INTRO | mki x es.edu

Hey there, thanks for buying this DIY kit! We – **Erica Synths** and **Moritz Klein** – have developed it with one specific goal in mind: teaching people with little to no prior experience how to design analog synthesizer circuits from scratch. So what you'll find in the box is not simply meant to be soldered together and then disappear in your rack.
Instead, we want to take you through the circuit design process step by step, explaining every choice we've made and how it impacts the finished module. For that, we strongly suggest you follow along using **LABOR[1],** which is an all-in-one circuit prototyping tool that allows you to experiment and play around with your components in a non-permanent way. To help you with this, we've included suggested breadboard layouts in select chapters.
In addition to this, you can also experiment with some of the chapter's circuits in a **circuit simulator** called CircuitJS. CircuitJS runs in your browser. You'll find weblinks in the footnotes which will direct you to an instance that already has example circuits set up for you. We strongly encourage you to fiddle with the component values and general structure of those circuits to get a better understanding of the concepts we're laying out.
Generally, this manual is intended to be read and worked through front to back, but there were a few things we felt should go into a dedicated appendix. These are general vignettes on electronic components & concepts, tools, and the process of putting the module together once you're done experimenting. Don't hesitate to check in there whenever you think you're missing an important piece of information. Most importantly though: have fun!

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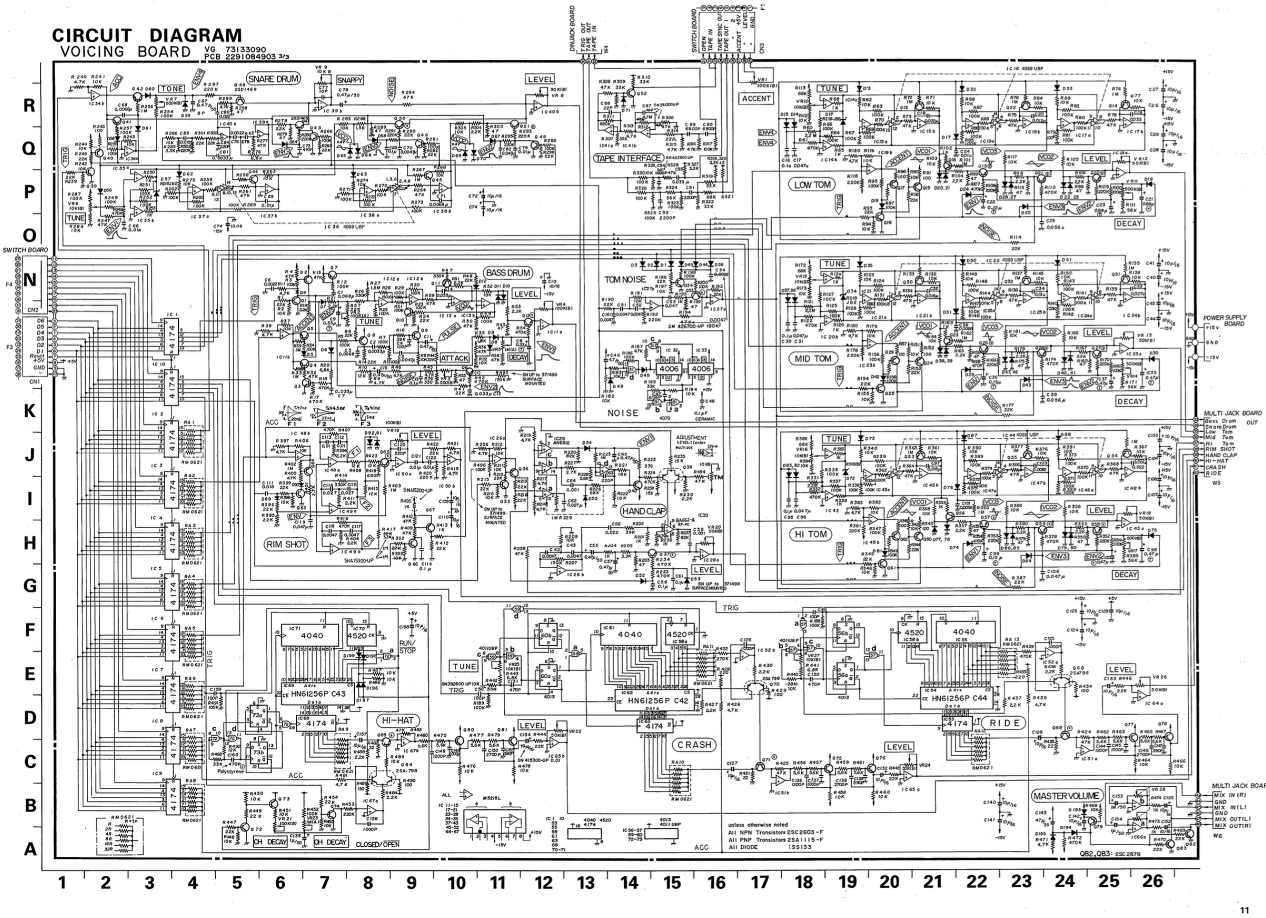
[1] You can also use a standard breadboard, but this will require you to get a little creative when adapting the suggested layouts. You'll also need to do some additional engineering to get the different supply voltages.

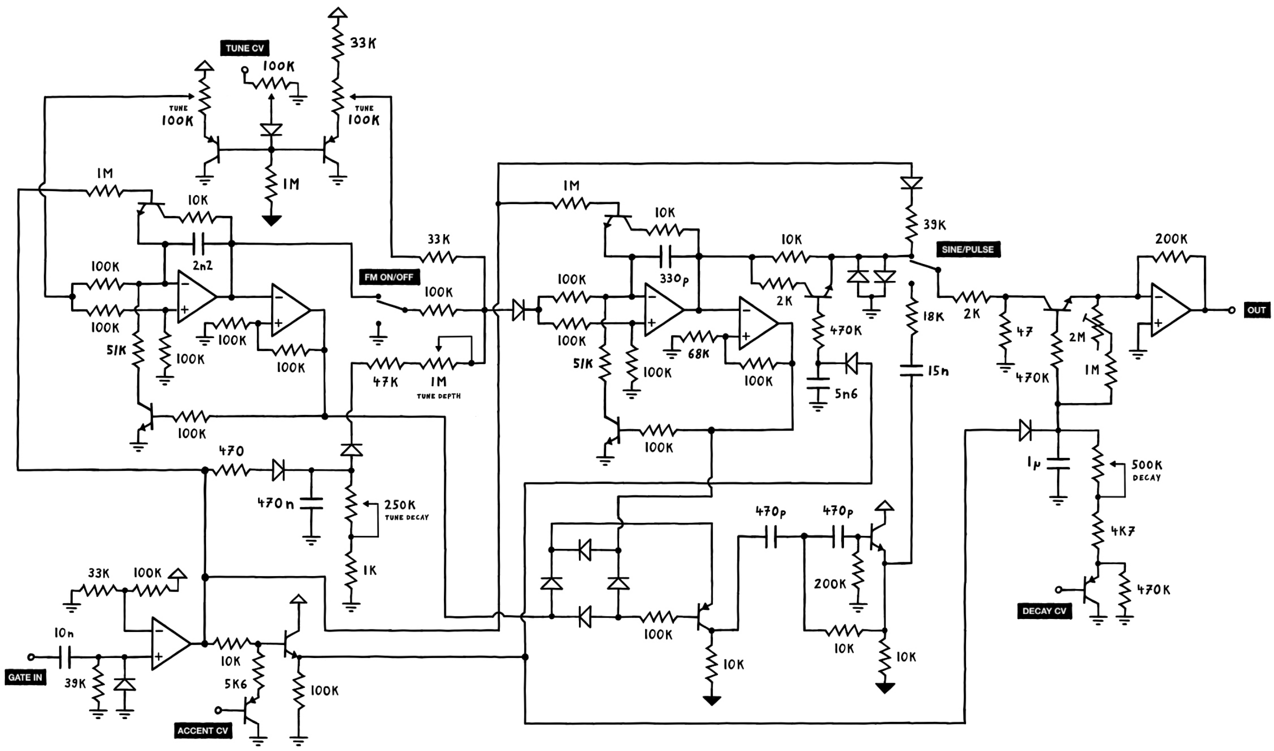
THE mki x es.edu FM DRUM

These are the schematics for Roland's TR-909, an iconic drum machine from the mid-1980s.



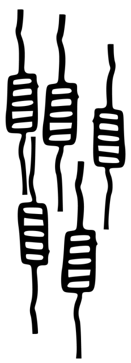
I've been staring at this absolute maze of components for a long time, trying to figure out how I can save a vintage 909 I bought a while ago. While it worked just fine initially, the kick, snare and toms quickly started to misbehave. To try and troubleshoot, I began breadboarding parts of the schematic, isolating different sections to understand what's wrong. But while doing that, I got a little carried away.

Instead of fixing my machine, I ended up creating something new: an FM drum circuit that combines the punchy character of the 909 with the metallic textures of FM synthesis.



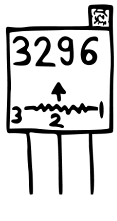
BILL OF MATERIALS

Before we start, please check if your kit contains all of the necessary components. In addition to a PCB, panel and power cable, your box should also contain:



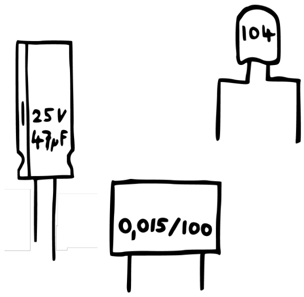
An array of resistors. The specific values (in ohms, which you should check for with a multimeter) are

1M	x4
470k	x3
330k	x1
200k	x1
100k	x16
68k	x1
51k	x2
47k	x1
39k	x2
33k	x3
18k	x1
10k	x7
5k6	x1
4k7	x1
2k	x2
1k	x4
470	x1
47	x1
10	x2



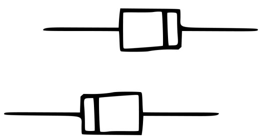
A trimmer potentiometer. The specific value (which is encoded & printed on top) is

2M (W205) x1



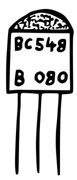
A bunch of capacitors. The specific values (which are printed onto their bodies) are

47 uF	x2
1 uF	x1
470n	x1
100 nF	x4
15 nF	x1
10 nF	x1
5.6 nF	x1
2.2 nF	x1
470p	x2
330p	x1



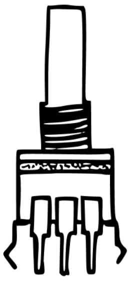
Some diodes. The specific model names (which are printed onto their bodies) are

1N4148 (signal)	x18
1N5819 (schottky)	x2



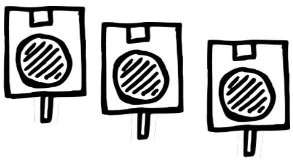
A couple transistors. The specific model names (which are printed onto their bodies) is

BC548B (NPN)	x8
BC558 (PNP)	x5



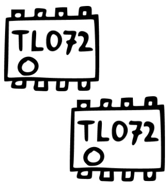
A handful of potentiometers. Their specific values (which may be encoded & printed onto their bodies) are

1M (B105) x1
500k (A504) x1
250k (B254) x1
100k (A104) x2
100k (B104) x1



A few jack sockets. The specific models (which you can identify by their color) are

Switched mono (black) x5



A couple chips. Their specific models (which are printed onto their bodies) are

TL074 (quad op amp) x1
TL072 (dual op amp) x1

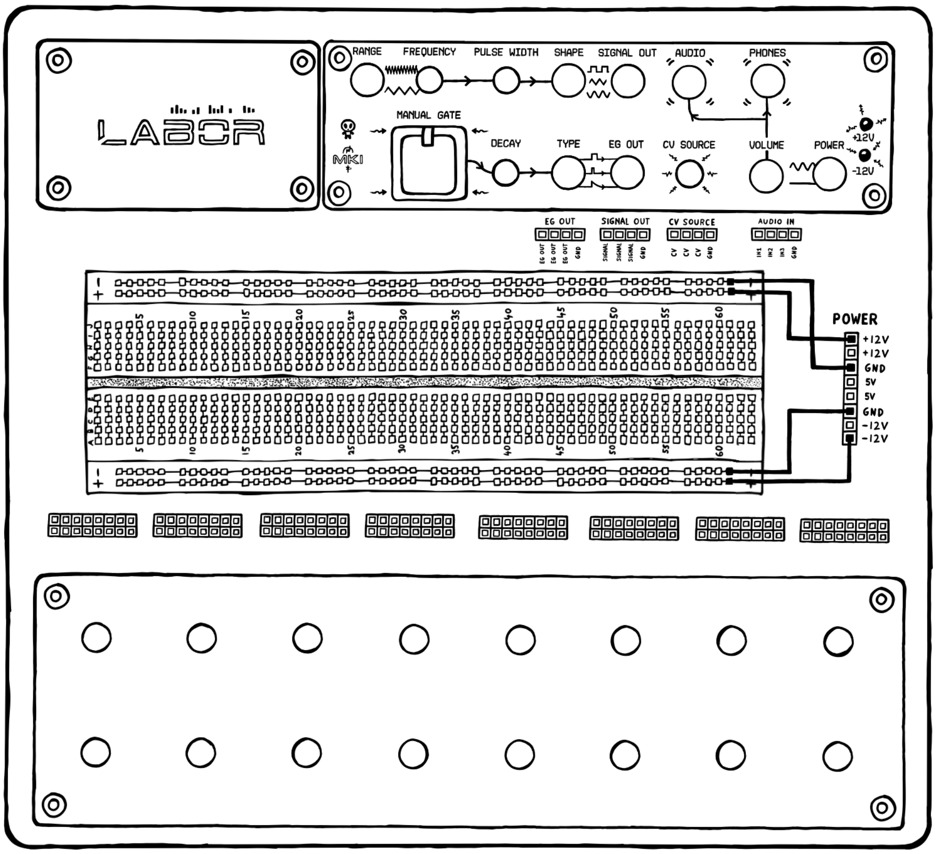
A few switches. The specific model is

Single pole, double throw x2

You will also find a few sockets that are only relevant when assembling the module in the end.

USAGE WITH MKI x ES LABOR

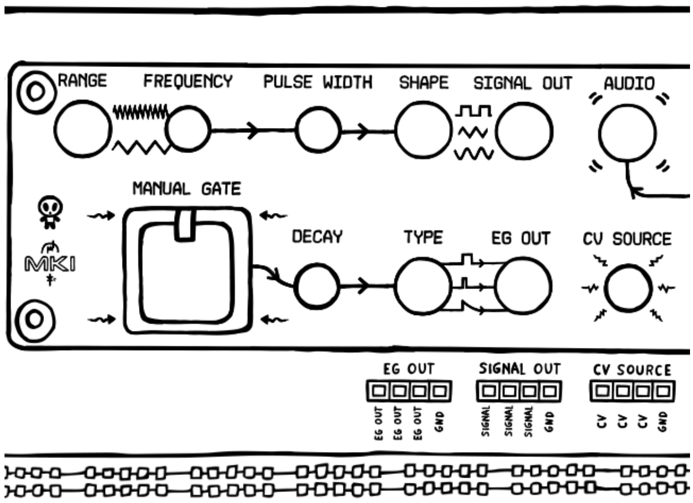
We recommend that you follow this guide using an **MKI x ES LABOR** prototyping board. **LABOR** comes equipped with everything you need for testing the circuits we lay out: a standard 830 tie point breadboard, an integrated dual power supply with over current protection, a manual gate/trigger/envelope generator, an LFO, a variable CV source, an output amplifier, and a modular interfacing section where you can insert all of your interfacing components like potentiometers, jack sockets, and switches.



Before you get started, connect the slots labeled **GND** on the power header to both breadboard rails labeled **–** (minus). Next, connect one slot labeled **+12 V** to the top breadboard rail labeled **+** (plus), and one slot labeled **–12 V** to the bottom breadboard rail labeled **+** (plus).

To listen to your circuit, you don't even need to set up an output jack socket. Instead, use the built-in output amplifier at the top of the device. Just plug your circuit's signal output into the header labeled **AUDIO IN**, and then connect your headphones to the **PHONES** output jack (or a line-level device like a standalone external speaker to the **AUDIO** output jack).

Sometimes, this guide will ask you to use external gear like sequencers or LFOs to send CV, audio signals, triggers or gates into your circuit. With **Labor**, there's no need for extra equipment – just use the built-in oscillator (audio/LFO), CV source or manual gate/trigger/envelope generator. You can grab all of those via the headers labeled **EG OUT**, **SIGNAL OUT**, and **CV SOURCE** and connect them to the designated points on the breadboard.

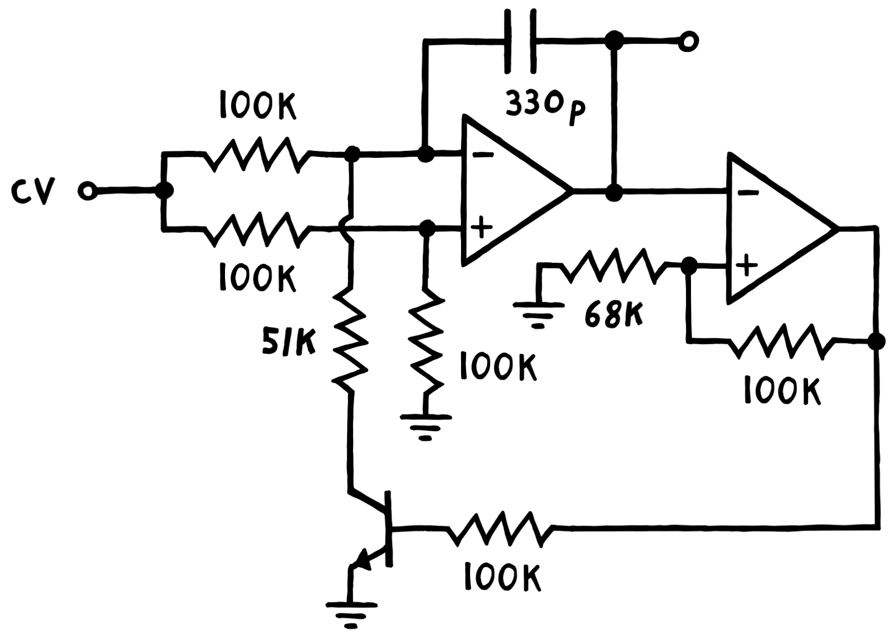


909-STYLE TRIANGLE VCO

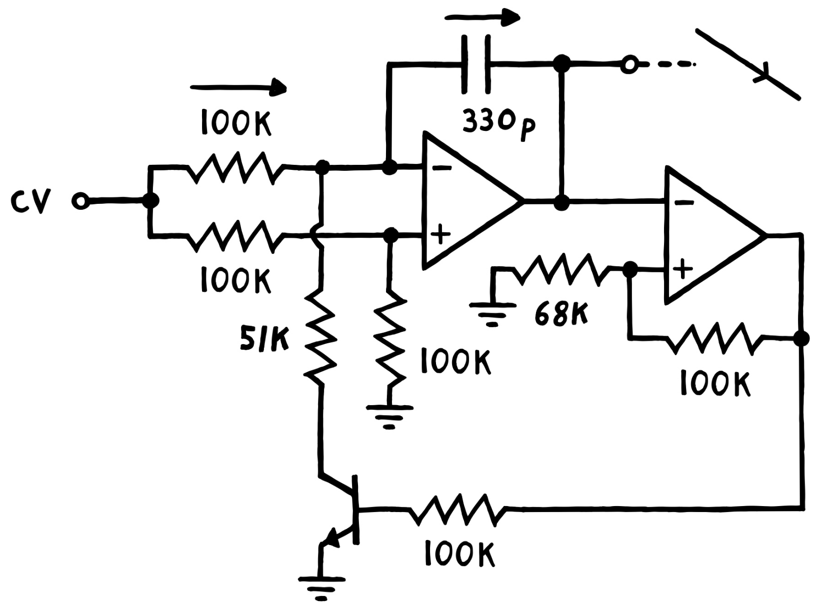
To understand how the FM Drum circuit works, we'll retrace my steps and start by isolating the core concepts of the TR-909's design. First up: the oscillators. Compared to the TR-808, Roland decided to step up their game here. **Because where the 808 used bridged-T oscillators – highly efficient little circuits capable of producing percussive sine wave hits with just a handful of components – the 909 uses full blown VCOs that generate continuous triangle waves.**

This shift was no accident, of course. Roland tried to compete with sample-based drum machines like the LinnDrum, so they needed the 909 to sound much more realistic than the robotic and sterile 808. This forced them to bite the bullet and trade efficiency (and lower manufacturing costs) for a more complex, but also more flexible subtractive synthesis approach. Which starts with a proper VCO.

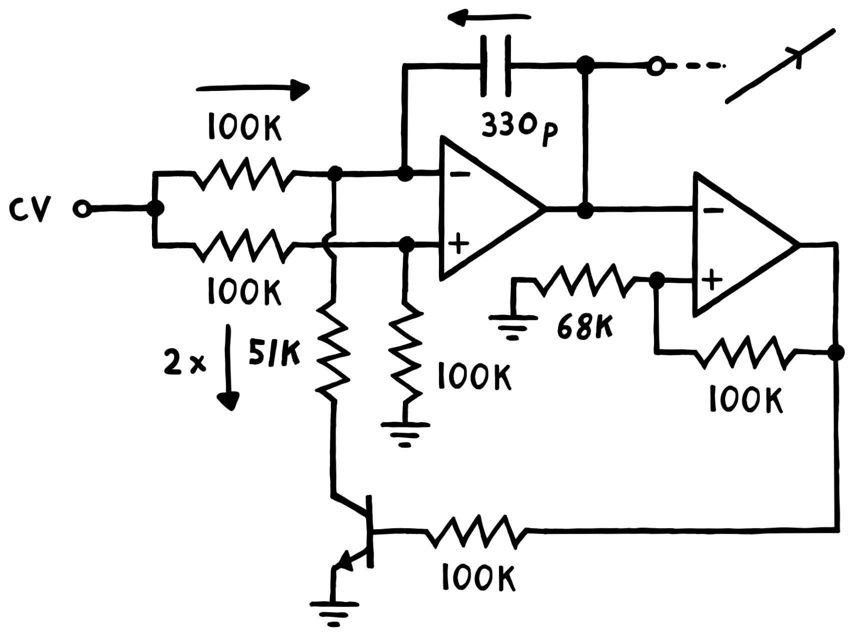
And though the 909's kick, snare and tom oscillators implement it in slightly different ways, they're all variations of the same basic idea: combining an integrator, a schmitt trigger inverter and an injection point for a control voltage in a feedback loop to generate a triangle wave. Here's my slimmed-down take on this idea.



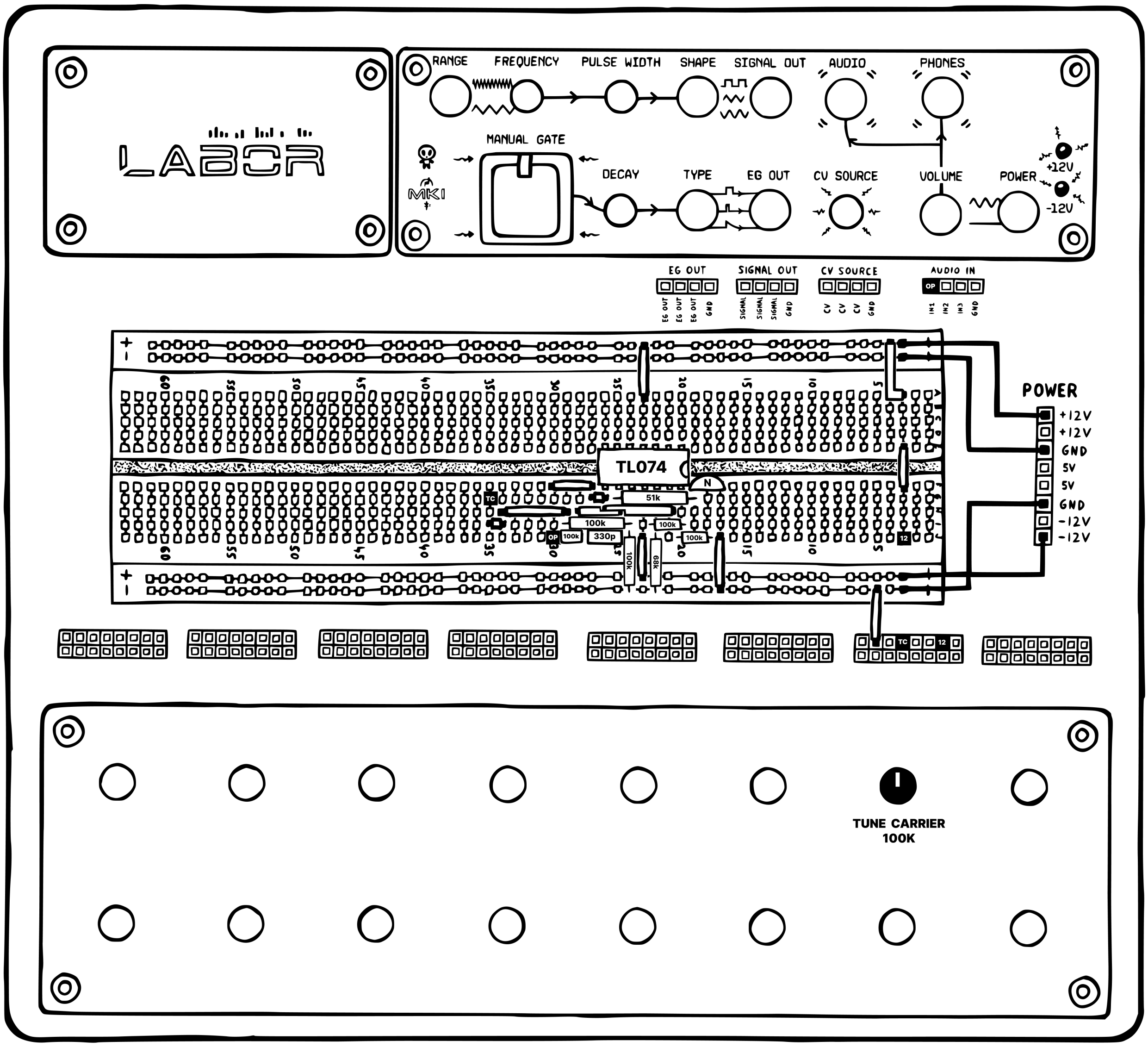
It works like this. We'll assume that the schmitt trigger inverter (right op amp) starts out latched into the low state, disabling the NPN transistor. At the integrator's (left op amp) non-inverting input, we get exactly half of the control voltage, and the op amp will do everything it can to keep the inverting input at the same level. So as the control voltage pushes a current through the upper resistor and into the capacitor, the integrator will pull the same amount of current out of the cap on the other side.



This way, the voltage on the left will stay at half of the CV, while the voltage on the right will steadily drop. This keeps going until the integrator's output crosses the schmitt trigger inverter's lower input threshold, causing it to latch into the high state.
This enables the NPN transistor, allowing current to flow from the integrator's inverting input to ground. How much current? Well, we know that the voltage difference between the CV and the inverting input is the same as the one between the inverting input and ground. Simply because the op amp will make sure the inverting input stays at half the CV.
We also know that the resistance on this path is (about) double the resistance on the other. So the current flowing from the inverting input to ground must be double the current flowing into the input. **And for this to work, the op amp needs to provide the missing amount of current**.



Which, conveniently, is exactly as much as is flowing in from the CV. **Causing the integrator's output to rise at the same rate it was previously dropping at.** Until it hits the schmitt trigger inverter's upper input threshold, and the whole cycle repeats.
At the integrator's output, this creates a triangle oscillation whose frequency is determined by the CV. Because if the CV rises, both the current flowing into and out of the integrator's capacitor increases by the same margin. To test our VCO in practice, here's how you could set it up on LABOR.



When you check the circuit's output on an oscilloscope (the Labor Scope add-on is great here), you should see that the integrator does produce a continuous triangle wave.[2] Using the CV potentiometer, which we've set up as a simple variable voltage divider between 12 V and ground, you should be able to adjust the oscillation frequency.

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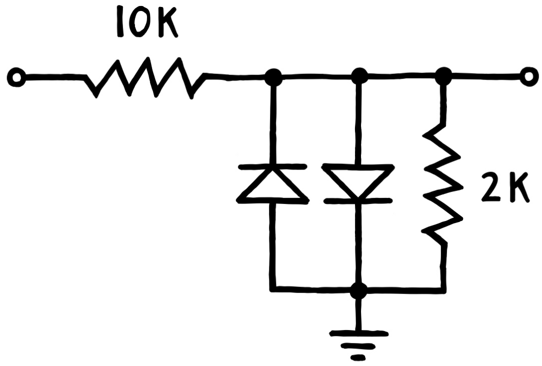
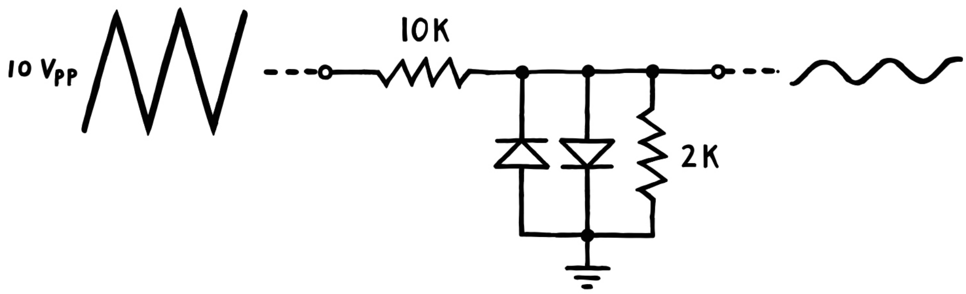
[2] You can try this chapter's circuit in a circuit simulator. I've already set it up for you right here: https://tinyurl.com/252vc9hj – you can change all values by double clicking on components.

TRI TO SINE WAVESHAPER

Since triangle waves sound a little too buzzy and bright compared to an actual drum hit, Roland needed a way to remove the harmonics above the base pitch. **But instead of a low pass filter, they decided to go with a dead simple waveshaper**. Which uses just a couple resistors and diodes to turn their triangle wave into a somewhat dirty sine wave.

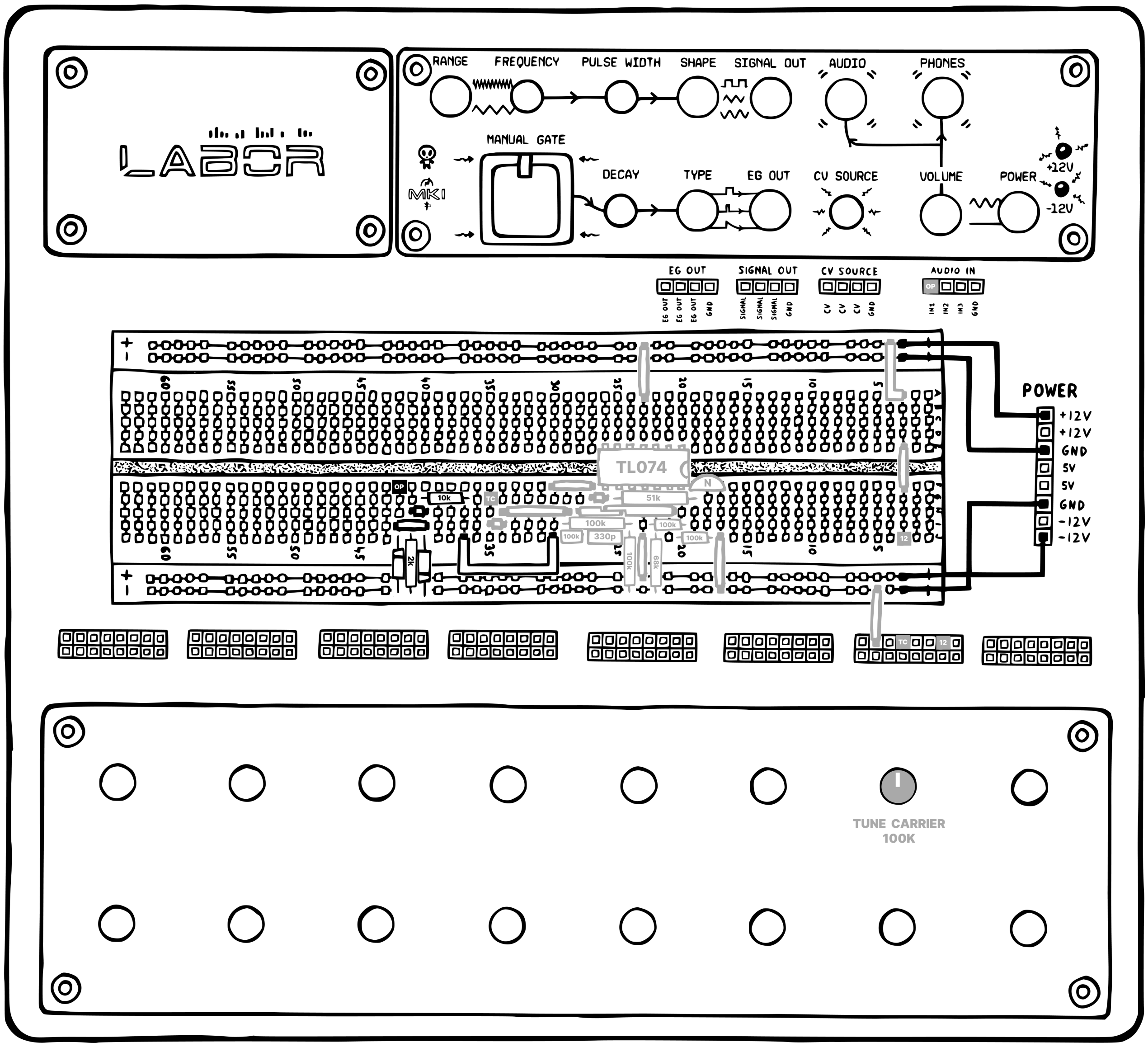


To understand how it works, let's assume that we apply our triangle to the resistor on the left. Then the other resistor going to ground will scale that triangle down: from around 10 V peak-to-peak to just 1.6 V peak-to-peak. Or rather, it would if it weren't for the diodes coming up from and going down to ground. **Because as the triangle tries to push past the 400mV line in either direction, the respective diode will gradually open up and sink (or source) more and more current**.



This will gently shave off the tips of the triangle, turning it into a rough sine wave. With the exact shape determined by the amount of scaling we apply: too little, and the tips become flat. Too much, and they become too pointy.

Here's how you can try this for yourself.



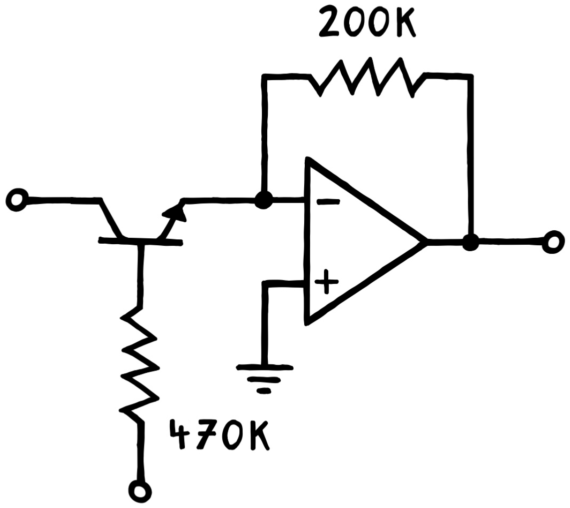
For our triangle wave, the amount of scaling is just right: the waveshaper should give you a pretty decent sine wave on the oscilloscope that's noticeably less buzzy.[3]

[3] You can try this chapter's circuit in a circuit simulator. I've already set it up for you right here: https://tinyurl.com/24p8j6hg – you can change all values by double clicking on components.

CRUDE VCA

There’s only one problem with it: it’s not percussive at all. This is one of the key advantages the TR-808’s bridged-T oscillators had – they produced sine waves with gradually decreasing amplitudes. To mimic this, we’ll have to set up a proper VCA and use it to modulate our sine wave’s amplitude.

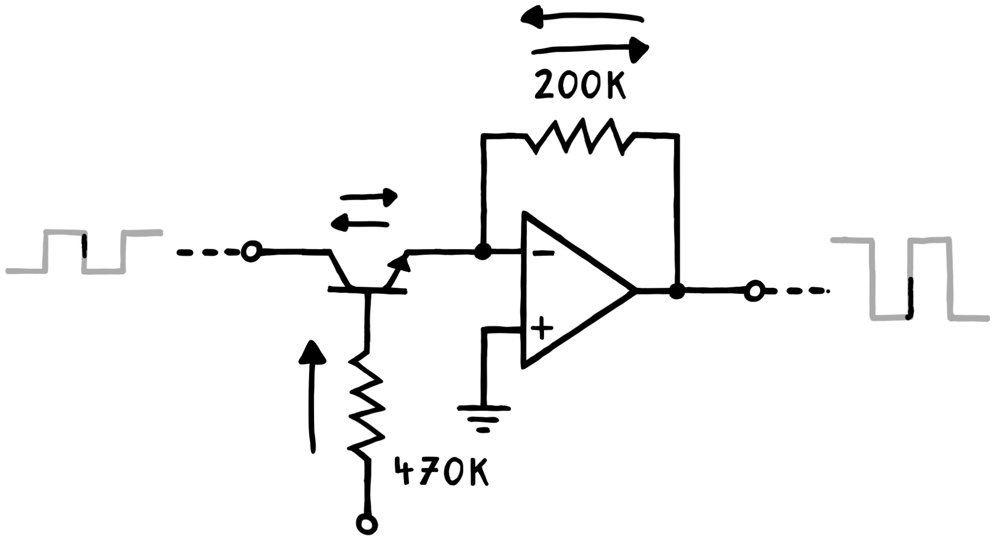
Normally, VCAs are pretty complex. Even the simplest designs use plenty of components. But in typical Roland fashion, the 909 engineers found a way to boil things down to the bare essentials. **Their VCA uses just 4 components: one op amp, one NPN transistor, and two resistors**.



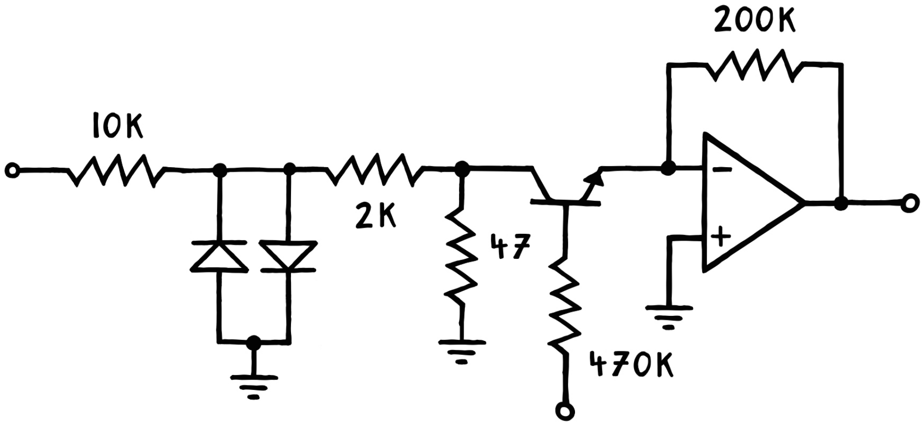
In essence, this design is just a regular inverting amplifier, but the input resistor has been replaced with an NPN transistor as a fake voltage controlled resistor. **For this to work, the input signal needs to be extremely low in volume: just around 20 mV peak-to-peak**. Then, the transistor will be locked into its saturation mode, where a large current flows into its base. This current’s ultimate direction is then modulated by the tiny oscillation at the transistor’s collector.

When the collector voltage goes high, some base current flows out of the emitter. And when it goes low, it flows entirely out of the collector – while also dragging current from the emitter with it. **This way, the transistor converts an oscillating voltage into an oscillating current of varying intensity depending on the current we push into the base**. In practical terms: if we increase the voltage applied to the base resistor, there is more base current to be modulated, and so the current swing at the emitter gets bigger.

Now, because the op amp wants to keep the voltage at its inverting input near 0, it will adjust its output voltage to neutralize the current swing. Essentially, it converts the swinging emitter current into a swinging voltage.



And since the emitter current depends on the base current, we can control the output signal's gain by changing the base voltage. And that's it: we've got a VCA![4] Of course, since most input signals will be way too loud to keep the transistor in saturation, we'll also add a strong voltage divider at the VCA's input. Which can, conveniently, double as the scaling resistance for our waveshaping circuit, since it's essentially just a 2k resistance to ground.



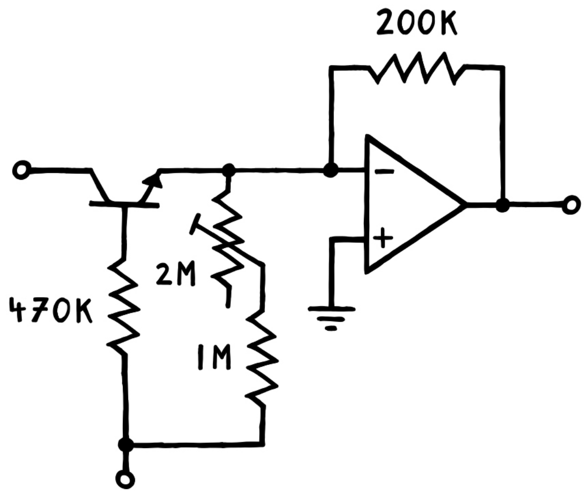
There is one small issue with this design. If you set this up on your breadboard, you'd notice that the output waveform has a positive DC offset that appears to be tied to the control voltage.[5] What's up with that?

Well, the issue is that when the collector voltage is 0, current is being pulled into the emitter. And the higher the control voltage, the more current flows backwards. This

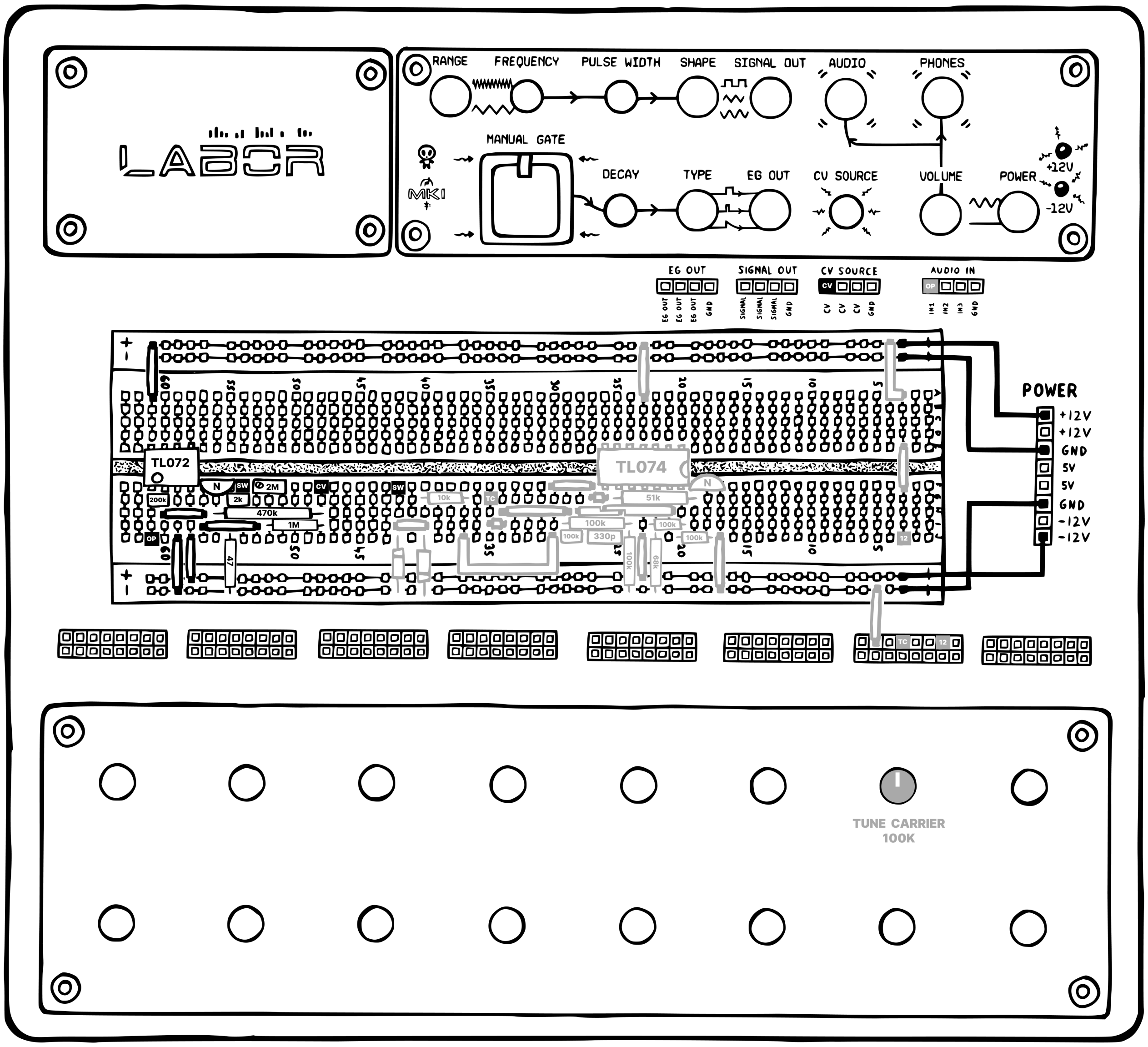
[4] You can try this chapter's circuit in a circuit simulator. I've already set it up for you right here: https://tinyurl.com/2d68mkl5 – you can change all values by double clicking on components.

[5] Please note that the offset could also be negative, depending on the characteristics of the specific op amp chip you've received in the kit. If it is, please try any other TL072 chips you might have laying around until you find one that gives you a positive offset. If you don't have one (or can't find one), just proceed with the chip you have. The offset is not that crucial to the final result. Still, sorry about the confusion here – I noticed this issue way too late in the development of the circuit to fix it in time.

forces the op amp to counteract, giving us an elevated output voltage when it should actually be neutral. How do we fix this? **Easy – we neutralize the backwards current flow by connecting our control voltage to the emitter via a big resistance**. Now, the issue here is that the exact amount of backwards current varies quite heavily between different transistors – even using the exact same model. So instead of trying to select the perfect resistor value here, we'll combine a 1M baseline resistor with a 2M precision trimmer.



This way, we can dial in the ideal amount of compensation current for our specific transistor.

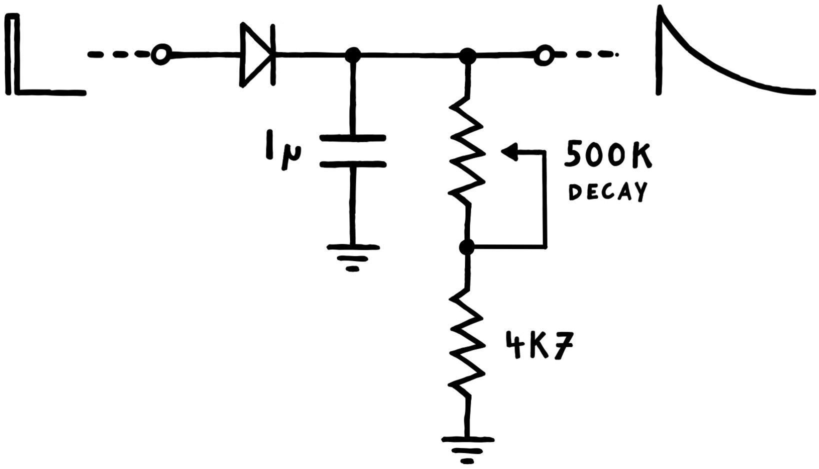


To center your sine wave, you'll need to view it on your oscilloscope and then fiddle with the trimmer until it's properly aligned with the 0 V line.[6]

[6] Please note that this will only work properly if your op amp causes the VCA to have a positive DC offset in the first place. Otherwise, you can only make the output look good enough – which might even mean omitting the trim pot and resistor entirely.

DECAY ENVELOPE

Great! But our circuit still doesn't sound like a drum. To get there, we'll need to add an envelope generator to drive our VCA and turn the static sine wave into a quick, percussive burst.c The 909's envelope generators are mostly just a diode, a capacitor, a potentiometer and a small resistor. We can copy this straight up.



If we now apply a trigger to the diode, the cap will fill up instantly and then slowly discharge via the potentiometer and resistor. This generates a falling voltage curve, whose steepness depends on the amount of resistance between the cap and ground. [7](The small resistor ensures that we don't create a short circuit at the pot's minimum setting.) If we now connect the envelope's output to our VCA's control voltage input, we should be able to shape the static sine wave into a percussive hit of varying length.

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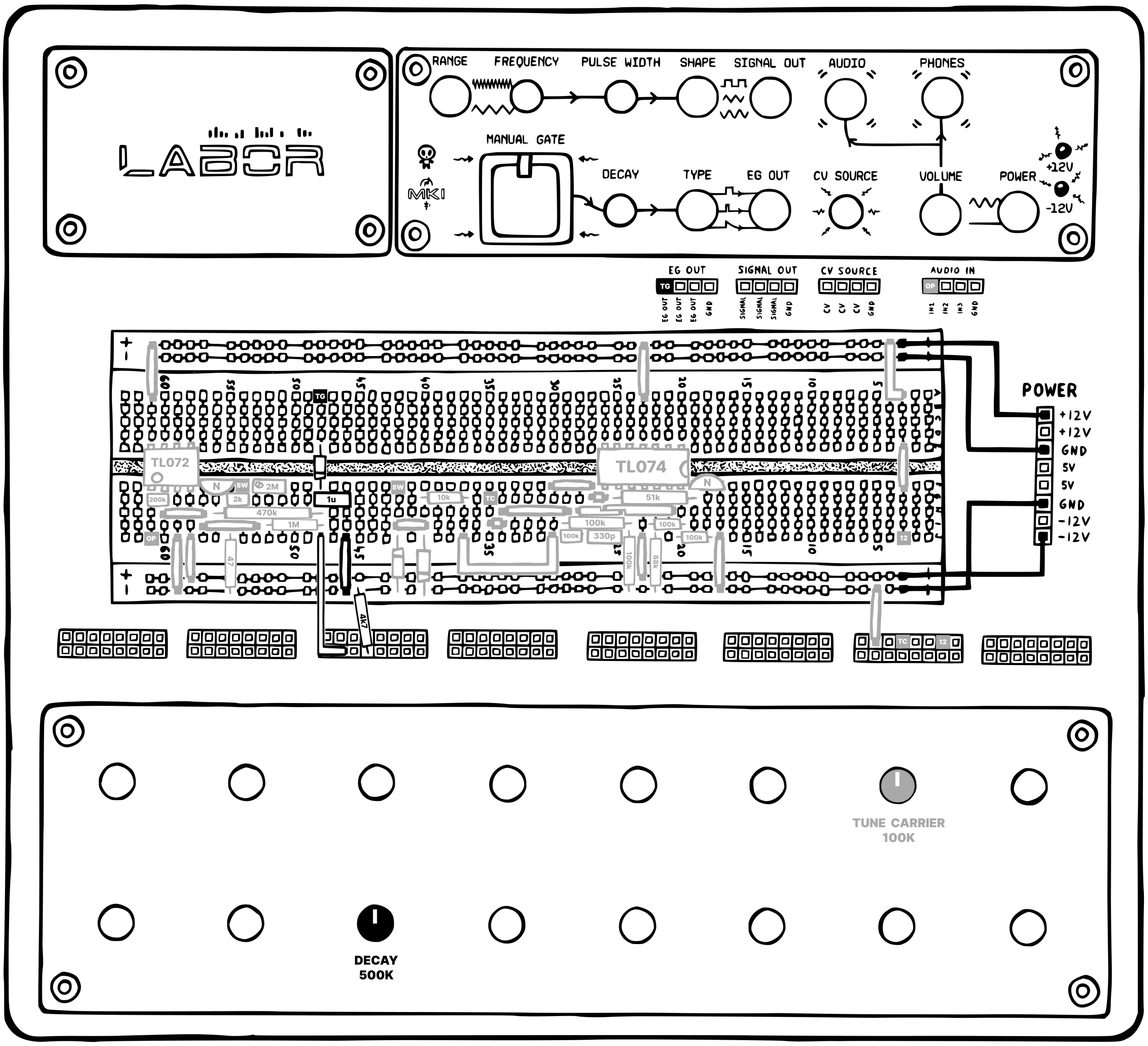
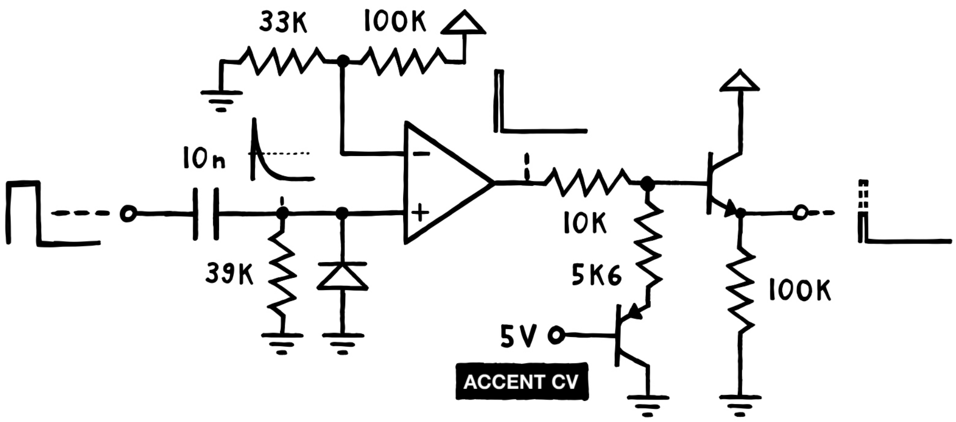
[7] You can try this chapter's circuit in a circuit simulator. I've already set it up for you right here: https://tinyurl.com/29vj9o3x – you can change all values by double clicking on components.



To test this, we'll use LABOR's manual gate, which has a dedicated setting for generating triggers. If you now push the button, you should get a percussive sine wave hit.

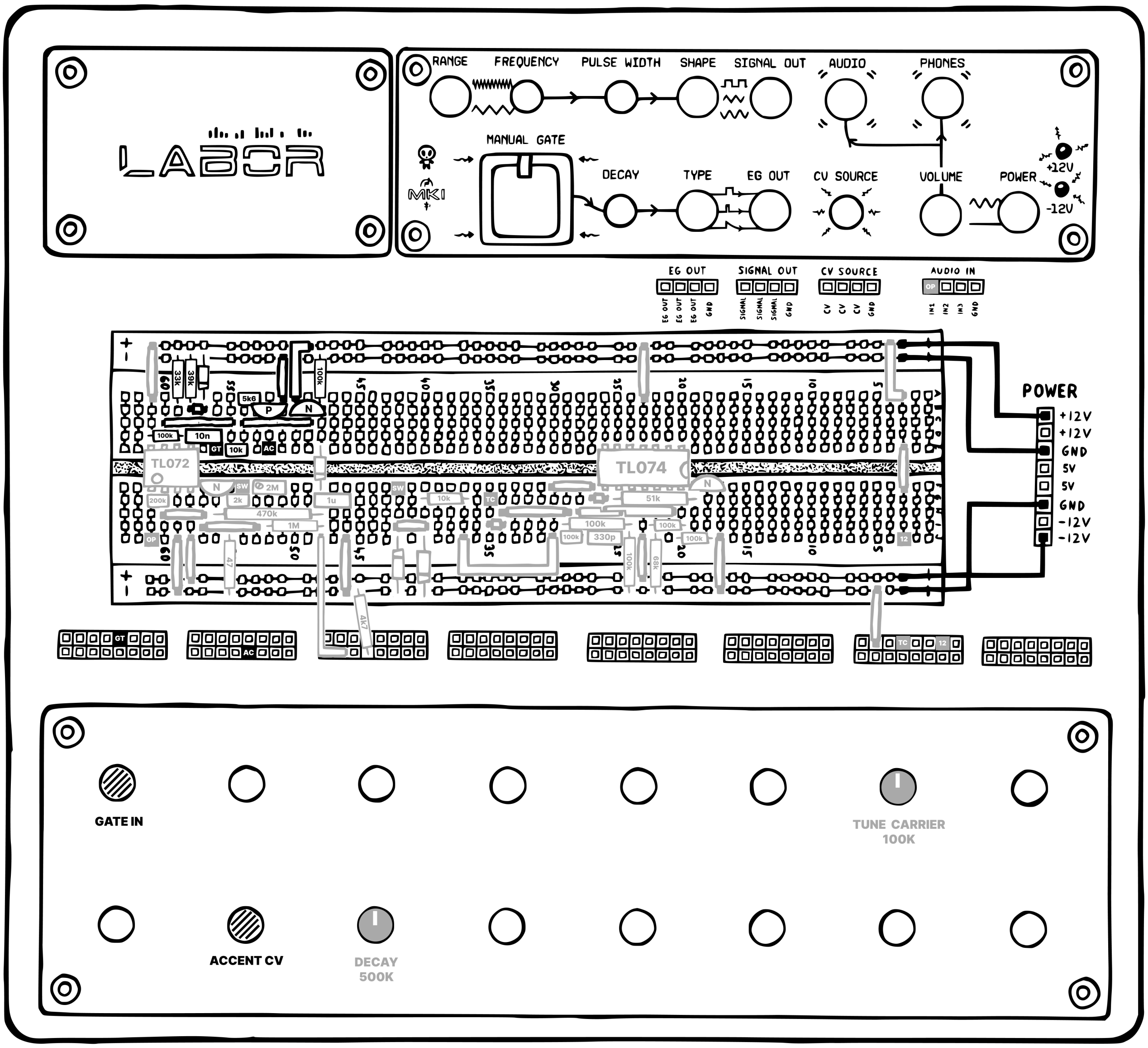
GATE TO TRIGGER CONVERTER

Sounds like a decent starting point to me! But before we can refine it, we'll need to talk about triggers some more. That's because unlike LABOR, most other gear sends out gates, not triggers. So to make our circuit compatible, we'll want to add a gate-to-trigger converter. For this, we can adapt the circuit I used in all my previous percussion circuits, which is a little easier to understand than Roland's solution for the same problem.



The Hi-Hat module's manual has an in-depth explanation, but here's the basic gist. The circuit uses a high pass with an added diode coming up from ground to turn an incoming gate into one quick, positive spike. **Then, that spike is converted into a proper trigger (a quick pulse jumping from low to high and back) using a comparator**. Finally, that trigger is sent into an accenting circuit which allows us to limit the height of the pulse to a given control voltage.

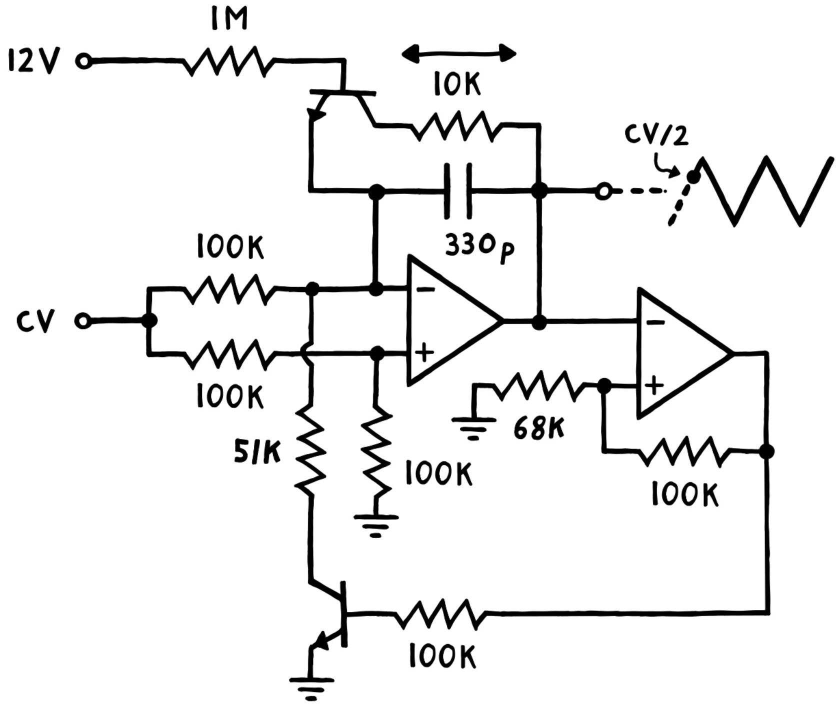
If we now use the accented trigger to drive our envelope generator, we can use the accent CV input to set the volume for each drum hit.



To test this, connect a sequencer to both the gate input and the accent CV input. (Alternatively, you can also use LABOR's EG OUT socket and the CV SOURCE.) You should be able to trigger the circuit and set its output volume simultaneously.

VCO RESET

While our circuit does work decently well, you probably noticed that the sound's transient (or attack) changes pretty dramatically between hits. We can also see this on the oscilloscope: the start of each hit looks different every time. That's because our VCO is always oscillating and could be at any point in its wavecycle when the trigger hits. **And the further it is from 0 V at that moment, the louder the clicking sound**.
Roland had the exact same issue, of course. They decided to fix it by resetting their oscillators whenever there's an incoming trigger. It works like this.



First, we bridge the integrator's capacitor with an NPN transistor and a 10k resistor. Then, we connect the NPN's base to the unaccented trigger via a 1M resistor. Now, as long as the trigger is low, the transistor will be disabled and the oscillator will operate as it normally would. **But if the trigger goes high, the capacitor discharges via the bridge, and the voltages on both sides will equalize**.[8] And when the trigger goes low again, the bridge will be disabled and the oscillator will start its wavecycle from this point. (Which is exactly 1/2 of the CV, by the way, since that's the voltage we apply to the non-inverting input.) This way, every trigger should produce the same output – because we always reset the VCO to the same starting point.[9]

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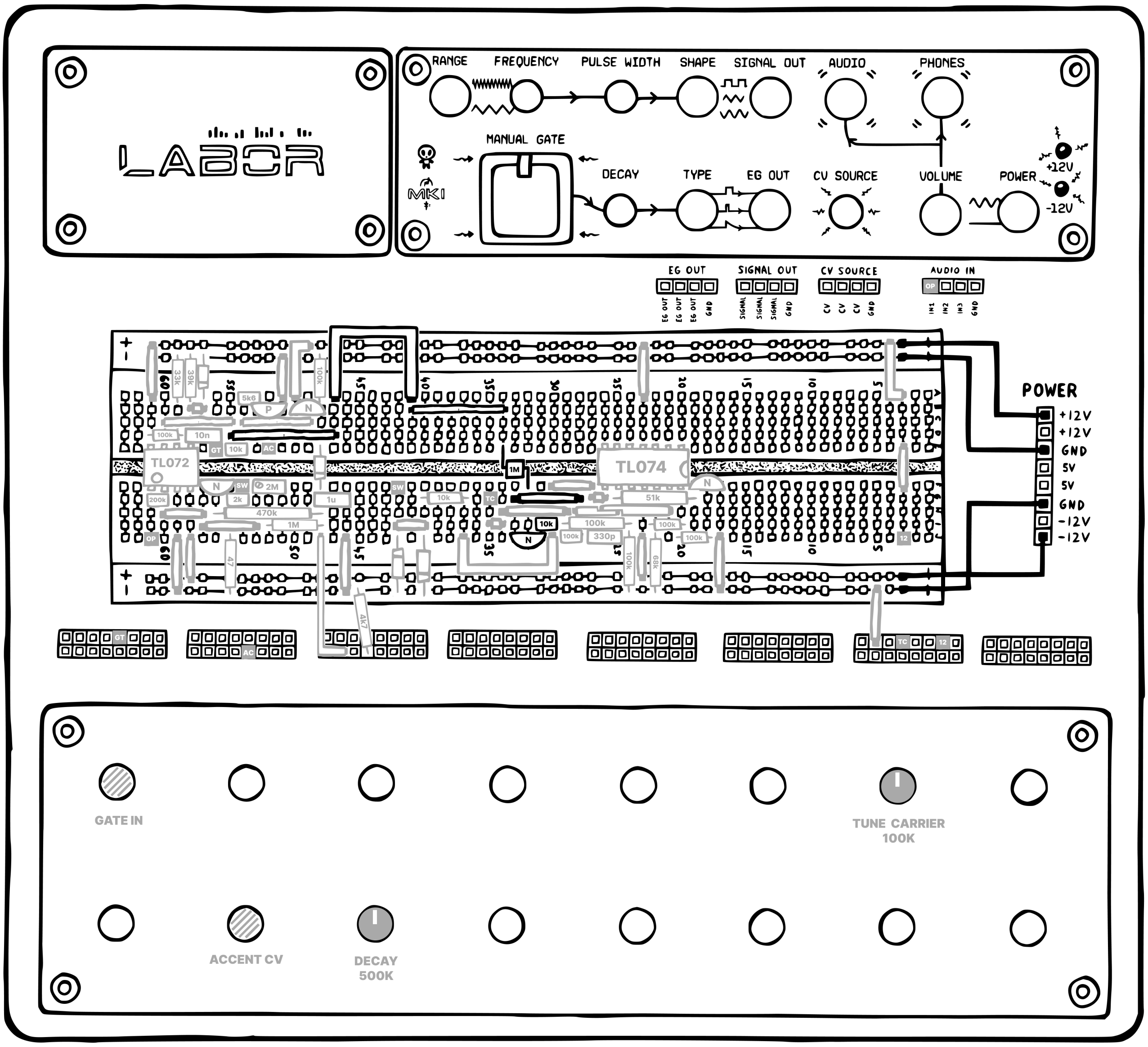
[8] You can try this chapter's circuit in a circuit simulator. I've already set it up for you right here: https://tinyurl.com/2623gohr – you can change all values by double clicking on components.

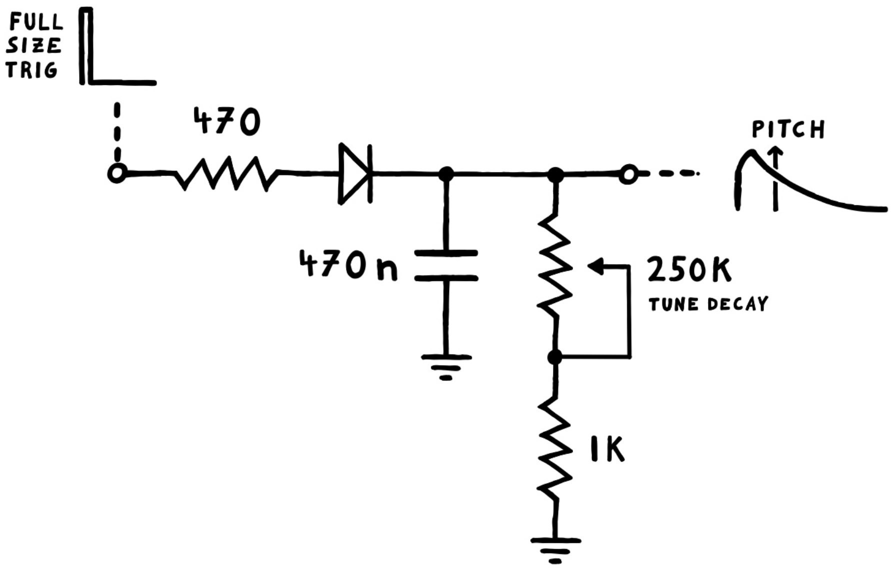
[9] Please note that the direction in which the oscillation heads after the VCO has been reset can differ, resulting in a slightly different waveform between hits. Still, the output should be much more consistent than before.



If you set this up and test it with your gate/CV sequence, all the individual hits should be much more consistent.

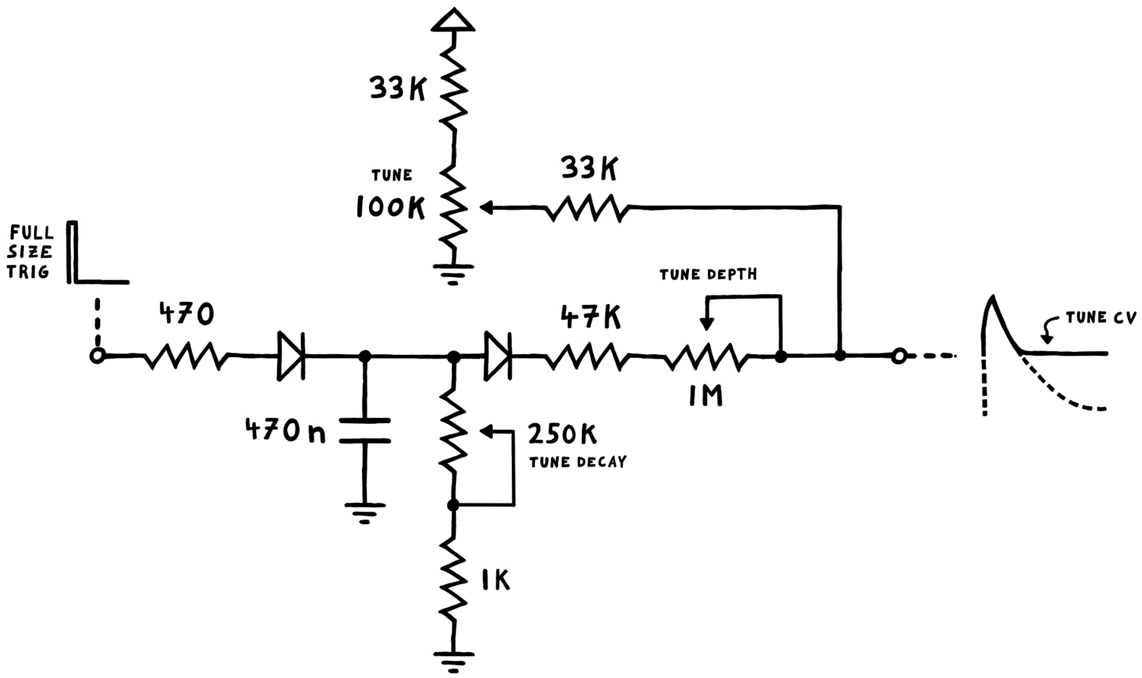
PITCH ENVELOPE

Unfortunately, our drum still sounds pretty flat. It's definitely missing that characteristic 909 punch. To add it, Roland used a simple pitch envelope to modulate the VCO's frequency. **Which is much easier to implement and more reliable than it was in the 808's bridged-t architecture**. Another big reason why Roland decided to shift gears here. Looking at the 909 schematics, the pitch envelope is basically identical to the other envelope we set up, except there's a resistance before the capacitor. Here's how it works.



When we apply a trigger to the input, the capacitor gets charged with a very slight delay, making the rising edge less steep. Then, the cap discharges just like it did in our previous envelope. By sending the output to the VCO's control voltage input, we force its pitch to quickly rise and then drop back down. **This is a basic simulation of what a beater hitting a drum would sound like**. And because an instant rise sounds too aggressive and artificial, we add the small resistor before the cap.

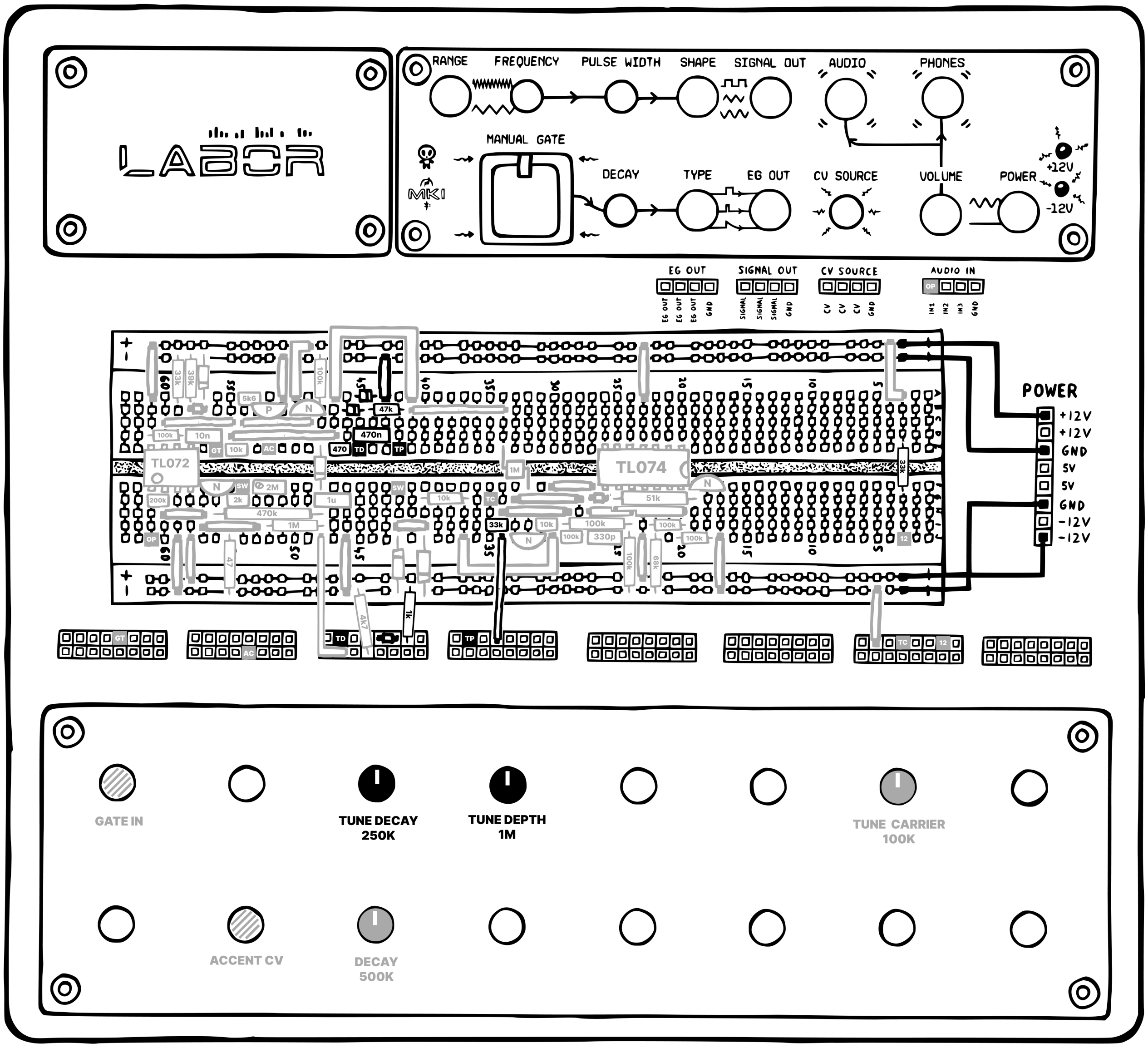
Now, here comes the tricky part. We need to mix the pitch envelope's output with the existing control voltage we use to set the oscillator's base pitch. In the 909, Roland set up a full blown summing mixer for this – which I'd like to avoid to keep our circuit small and lean. So instead, I'll go for a passive mixer using two resistors. **The relation between those resistors will determine the intensity with which the envelope affects the mixed output**. Because this is an important sound design parameter, we'll also add a big 1M series potentiometer to the pitch envelope path, which will allow us to dial its intensity down to almost 0.



Also, we'll set up another diode after the envelope's output – so that it only affects the total mix if the envelope is higher than the tune CV. **This way, we will always drop down to the set base pitch (and not lower)**. For the trigger, we'll use the unaccented one again, since the pitch bend is integral for the output to sound like a drum. Making it less noticeable on an un-accented hit doesn't sound quite right to me (though maybe it does for you – give it a shot!).

One final thing. For the pitch envelope to have an effect at all, its peak needs to be higher than the tune CV. Otherwise, the diode will block and the output will stay the same. Because of this, we add a 33k resistor above the tune CV potentiometer, limiting its maximum output voltage to around 9 V. This saves us a little headroom for the envelope to do its job.[10]

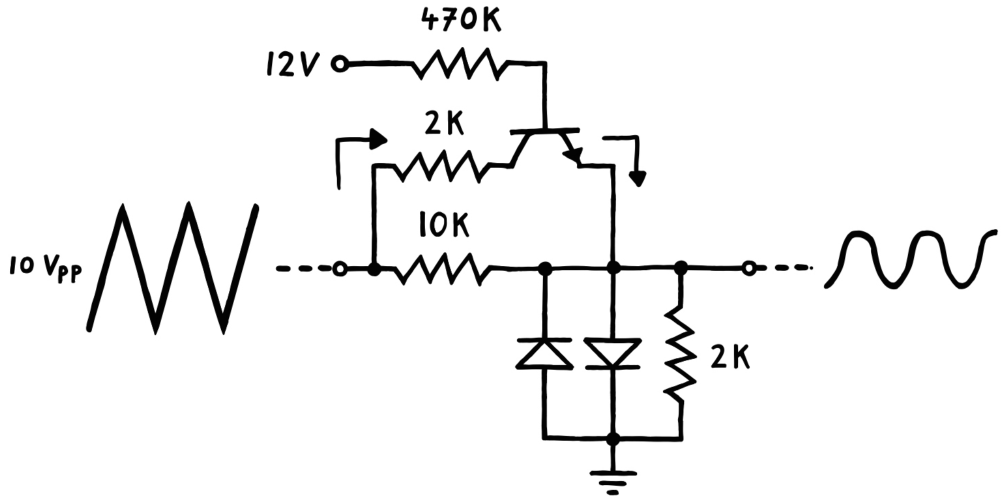
[10] You can try this chapter's circuit in a circuit simulator. I've already set it up for you right here: https://tinyurl.com/2d5pt25g – you can change all values by double clicking on components.



By playing with the tune decay and tune depth potentiometers, you should be able to get sounds that are way more 909-ish already. If you tune the VCO really low, you can get some decent kicks out of our circuit.

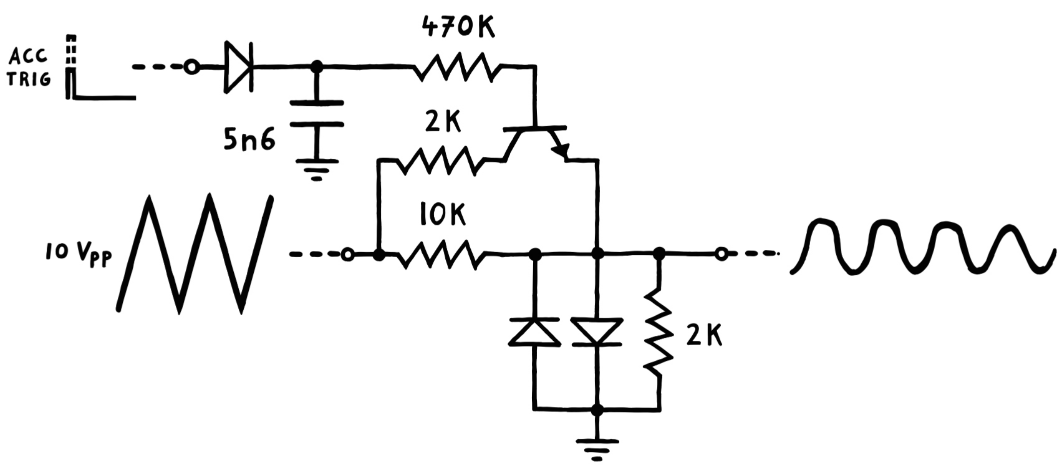
IMPACT DISTORTION

Still, the sound is missing the 909's sharp attack. There are multiple things that Roland did to add it – and I've cherry picked two of them. We'll start with an enhancement to the triangle to sine waveshaper (which Roland kept exclusive to the toms). Its purpose is to distort the sine wave momentarily when the trigger hits, adding high frequency content and extra volume for an aggressive impact.

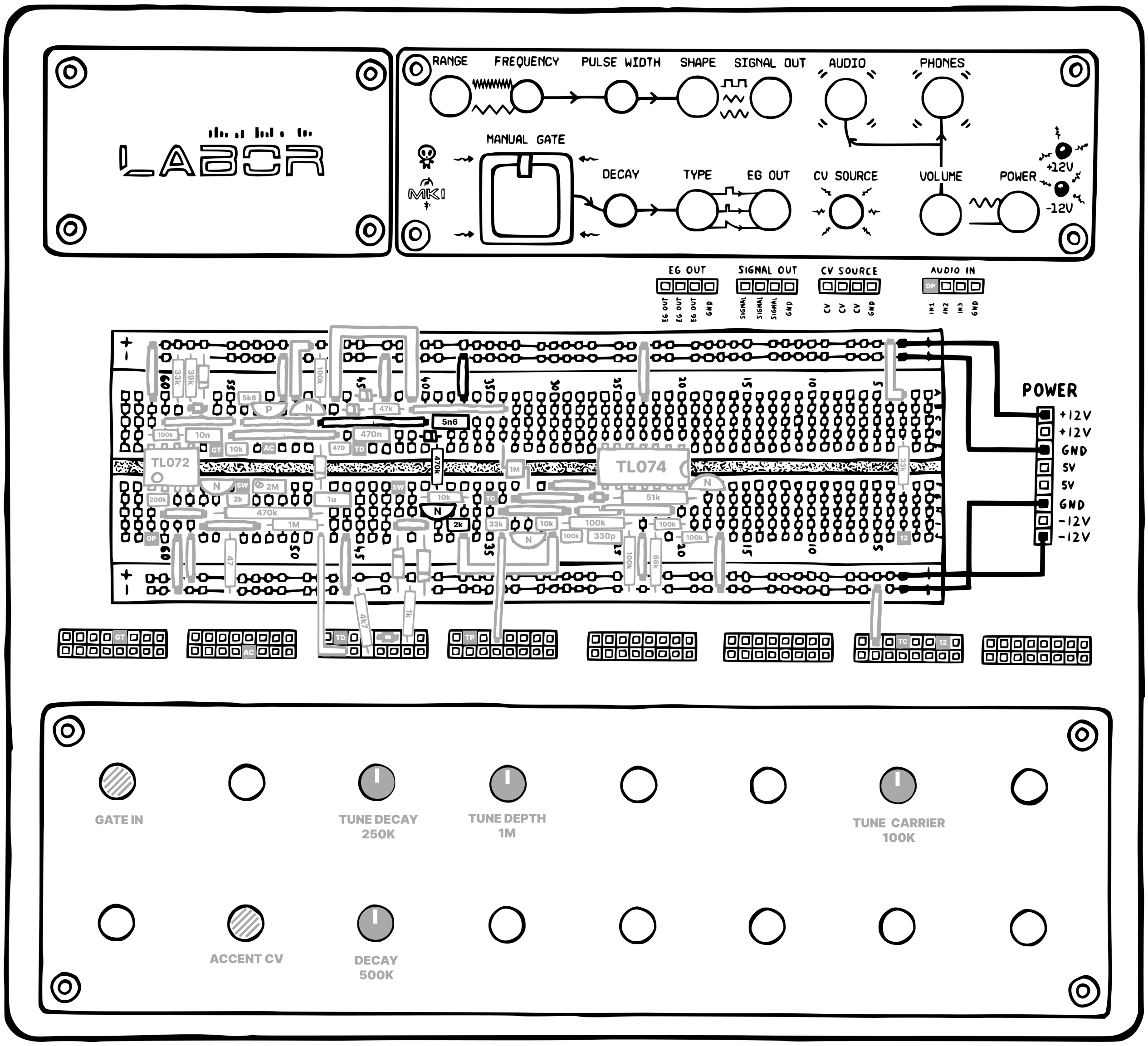


For this, we bridge the 10k resistor with a smaller resistor and another NPN transistor. Now when a voltage is applied to the NPN's base via a big resistor, it allows (forward) current to bypass the 10k resistor. **This selectively reduces the amount of scaling we apply to the input, which causes one diode to open more quickly and widely, resulting in a much sharper cut-off at the positive tips of our triangle wave**.

Since we want this to happen for a little bit longer than the trigger duration, we'll set up a basic envelope. While experimenting with this, I found that there is a sweet spot for the ideal duration – so we'll omit a decay potentiometer and let the cap discharge purely via the transistor instead.



For the trigger, I think it makes sense to use the accented version here. This way, unaccented hits stay a bit more mellow, while accented ones become more harsh.[11]

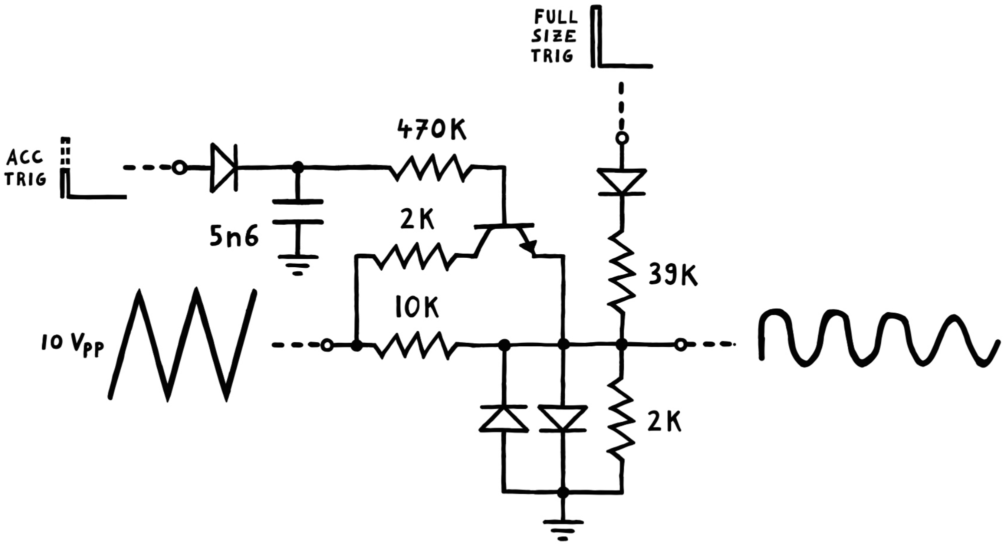


Because the effect is subtle, you can do a before/after comparison by removing the diode before the 5n6 capacitor while testing the circuit.

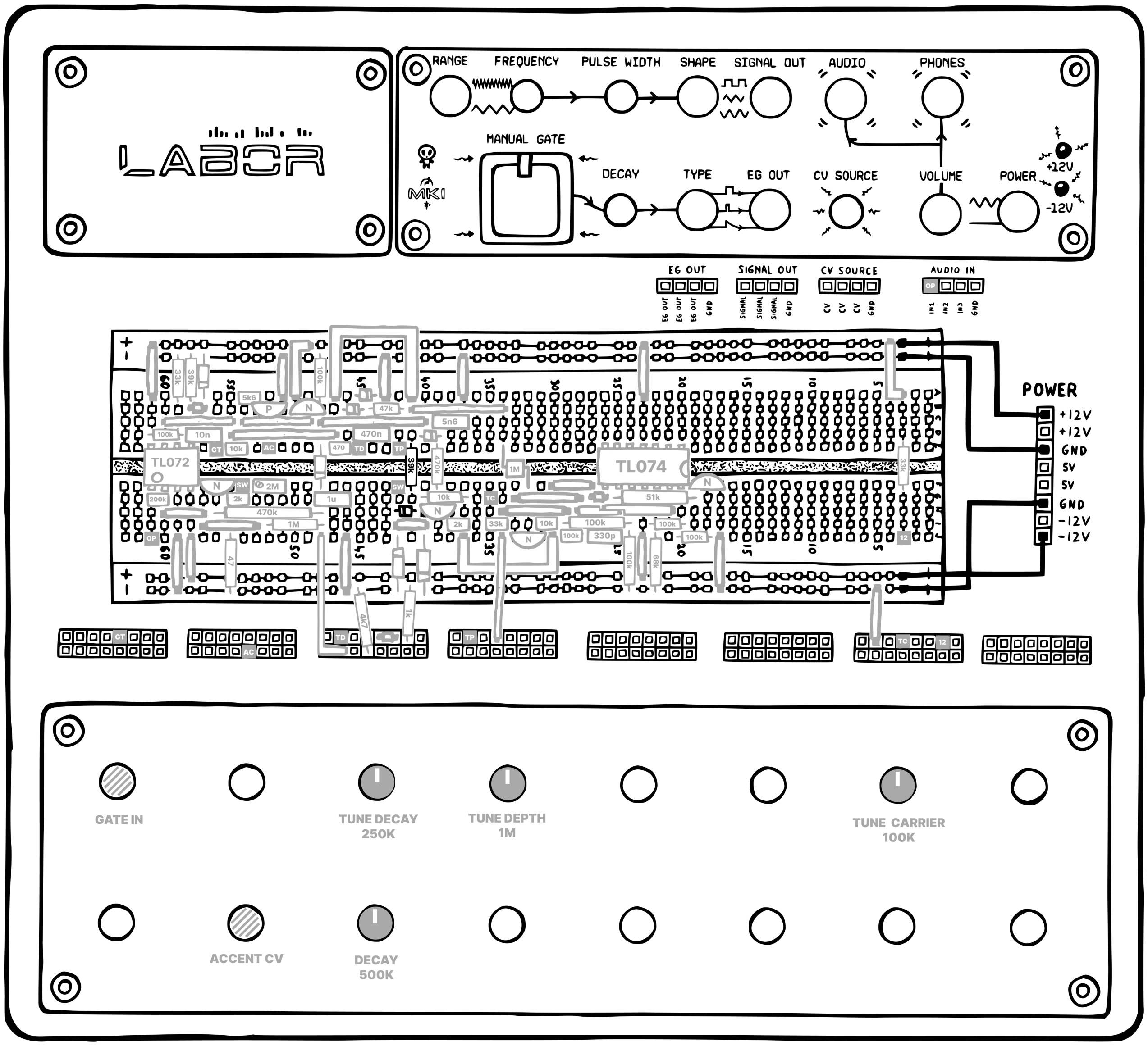
[11] You can try this chapter's circuit in a circuit simulator. I've already set it up for you right here: https://tinyurl.com/2ys2rwx5 – you can change all values by double clicking on components.

ADDED CLICK

For me, the distortion adds some much needed complexity to the sound's transient, making it much more characterful. Though I'd still like to add a little more spice to it. The 909's kick injects white noise mixed with the trigger pulse into the output to achieve this. I'm not keen on adding a dedicated noise generator, though. So let's try and see how far we get with injecting only the trigger.



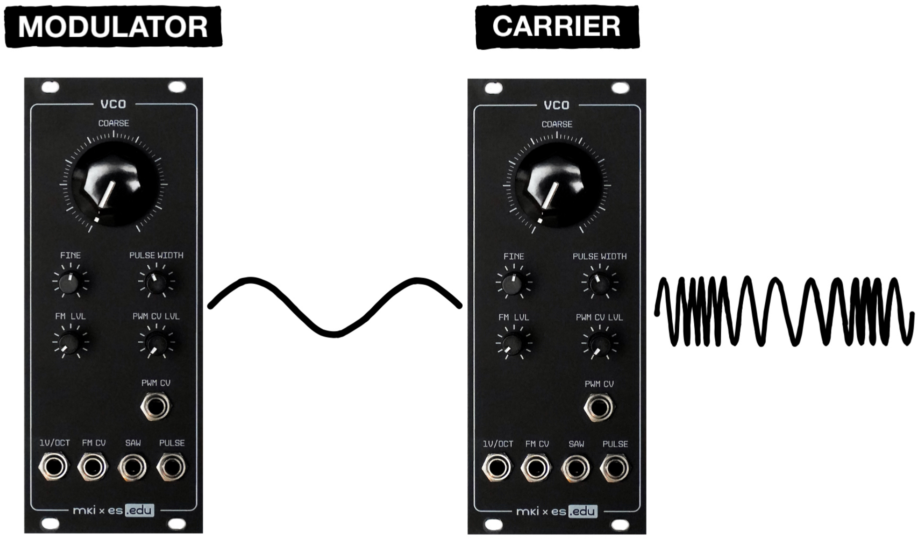
Doing this is really easy – we only need to connect our trigger to the waveshaper's output via a diode and a relatively big resistor. For me, a 39k seems to be the sweet spot: the added attack is noticeable, but the trigger doesn't overpower the rest of the sound. The diode is necessary because the trigger drops down to -12 V when it's inactive – which would noticeably tamper with the waveshaper's operation. Also, I decided to use the unaccented trigger here, since its volume is already affected by the VCA itself. No need to double this up!



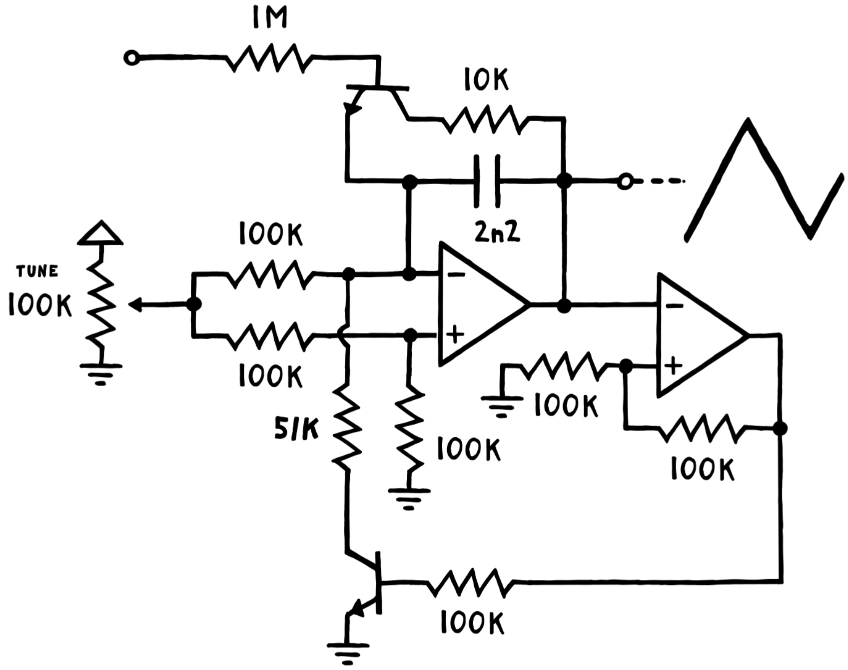
Since the effect is, again, pretty subtle, you can do another before/after comparison by removing the new diode while testing the circuit.

FREQUENCY MODULATION

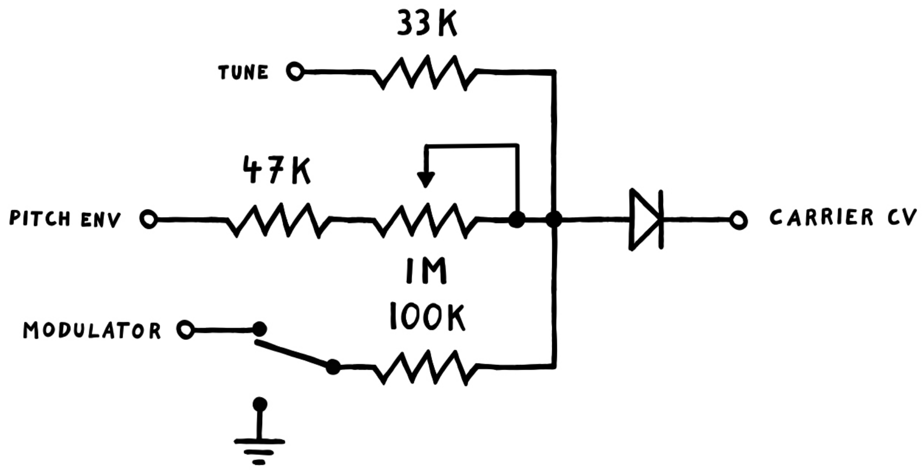
At this point, our circuit is essentially a hybrid between the 909's kick and toms. This would've been enough exploration for me to go and fix my machine. But I wasn't done yet: I started thinking about ways to expand on the 909's sounds. **And one technique that's absent from all of Rolands vintage drum machines is Frequency Modulation.** Frequency Modulation (or FM for short) can create very rich, metallic textures. It works like this.



You take one VCO, which we'll call the carrier, and you modulate its frequency using the output of another VCO, which we'll call the modulator. This creates complex patterns in the output waveform of the carrier. Which, depending on the relation between the two oscillators' frequencies, can sound really harsh and aggressive. To implement this in our circuit, we'll simply clone our existing VCO to create a modulator – with two slight differences.

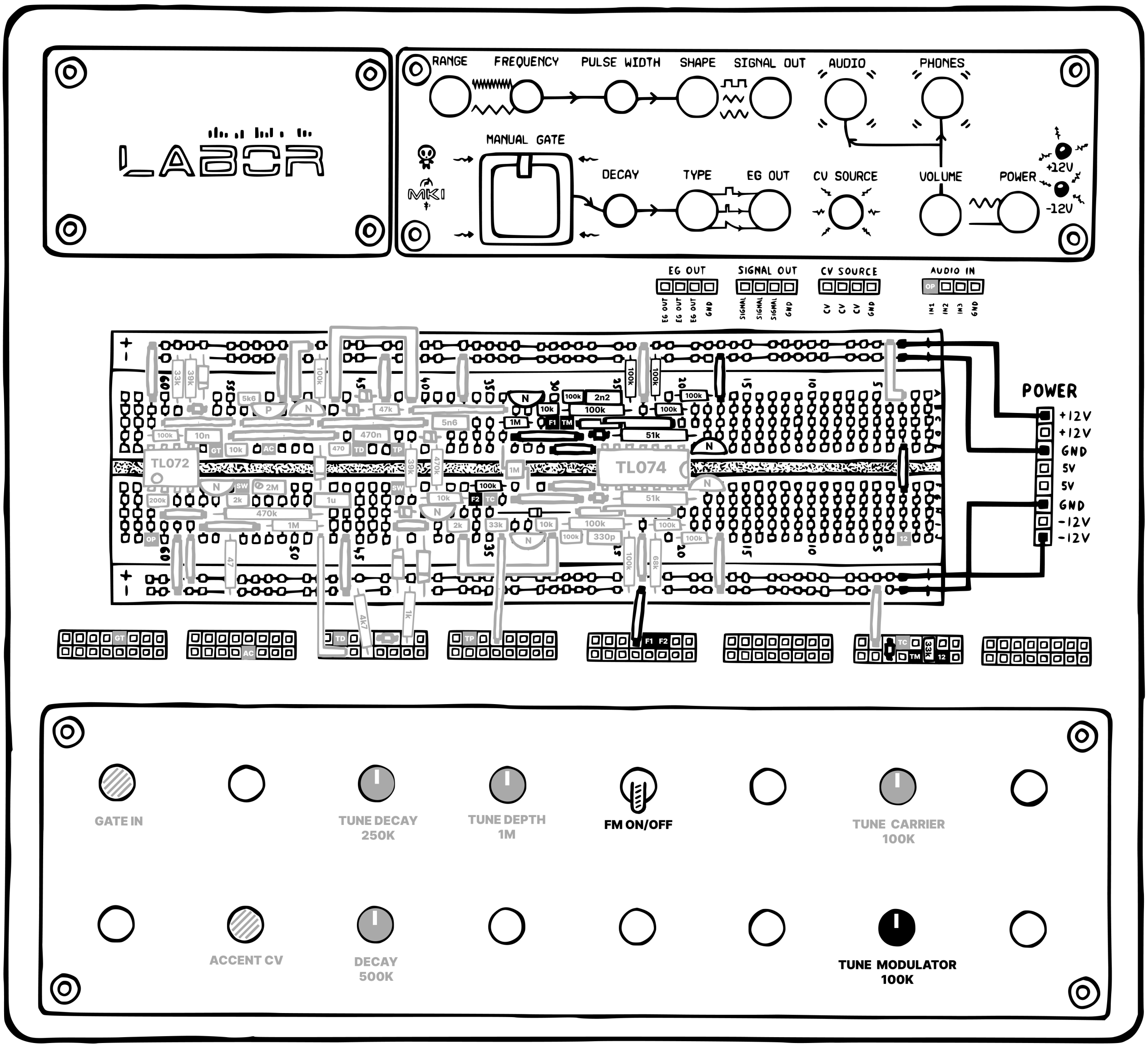


First, we'll use a significantly bigger capacitor in our integrator. This shifts the oscillator's frequency range downwards. **This is desirable because it generally sounds more interesting if the modulator's frequency is lower than that of the carrier – at least in my opinion**. And second, we'll push the schmitt trigger inverter's thresholds further apart so we get a stronger output to work with.
To mix the modulator's output with the tune CV and the pitch envelope, we'll simply use a 100k resistor.[12] The resistor value directly determines the FM intensity here: the bigger the resistor, the less noticeable the effect.



[12] You can try this chapter's circuit in a circuit simulator. I've already set it up for you right here: https://tinyurl.com/29pjo7lm – you can change all values by double clicking on components.

Also, we'll add a simple switch that lets us choose between applying the modulator's output or ground – so we're able to use the circuit in regular 909 or FM mode. To prevent the carrier from crashing into either supply rail when the input CV goes negative, we also add a diode at the carrier's CV input. This will prevent it from going negative, even if the mixer's output does.

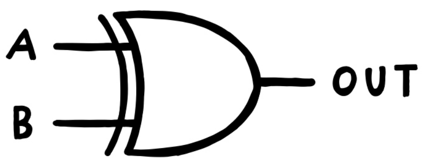


By flipping the FM ON/OFF switch, you should be able to turn on the circuit's FM mode. Try tuning carrier and modulator to different intervals and see what kinds of glassy/ metallic sounds you can create!

XOR PULSE

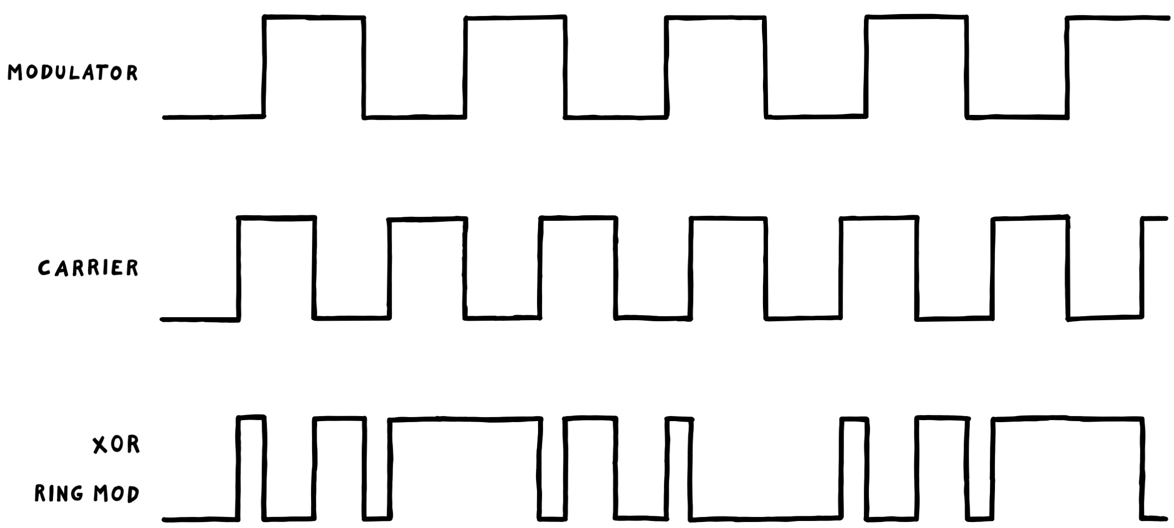
But while the circuit is already pretty versatile, there is still some untapped potential here: So far, we're only using our VCOs' triangle outputs. They also generate square waves, though! And square waves, as I've learned by studying the TR-606 and TR-808 schematics, are great for synthesizing analog hi-hats and cymbals. At least if you use a bunch of them.

We only have two, though – and simply mixing them together would not give us a signal that's complex enough. **So we'll need to fake having more oscillators than we actually do**. For this, we'll use something called an exclusive OR gate (or XOR gate for short). An XOR gate is a logic gate that accepts two inputs and produces one output.

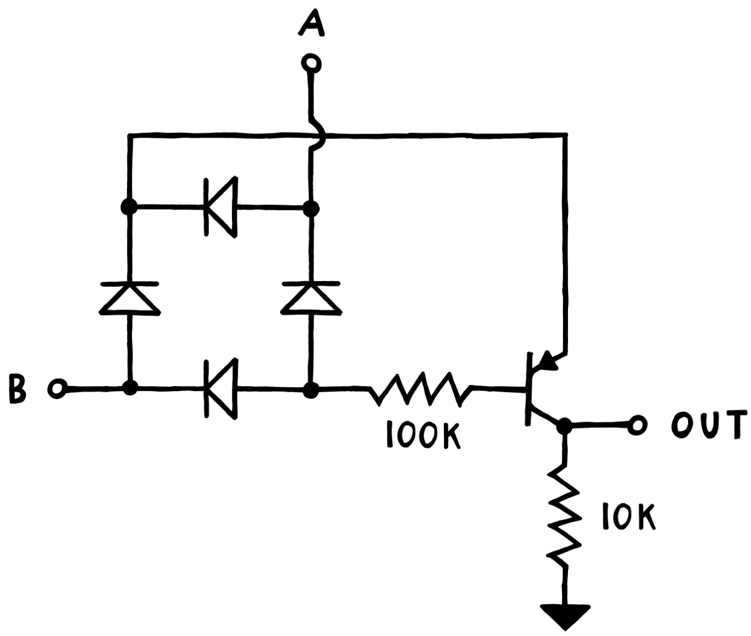


A	B	OUT
LO	LO	LO
HI	LO	HI
LO	HI	HI
HI	HI	LO

If either input is high while the other is low, the XOR's output will be high. But if both inputs are the same value, its output will be low. With two dissonant square waves as our inputs, this creates a pulse wave with semi-chaotic pulse width that sounds harsh and metallic.

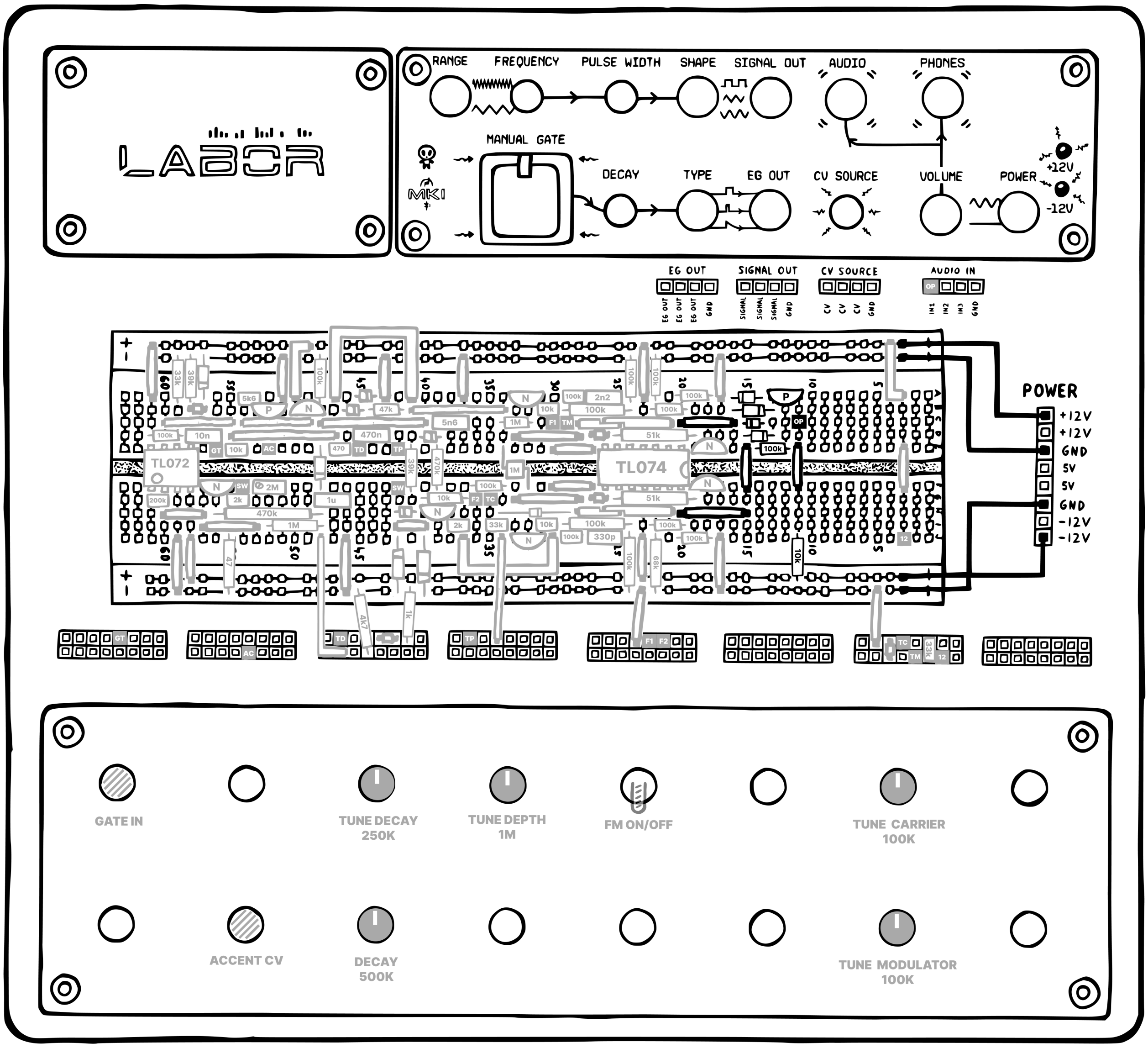


Coincidentally, this signal is identical to what you would get from a ring modulator. Which is why XOR gates are often used as a cheap substitute for true ring modulation. (The Korg MS-20 is one famous example.) To implement an XOR gate, we only need 4 diodes, 2 resistors and a PNP transistor.



It works like this. If both inputs (A/B) are low, both the transistor's base and emitter are low. So no current flows through it, and the output is pulled low by the resistor. If the top input (A) is high and the left input (B) is low, the transistor's emitter is pushed high and the base is pulled low, causing current to flow and the output to be pushed high as well. Same deal if the input states are flipped – emitter is high, base is low, current flows and the output goes high. But if both inputs are high, both emitter and base will be high, cutting off the current flow and dropping the output voltage back down.[13]

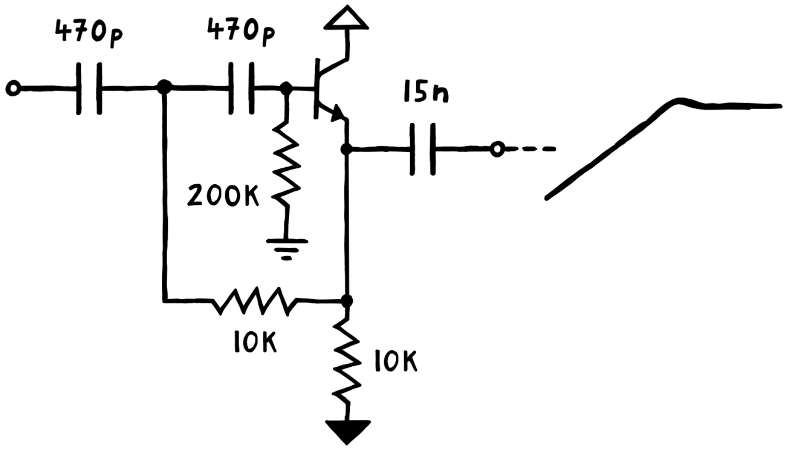
[13] You can try this chapter's circuit in a circuit simulator. I've already set it up for you right here: https://tinyurl.com/2csqtvo9 – you can change all values by double clicking on components.



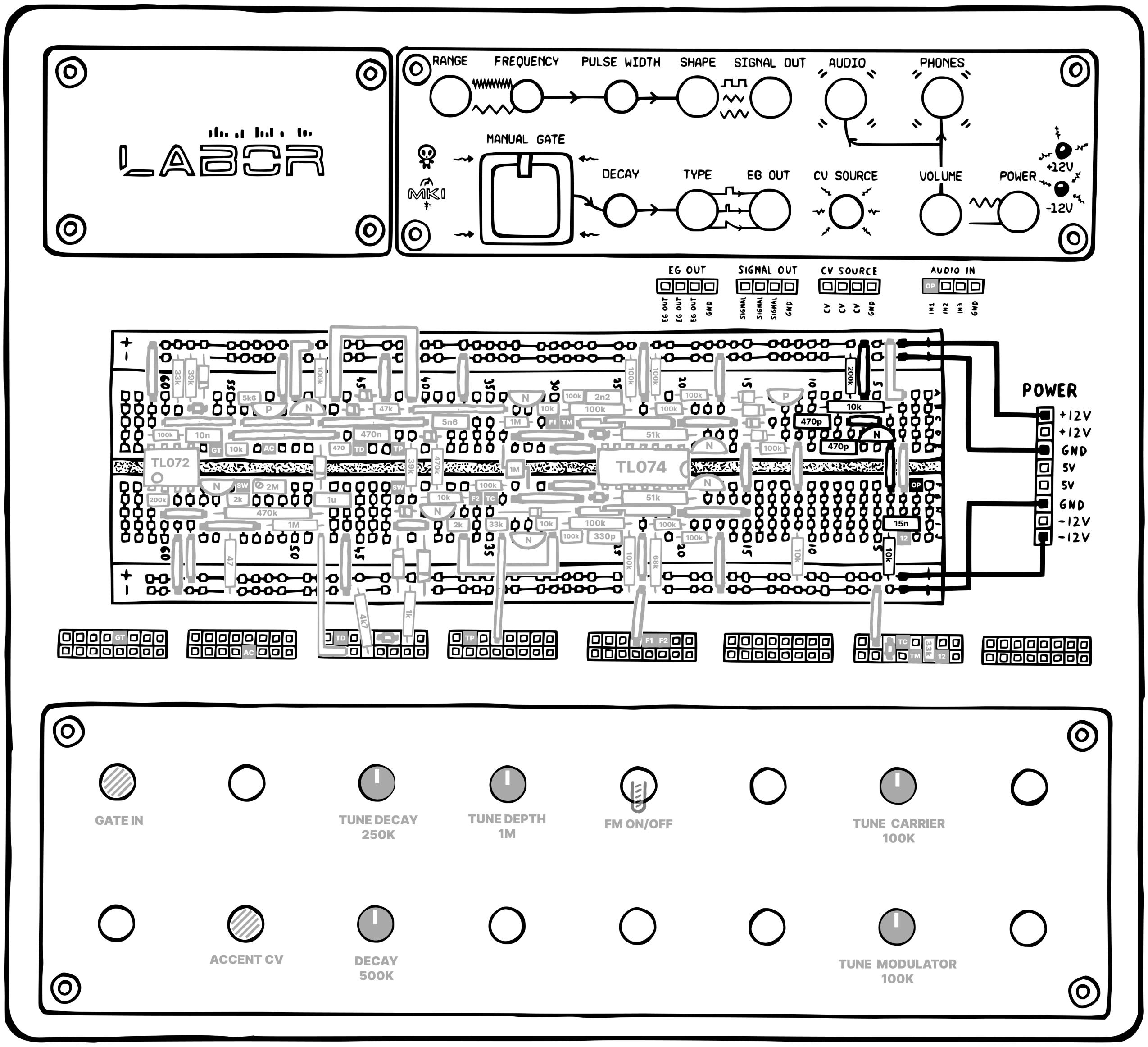
As expected, you should get a decently chaotic output from our XOR circuit. Try adjusting the frequency relationship between the two oscillators to find some interesting sweet spots.

HIGH PASS FILTER

The XOR gate does add some strange metallic overtones, which sound great. But for proper cymbal sounds, the signal has way too much low-end. So we'll remove it with a simple sallen-key high pass filter. You can check the Hi-Hat module's manual for an in-depth explanation, but here's the basic gist.



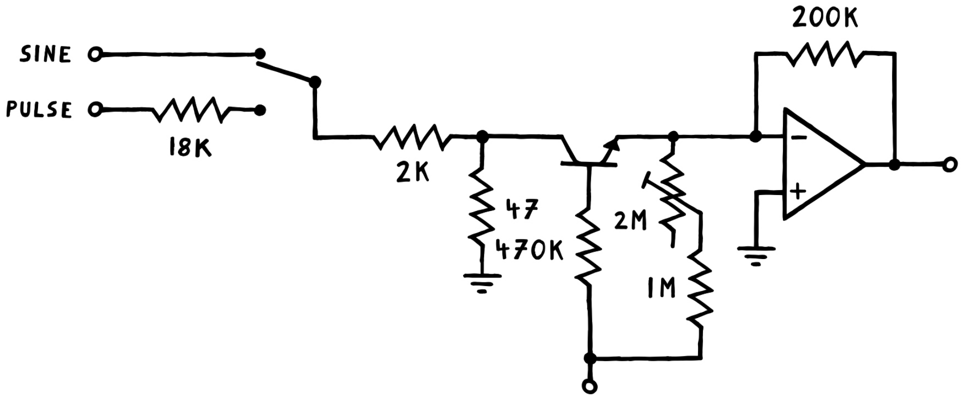
A sallen-key high pass is just a regular two-stage high pass with an added amplification stage – whose output is then fed back into the first filter stage. This makes the circuit's rolloff more steep, and it adds a bit of resonance. In our case here, we get a 12 db/oct rolloff starting at 7.5 kHz and a pronounced bump at that same frequency. Which should give the output a little extra bite compared to a completely flat filter. Because a transistor-based sallen-key filter introduces a significant DC offset, we remove it with a simple (15n) coupling capacitor.



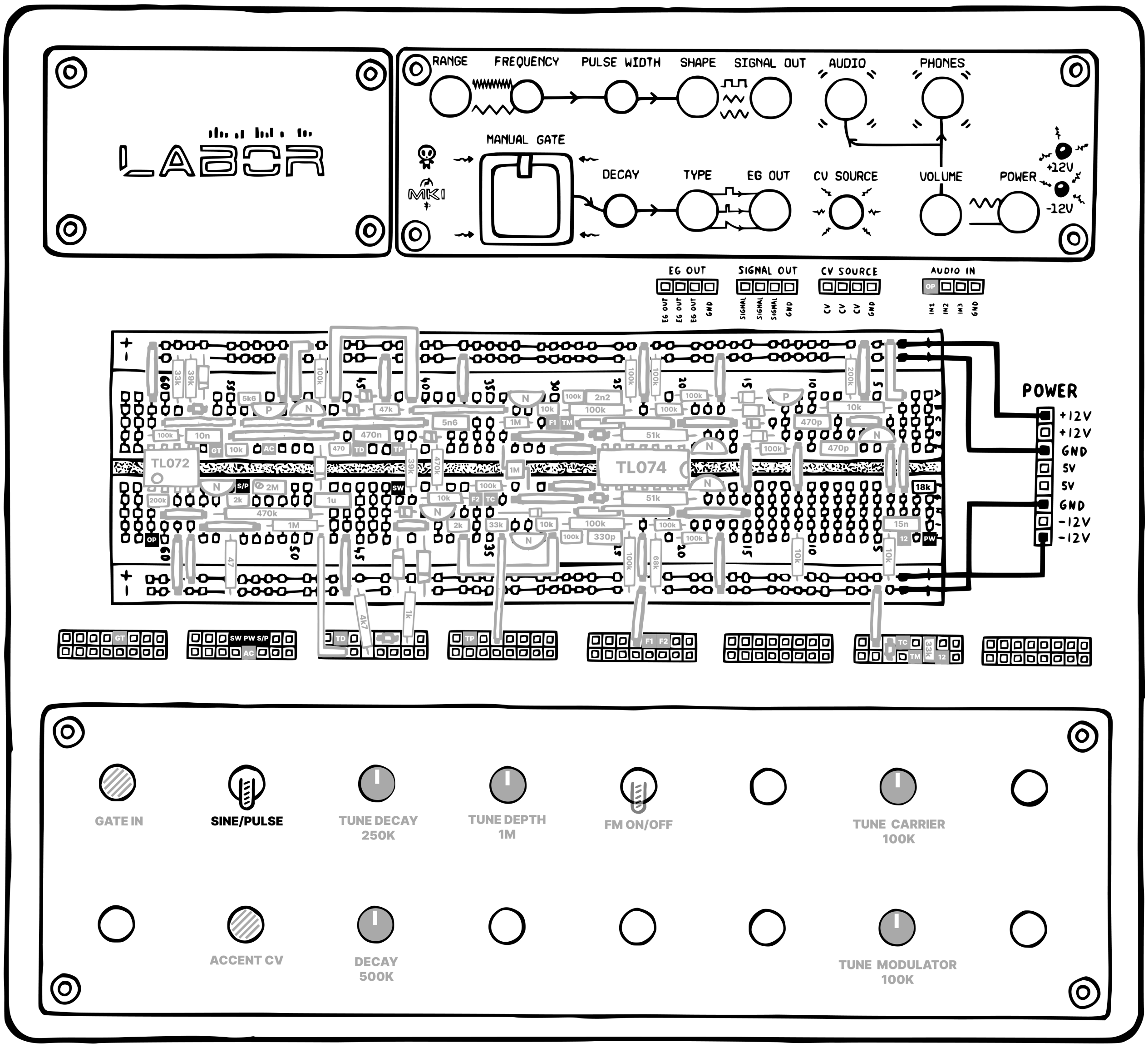
If you try this, the output should sound a lot thinner (and more cymbal-like) than the unfiltered signal.

SINE/PULSE SWITCH

Okay, so now we'll need to turn this static cymbal noise into a percussive hit. Thankfully, we can repurpose our existing VCA and decay envelope for this. All we need to do is set up a simple SPDT switch before the VCA's input. Then, we apply our sine wave output to one side of the switch – and the pulse wave output to the other.



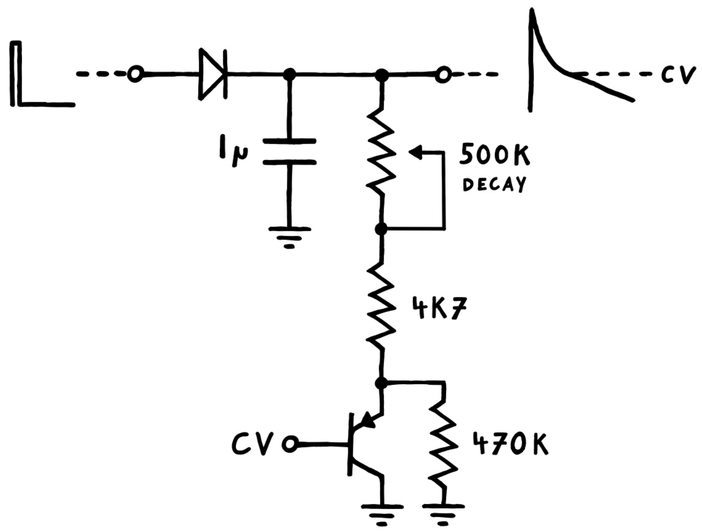
Because the pulse wave is much louder than the sine wave, we also need an additional 18k resistor to compensate. In combination with the VCA's voltage divider, this scales the pulse down to the proper input range for our VCA.



To test this, first set the VCA's input to pulse wave and turn the FM mode off. By tuning the two oscillators to different intervals, you can get a decent variety of cymbal- and hihat sounds. Cool – but what if you also turn on the FM mode? The difference is subtle, but you should get a different *flavor* of cymbal.

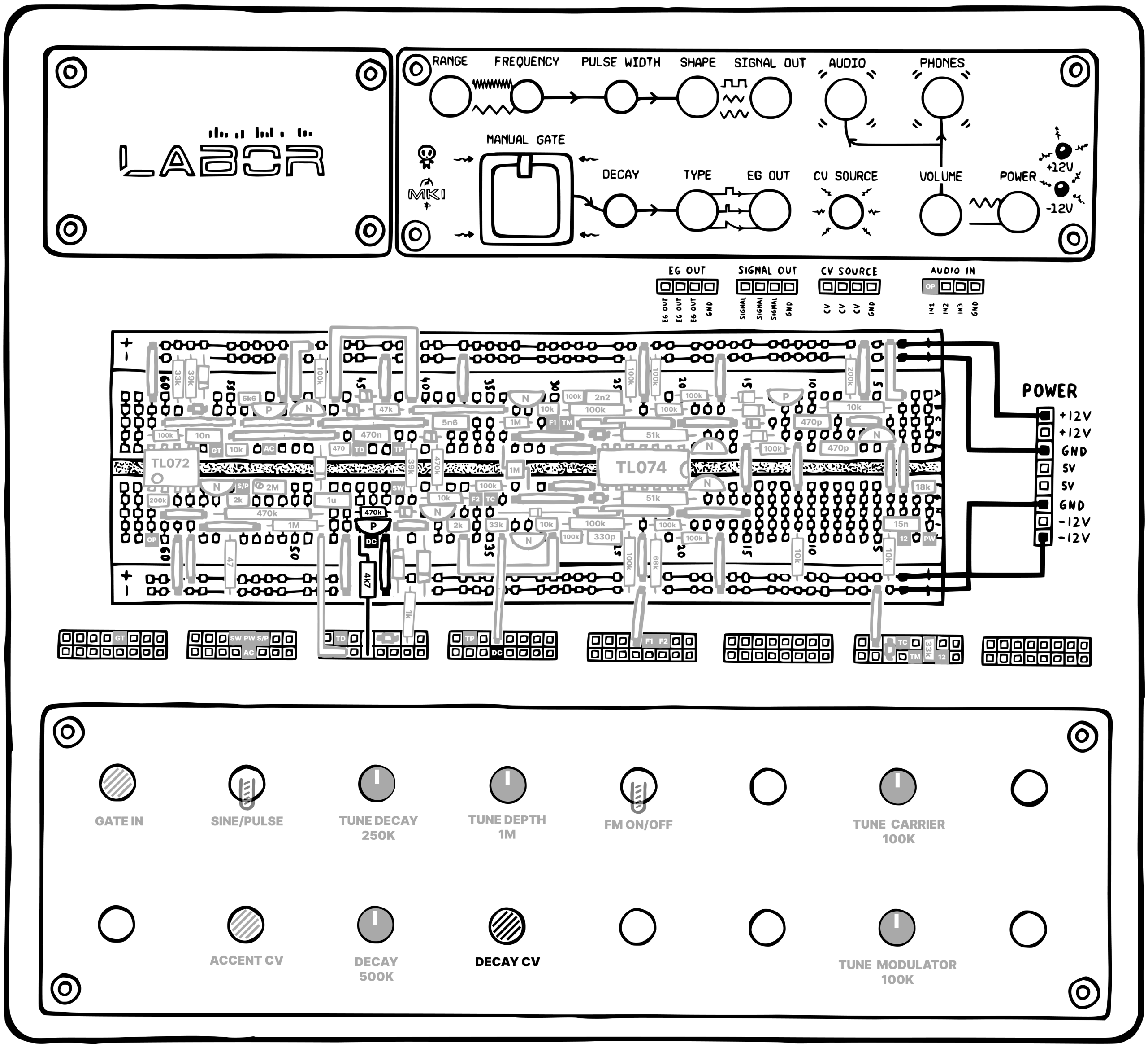
DECAY CV

With 4 different modes (single oscillator sine, FM sine, dual oscillator pulse, and FM pulse) our circuit now gives us plenty of manual sound design options. It's a little light on voltage control, though. So let's do something about that! First, I'd like to add a CV input for the decay envelope. Luckily, we can re-use another idea from my Hi-Hat module here.



If we put a PNP transistor between the envelope's resistance and ground, the capacitor can only discharge to the voltage at the PNP's base – keeping the VCA open at that level. Since we still want the volume to decay further from here, if only slowly, we bridge the transistor with a big 470k resistor. This allows the cap to gradually discharge past the set CV level.[14]

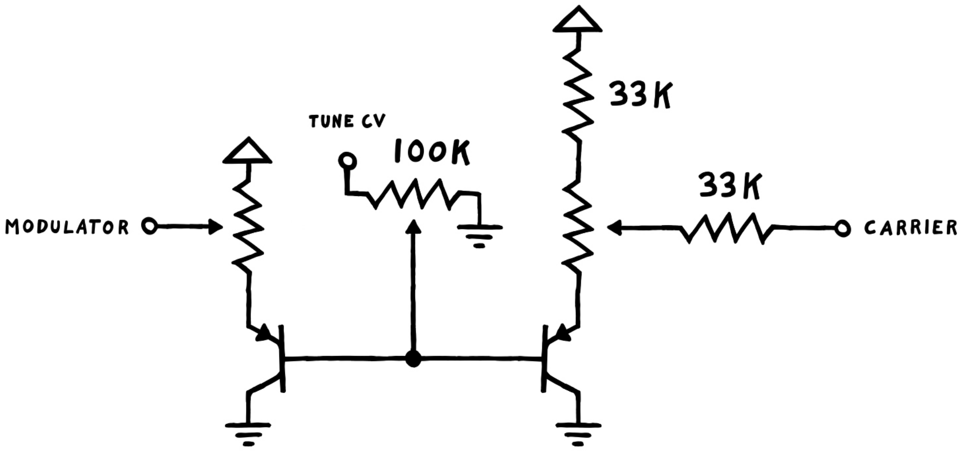
[14] You can try this chapter's circuit in a circuit simulator. I've already set it up for you right here: https://tinyurl.com/2473cgwl – you can change all values by double clicking on components.



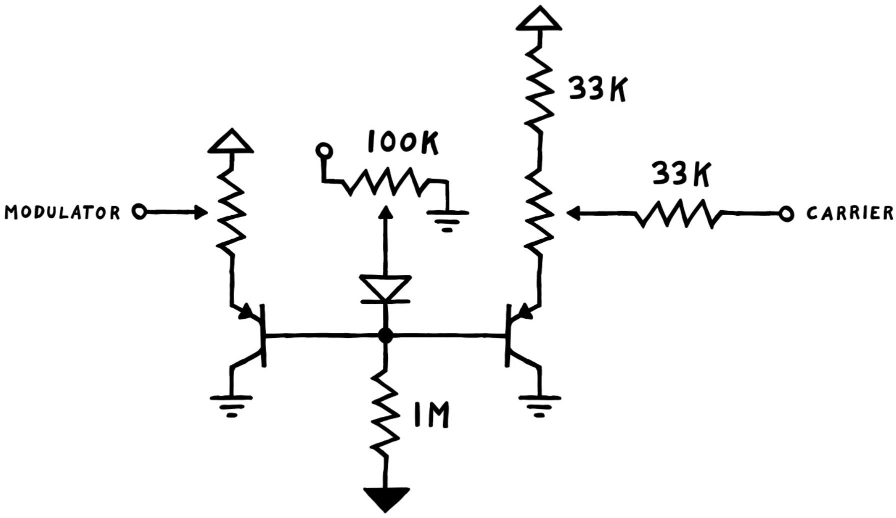
You should now be able to adjust the output's tail by applying different CV levels to the new DECAY CV input. Try it in the pulse wave mode as well, for simulating opening and closing hi-hats!

TUNE CV

Besides a decay CV input, it would also be great to have a tune CV input that controls both oscillators' frequencies at once – so we can play some simple melodies and bass lines. Good thing our circuit uses proper VCOs! Because this allows us to simply mix our existing control voltages for VCO 1 and 2 with an external control voltage. There's only one small issue. **Since we're using passive mixers for the tune CV inputs, the external voltage would interact with the other control voltages in unintended ways**. Check the Mixer module's manual for a deep dive into this issue.
To work around it, we have to get a little creative. Instead of mixing the external voltage in, we can use it to modify the internal tune CV levels. It works like this: we add two PNP transistors between the tune CV potentiometers and ground.

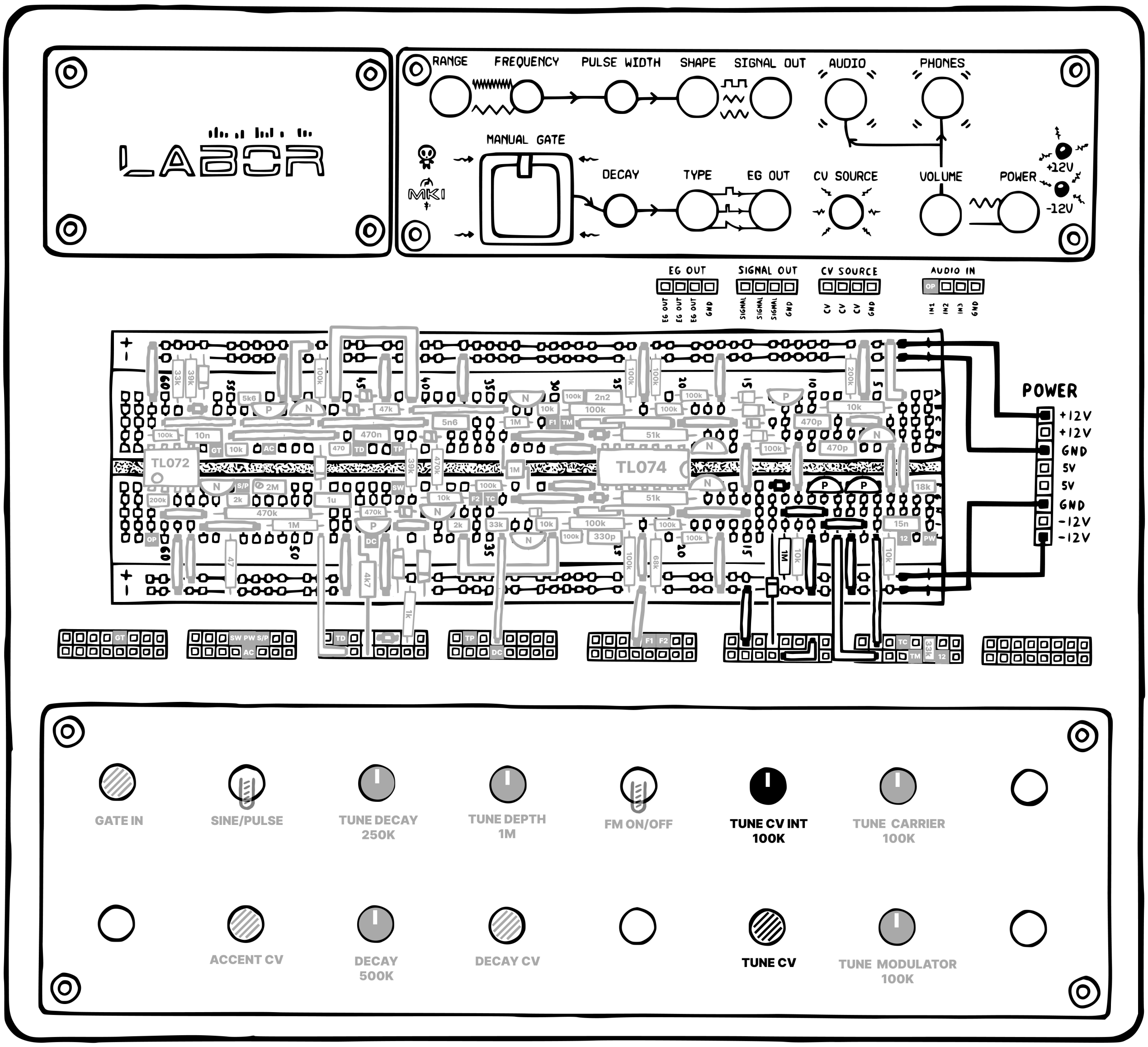


If we then apply a positive voltage to the transistors' bases, that voltage will be mirrored at the PNPs' emitters. **And if the voltage below the potentiometers is raised above ground, their output voltages will increase proportionally**. There's one caveat to this solution: since it's kind of a messy hack, we can't expect to get a v/oct response. Still, we can make it somewhat tuneable by adding a simple input attenuator.
There's one issue with this setup. If we'd try to set both oscillators to their lowest pitch, that lowest pitch will be much higher than what we got previously. What's up with that? Well, the issue is that our PNP transistors don't mirror their base voltages exactly. Instead, they add what we call a diode drop. So if we apply 0 V to their bases, we get around 500 mV at both emitters – meaning that our voltage dividers can never go down to 0 V. To fix this, we'll simply subtract a diode drop from the external CV using a diode and a big 1M resistor.



It works like this. The resistor ensures that a small current flows through the diode at all times. And when a current flows through a diode, there will be a diode drop across it. So for 0 V external CV, we'll get around -500 mV at the PNPs' bases. Which should then give us about 0 V at their emitters.[15]

[15] You can try this chapter's circuit in a circuit simulator. I've already set it up for you right here: https://tinyurl.com/2bahf9kd – you can change all values by double clicking on components.



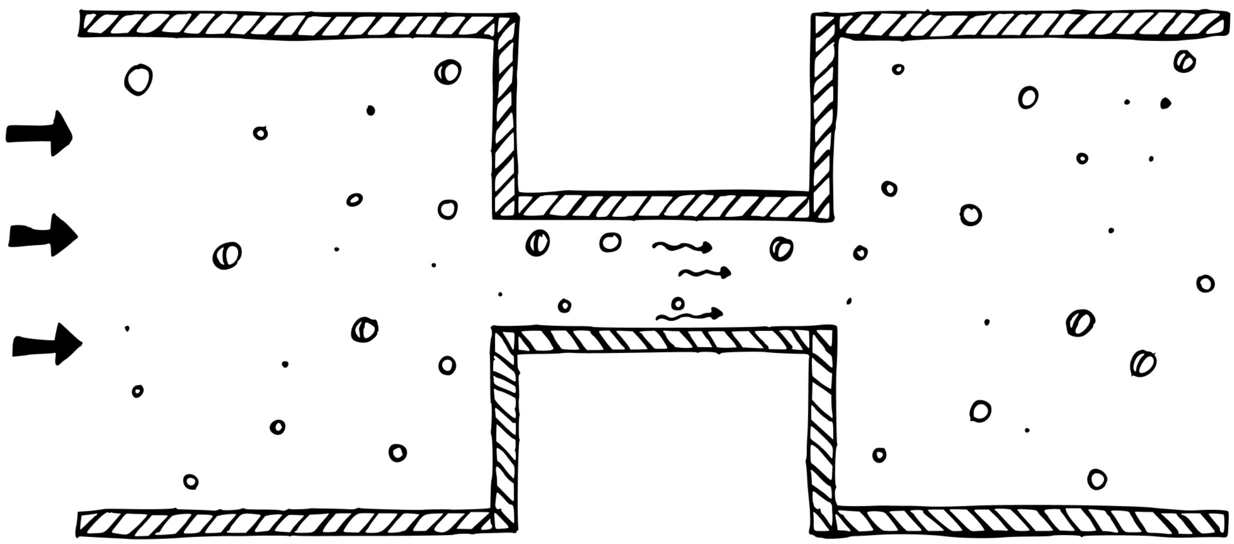
Try sending a CV sequence into the new TUNE CV input. By playing with the individual steps' CV values in combination with the TUNE CV INT potentiometer, you should be able to dial in some basic melodies. Great! And with this, our FM Drum is complete. Once you're done experimenting, dig out the panel and PCB from the kit, heat up your soldering iron and get to building. You can find more information on how to populate the board & how to solder in the enclosed appendix.

COMPONENTS & CONCEPTS APPENDIX

In this section, we'll take a closer look at the components and elemental circuit design concepts we're using to build our module. Check these whenever the main manual moves a bit too fast for you!

THE BASICS: RESISTANCE, VOLTAGE, CURRENT

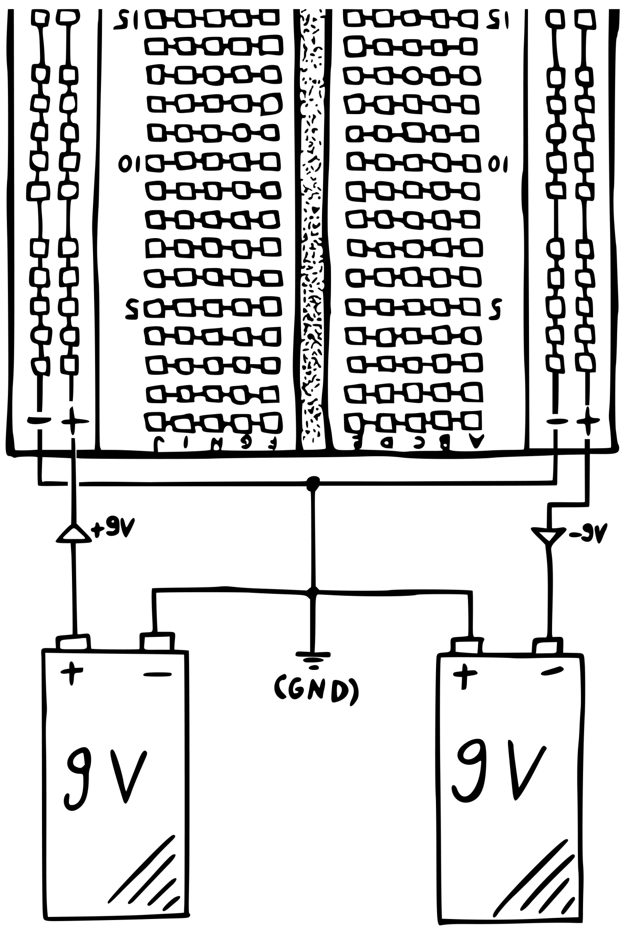
There are three main properties we're interested in when talking about electronic circuits: resistance, voltage and current. To make these less abstract, we can use a common beginner's metaphor and compare the flow of electrons to the flow of water through a pipe.



In that metaphor, resistance would be the width of a pipe. The wider it is, the more water can travel through it at once, and the easier it is to push a set amount from one end to the other. Current would then describe the flow, while voltage would describe the pressure pushing the water through the pipe. You can probably see how all three properties are interlinked: **more voltage increases the current, while more resistance to that voltage in turn decreases the current.**

USING TWO 9 V BATTERIES AS A DUAL POWER SUPPLY

Dual power supplies are great – and if you want to get serious about synth design, you should invest in one at some point. But what if you're just starting out, and you'd like to use batteries instead? Thankfully that's totally doable. **You just need to connect two 9 V batteries like shown here.** For this, you should use 9 V battery clips, which are cheap & widely available in every electronics shop.

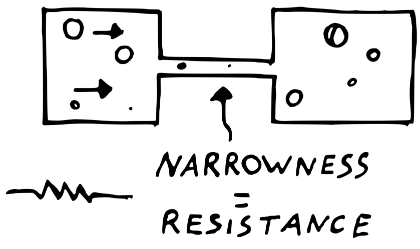


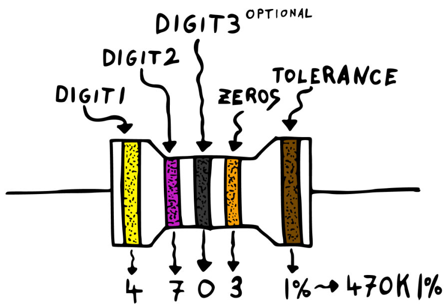
By connecting the batteries like this, the positive terminal of the left battery becomes your +9 V, while the negative terminal of the right is now your –9 V, and the other two combine to become your new ground.[16] **Please make sure you disconnect the batteries from your breadboard when you make changes to the circuit!** Otherwise you run the risk of damaging components.

[16] If you're struggling with setting this up, you can watch me do it here: https://youtu.be/XpMZoR3fgd0?t=742

RESISTORS

While a conductive wire is like a very big pipe where lots of water can pass through, **a resistor is like a narrow pipe that restricts the amount of water that can flow**. The narrowness of that pipe is equivalent to the resistance value, measured in ohms (Ω). The higher that value, the tighter the pipe.





Resistors have two distinctive properties: linearity and symmetry. Linearity, in this context, means that for a doubling in voltage, the current flowing will double as well. Symmetry means that the direction of flow doesn't matter – resistors work the same either way.

On a real-life resistor, you'll notice that its value is not printed on the outside – like it is with other components. Instead, it is indicated by colored stripes[17] – along with the resistor's tolerance rating. In addition to that, the resistor itself is also colored. Sometimes, depending on who made the resistor, this will be an additional tolerance indicator.

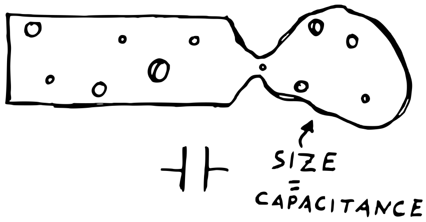
For the resistors in this kit, a yellow body tells you that the actual resistance value might be ±5 % off. A dark blue body indicates ±1 % tolerance. Some kits will also contain light blue ± 0.1% resistors to avoid the need for manual resistor matching.

While in the long run, learning all these color codes will be quite helpful, you can also simply use a multimeter to determine a resistor's value.

[17] For a detailed breakdown, look up resistor color coding. There are also calculation tools available.

CAPACITORS

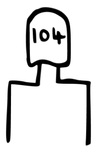
A capacitor is a bit like a balloon that you can attach to the open end of a pipe. If there's some pressure in the pipe, the balloon will fill up with water until the pressure equalizes. (Since the balloon needs some space to expand into, both of the capacitor's legs need to be connected to points in your circuit.)

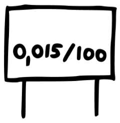


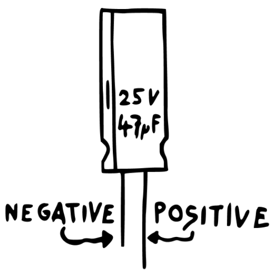
Then, should the pressure in the pipe drop, the balloon releases the water it stored into the pipe. The maximum size of the balloon is determined by the capacitor's capacitance, which we measure in farad (F). There are quite a few different types of capacitors: electrolytic, foil, ceramic, tantalum etc. They all have their unique properties and ideal usage scenarios – but the most important distinction is if they are polarized or not.

You shouldn't use polarized capacitors against their polarization (applying a negative voltage to their positive terminal and vice versa) – so they're out for most audio-related uses like AC coupling, high- & low-pass filters etc.

Unlike resistors, capacitors have their capacitance value printed onto their casing, sometimes together with a maximum operating voltage. **Be extra careful here!** That voltage rating is important. Your capacitors can actually explode if you exceed it! So they should be able to withstand the maximum voltage used in your circuit. If they're rated higher – even better, since it will increase their lifespan. No worries though: the capacitors in this kit are carefully chosen to work properly in this circuit.







Ceramic capacitors usually come in disk- or pillow-like cases, are non-polarized and typically encode their capacitance value.[18] Annoyingly, they rarely indicate their voltage rating – so you'll have to note it down when buying them.

Film capacitors come in rectangular, boxy cases, are non-polarized and sometimes, but not always, directly indicate their capacitance value and their voltage rating without any form of encoding.[19]

Electrolytic capacitors can be identified by their cylinder shape and silver top, and they usually directly indicate their capacitance value and their voltage rating. They are polarized – so make sure you put them into your circuit in the correct orientation.

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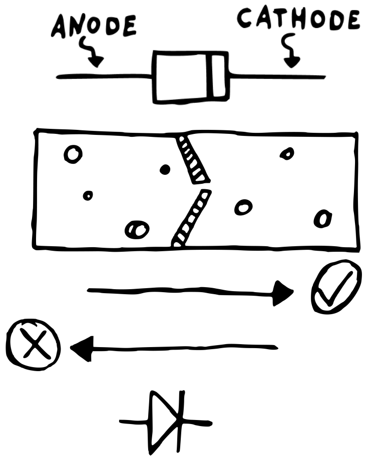
[18] For a detailed breakdown, look up ceramic capacitor value code. There are also calculation tools available.

[19] If yours do encode their values, same idea applies here – look up film capacitor value code.

DIODES

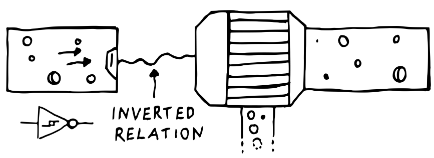


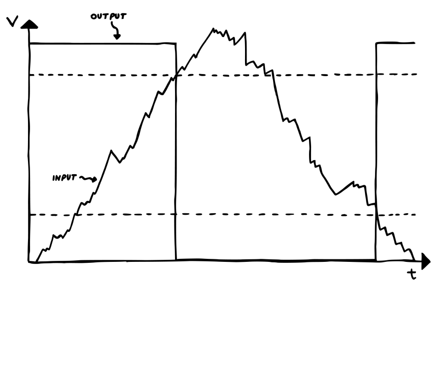
Diodes are basically like one-way valves. Current can only pass through in one direction – from anode to cathode. That direction is indicated by the arrow in the diode symbol and by a black stripe on the diode's casing. So any current trying to move in the opposite direction is blocked from flowing.

There are a few quirks here, though. For one, the diode will only open up if the pushing force is strong enough. Generally, people say that's 0.7 V**,** but in reality, it's usually a bit lower. Also, diodes don't open up abruptly – they start conducting even at much lower voltages, although just slightly.

There are a lot of different diode types: Zener, Schottky, rectifier, small signal etc. They all have their unique properties and ideal usage scenarios – but usually, a generic 1N4148 small signal diode will get the job done.

SCHMITT TRIGGER INVERTERS

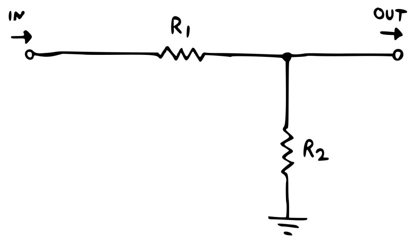




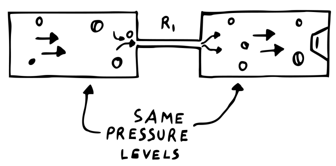
You can think of a Schmitt trigger inverter as two separate things. On the left, there's a sensor that measures the pressure inside an attached pipe. On the right, there is a water pump. This pump's operation is controlled by the sensor. Whenever the pressure probed by this sensor is below a certain threshold, the pump will be working. If the pressure is above a second threshold, the pump won't be working.

Here's a quick graph to visualize that. The squiggly line represents the voltage at the input, while the dotted line shows the voltage at the output. So every time we cross the upper threshold on our way up, and the lower one on our way down, the output changes its state. One thing that's very important to keep in mind: no current flows into the sensor! It's really just sensing the voltage without affecting it.

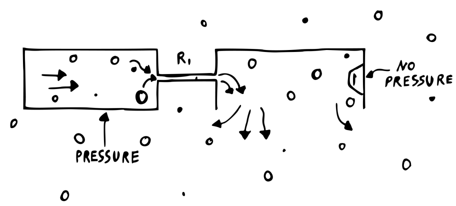
VOLTAGE DIVIDERS



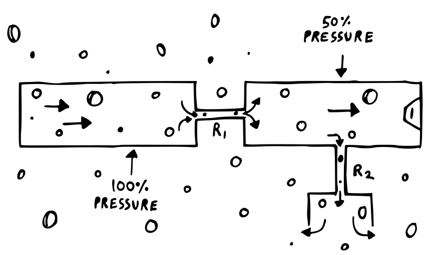
A voltage divider is really just two resistors set up like this: input on the left, output on the right. If R1 and R2 are of the same value, the output voltage will be half of what the input voltage is. How does it work?
Let's use our analogy again: so we have a pipe on the left, where water is being pushed to the right with a specific amount of force. Attached to it is a narrow pipe, representing R1, followed by another wide pipe. Then at the bottom, there's another narrow pipe, representing R2, where water can exit the pipe system. Finally, imagine we've set up a sensor measuring the voltage in the right hand pipe.
First, think about what would happen if R2 was completely sealed off. Our sensor would tell us that **the pressure on the right side is exactly the same as the pressure on the left**. Because the pushing force has nowhere else to go.



On the other hand, imagine R2 would just be a wide opening. Then **the pressure on the right would be 0**, because it'd all escape through that opening. But what happens if R2 is neither completely closed off nor wide open? Then the pressure would be retained to varying degrees, depending on the narrowness of the two resistor paths.

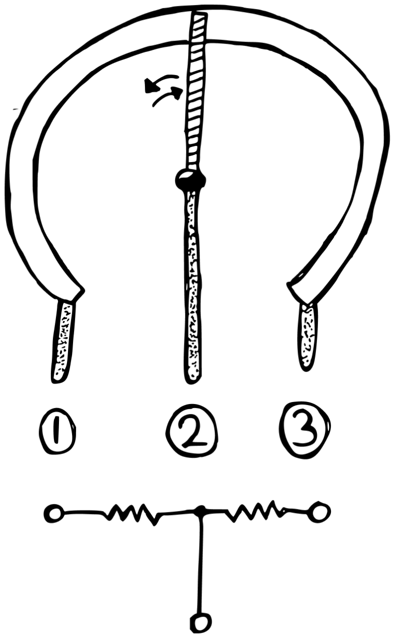


If pipe R1 is wide and pipe R2 is narrow, most of the pressure will be retained. But if it's the reverse, the pressure level will be only a tiny fraction. And if R1 and R2 are identical, **the pressure will be exactly half of what we send in**.



POTENTIOMETERS

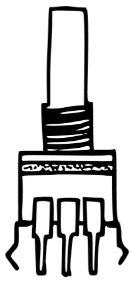
Potentiometers can be used as variable resistors that you control by turning a knob. But, and that's the handy part, they can also be set up as variable voltage dividers. To see how that works, let's imagine we open one up.

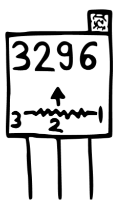


Inside, we would find two things: a round track of resistive material with connectors on both ends plus what's called a wiper. This wiper makes contact with the track and also has a connector. It can be moved to any position on the track. Now, the resistance value between the two track connectors is always going to stay exactly the same. That's why it's used to identify a potentiometer: as a 10k, 20k, 100k etc. But if you look at the resistance between either of those connectors and the wiper connector, you'll find that this is completely dependent on the wiper's position.

The logic here is really simple: **the closer the wiper is to a track connector, the lower the resistance is going to be between the two**. So if the wiper is dead in the middle, you'll have 50 % of the total resistance between each track connector and the wiper.

From here, you can move it in either direction and thereby shift the ratio between the two resistances to be whatever you want it to be. By now, you might be able to see how that relates to our voltage divider. If we send our input signal to connector 1 while grounding connector 3, we can pick up our output signal from the wiper. Then by turning the potentiometer's knob, we can adjust the voltage level from 0 to the input voltage – and anything in between.





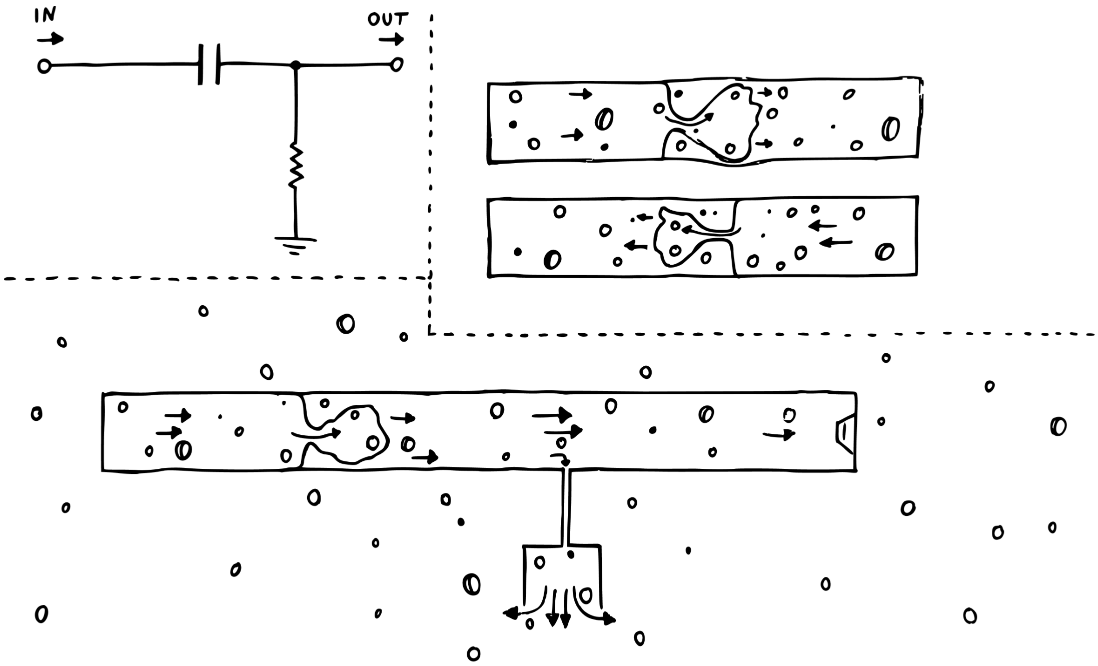
In these kits, you will encounter different types of potentiometers. First, there's the regular, full-size variant with a long shaft on top. These are used to implement user-facing controls on the module's panel and they usually – but not always – indicate their value directly on their casing. Sometimes, they'll use a similar encoding strategy as capacitors, though.[20]

Second, we've got the trimmer potentiometer, which is usually much smaller and doesn't sport a shaft on top. Instead, these have a small screw head which is supposed to be used for one-time set-and-forget calibrations. Trimmers usually encode their value.

[20] Look up potentiometer value code for a detailed breakdown.

AC COUPLING

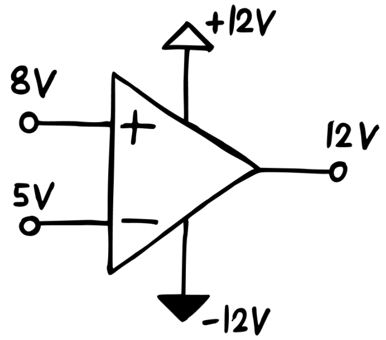
What is AC coupling – and how does it work? Imagine two adjacent pipes with a balloon between them. Now, no water can get from one pipe into the other, since it's blocked by the balloon. But, and that's the kicker, **water from one side can still push into the other by bending and stretching the balloon, causing a flow by displacement**.



Next, we'll bring in a resistor after the coupling point, going straight to ground. **This acts like a kind of equalizing valve**. Now imagine we apply a steady 5 V from one side. Then on the other side, we'll read 0 V after a short amount of time. Why? Because we're pushing water into the balloon with a constant force, causing it to stretch into the other side, displacing some water. If we didn't have the equalizing valve there, we'd simply raise the pressure. But since we do have it, the excess water can drain out of the system. Until the pressure is neutralized, and no water is actively flowing anymore.
Okay, so now imagine that the voltage on the left hand side starts oscillating, let's say between 4 V and 6 V. When we start to go below 5 V, the balloon will begin contracting, basically pulling the water to the left. This will create a negative voltage level in the right hand pipe – like as if you're sucking on a straw, making the voltage there drop below 0 V. Then, once the pressure on the other side rises above 5 V, the balloon will inflate and stretch out again, pushing water to the right. And the pressure in the right hand pipe will go positive, making the voltage rise above 0 V. **We've re-centered our oscillation around the 0 V line.** Okay, but what about the resistor? If current can escape through it, doesn't that mess with our oscillation? Well, technically yes, but practically, we're choosing a narrow enough pipe to make the effect on quick pressure changes negligible!

OP AMPS

Op amps might seem intimidating at first, but they're actually quite easy to understand and use. The basic concept is this: every op amp has two inputs and one output. Think of those inputs like voltage sensors. You can attach them to any point in your circuit and they will detect the voltage there without interfering. **No current flows into the op amps inputs – that's why we say their input impedance is very high**. Near infinite, actually. Okay, but why are there two of them?



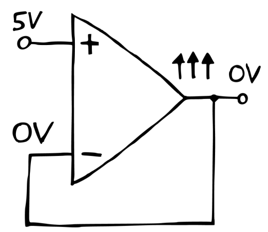
The key here is that op amps are essentially differential amplifiers. This means that they only amplify the difference between their two inputs – not each of them individually. If that sounds confusing, let's check out a quick example. So we'll imagine that one sensor – called the non-inverting input – is reading 8 V from somewhere. The other sensor – called the inverting input – reads 5 V. Then, as a first step, the op amp will subtract the inverting input's value from the non-inverting input's value. Leaving us with a result of 3. (Because 8 minus 5 is 3.) **This result then gets multiplied by a very large number – called the op amp's gain**. Finally, the op amp will try to push out a voltage that corresponds to that multiplication's result.

But of course, the op amp is limited here by the voltages that we supply it with. If we give it –12 V as a minimum and +12 V as a maximum, the highest it can go will be +12 V. So in our example, even though the result of that multiplication would be huge, the op amp will simply push out 12 V here and call it a day.

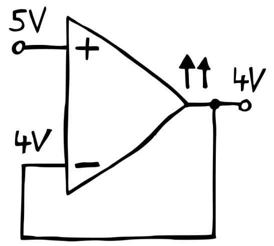
The handy thing though about op amps outputs is that they draw their power directly from the power source. This means that they can supply lots of current while keeping the voltage stable. **That's why we say an op amp has a very low output impedance**.

OP AMP BUFFERS/AMPLIFIERS

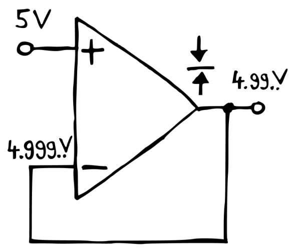
Buffering, in the world of electronics, means that we provide a perfect copy of a voltage without interfering with that voltage in the process. With an op amp-based buffer, the buffering process itself works like this. We use the non-inverting input to probe a voltage, while the inverting input connects straight to the op amp's output. **This creates what we call a negative feedback loop**. Think of it this way. We apply a specific voltage level to the non-inverting input – let's say 5 V.



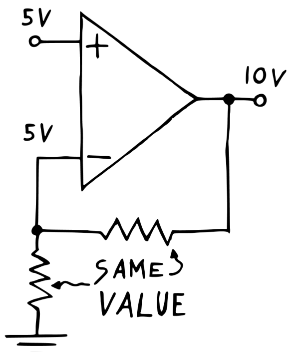
Before the op amp starts processing the voltages at its inputs, the output will be switched off. This means that **output and inverting input sit at 0 V at first**. So then, the op amp will subtract 0 from 5 and multiply the result by its gain. Finally, it will try and increase its output voltage to match the calculation's outcome.



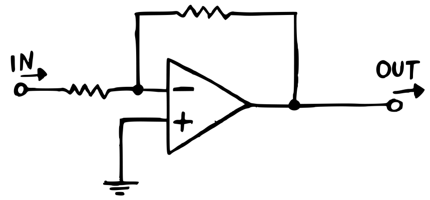
But as it's pushing up that output voltage, the **voltage at the inverting input will be raised simultaneously**. So the difference between the two inputs is shrinking down. Initially, this doesn't matter much because the gain is so large. As the voltage at the inverting input gets closer to 5 V though, the difference will shrink so much that in relation, the gain suddenly isn't so large anymore.



Then, the output will **stabilize at a voltage level that is a tiny bit below 5 V**, so that the difference between the two inputs multiplied by the huge gain gives us exactly that voltage slightly below 5 V. And this process simply loops forever, keeping everything stable through negative feedback. Now if the voltage at the non-inverting input changes, that feedback loop would ensure that the output voltage is always following. So that's why this configuration works as a buffer: the **output is simply following the input**.



How about amplifying a signal though? To do that, we'll have to turn our buffer into a proper non-inverting amplifier. We can do that by replacing the straight connection between inverting input and output with a voltage divider, forcing the op amp to work harder. Here's how that works. Say we feed our non-inverting input a voltage of 5 V. Now, **the output needs to push out 10 V in order to get the voltage at the inverting input up to 5 V**. We call this setup a non-inverting



amplifier because the output signal is in phase with the input.

For an inverting buffer/amplifier, the input signal is no longer applied to the non-inverting input. Instead, that input is tied directly to ground. So it'll just sit at 0 V the entire time. The real action, then, is happening at the inverting input. Here, we first send in our waveform through a resistor. Then, the inverting input is connected to the op amp's output through another resistor of the same value.

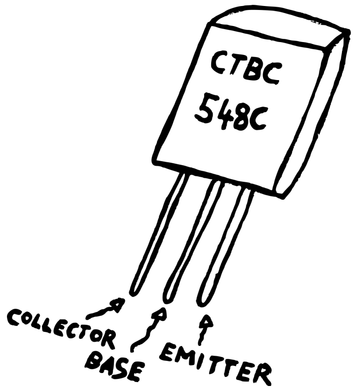
How does this work? Well, let's assume that we're applying a steady voltage of 5 V on the left. Then, as we already know, the op amp will subtract the inverting input's voltage from the non-inverting input's voltage, leaving us with a result of –5 V. Multiply that by the huge internal gain, and the op amp will try to massively decrease the voltage at its output.

But as it's doing that, an increasingly larger current will flow through both resistors and into the output. Now, as long as the pushing voltage on the left is stronger than the pulling voltage on the right, some potential (e.g. a non-zero voltage) will remain at the inverting input. Once the output reaches about –5 V though, we'll enter a state of balance. Since both resistors are of the same value, the pushing force on the left is fighting the exact same resistance as the pulling force on the right. **So all of the current being pushed through one resistor is instantly being pulled through the other**.

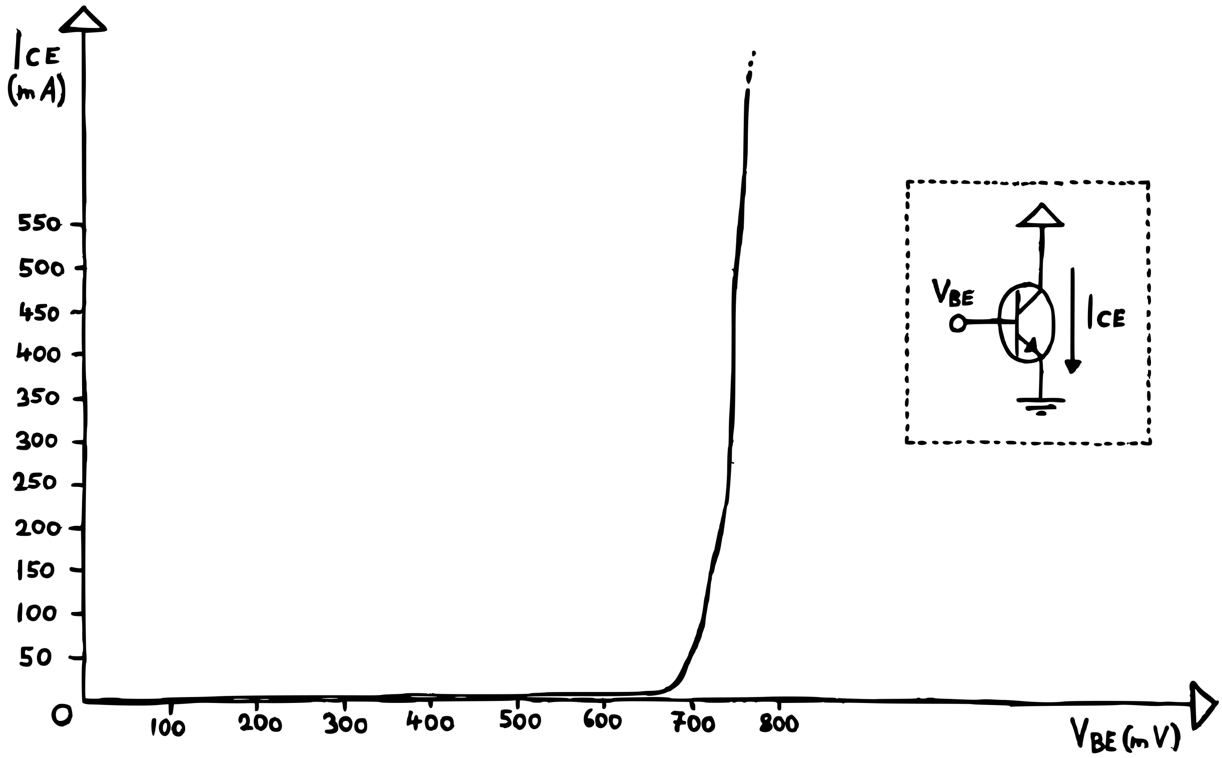
And that means that the voltage at the inverting input will be lowered to about 0 V, allowing our op-amp to settle on the current output voltage level. So while we read 5 V on the left, we'll now read a stable –5 V at the op amp's output. Congrats – we've built an inverting buffer! **If we want to turn it into a proper amplifier, we'll simply have to change the relation between the two resistances**. By doing this, we can either increase (if you increase the right-hand resistor's value) or reduce (if you increase the left-hand resistor's value) the gain to our heart's content.

BIPOLAR JUNCTION TRANSISTORS

Bipolar junction transistors (or BJTs for short) come in two flavors: NPN and PNP. This refers to how the device is built internally and how it'll behave in a circuit. Apart from that, they look pretty much identical: a small black half-cylinder with three legs.



Let's take a look at the more commonly used NPN variant first. Here's how we distinguish between its three legs. **There's a collector, a base and an emitter.**[21] All three serve a specific purpose, and the basic idea is that you control the current flow between collector and emitter by applying a small voltage[22] to the base. The relation is simple: **more base voltage equals more collector current**. Drop it down to 0 V and the transistor will be completely closed off. Sounds simple – but there are four important quirks to this.



First, the relation between base voltage and collector current is exponential. Second, unlike a resistor, a BJT is not symmetrical – so we can't really reverse the direction of the

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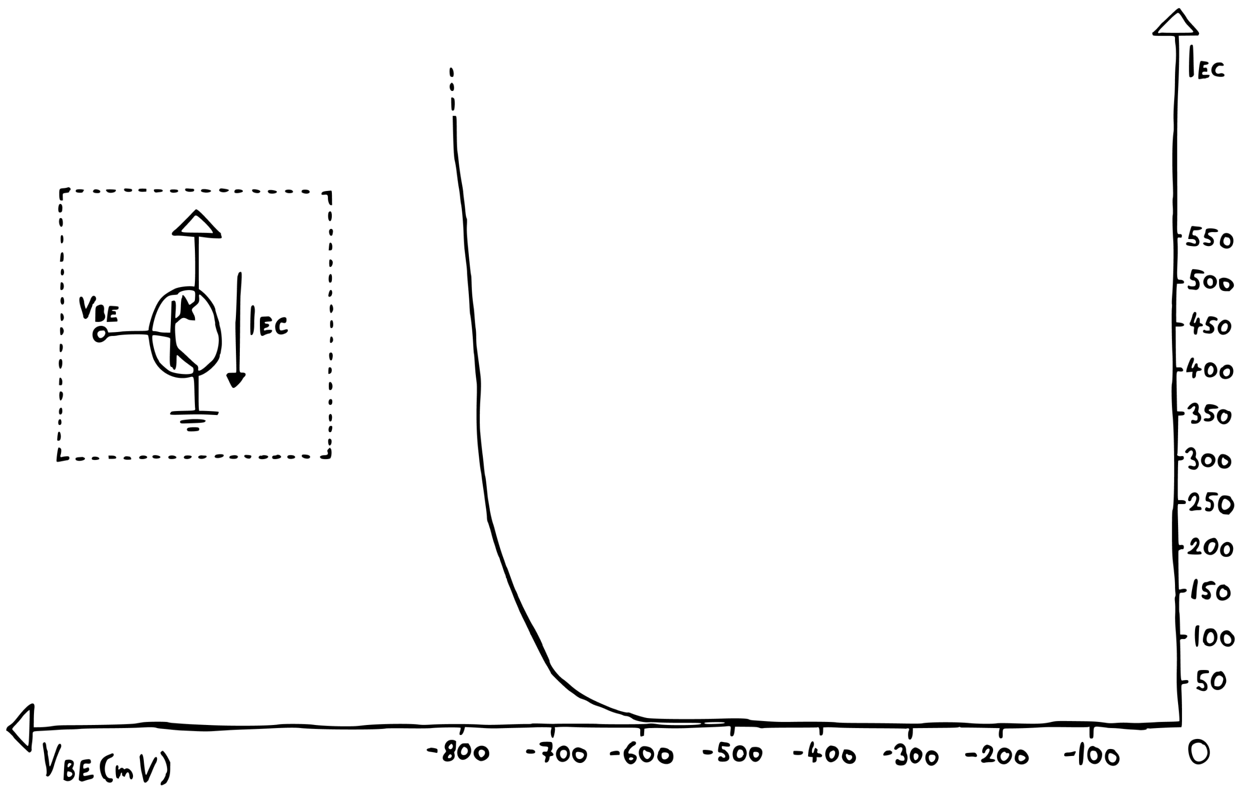
[21] Please note that the pinout shown here only applies for the BC series of transistors. Others, like the 2N series, allocate their pins differently.

[22] The voltage is measured between base and emitter. So „a small voltage" effectively means a small voltage **difference** between base and emitter!

collector current. (At least not without some unwanted side effects.) Third, also unlike a resistor, a BJT is not a linear device. Meaning that a change in collector voltage will not affect the collector current. And fourth, the collector current is affected by the transistor's temperature! The more it heats up, the more current will flow.

Now, for the PNP transistor, all of the above applies, too – except for two little details. Unlike with the NPN, **the PNP transistor decreases its collector current when the voltage at its base increases**[23]. So you have to bring the base voltage below the emitter to open the transistor up. Also, that collector current flows out of, not into the collector!

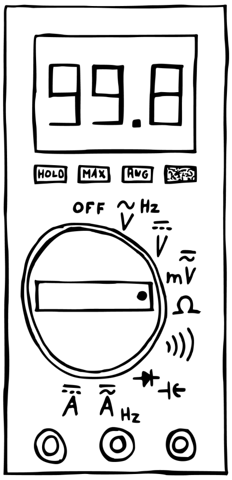


[23] Again, the voltage is measured between base and emitter.

TOOLS APPENDIX

There are two types of tools that will help you tremendously while designing a circuit: multimeters and oscilloscopes. In this appendix, we'll take a quick look at each of these and explore how to use them.

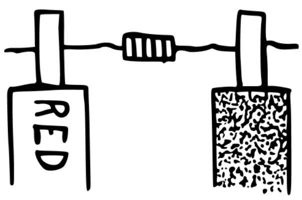
MULTIMETERS

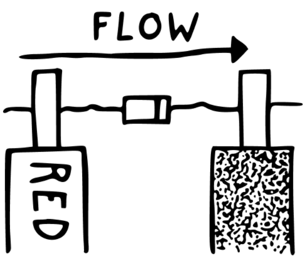


Multimeters come in different shapes and sizes, but the most common type is probably the hand-held, battery powered variant. It can measure a bunch of different things: voltage, current, resistance, continuity. Some have additional capabilities, allowing you to check capacitance, oscillation frequency or the forward voltage drop of a diode.

When shopping for one, you'll probably notice that there are really expensive models boasting about being TRUE RMS multimeters. For our purposes, this is really kind of irrelevant, so don't feel bad about going for a cheap model!

Using a multimeter is actually really straightforward. Simply attach two probes to your device – the one with a black cable traditionally plugs into the middle, while the red one goes into the right connector. Next, find whatever you want to measure and select the corresponding mode setting.





In some cases, it doesn't matter which probe you connect to which component leg or point in your circuit. This is true for testing resistors, non-polarized capacitors (foil/film, ceramic, teflon, glass etc.), continuity[24] or AC voltage.

In others, you'll have to be careful about which probe you connect where. For testing the forward voltage drop of a diode, for example, **the multimeter tries to push a current from the red to the black probe**. Here, you'll have to make sure the diode is oriented correctly, so that it doesn't block that current from flowing. For testing a DC voltage, you want to make sure the black probe is connected to ground, while you use the red one to actually take your measurement.

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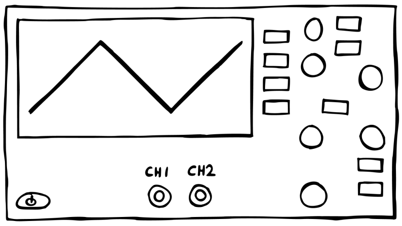
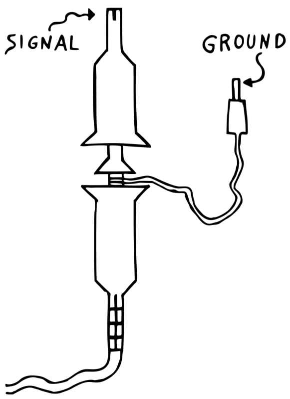
[24] Just a fancy word for saying that two points are electrically connected.

OSCILLOSCOPES



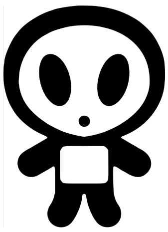


While multimeters are fairly cheap and compact, oscilloscopes are usually somewhat pricey and bulky. **If you're willing to make the investment, they are a huge help with the troubleshooting process, though**. Using one is, again, surprisingly straightforward – if you manage to work your way through the sometimes quite convoluted UI, especially on digital models.

To start using your scope, simply attach a probe to one of the channel inputs. These probes usually have two connectors on the other end: a big one that you operate by pulling the top part back – and a smaller one, which is usually a standard alligator clip. The latter needs to be connected to your circuit's ground rail, while you probe your oscillation with the former. Now what the oscilloscope will do is **monitor the voltage between the two connectors over time and draw it onto the screen as a graph**. Here, the x-axis is showing time, while the y-axis is showing voltage. You can use the device's scaling controls to zoom in on a specific part of your waveform.

Usually, digital oscilloscopes will also tell you a couple useful things about the signal you're currently viewing: minimum/maximum voltage level, oscillation frequency, signal offset. Some even offer a spectrum analyzer, which can be useful to check the frequencies contained in your signal.

BUILD GUIDE



MODULE ASSEMBLY APPENDIX

ecause the FM Drum is the most advanced project in our DIY.EDU series, it requires maximum attention during the assembly. Before we start building, let's take a look at the complete **mki x es.edu FM Drum** schematics (see next page) that were used for the final module's design and PCB fabrication. Most components on the production schematics have denominations (a name – like R1, C1, VT1, VD1, etc.) and values next to them. Denominations help identify each component on the PCB, which is particularly useful during **calibration, modification** or **troubleshooting.**

XS1 is the **Trigger input** jack socket, **XS2** is the **Accent input** jack sockets; it requires +5V gate signal to initiate the accent. **XS3** is the **Tune CV input, XS4 is the Decay CV input**, and **XS5** is the **Audio output** jack socket – these are the very same we've already been using on the breadboard for interfacing with other devices. In our designs, we use eurorack standard 3,5mm "Thonkicon" jack sockets (part number WQP-PJ301M-12).

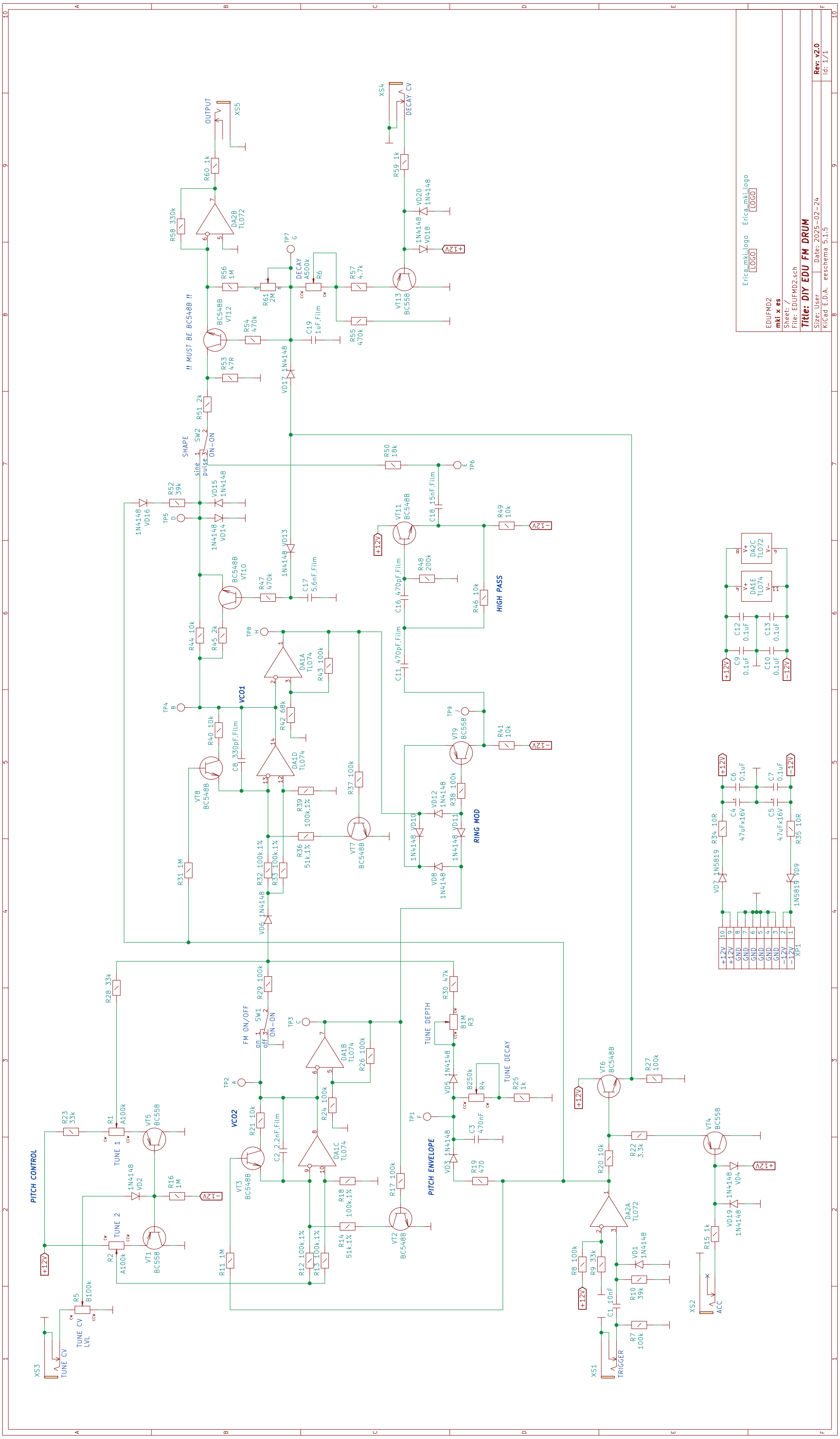
XP1 is a standard eurorack **power connector.** It's a 2x5 male pin header with a key (the black plastic shroud around the pins) to prevent accidental reverse polarity power supply connection. This is necessary because connecting the power incorrectly will permanently damage the module.

VD7 and **VD9** are **schottky diodes** that double-secure the reverse polarity power supply protection. Diodes pass current only in one direction. Because the anode of VD7 is connected to +12 V on our power header, it'll only conduct if the connector is plugged in correctly. If a negative voltage is accidentally applied to the anode of VD7, it closes, and no current passes through. The same goes for VD9, which is connected to -12 V. Because schottky diodes have a low forward voltage drop, they are the most efficient choice for applications like this.

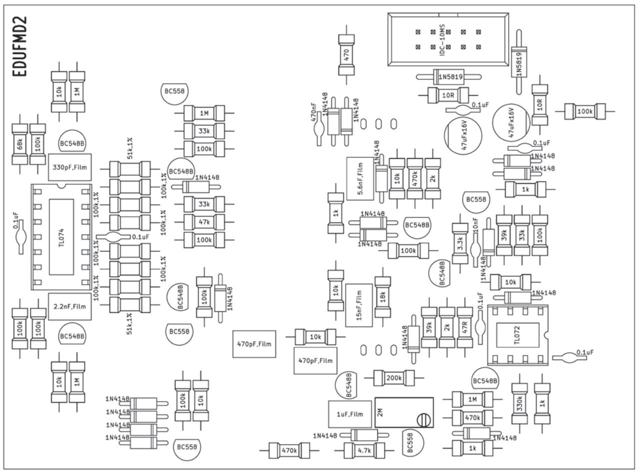
Next, we have two **10 Ohm resistors (R34** and **R35)** on the + and – 12 V rails, with **decoupling** (or **bypass-**) capacitors **C6, C7**. These capacitors serve as energy reservoirs that keep the module's internal supply voltages stable in case there are any fluctuations in the power supply of the entire modular system. In combination with R34 and R35, the large **47 microfarad pair (C4 and C5)** compensates for low frequency fluctuations, while C6 and C7 filter out radio frequencies, high frequency spikes from switching power supplies and quick spikes created by other modules. Often another component **– a ferrite bead** – is used instead of a 10 Ohm resistor and there's no clear consensus among electronic designers which works best, but generally for analogue modules that work mostly in the audio frequency range (as opposed to digital ones that use microcontrollers running at 8 MHz frequencies and above), resistors are considered to be superior.

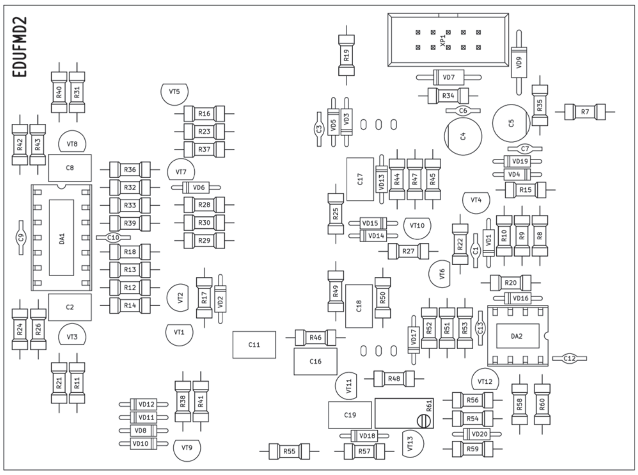
Another advantage of 10 Ohm resistors is that they will act like **slow "fuses"** in case there's an accidental short circuit somewhere on the PCB, or an integrated circuit (IC) is inserted backwards into a DIP socket. The resistor will get hot, begin smoking and finally break the connection. Even though they aren't really fuses, just having them there as fuse substitutes is pretty useful - **you'd rather lose a cent on a destroyed resistor than a few euros on destroyed ICs.**

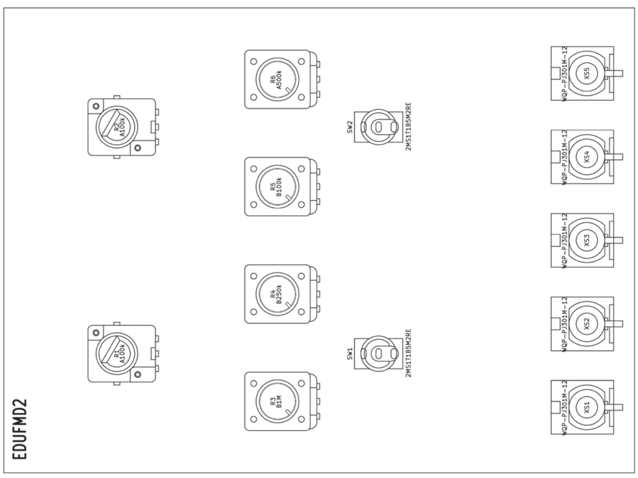
Capacitors **C9, C10, C12, C13** are additional decoupling capacitors. If you inspect the PCB, you'll see that these are placed as close to the power supply pins of the ICs as possible. For well-designed, larger PCBs you will find decoupling capacitors next to each IC. Like the others, their job is to simply compensate for any unwanted noise in the supply rails. If the input voltage drops, then these capacitors will be able to bridge the gap to keep the voltage at the IC stable. And vice-versa - if the voltage increases, then they'll be able to absorb the excess energy trying to flow through to the IC, which again keeps the voltage stable. Typically, 0.1 uF capacitors are used for this purpose.

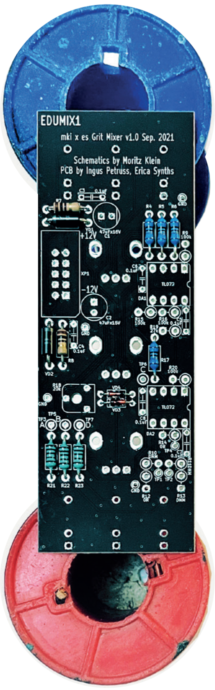

Title: DIY EDU FM DRUM
EDUFMD2
mki x es
File: EDUFMD2.sch
Size: User
Date: 2025-02-24
KiCad E.D.A. eeschema 5.1.5
Rev: v2.0
Id: 1/1
PITCH CONTROL
PITCH ENVELOPE
VCO1
VCO2
RING MOD
HIGH PASS
!! MUST BE BC548B !!
OUTPUT
DECAY CV
TRIGGER
ACC
TUNE CV

Before you start soldering, we highly recommend printing out the part placement diagrams with designators and values and follow step-by step instructions below. As mentioned above, FM Drum is the most complex project in our DIY.EDU line, so, this will help you to avoid mistakes in the build process.

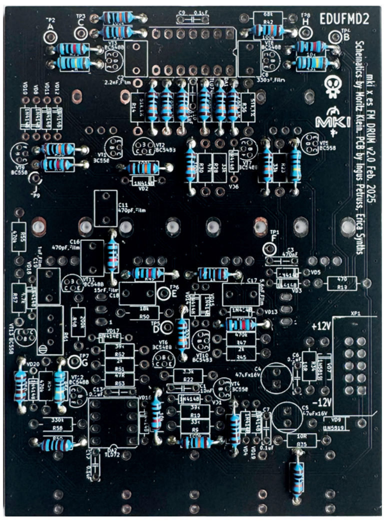




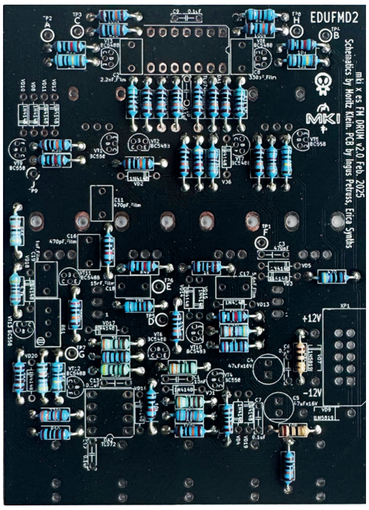




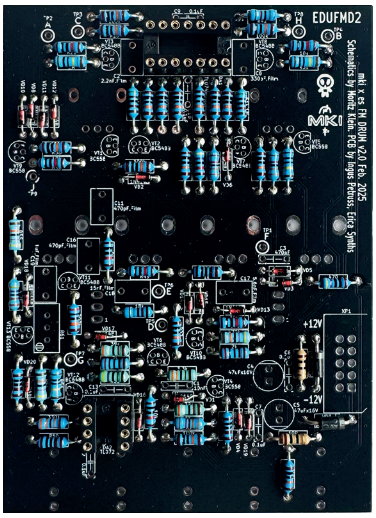
Place the FM Drum PCB in a PCB holder for soldering or simply on top of some spacers (I use two empty solder wire coils here).



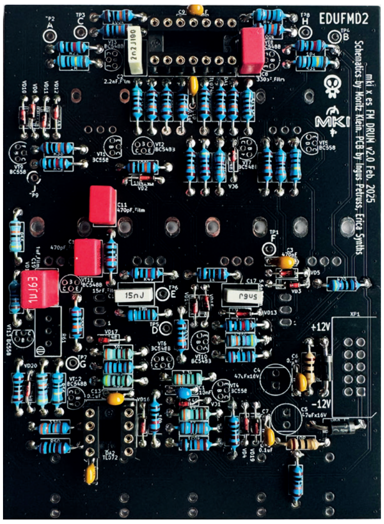
I usually start populating PCBs with lower, horizontally placed components. In this case we have lot of resistors, so, for sake of clarity, let's start with **100k and 10k resistors.** There are six 100k 1% accuracy resistors in the BOM, but because all resistors in the FM Drum kit are 1% accuracy, so you can just solder them all. Bend the resistor leads and insert them in the relevant places according to the part placement diagram above. Flip the PCB over and solder all components. Then, use pliers to cut off the excess leads.



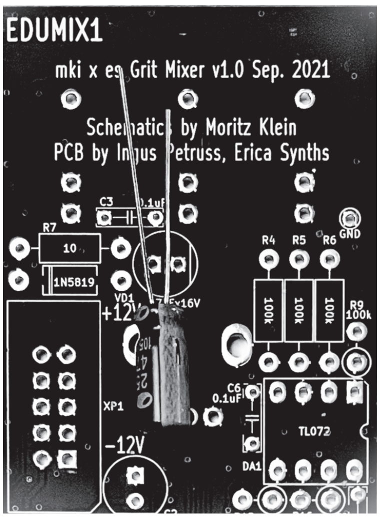
Next, populate remaining **resistors.**



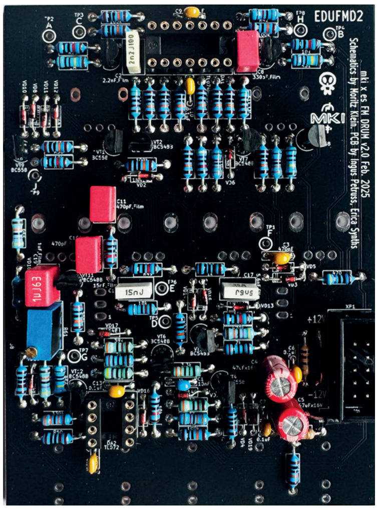
Now, proceed with **switching diodes** and the **power protection diodes.** Remember – **when inserting the diodes, orientation is critical!** A thick white stripe on the PCB indicates the cathode of a diode – match it with the stripe on the component. Solder all diodes. Also, **insert the first DIP socket,** hold it in place and solder one of the pins. Continue with the **next DIP socket.** Make sure the DIP sockets **are oriented correctly** – the notch on the socket should match the notch on the PCB's silkscreen. Now, turn the PCB around and solder all remaining pins of the DIP sockets. Compare your component placement with the one on the picture below.



Then proceed with the ceramic capacitors. Start with soldering 0,1uF capacitors - place the PCB in your PCB holder or on spacers, insert the capacitors and solder them like you did with the resistors & diodes before
Now, solder other ceramic and film capacitors. When completed, your PCB should look like this:

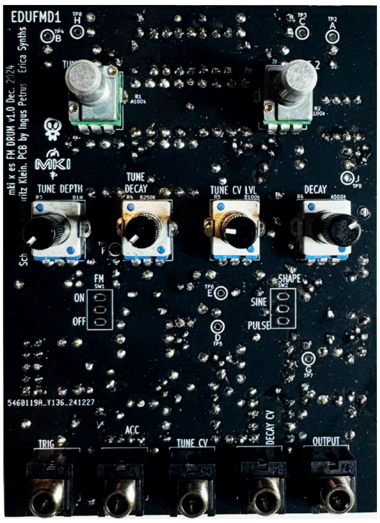


Insert & solder the electrolytic capacitors. Electrolytic capacitors are bipolar, and you need to mind their orientation. The positive lead of each electrolytic capacitor is longer, and there is a minus stripe on the side of the capacitor's body to indicate the negative lead. On our PCBs, the positive pad for the capacitor has a square shape, and the negative lead should go into the pad next to the notch on the silkscreen.
Next, insert and solder **transistors.** There are PNP and NPN transistors in the kit, therefore before soldering them, I highly recommend to sort them. Make sure you install them in correct places and pay attention on the orientation of the transistors – notch on the silkscreen has to match the flat part of the transistor.

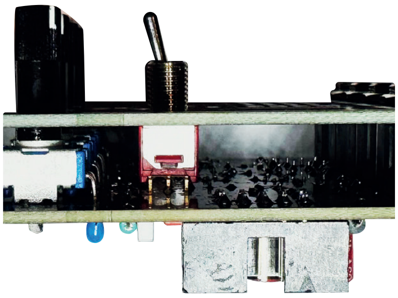


Then populate the trimpot and **2x5 PSU socket.** Make sure the orientation of the socket is as shown in the picture below – the arrow pointing to the first pin is aligned with a notch on the silkscreen. The key on the socket will be facing outwards the PCB. Now your PCB should look like this:

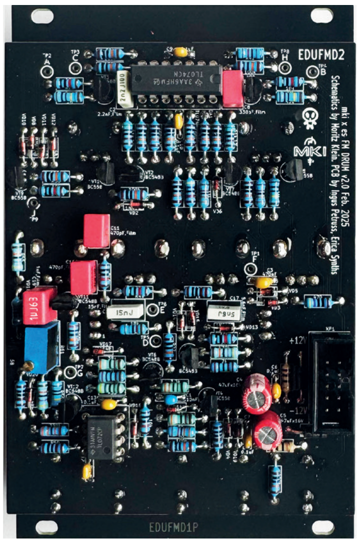
Now, turn the PCB around and inspect your solder joints. **Make sure all components are soldered properly and there are no cold solder joints or accidental shorts.** Clean the PCB to remove extra flux, if necessary.



Next, **insert the top potentiometers and jack sockets,** then fit the panel to align components, you just installed, and solder them. Remove the front panel, insert other potentiometers, but don't solder them yet! Fit the front panel, screw the nuts on the top potentiometer and jack sockets and make sure that the potentiometer shafts are aligned with the holes in the panel – and that they're able to rotate freely. Now, go ahead and solder the potentiometers.



The switches need special attention. Insert the **switches** in relevant places, but **do not solder** them, yet. Place the front panel, fix it with a potentiometer and jack socket nuts. Then tighten switch nuts, so that switches are pushed against the front panel. Now, solder the switches.



Now, insert the ICs into their respective DIP sockets. Mind the orientation of the ICs – match the notch on each IC with the one on its socket.

Finally fit the Tune potentiometer knobs and we are done!

Congratulations! **You have completed the assembly of the mki x es.edu FM Drum module!** Connect it to your eurorack power supply and switch it on. If there's no "magic smoke", it's a good sign that your build was successful.

The module has one trimpot that needs to be adjusted. Patch trigger signal (the gate output of your DIY.EDU Sequencer will work fine, but the Erica Synths Drum Sequencer is the best choice) to the input of the module and connect the output of the module to a mixer. You should hear the percussion sound. The intended purpose of the trimpot is to center the output waveform around 0V, if the VCA adds a positive DC offset. (The best is to test the FM Drum built on the DIY.EDU Labor, before building it on the PCB.) Connect the output of the FM Drum to the oscilloscope in DC coupling mode (the Labor Scope is good choice), apply some triggers, and observe the waveform on the output.

If the offset is negative, either try a different TL072 chip, or remove the trimpot.
If the offset is negative, but close to 0, also remove the trimpot.
If the offset is positive, adjust the trimpot so that the center of the waveform on the output of the module is at 0V. The effect of a non-centered wave is additional low-frequency content at the output signal, which may not be nice in the cymbal modes.

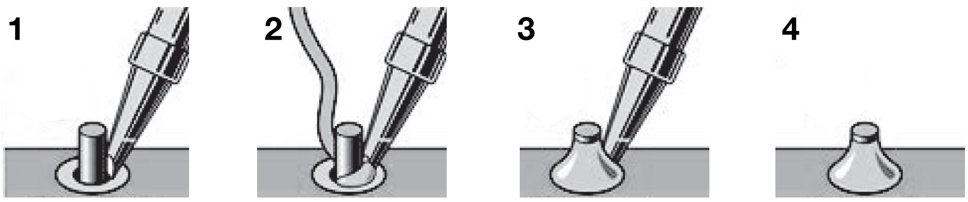
Once calibration is completed, it's time to make some music. Turn gates on the sequencer on and off in order to achieve a desired drum pattern and tweak some knobs and flip switches on the module to observe change of the sound.

Enjoy!

SOLDERING APPENDIX

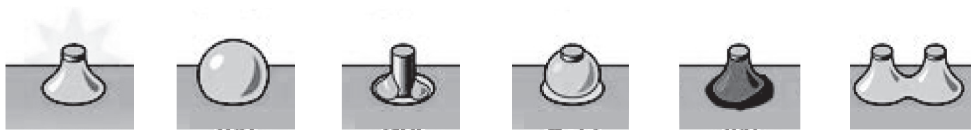
If you've never soldered before – or if your skills have become rusty – it's probably wise to check out some **THT** (through-hole technology) **soldering tutorials on YouTube**. The main thing you have to remember while soldering is that melted solder will flow towards higher temperature areas. So you need to make sure you apply equal heat to the component you are soldering and the solder pad on the PCB. The pad will typically absorb more heat (especially ground-connected pads which have more thermal mass), so keep your soldering iron closer to the pad on the PCB. It's critically important to dial in the right temperature on your soldering station. I found that about 320 °C is the optimal temperature for most of parts, while for larger elements like potentiometers and sockets, you may want to increase that temperature to **370 °C.**

Here's the recommended soldering sequence:



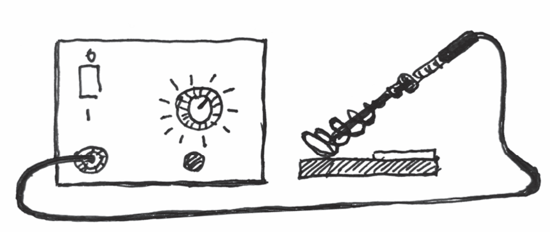
Heat part and pad 2 - 3 sec | Add solder | Continue heating 1 -2 sec. | Let cool

After you have completed soldering, inspect the solder joint:

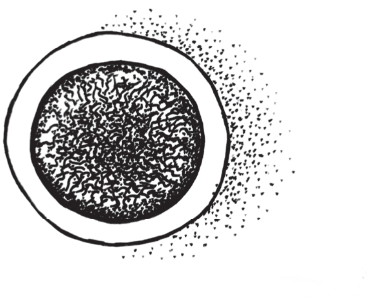


Perfect | Too much solder | Not enough solder | Cold joint | Too much heat | Short

DIY electronics is a great (and quite addictive) hobby, therefore we highly recommend you invest in good tools. In order to really enjoy soldering, you'll need:

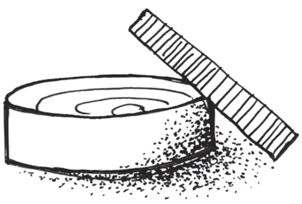


A decent soldering station. Top-of-the-line soldering stations (brands like Weller) will cost 200€ and above, but cheaper alternatives around 50€ are often good enough. Make sure your soldering station of choice comes with multiple differently-sized soldering iron tips. The most useful ones for DIY electronics are flat, 2mm wide tips.



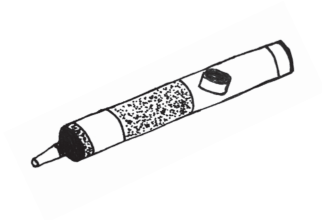
When heated up, the tips of soldering irons tend to oxidize. As a result, solder won't stick to them, so you'll need to clean your tip frequently. Most soldering stations come with a **damp sponge for cleaning the iron tips** – but there are also professional solder tip cleaners with **golden curls** (not really gold, so not as expensive as it sounds). These work much better because they do not cool down the iron.

Solder wire with flux. I find 0,7mm solder wire works best for DIY projects.

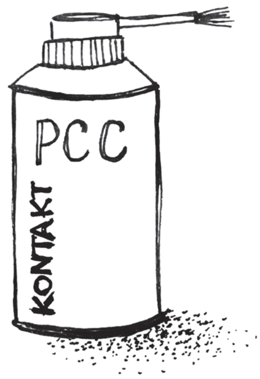


Some **soldering flux** paste or pen will be useful as well.

Cutting pliers. Use them to cut off excess component leads after soldering.



A solder suction pump. No matter how refined your soldering skills are, you will make mistakes. So when you'll inevitably need to de-solder components, you will also need to remove any remaining solder from the solder pads in order to insert new components.



Once you have finished soldering your PCB, it's recommended to remove excess flux from the solder joints. **A PCB cleaner** is the best way to go.

All of these tools can be found on major electronic components retailer websites, like Mouser, Farnell and at your local electronics shops. As you work your way towards more and more advanced projects, you'll need to expand your skillset and your tool belt – but the gratification will be much greater.