroaction. Fondamentalement, cet op-amp n'est pas du tout nécessaire, car une résistance de Vcc à la jonction de sommation de U1B aurait fait la même chose. U9A est encore plus curieux. Il prend le pic du signal du VCF et essaie d'ajuster la référence basse en conséquence. Cela pourrait agir comme une sorte de compression sur le signal audio, plutôt que simplement sur le CV. Cela pourrait être assez utile pour traiter les changements de volume liés à la résonance, ou s'il y avait un circuit de distorsion avant cela. Mais, comme il n'y a pas de résistance à travers C58, il se charge simplement et reste chargé pour toujours, donc il ne peut pas répondre aux changements d'amplitude. C'est juste la valeur de crête qu'il a jamais vue. Je ne suis pas sûr de ce que C55 représente, car elle ne permettrait qu'une compression temporaire. La chose la plus étrange à ce sujet est que la rétroaction est inversée, de sorte que des signaux plus importants augmentent en fait le gain du VCA. Inverser D8 corrigerait cela. Mais, finalement, le circuit de compression CV surpasse surtout tout effet que cela peut avoir..

Modifications

Schéma modifié

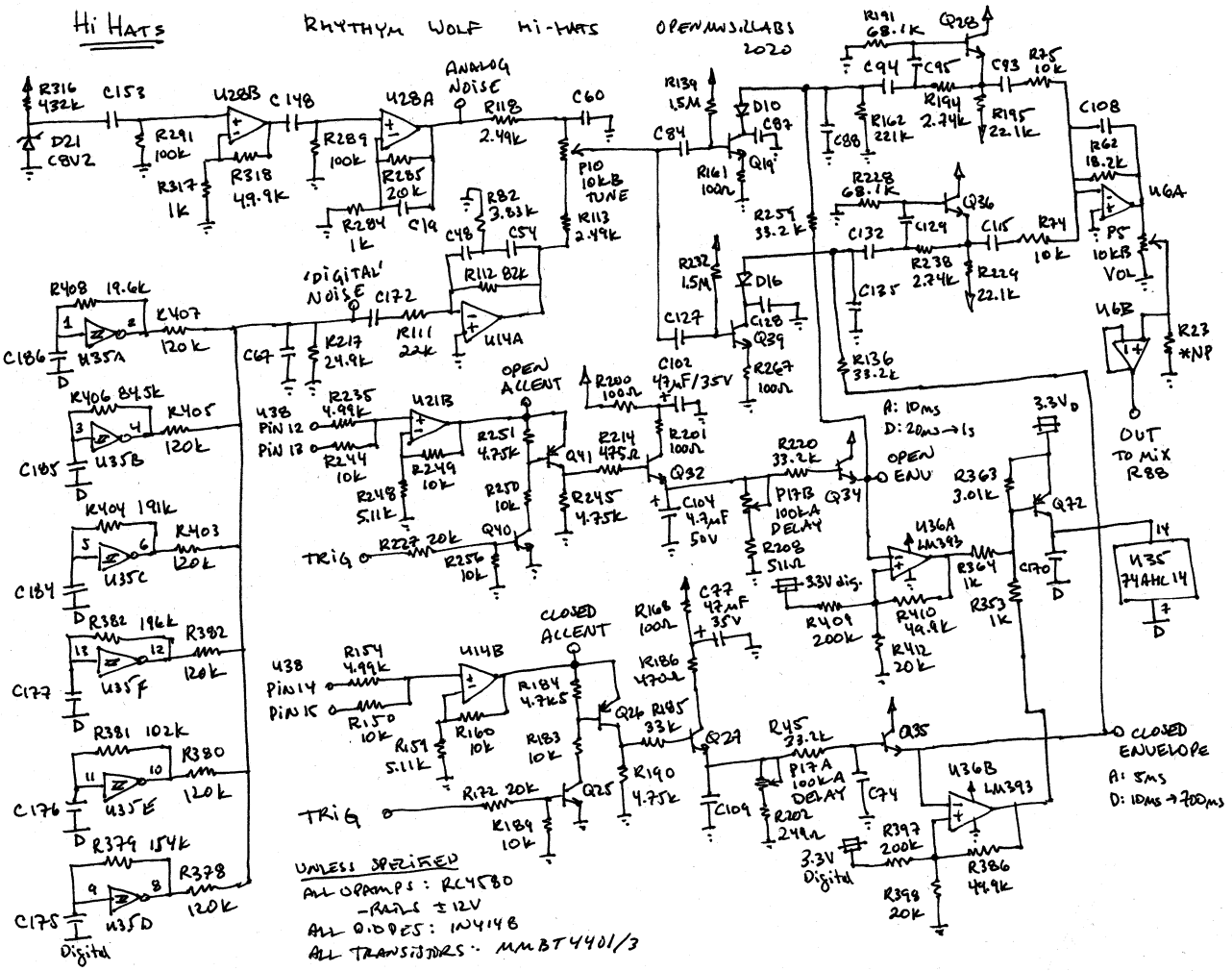


Cela a été converti en un VCA linéaire standard. Étant donné que le circuit de compression ne fonctionnait pas, je l'ai laissé en place, et R73 et R76 ont été retirés pour que le miroir de courant fonctionne correctement. R58 a été réduit à 22k, mais cela n'était pas strictement nécessaire, cela a simplement élevé le signal pour le rendre un peu plus conforme aux autres voix du synthétiseur.

Charlestons (HH)

Vue d’ensemble

Schémas d’origine



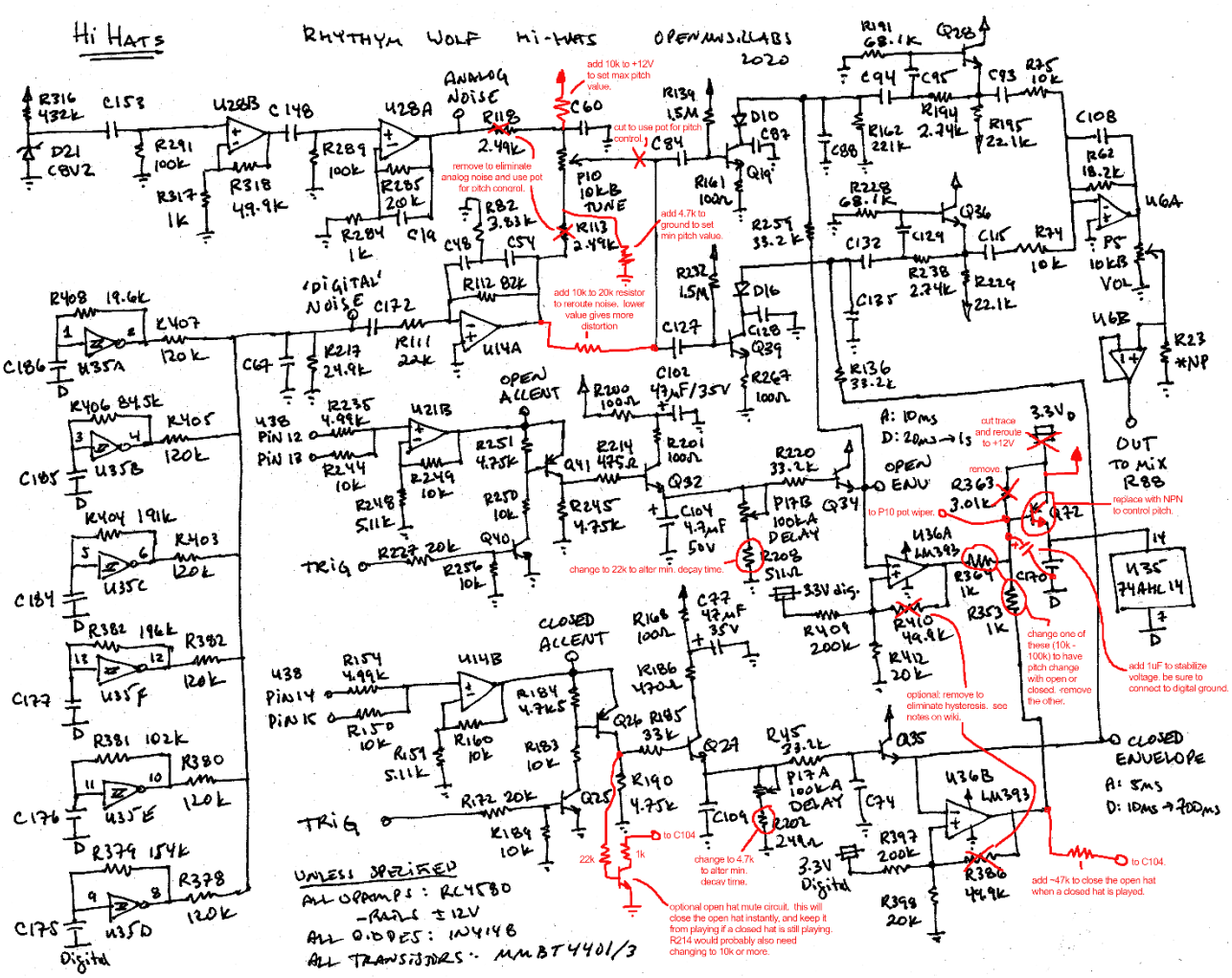
The RW utilizes both an analog and digital noise generator, and the tune doesnt actually tune anything, and rather sets the mix of these two. R118,113 limit the extent of this mix, and the relative volume of each is quite different. The digital noise is around 5Vpp max, whereas the analog is 500mVpp or so. There are two lowpass filters on the analog noise (C19,60) and the digital noise has a resonant lowpass and a regular RC lowpass (C48,54,67). This combined noise source goes to two pretty standard drum 'VCAs' (Q19,39 - one each for open/closed). The envelopes for these are generated with the same gate/accent setup as the bass synth. The envelope's peak value tracks the accent voltage. The decay time is half for the closed hat. The envelopes also go to comparators to shut off the digital noise source. The output of the VCAs go to a sallen-key highpass filter, using a single transistor. The output then goes to a mixer stage and the volume pot.

I find it odd that the digital noise source is connected to digital powersupplies. This means any noise in the digital side will be coupled into the hihat outputs. The amount of current drawn by the HC14 is really low, so i'm not sure it would have been splattering noise onto the analog supplies if it was connected to them. The other thing is that the comparators do not work as intended here. Since there are diodes and capacitors in the VCAs which hold their charge after the envelope is off, the emitters of the envelope buffers sometimes do not get down below the threshold to shut off the comparator. A pulldown resistor is needed at the emitters to ground. i'm not sure this matters much, though. This section could be modified to apply a sort of 'mute' to the voices, so they end more abruptly. This might be good for the closed hat. Another oddity is that as the decay knob is turned down, it loads down the charging circuit, and the envelope actually gets completely eliminated, so the voice pretty much stops sounding at really low settings. R208,202 could be increased to basically set a more reasonable limit on the decay time.

All in all, i like the sound of this, and fixing the decay times is a pretty simple mod. i would like to be able to alter the pitch in some way, and i think the easiest way would be to alter the supply voltage to the HC14. These can run from 2V to 5.5V, and the pitch of the oscillators change with the supply voltage, as the hysteresis increases with increasing supply voltage. A little modulation on it might be nice as well. If i felt that the mute function was needed, then this voltage would be applied to the emitter of Q72, otherwise it would be easier to just remove Q72 and apply it directly to the supply pin of the HC14.

Modifications

[Schéma modifié]



i tried a number of different mods, and ended up using a small set of them. The ones i kept were as follows:

7. Minimum decay time - change R208 to 10k and R202 to 4.7k. This is very much to your taste, so i would suggest trying a few different values to see how they sound.
8. Tune knob rework - i eliminated the analog noise, as it didnt sound as good as the digital noise, and repurposed the tune knob to tune the digital noise. This is accomplished by varying the voltage to the 74AHC14 IC that creates the digital noise. To do this, remove Q72 and replace with an NPN transistor. The trace the 3.3V rail does not need to be cut if that pad is no longer used. Instead, run a trace to the +12V supply. i connected up at the comparator on the backside of the board, but any +12V point would do. Remove R363 and place a 1uF capacitor between the base of the transistor and digital ground. Digital ground can be found closest at C177. If you are not planning on doing

pitch bending, i would suggest a 10uF capacitor instead, as it will be more stable. Run a wire to the wiper of P10.

9. To rework the pot, cut the wiper trace and wire it to the transistor on the bottom of the board. Remove R118 and replace it with a 10k to +12V. Remove R113 and replace it with a 4.7k to ground. You can use a smaller value for the 4.7k, but it might not work. i installed a 3.9k, and it sounded great for a while, but then the digital noise started to cut out at low pot settings. The 74AHC14 is only rated down to 2V, and a 3.9k supplied 1.5V or so. i'm not sure if it was the low voltage or something else i had done that messed it up. A 3.6k is the lowest that will work at all. This will allow a wide frequency sweep. You can reduce the 10k value to get higher frequencies, but it seemed high enough to me at 10k.
10. Reconnect the noise source to the VCAs with a 10k resistor from the output of U14A (pin1) to the input of the VCAs (C84,127). This value can be made smaller or larger to give varying amounts of clipping in the VCA output. Larger values will give a softer sound, but at some point will make the volume very quiet. Smaller values will give a harsher tone.
11. Pitch shift - you can make the closed hats a different pitch than the open hats by adjusting R364 and R353. When one of these resistors is engaged, it pulls down the voltage on the 74AHC14, lowering the pitch. So, if you want the closed hat to be a lower pitch, put a resistor in R353 and not in R364. To make the open hat lower pitched, do the reverse. The value of the resistor determines the pitch depth, with values from 10k to 100k being reasonable. A 47k would be a good starting point. If you use a larger value than 1uF for the transistor base capacitor in (1), the pitch shift will have a slew to it. its interesting, but not something i wanted as a fixed option. See discussion below for tweaks to this mod.
12. Open hat closure - to close the open hat when the closed hat triggers, run a 22k resistor from the output of U36B (pin7) to C104. This will discharge the envelope of the open hat whenever the closed hat is on, very similar to how the 606 or 808 works. Smaller values of this resistor will close the hat faster, but will also attenuate the hat more on subsequent retriggers when the closed is still sounding. An alternate version of this involves a transistor and a few resistors off the trigger signal. This latter method is far more abrupt, and mutes the open hat whenever the closed is activated. i didn't like it in comparison to the former version, so i didnt explore it completely. R214 would need to increase in order to keep Q32 from over-powering the pulldown resistor on the mute. In theory, it should be >10x the value of the pulldown resistor.

Autres modifications

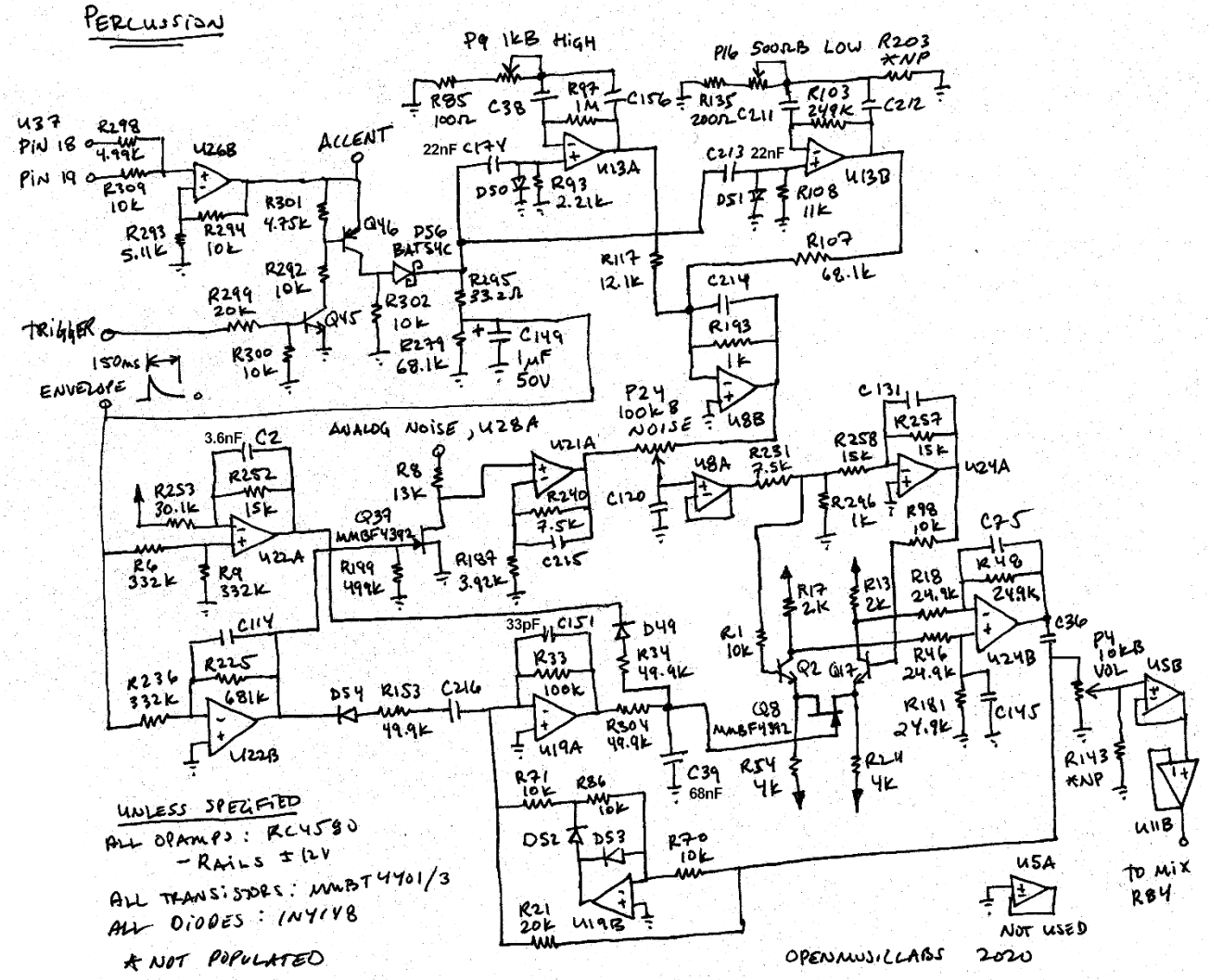
There is a lot of design space here, and i did not explore it fully. i tried adjusting one of the oscillators on the 74AHC14, as there were two set to 1kHz, but after changing it, it didn't sound as good, so i put it back. The main area that could use some work, is the settings on the comparators. i removed R386 and R410, which eliminated the hysteresis. This raised the threshold for detecting note off, which worked well in keeping the pitch from always hanging low. For pitch shifting purposes, it depends upon which note is being shifted down. I found that triggering off the closed hat made the most sense, as it was the shorter of the two envelopes. But, this sends some voltage to the open hat envelope if you have the open-hat closure mod. Instead, cut the signal from the open hat to U36A, and run the output of U36B to the U36A input. This basically gives you an inverter with a buffered output that can be patched to the pitch mod. As it also inverts which hat is pitched down, you can also send the signal to the non-inverting input of U36A, remove R412, and tie the inverting input to something around 1V. i found that the pitch shift had a bit of slew, and also turned off halfway through note at times, which was rather cheesey, so i didn't use this mod. For open hat muting, you might not want to hold the mute too long, so a higher threshold would be preferred, which can

be achieved by increasing R398. If the mute comes off too early, it won't matter as much as the envelope will have probably decayed to a very low level by that point anyways.

Percussion

Vue d'ensemble

[Schéma d'origine]



This has three main sections. The first is a standard pinged resonant filter (U13A,B) for the low and high percussion sounds. The second is a noise source gated by a transistor (Q37), and the third is a compressor built around a few opamps (U19,24) and transistors (Q2,17,8). The pinged filters work as they should and sound fine, but are limited by D50/51 such that accent does change the volume much. The gated noise source sort of works. It turns the noise on butonly moderately follows the envelope, and it doesnt really shut off. The JFET has a nominal on resistance of 60ohms, so with the 13k drain resistor, that should give -45dB attenuation for "off", which it might do, but that's not really enough. You can definitely still hear it at the ouput, even when the voice is not triggered. This is compounded by the fact that it goes through a compressor, which amplifies it when its "off". If you use the "howl" function, this noise is completely unreasonable. If you have the percussion voice turned up at all, the noise is constantly at the output, and its distorted, so it splatters all across the audible range.

The compressor is the most sonically interesting part of the circuit. It removes the usual exponential decay of the envelope, and instead keeps the amplitude constant for the majority of the range. This also has the effect of extending the apparent length of the notes, increasing the gain as the signal decays out. This makes a more "hollow" sound as compared to the typical "ping" of the resonant filter.

The compressor is driven by three signals: feedback from the output, a short attack spike, and the envelope. The envelope is a simple RC decay set by C149 and R279, and this is intended to serve as both an envelope, and a gate to turn the compressor off when no notes are played (U22A). Whether due to JFET variations, or just poor design, it doesn't do either on the unit i tested, as the level is too low to turn the compressor off. The attack spike (U22B) also has a failing, in that the blocking capacitor C216 has no discharge path, so it charges once on the first gate, and then never passes the attack signal again. The compressor's main input is from the fullwave rectifier (U19B) which is fed from the output. This has a gain of five, and a lowpass filter on the output. The heart of the compressor is a VCA composed of Q2,17,8, where Q8 is JFET used as a variable resistor to vary the gain of the VCA.

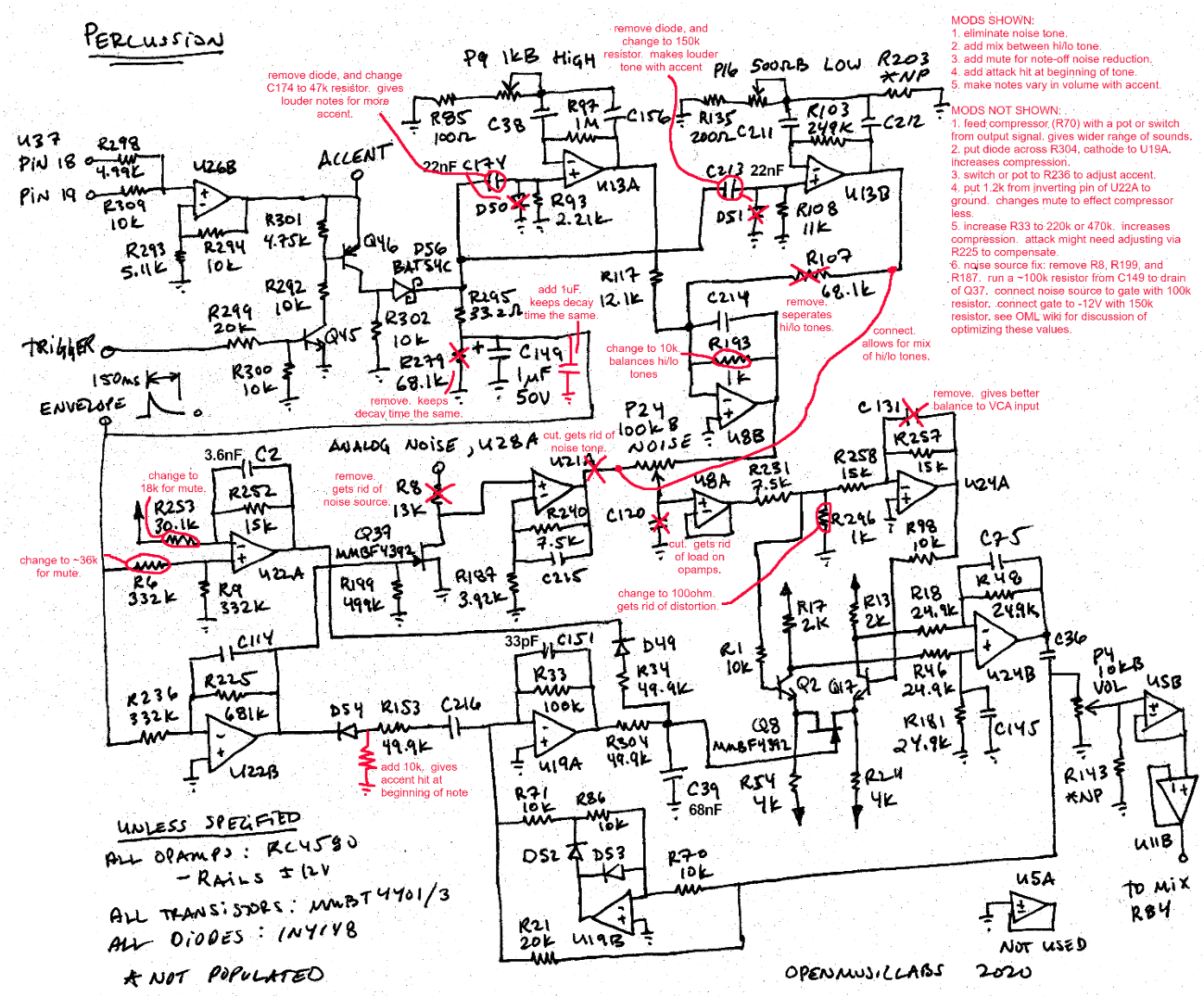
The inclusion of JFETs is a bit strange as they could have used a standard diffpair with a current source. The JFET in the noise gate also doesn't really do its job, and they would have been better off using the gating scheme from the hi-hats. If those introduced too much distortion, another diffpair could be used. It suprises me to see a six voice analog synthesizer with no trimmers inside. I have to imagine a bunch of things are just out of spec.

And now on to the electrical oddities:

5. There is a cap on the mixing pot wiper. Either end of the pot is driven by an opamp, so when its at the extremes it puts a hefty capacitive load on the opamps. This makes them less stable. A series resistor on the wiper would have fixed this.
6. The gate of Q39 is driven directly by an opamp output. This means that any excursion above 0.6V will destroy the JFET. Normal operation precludes this, but who knows what happens at startup. A series resistor would have fixed this, and reduced capacitive feedthrough on the attack phase.
7. D54 discharges C126, but there is no return current path, so it only happens once on the first trigger, and then never effects the circuit again. A current drain resistor is need on D54. This pathway is intended to gate the compressor open for the attack phase, but this is already done with the D49 pathway, and could have been replicated with a capacitor in parallel with D49.
8. U11B is completely redundant, and off in a corner of the board above the digital section. U5A is not used. i'm guessing they originally ran the signal all the way over to the corner and then back to the mixing section, but later decided this was not worth it, and added another opamp (U5), but then did not unroute the old opamp. So you have half an opamp going unused and a signal being double buffered.

Modifications

Schéma modifié



J'ai décidé pour mon boîtier de couper complètement le son du bruit. Cela ne sonnait pas bien, était beaucoup de travail pour obtenir un son correct, et reproduisait les mêmes sons que j'obtenais des autres voix. À la place, j'ai utilisé le pot de mixage pour mélanger entre les tons hi/lo. Cela donne une gamme de sons beaucoup plus large. Cela impliquait de désactiver la source de bruit pour qu'elle ne se propage pas partout dans le circuit, et de couper la piste vers le pot de mixage. Sur le mien, j'ai coupé les deux extrémités du pot, et câblé directement depuis les sorties de U13. Mais, il est plus facile de couper un côté et d'ajuster le volume sur U8B. C120 est retiré pour éviter que les amplis opérationnels n'aient à piloter une charge capacitive importante lorsque le pot est aux extrêmes. R296 devait être réduit car les niveaux étaient beaucoup plus élevés maintenant. 100 ohms était une bonne valeur pour le mettre juste à la limite de la distorsion sur l'ensemble de la plage. Vous pouvez aller un peu plus petit si vous voulez qu'il soit parfaitement propre, mais le rapport signal/bruit empire.

Ensuite, j'ai modifié les tons hi/lo pour qu'ils soient plus forts avec plus d'accent. Je pense que cela a été initialement fait comme un volume fixe parce que le circuit de coupure faisait également le contrôle de l'enveloppe. Mais une fois que j'ai activé la fonction de coupure, j'ai décidé que j'aimais quand même l'effet de volume variable. Cela se fait en retirant les diodes aux entrées de U13, et en remplaçant les condensateurs par des résistances. J'ai choisi les valeurs pour donner un peu de distorsion au niveau le plus élevé. Les résistances peuvent être augmentées d'environ 20 % pour éliminer cela, ou réduites de 20 % pour augmenter cette distorsion. En raison du remplacement de ces condensateurs par des résistances, le temps de déclin a été raccourci. C'est correct, mais les notes ont perdu un peu de ce son creux, alors j'ai retiré R279

et ajouté un autre condensateur de 1uF pour ramener le temps de déclin à son niveau initial. En utilisant un condensateur CMS, j'ai pu simplement remplacer R279 par ce condensateur.

Ensuite, le circuit d'accentuation, qui avait simplement besoin d'une résistance de vidange de 10k pour que C216 se décharge. Cela donne un clic au début des notes, qui est plus fort avec plus d'accent. Je ne suis pas sûr de l'aimer complètement, mais cela peut être facilement désactivé en déconnectant la résistance de vidange (ou toute autre méthode pour bloquer le signal).

Enfin, le circuit de coupure, qui a probablement fonctionné sur leur banc d'essai mais pas en production en raison de la grande dispersion des paramètres JFET. J'ai mesuré la plage active du JFET pour mon compresseur VCA, et c'était -4V pour le plein arrêt, et -3V pour un gain de 1/2. Le gain complet était à -0.5V. J'ai modifié R253 pour que la sortie soit d'environ -8.5V sans signal. Cela, combiné avec la sortie à 0V du redresseur pleine onde, donne -4.2V, ce qui garantit que le VCA est éteint lorsqu'il n'y a pas de signal. Cela réduit considérablement le souffle de fond de cette voix. Ensuite, j'ai diminué R6 à 36k pour augmenter l'enveloppe. Cela a permis au compresseur de s'ouvrir, donnant des notes plus longues qui étaient plus fortes. C'est très similaire au son d'origine, car l'enveloppe ne fonctionnait pas du tout avant la modification de R253. Si vous ajoutez une résistance de 1.2k de la broche d'inversion de U22A à la masse, le gain augmente et élimine complètement l'effet d'enveloppe, en coupant simplement le VCA.

Autres modifications

Autres choses que j'ai essayées, mais pour lesquelles que je ne voulais pas ajouter de commutateurs/potentiomètres pour le moment :

1. Mettre un potentiomètre ou un interrupteur sur le signal allant de la sortie vers le compresseur. Un potentiomètre ici est excellent. Pour les tests, j'ai simplement câblé depuis le curseur du potentiomètre de volume et j'ai été tenté de le laisser ainsi. Cela permet une très large gamme de sons de la voix. Sans le compresseur, ils ressemblent beaucoup à des résonances, très aigus et courts. Avec le compresseur, ils ont un son creux agréable.
2. Mettre une diode à travers R304. Cela fait que le compresseur s'allume instantanément, ce qui sonne assez bien lorsqu'aucun accent n'est ajouté. Cela augmente également l'effet de compression, mais déforme le début des notes.
3. J'ai essayé des réglages de gain de compression plus élevés (augmentation de R33), et c'était amusant, mais finalement j'ai décidé que c'était trop pour un réglage fixe, et ils déformaient beaucoup. Un potentiomètre ici pourrait permettre quelques variations amusantes.
4. Réparation de la source de bruit. C'est possible, mais compliqué. Fondamentalement, le JFET doit être câblé différemment. Actuellement, l'enveloppe est envoyée à la grille et le bruit est au drain. Mais, en raison de la nature non linéaire de l'entrée de grille, il n'y a qu'une petite fenêtre où le JFET est actif, et l'enveloppe ne fait que couper le signal car il ne passe pas beaucoup de temps dans cette région. La source de bruit devrait aller à la grille, et l'enveloppe devrait aller au drain. Comme les JFET varient beaucoup, vous devrez tester les valeurs pour obtenir la réponse que vous aimez.
5. Tout d'abord, commencez par retirer R8 et R199 et connectez une résistance de 100k de l'enveloppe à C149 au drain. Avec une résistance de drain de 100k à C149, l'enveloppe se raccourcira de dix pour cent ou plus, donc la capacité pourrait avoir besoin d'augmenter de 220nF ou plus. Si le temps d'attaque est un peu trop long à ce stade, court-circuitez R295. Ensuite, coupez la piste entre le drain et U21A et mettez un condensateur de 0.1uF et une résistance de 10k à la masse. Cela agira comme un filtre passe-haut et réduira le bruit continu. Utiliser un condensateur ou une résistance plus importante ici laissera passer plus de basses fréquences, mais laissera également passer plus de bruit continu. Enfin, prenez une résistance de 10k et attachez une extrémité à la source de bruit et l'autre

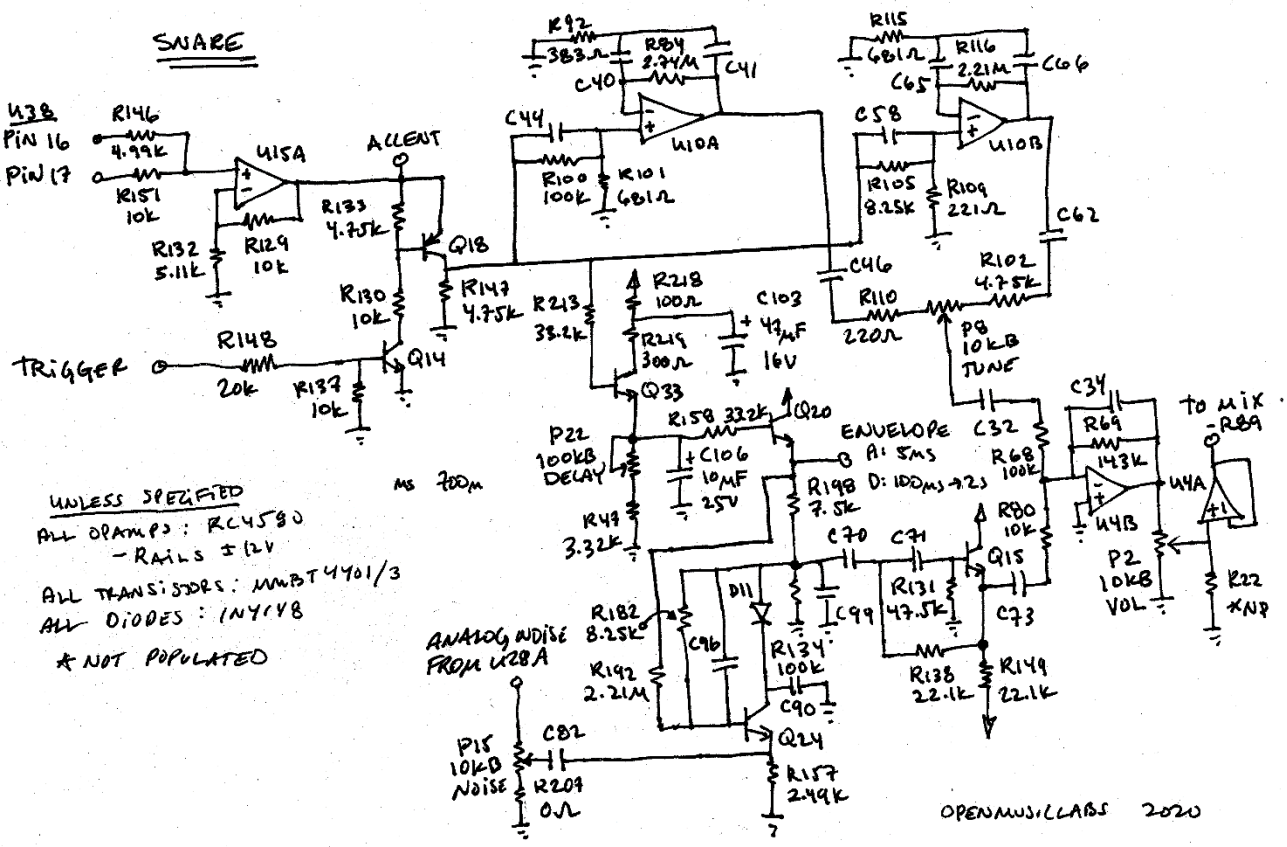
à la grille. Ensuite, connectez la grille avec un potentiomètre de 100k à -12V. Ajustez le potentiomètre jusqu'à ce que la source de bruit sonne correctement. Vérifiez la valeur du potentiomètre et soudez cette valeur à la place du potentiomètre.

6. Deuxièmement, vérifiez le volume de sortie. Il sera très probablement en distorsion ou trop fort. S'il est trop fort, retirez R187. S'il est toujours trop fort ou en distorsion, diminuez la valeur de la résistance de 10k dans le filtre passe-haut sur le drain. Vous devrez très probablement augmenter la valeur du condensateur en conséquence. Pour façonner davantage le bruit, un condensateur peut être placé entre la grille et la masse pour éliminer les hautes fréquences.

Caisse claire

Vue d'ensemble

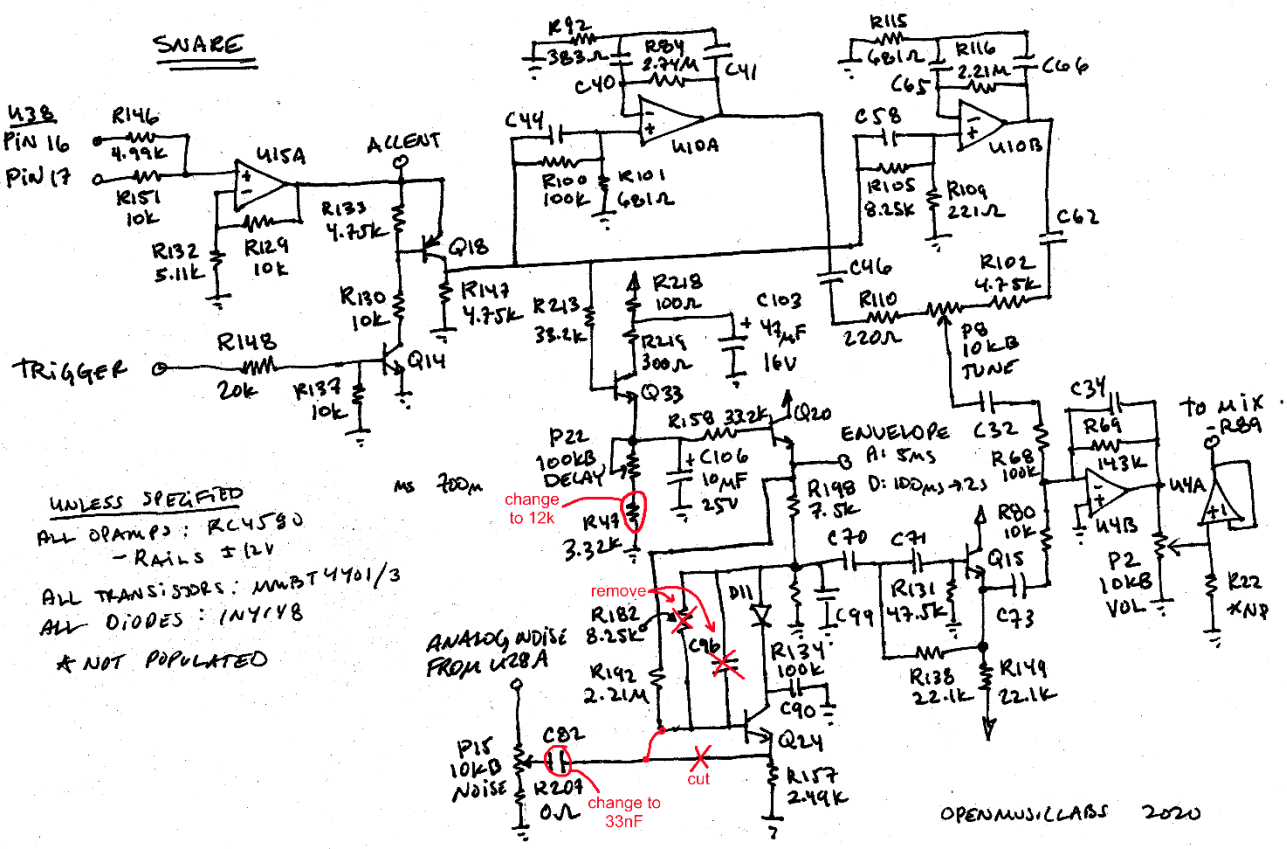
[Schéma d'origine]



Ceci est très similaire à la caisse claire de la TR-808 (on ne peut pas vraiment comparer les filtres pingés sans les valeurs des condensateurs), sauf que le VCA de bruit est un étage d'amplification à base commune, qui ne fonctionne pas du tout. L'enveloppe agit pratiquement comme une porte pendant 90 % du temps, puis fait la décroissance pour les 10 % restants. De plus, cette topologie a une tension plus élevée pour fonctionner car l'émetteur est plus élevé (ils auraient pu le polariser à -12 V pour éliminer cela), donc l'enveloppe coupe plus tôt pour les niveaux d'accentuation plus bas. Cela prend environ le double de temps entre les accents bas et élevés. Il serait assez simple de convertir cela en un VCA de style 808.

Modifications

[Schéma modifié]



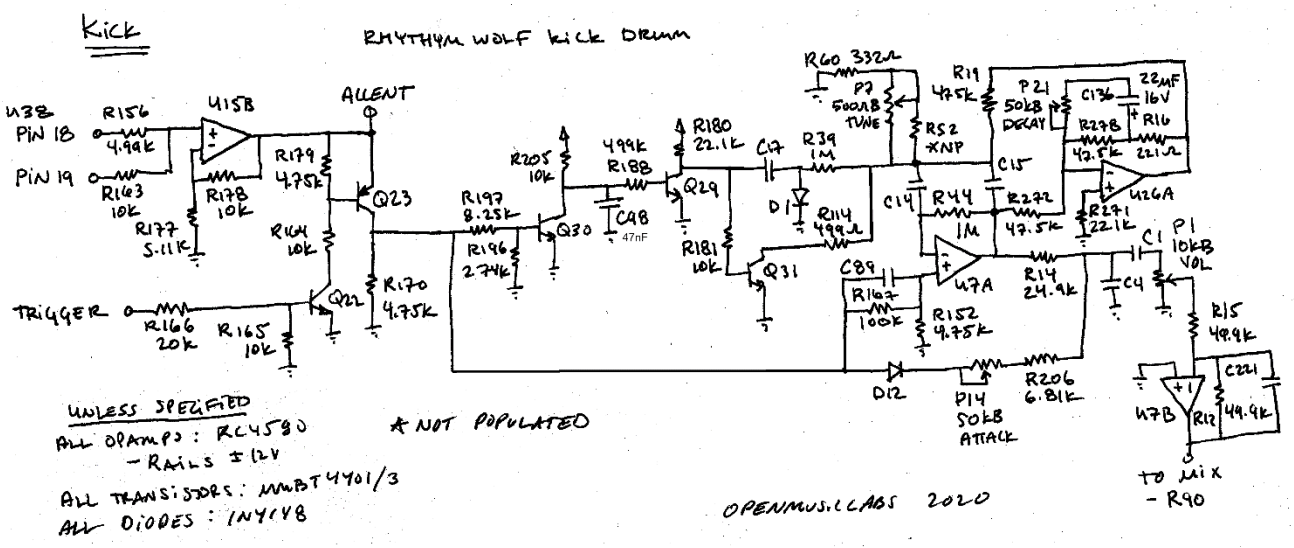
Je n'ai pas modifié cette partie trop intensément, principalement parce que je commençais à manquer d'énergie pour ce projet. Je l'ai ramenée en grande partie à une caisse claire de style TR-808.

1. Retiré C82 et relié le côté de la source de bruit à la base de Q24 avec un condensateur de 33nF.
2. Supprimé R182 et C96, ce qui en fait un VCA standard.
3. Changé R157 à 100 ohms.
4. Changé R47 (erreur sur le schéma - indiqué comme R147) à 12k pour que le déclin ne coupe pas complètement le bruit..

Grosse caisse

Vue d’ensemble

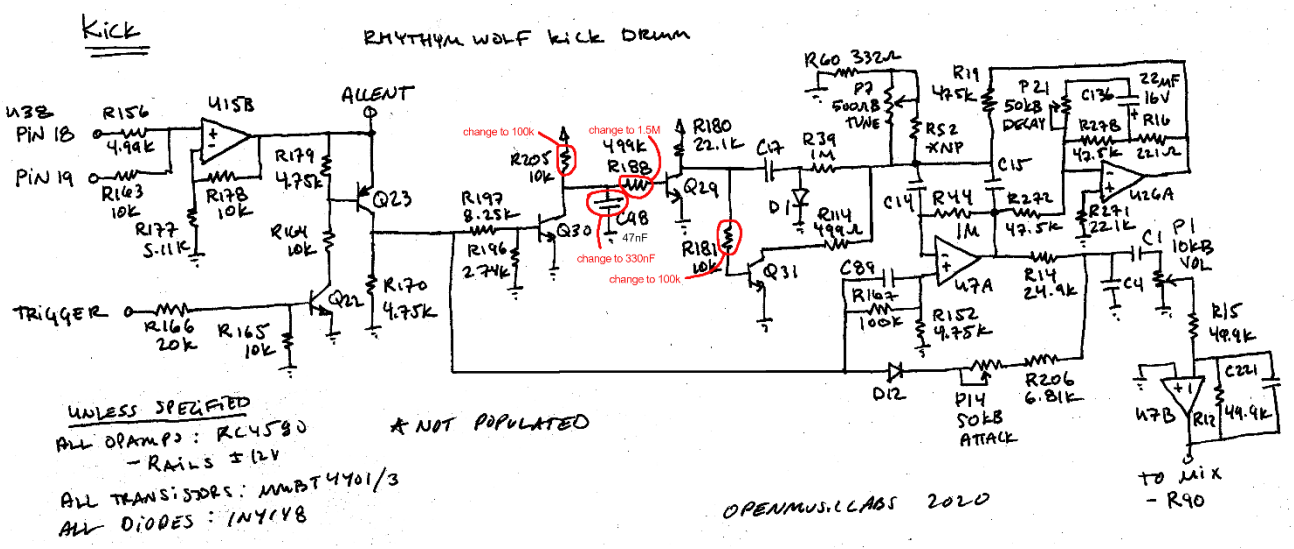
Schéma d’origine



Il est intéressant de voir le niveau auquel ils sont allés pour copier la TR-808 : sur l'amplificateur de régénération (U26A), il y a une résistance de l'entrée non inverseuse à la masse. Cela est généralement fait pour contrer la tension de décalage générée par les courants de polarisation sur les amplificateurs opérationnels, et cela se fait à travers la TR-808. Mais, c'est le seul ampli-op de la Rhythm Wolf qui utilise cette technique. De plus, le potentiomètre de sortie va à un étage d'amplification inverseur plutôt qu'à un tampon, ce qui est également le seul cas de ce genre ; je ne suis pas sûr pourquoi. Il y a l'ajout de l'"attaque", qui est une petite impulsion juste au début de la frappe de la batterie (via D12). La fréquence de la voix de batterie ne change pas tout au long de la note, comme c'est le cas sur la TR-808 en raison de ce qui semble être un échange de R205 et R188, rendant la décroissance trop rapide pour être perceptible. Dans l'ensemble, c'est la partie la mieux conçue de la Rhythm Wolf.

Modifications

Schéma modifié



J'aimais le son de la grosse caisse, donc je n'ai pas apporté beaucoup de modifications, ajustant seulement le balayage de tonalité. Après avoir effectué la modification, j'ai décidé que je l'aimais sans le balayage de tonalité pour une grande partie de la plage, mais que les basses sonnaient mieux avec le balayage de tonalité, alors j'ai fait un compromis (C98 de 100nF). Je recommande d'essayer plusieurs valeurs différentes, car le son change assez drastiquement en fonction de la profondeur et du temps de décroissance.

Idéalement, vous pourriez mettre un potentiomètre sur ce paramètre. Je pensais réutiliser le bouton d'attaque pour cela, mais je le voulais aussi, alors peut-être plus tard, j'en mettrai un, ou du moins un interrupteur. Un avertissement avec cette modification : cette partie du circuit dépend entièrement du bêta du transistor, ce qui signifie qu'il changera avec la température et entre les synthétiseurs. Pour cette raison, j'ai choisi des valeurs conservatrices et intermédiaires, de sorte que lorsque le bêta change, le circuit reste dans un point de fonctionnement raisonnable. C'est particulièrement important si vous choisissez des résistances de grande valeur, car le bêta pourrait se déplacer et les transistors ne s'allumeront plus.

1. Changer R181 à 100k. Cela ajuste la quantité de décalage de hauteur. Les tons bas et haut sont respectivement de 40 Hz et 62 Hz. Avec le décalage de hauteur, ils peuvent monter jusqu'à 64 Hz/83 Hz au début d'une note. 10k est le plus bas que j'utiliserais ici, et cela donne le décalage complet avec une transition nette entre l'extrémité haute/basse de la note. 47k donne également le décalage complet, mais avec une transition plus douce. 100k ne monte qu'à 58 Hz à l'extrémité basse et donne une transition plus longue. 220k monte jusqu'à 55 Hz et donne la transition la plus longue. Aller au-delà de 220k rend l'effet assez subtil. J'ai décidé d'opter pour 100k car les décalages vraiment élevés semblaient un peu comiques.
2. Changer R188 à 1.5M. N'importe quoi entre 500k et 3.3M serait probablement bien. Plus la valeur est grande, plus le décalage devient long pour une valeur de condensateur donnée. J'ai choisi une valeur intermédiaire pour assurer le fonctionnement à travers la température.
3. Changer R205 à 100k. Cela peut être n'importe quoi de 10k à 1M, avec des valeurs plus grandes donnant des temps de déclin plus longs. Mais, lorsque R205 devient supérieur à la moitié de la valeur de R188, le décalage maximal diminue, donc il y a des rendements décroissants. 100k à 500k est probablement idéal. Cela peut être remplacé par un potentiomètre pour rendre le temps de décalage réglable. Le potentiomètre de 50k pourrait être adapté ici, avec une résistance de 2,2k en série pour limiter le courant à la valeur maximale. Idéalement, un potentiomètre de 500k serait utilisé, car le courant est toujours assez élevé avec le 50k/2,2k.
4. Changer C98 à 330nF. Cela modifie le temps de décalage avec R205. Pour une R205 de 100k, des valeurs de 100nF à 2,2uF sont raisonnables. 330nF donnera un temps de décalage de 40ms. La façon dont je mesure le temps de décalage est le temps qu'il faut pour que le collecteur de Q29 passe de 10V à la masse. Environ 5ms, c'est quand l'effet n'est plus perceptible, à 10ms il est subtil, et 50ms - 200ms sont très perceptibles. Après cela, il est difficile d'entendre l'effet, car le ton s'est estompé. Cette tension diminue linéairement, en augmentant R188, elle peut être rendue plus exponentielle..

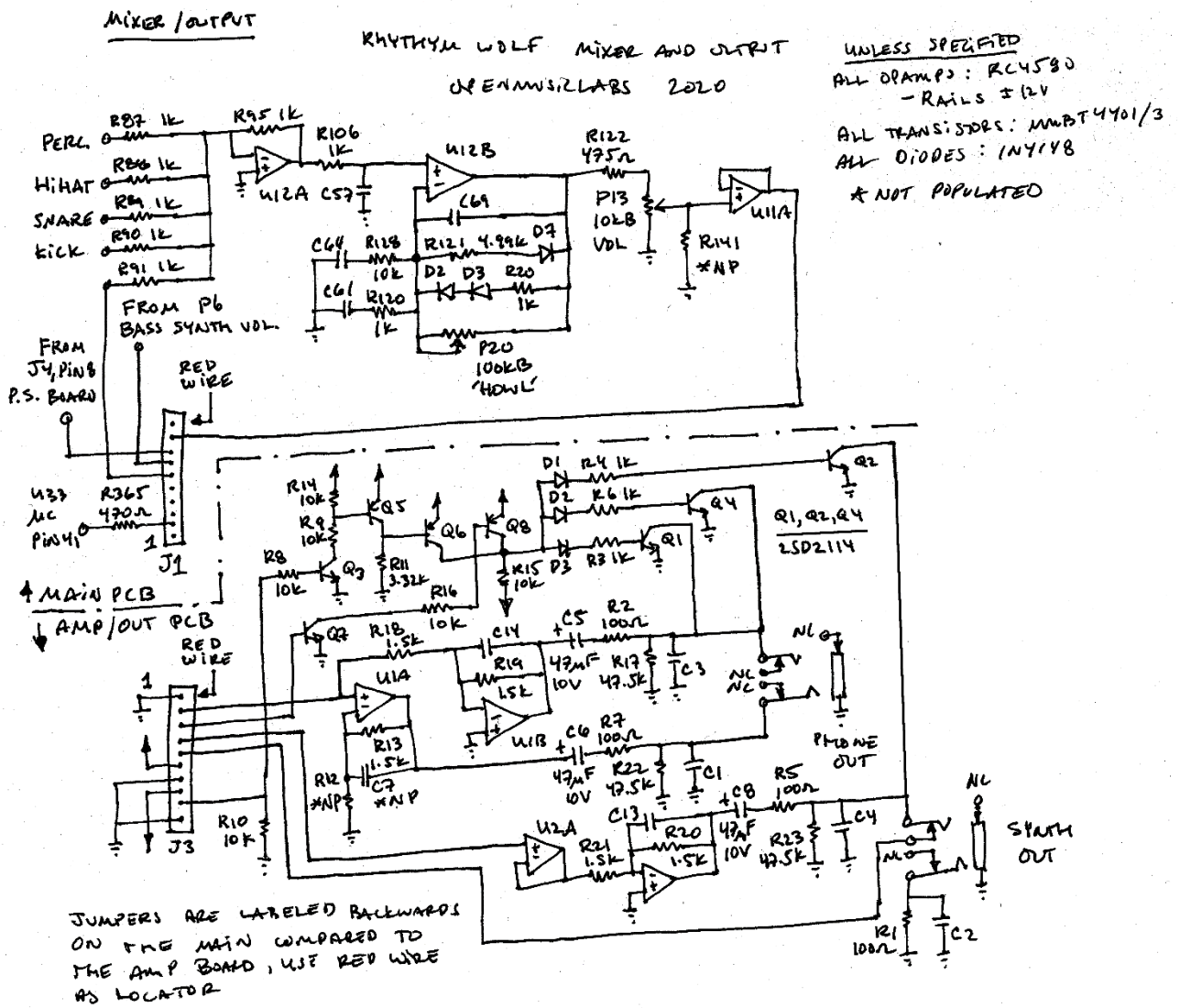
Comparison to the 808

Avec le niveau de détail qu'ils ont apporté en essayant de reproduire la TR-808, il est intéressant de noter qu'ils ont choisi une valeur du potentiomètre de TUNE qui est 1/100 de celle de la TR-808. Cela a des conséquences graves sur le comportement, et si votre objectif est de reproduire la TR-808, la seule façon de le faire est de changer P7, et R60,114 et C14,15. En raison de l'impédance basse à cette jonction de ces composants, le niveau de tension n'atteint jamais une valeur suffisamment élevée pour faire passer suffisamment de courant à travers R39 pour ajuster la hauteur du son de la batterie. D1 ne s'allume jamais. Ce changement de hauteur peut être compensé avec les modifications ci-dessus, en réglant le temps de décroissance à 150 ms. Mais, le changement de hauteur sera légèrement différent, car la TR-808 change principalement de hauteur lorsque l'équivalent de D1 s'allume, ce qui se produit uniquement lors des excursions positives, donc sur la moitié du cycle. Il existe donc deux mécanismes différents pour le changement de hauteur, l'un au début du son pendant environ 10 ms, et l'autre pendant la durée du son, ce qui est un effet beaucoup plus subtil.

Section mixage

Vue d’ensemble

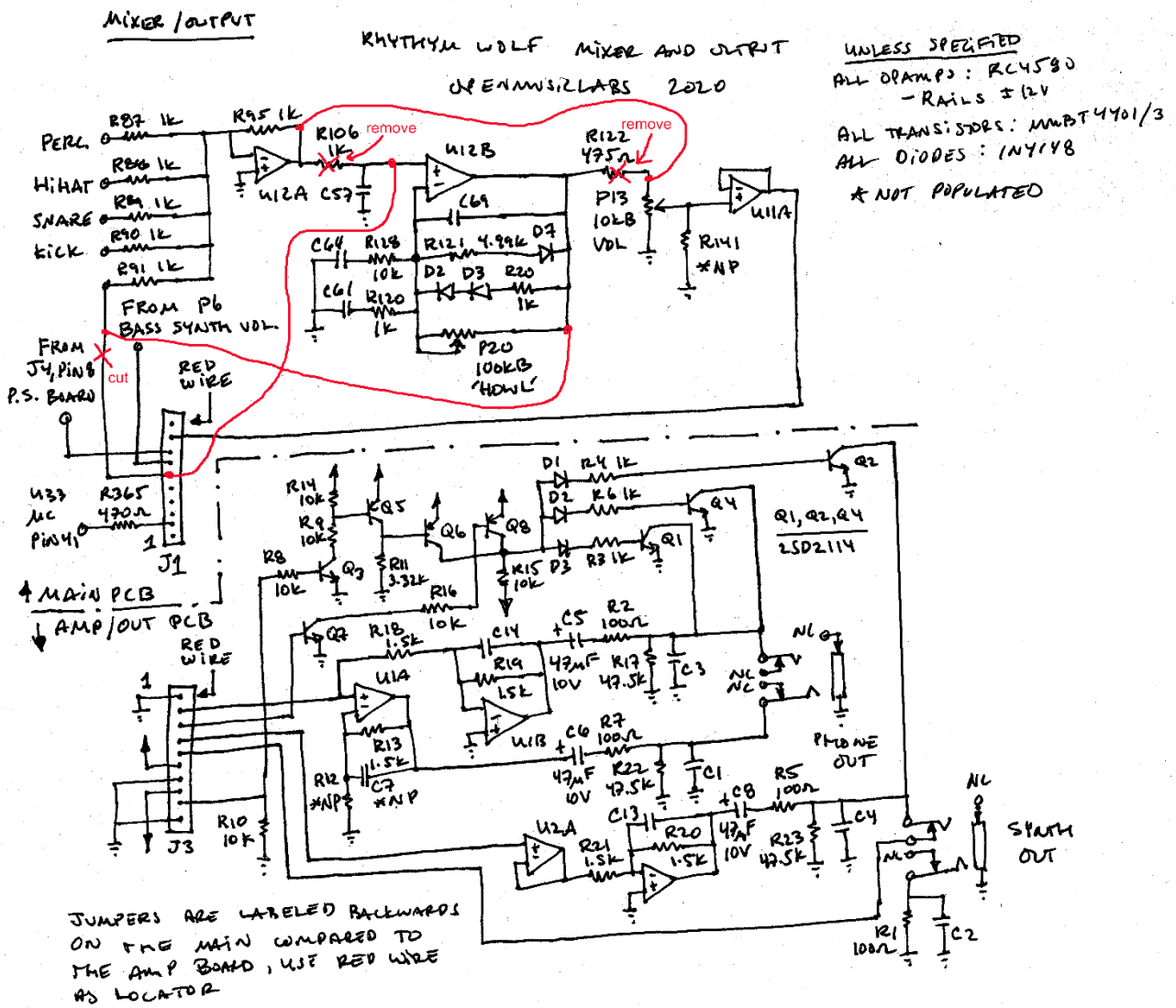
[Schéma d’origine]



La section du mélangeur est un amplificateur sommateur assez simple suivi d'un amplificateur de sortie différentiel pour la sortie casque. La sortie du synthé est envoyée à une prise séparée avant le mélange, et n'est incluse dans le mélange que si aucun connecteur n'est inséré. Il y a un circuit de sourdine sur les sorties qui est contrôlé à la fois par la carte d'alimentation et le microcontrôleur. Cela semble un peu redondant, car le microcontrôleur contrôle également la génération de la tension analogique, et je ne vois pas pourquoi vous voudriez annuler la sourdine de la sortie avant d'allumer l'alimentation analogique. Entre le mélangeur et l'amplificateur de sortie se trouve le circuit de distorsion "howl". Celui-ci a une distorsion asymétrique, un filtre passe-bas et un filtre passe-haut avec une pente inférieure à 6 dB/octave.

Modifications

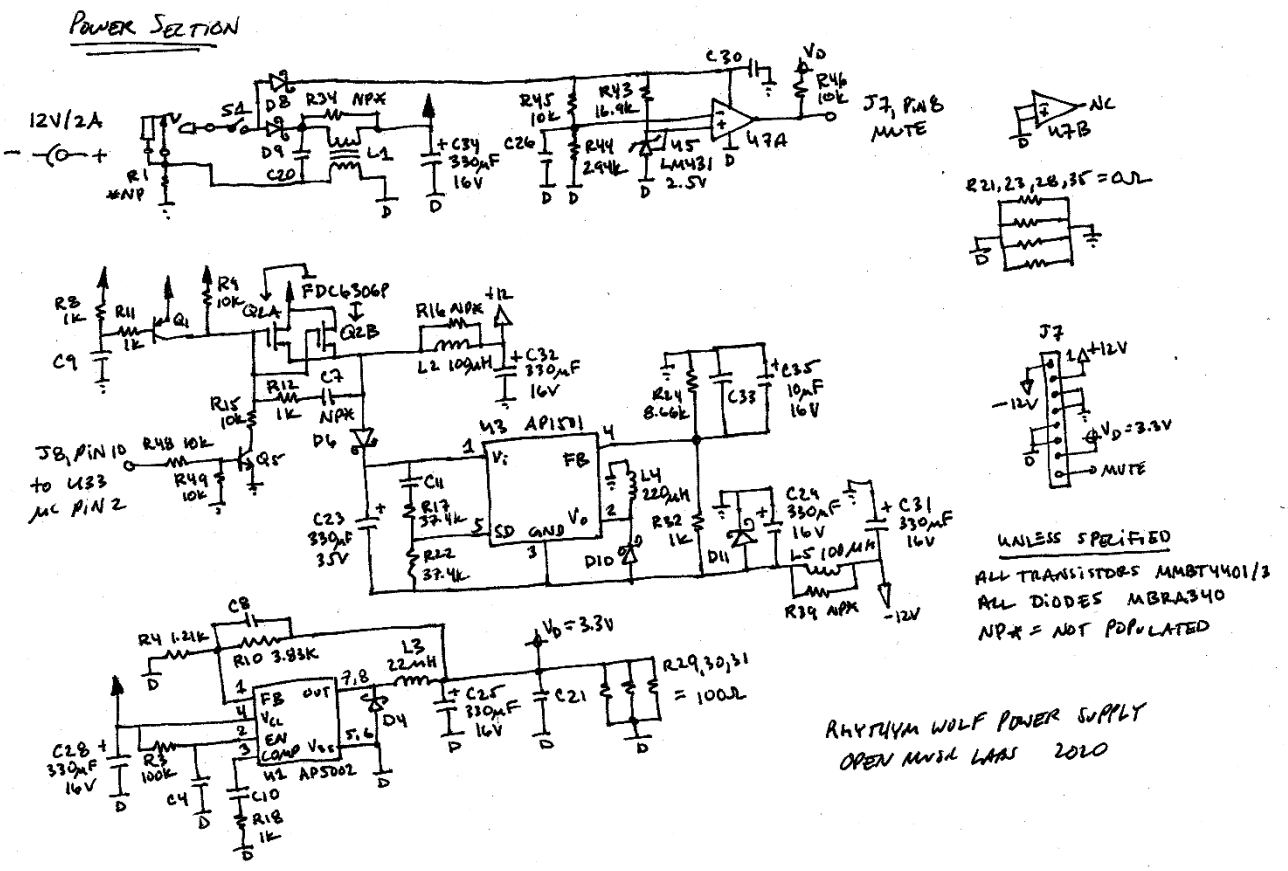
[Schéma modifié]



L'application de la fonction "howl" à toutes les voix n'a aucun sens, et je l'appliquerais soit à la grosse caisse, soit à la basse. C'était une décision difficile, mais j'ai finalement décidé de l'appliquer à la basse. Cela a été une affaire simple de couper la piste du potentiomètre de basse, de retirer R106 et R122, puis de câbler la basse vers U12B et revenir à son entrée normale dans le mélangeur. La sortie du mélangeur est ensuite câblée pour contourner U12B et aller directement à U11A et à la sortie. Une chose similaire pourrait être faite pour la grosse caisse.

Section d'alimentation

Schémas



La RW tire 0,5 A du transformateur mural 12V (il est évalué à 2A). Ce transformateur mural à commutation 12V passe par un filtre LC pour la ligne d'alimentation analogique interne +12V. Il alimente également un régulateur à découpage pour créer l'alimentation analogique -12V et l'alimentation numérique 3,3V. Il y a un comparateur utilisé pour couper le son de la sortie audio pendant un moment au démarrage, et un autre signal provenant du microcontrôleur pour retarder le démarrage des rails analogiques jusqu'à ce que la ligne numérique soit configurée.