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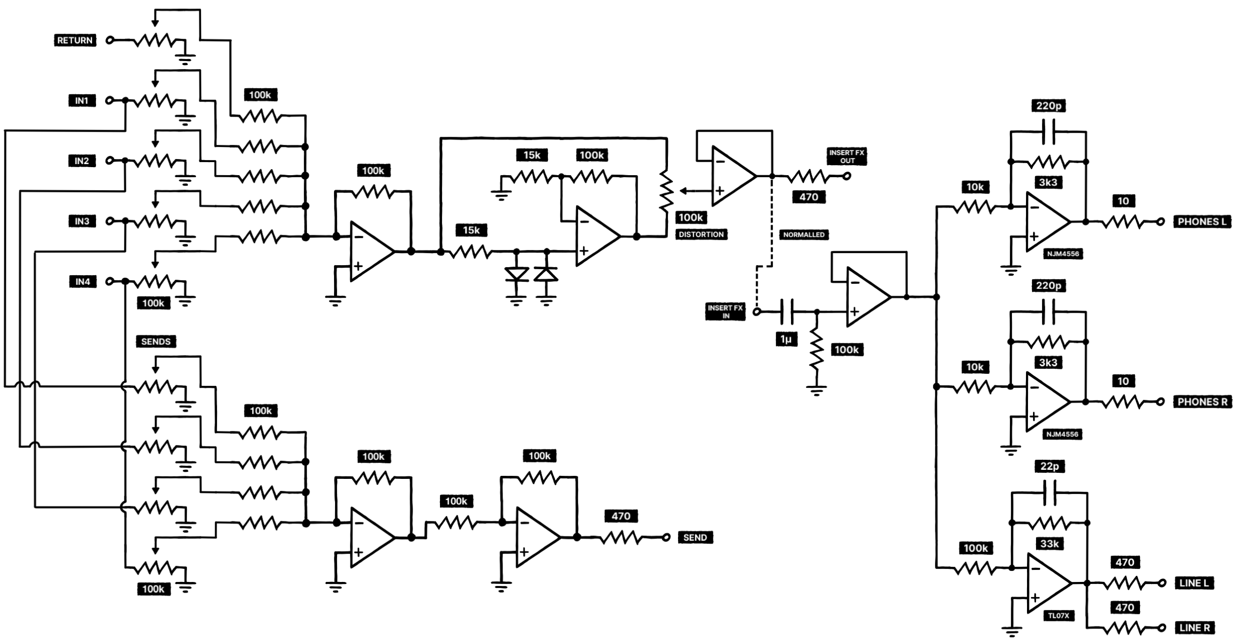
[1] You can also use a standard breadboard, but this will require you to get a little creative when adapting the suggested layouts – plus you'll need to do some additional engineering to get the different supply voltages.

THE mki x es.edu OUTPUT MIXER 2

Most output mixer modules are simple: a few level knobs, maybe an aux input, and that's it. But for our DIY Drums System, I wanted something more flexible. **This is the final piece of the puzzle – the module that brings all four drum voices together and lets us shape the overall sound.**

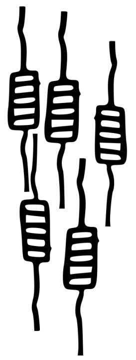
The system includes two effects: a bucket brigade delay and a diode-based compressor. Rather than slapping those onto the entire mix, I wanted to be able to send individual voices into them, blend the results back in, and explore more complex textures that way. **So I decided to set up a complete send/return path with individual send controls for each channel**.

And since I love adding a bit of grit to my drum mix, I also implemented a diode-clipping overdrive stage as a default insert effect. It's a simple circuit, but it opens up a whole new range of tones.



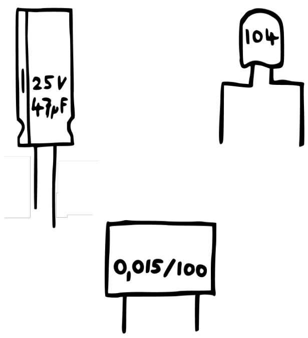
BILL OF MATERIALS

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:



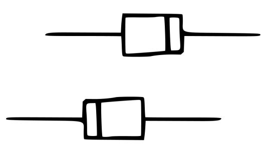
A bunch of resistors. The specific values (in ohms, which you should check for with a multimeter) are

100k	x16
33k	x1
15k	x2
10k	x2
3k3	x2
470	x4
10	x4



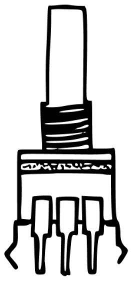
A couple capacitors. The specific values (which are printed onto their bodies) are

47 µF	x2
1 µF	x1
100 nF (0,1)	x8
220 pF (220/221)	x2
22 pF (22/220)	x1



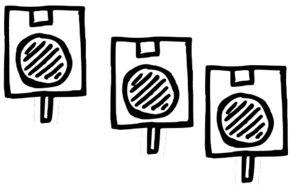
Some diodes. The specific model names (which are printed onto their bodies) are

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



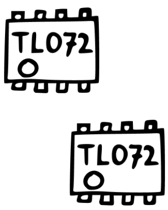
A handful of potentiometers. Their specific values (which may be encoded & printed onto their bodies) are

100k (A104) x9
100k (B104) x1



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

Switched mono (black) x8
Switched stereo (green) x2



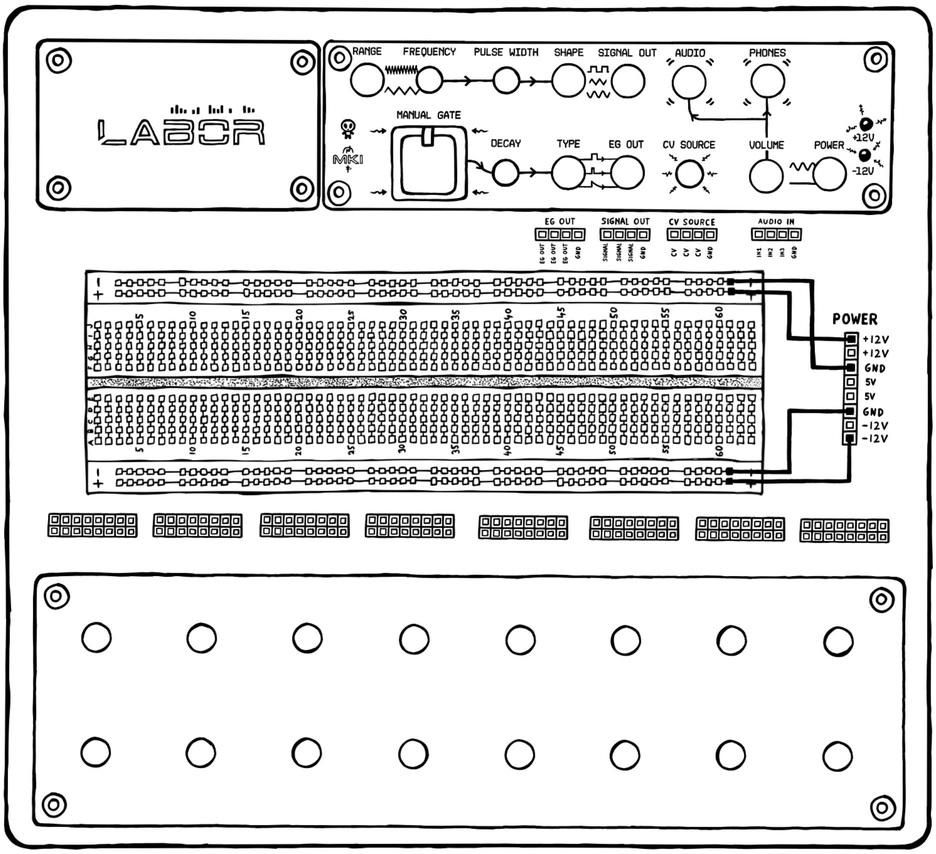
A couple chips. Their specific models (which are printed onto their bodies) are

TL074 (quad op amp) x2
NJM4556 (dual power op amp) x1

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

USAGE WITH MKI x ES LABOR

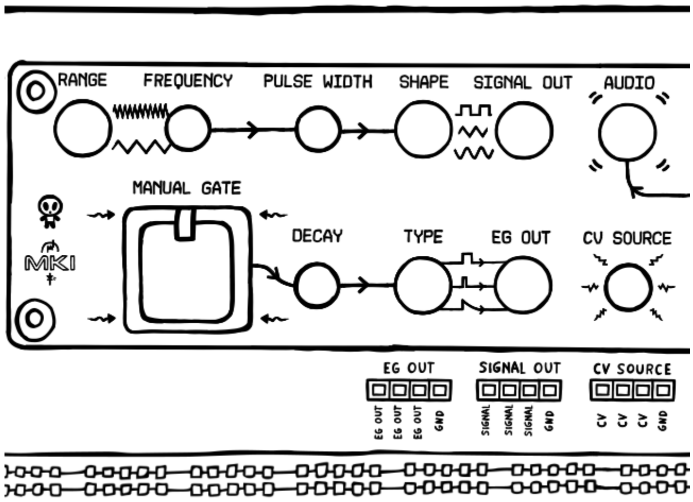
We highly recommend that you follow this guide using an **MKI x ES LABOR** prototyping board. **LABOR** comes equipped with everything you need for testing the circuits we lay out: a standard 830 tie point breadboard, an integrated dual power supply with short circuit protection, a manual gate/trigger/envelope 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).

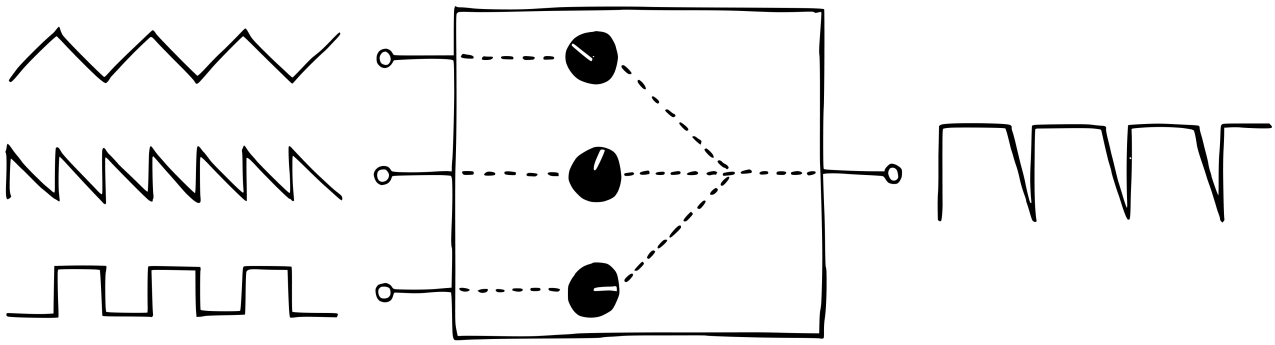
To listen to your circuit, you don’t even need to set up an output jack socket. Instead, use the built-in output amplifier at the top of the device. Just plug your circuit’s signal output into the header labeled **AUDIO IN**, and then connect your headphones to the **PHONES** output jack (or a line-level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



WHAT'S A MIXER?

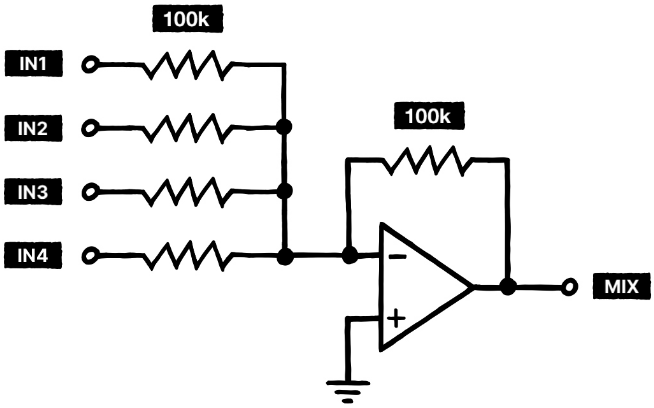
Before we talk about the more intricate parts of the circuit's design, we'll need to understand what a mixer does, exactly. Thankfully, compared to other modules like kick drums, delays, and sequencers, mixers are pretty straightforward and simple devices. **They've got one basic purpose: take multiple input signals and combine them into a single output signal.**



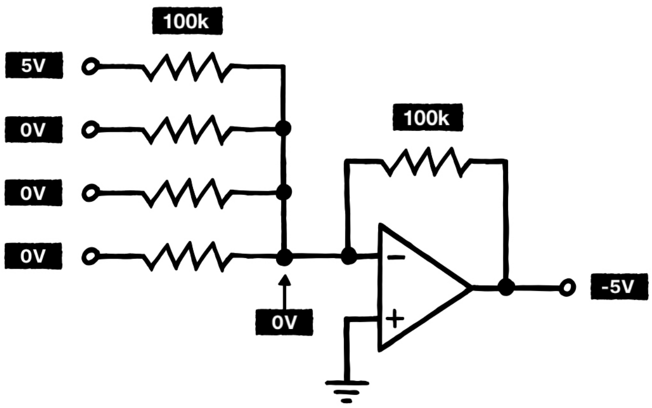
In our case, we want to mix 4 signals: Kick Drum, Hi-Hat, Snare Drum, and FM Drum. The resulting drum mix should then be made available via a single output. Ideally, we'll also want to adjust the individual volume levels for every input signal. Okay, so how do we do that?

THE SUMMING AMPLIFIER

Let's start by focusing on mixing first, and worry about volume controls later.[2] For this, all we need is five 100k resistors and one op amp. **Set them up like this, and you get what's called an inverting summing amplifier with 4 input channels.**[3]



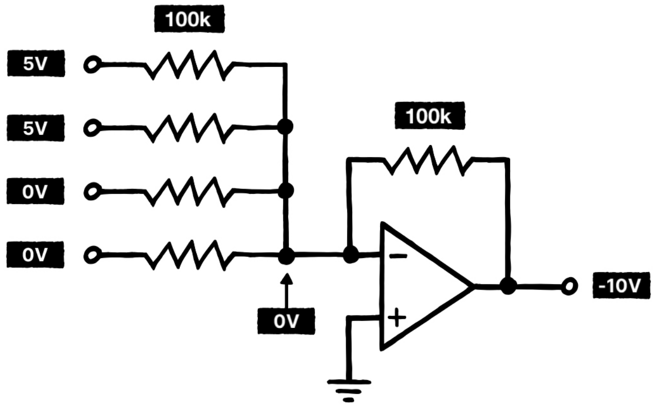
Here's how it works. The op amp is wired so that its inverting input (–) stays as close to 0 V as possible. That's because the circuit only reaches a state of balance when the voltage at the inverting input matches the voltage at the non-inverting input (+). **So whenever the inverting input moves away from 0 V, the op amp adjusts its output in the opposite direction to restore balance.**
Let's assume that one input suddenly jumps up to 5 V while all others stay at 0 V.



[2] For a deeper look at how mixing works in detail, see our DIY Mixer manual: https://www.ericasynths.lv/media/MIXER_MANUAL.pdf

[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/2yzl2of6 – you can change all values by double clicking on components.

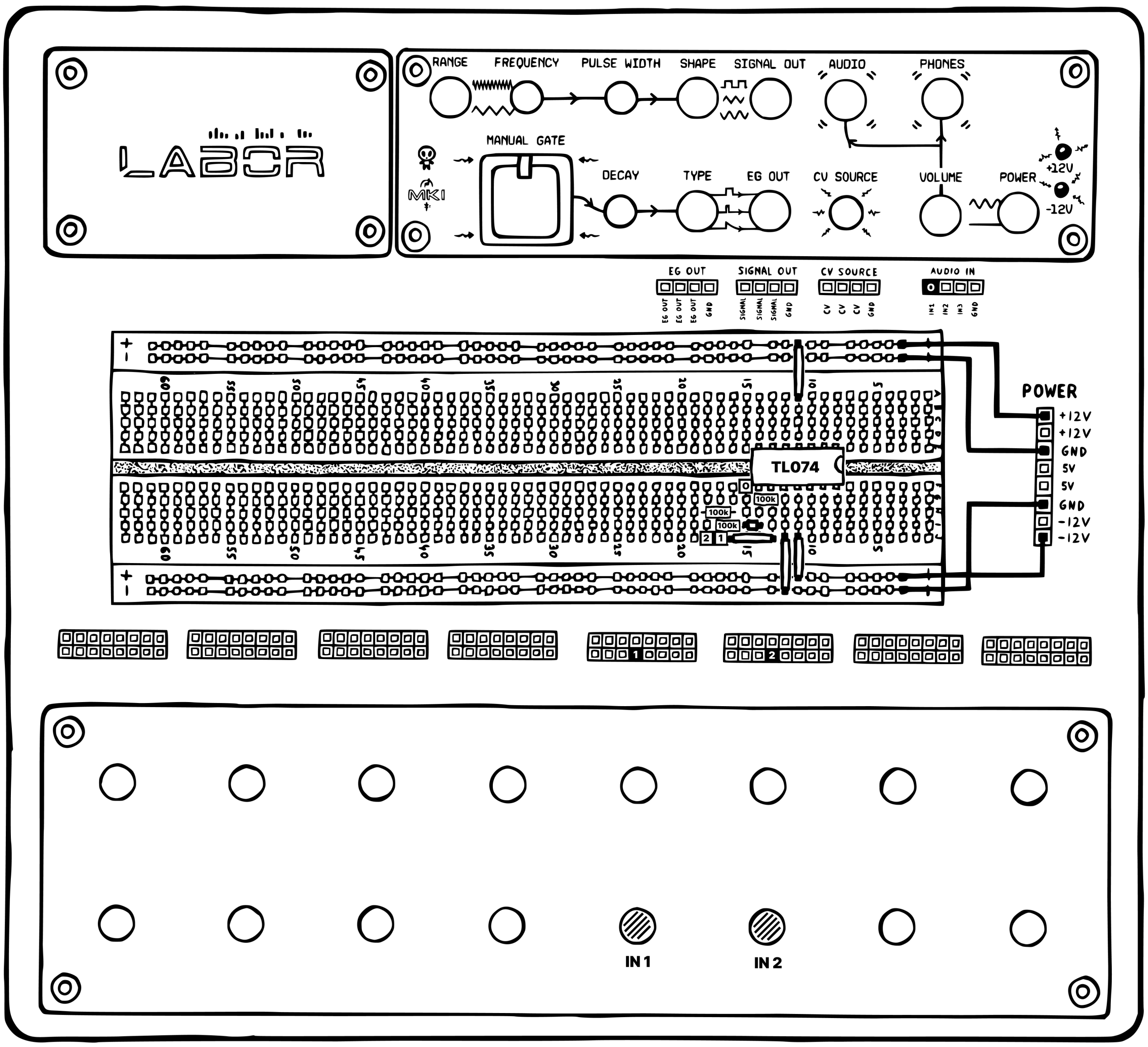
The op amp will respond by pulling its output down to –5 V – this way, the two 100k resistors form a 50% voltage divider whose midpoint (the inverting input) sits right at 0 V. Next, imagine another input also jumps to 5 V.



The op amp now has to pull twice as hard to neutralize both signals, so its output drops down to –10 V. **That's why we call this circuit an inverting summing amplifier: it sums the inputs while inverting the result.** The inversion isn't something we chose intentionally – it's simply a necessary byproduct.[4] But for audio, that doesn't matter at all. A flipped waveform sounds exactly the same, as long as we don't later mix it with a non-inverted version (which would cause phase cancellation issues).

For testing our summing amplifier on the breadboard, we can make life a bit easier by setting up just 2 channels instead of 4. This keeps things tidy – which will become more important as we add more functionality later – while still letting us verify that the circuit works as intended.

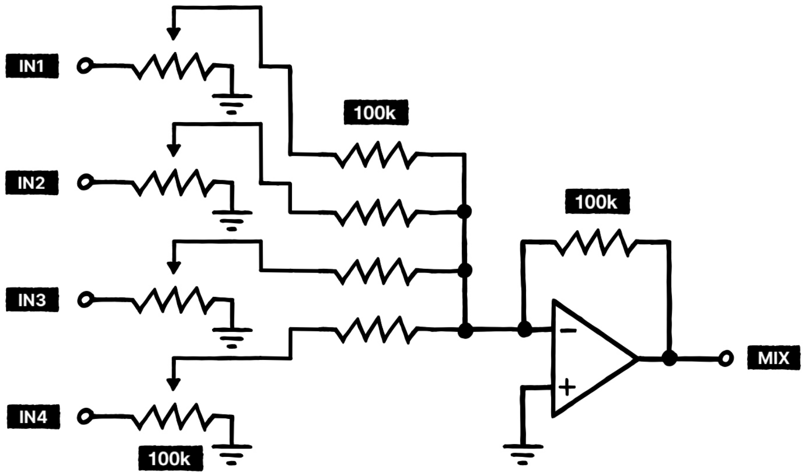
[4] This is also explored more extensively in the DIY Mixer manual.



If you set this up and feed a couple audio signals into it, you should get a clean mix at Labor's audio outputs. Great! Now what about those volume controls?

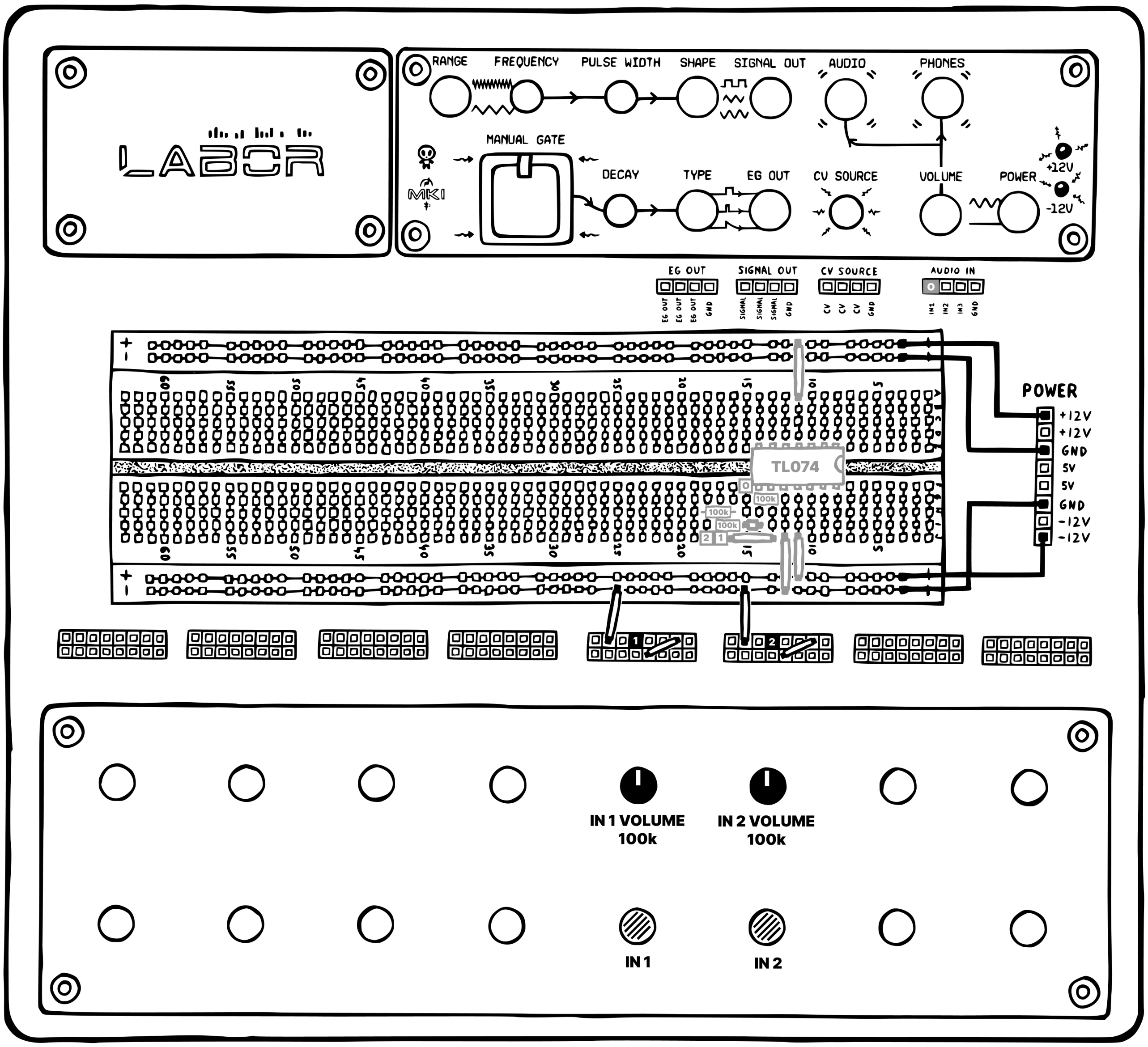
PER-CHANNEL VOLUME CONTROLS

Thankfully, adding them is fairly straightforward. All we need to do is put one 100k potentiometer between each input socket and its respective 100k resistor like this.[5]



Each potentiometer acts as a variable voltage divider – or in audio terms, an attenuator. Here's the idea: when we send an audio signal through a voltage divider, we can pick up a scaled-down version of that signal at the divider's output. The amount of scaling depends on the ratio between the two resistors in the divider. **Conveniently, a potentiometer is just that – a voltage divider built into a single, adjustable component.** Turning the knob changes the resistance ratio, and therefore the output level. This lets us set each channel's signal level to anywhere between full volume and complete silence, giving us per-channel control over the mix.

[5] You can try this chapter's circuit in a circuit simulator. I've already set it up for you right here: https://tinyurl.com/2yoe5srk – you can change all values by double clicking on components.

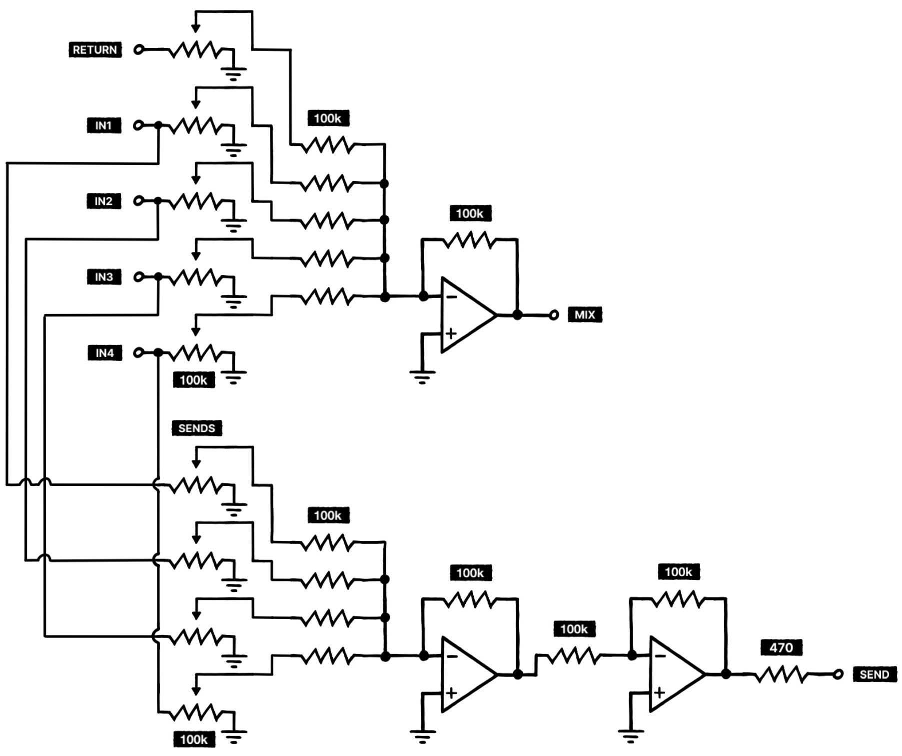


Once you've set this up, try connecting audio signals to both inputs and play with the two potentiometers. You should be able to smoothly adjust the volume of each channel.

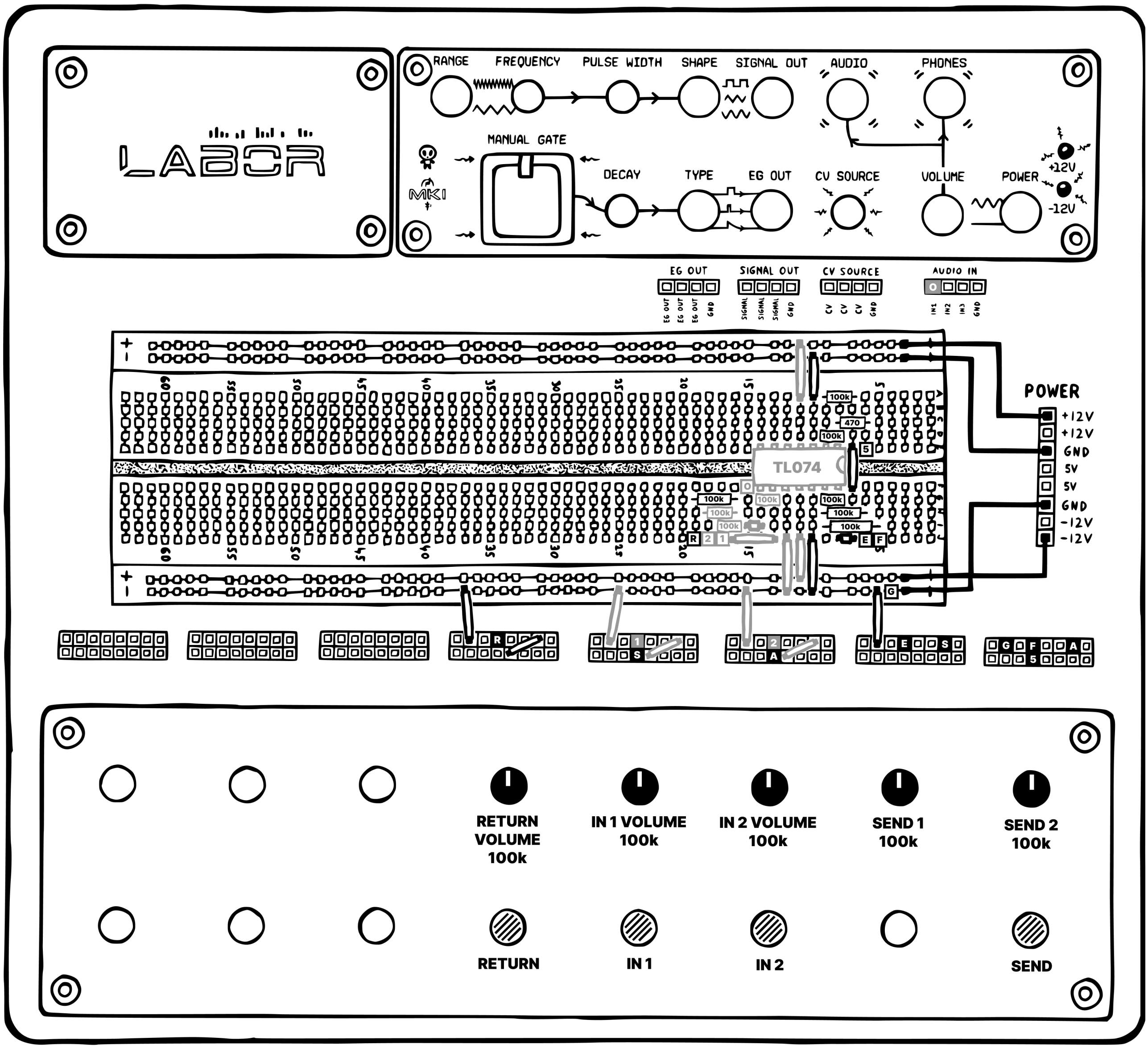
THE SEND/RETURN LOOP

Now that we've got mixing fully covered, let's see how we can add external effects – but only to the channels we want. If we simply took the mixer's output and ran it through an effect like our bucket brigade delay or compressor, every sound would be processed the same way. That's not ideal – maybe you only want delay on your snare and hi-hat, not on the entire drum mix.

The way to solve this is with a send/return loop. **A send/return loop lets you decide which channels get sent to an external effect, and then mix the processed signal back in**. Here's how it works. First, we clone our existing mixer – including the input potentiometers. Then, we feed our four input signals into that new mixer. The signal we get from this is our send output, and the four input potentiometers are our per-channel send controls. Finally, we add a fifth input channel to the original mixer as our return input – this is where the processed signal from the effect will come back in.



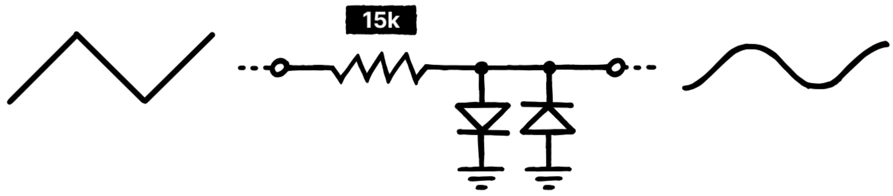
There's one thing to watch out for: since our send mixer is inverting, the signal coming back from the effect would be flipped in polarity compared to the dry signals. **To avoid phase issues when blending them, we'll pass the send mixer's output through a simple inverting buffer – essentially the same circuit as before, just with a single input**. This cancels the inversion, bringing the send output back in phase with the four inputs. Finally, we'll add a 470 ohm resistor in series with the output. Long patch cables have capacitance, which can make the op amp ring or oscillate if driven directly. **This resistor helps dampen that effect and keep the output stable**.



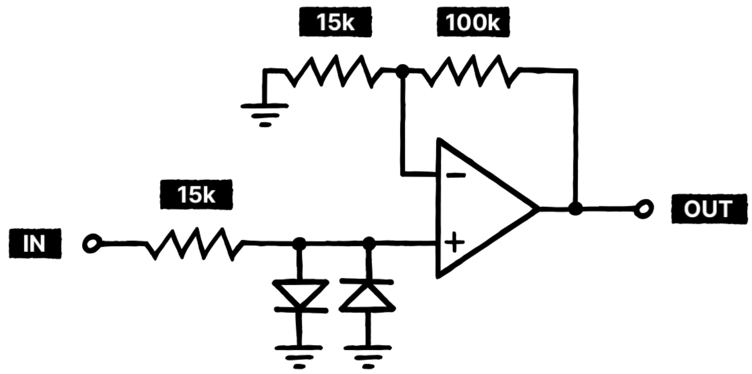
Try connecting the send output to an effect and routing its output back to the return input. You should now be able to apply the effect only to specific channels by adjusting their send potentiometers.

INSERT DISTORTION

One effect that I think actually sounds best when it affects all drum voices equally is distortion. Since our DIY Drums System doesn't have a dedicated distortion module, we'll simply build one right into the mixer – as an insert effect, meaning it processes the entire mix at once. **The simplest way to distort a signal is through diode clipping**. You only need three components for this: two diodes and one resistor.[6]

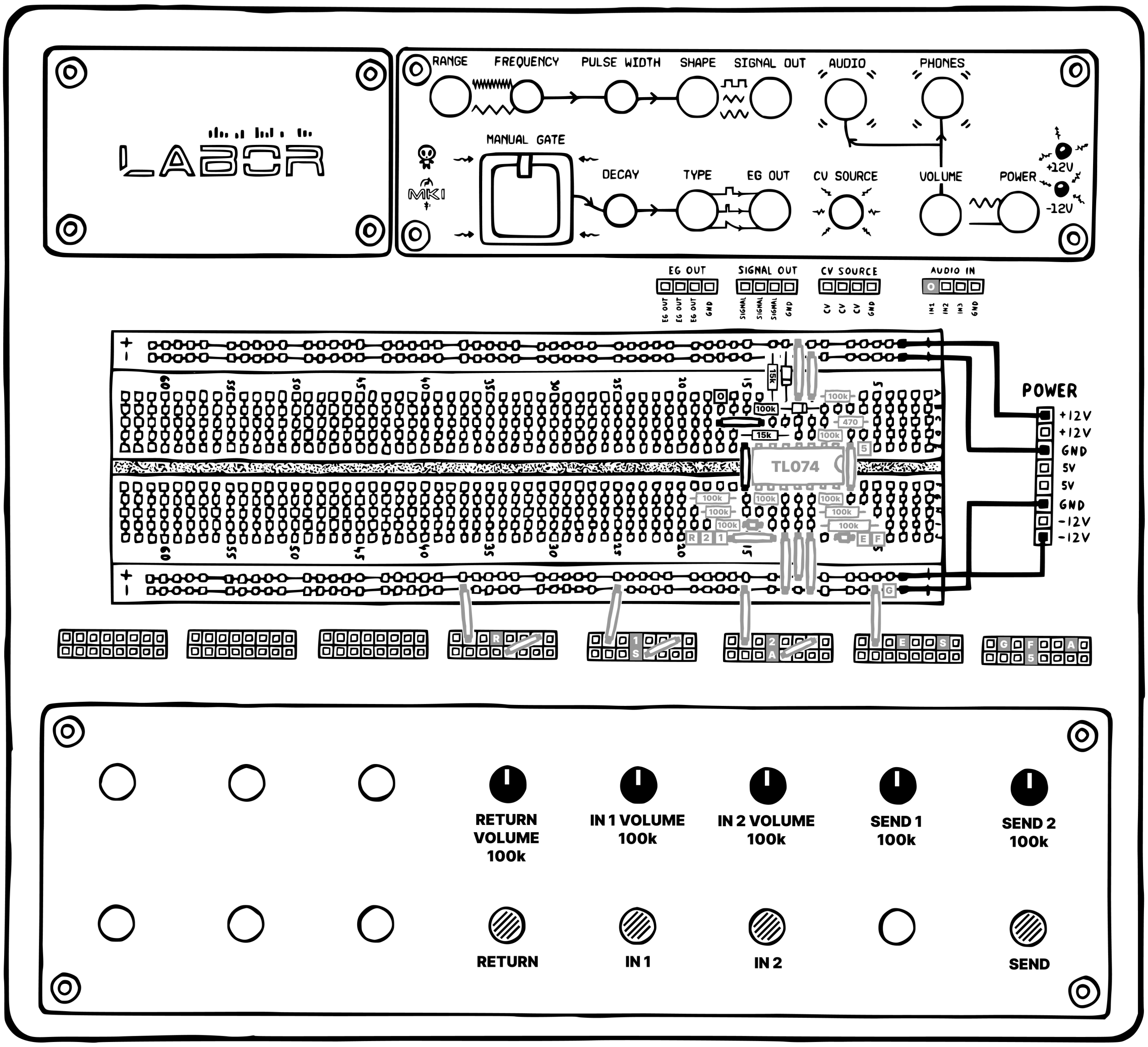


Here's how it works. Let's assume we apply a 2 V peak-to-peak triangle wave to the resistor. The 1N4148 diodes we use only start conducting once the voltage at their anode (input) is roughly 500 mV higher than the voltage at their cathode (output). So, within the range between –500 mV and +500 mV, nothing happens – the output follows the input perfectly. Once the input swings beyond that range, the diodes gradually open up, allowing current to flow down to (or up from) ground. **This prevents the output voltage from going much higher or lower, effectively clipping off the top and bottom of the waveform – and that clipping is what creates audible distortion**.
The louder the input signal, the stronger the clipping and the harsher the sound.[7] But because the signal is now limited to about ±500 mV (around 1 V peak-to-peak), the output volume is a lot lower than the original mix. **To bring it back up, we'll add a non-inverting amplifier stage at the circuit's output**.



[6] You can try this chapter's circuit in a circuit simulator. I've already set it up for you right here: https://tinyurl.com/28hqvvfn – you can change all values by double clicking on components.

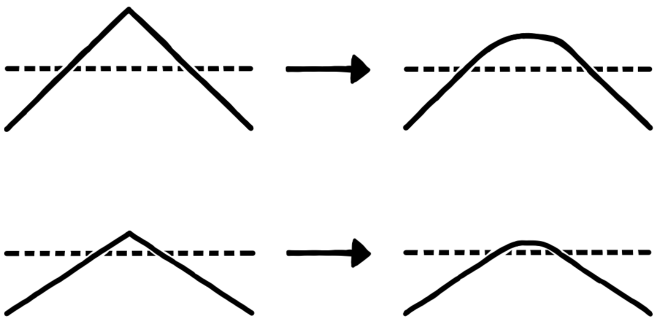
[7] The input resistance also has an effect on the outcome: more resistance means more clipping, but also less output volume.



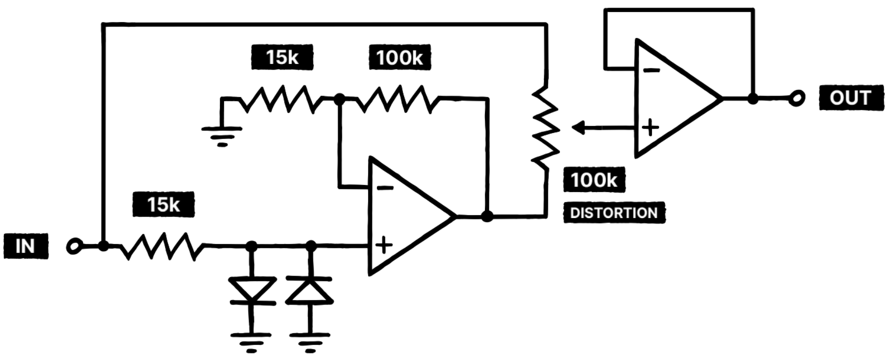
Try sending two drum voices into the mixer and crank their levels up all the way – the output should sound heavily distorted. Great, but it might be a bit too much in some scenarios. So how do we reduce the amount of distortion we apply?

VARIABLE DISTORTION

At first, the obvious solution might be to simply turn the drum mix down before it reaches the distortion circuit. After all, the quieter the signal, the less of it will get clipped – and the cleaner it will sound. That approach does work, but it comes with an issue: it also lowers the output volume. **That's because our circuit uses soft clipping, not hard limiting**. The diodes don't suddenly chop the waveform's top and bottom off at ±500 mV – they let it push slightly past that point, depending on how strong the input signal is.[8]



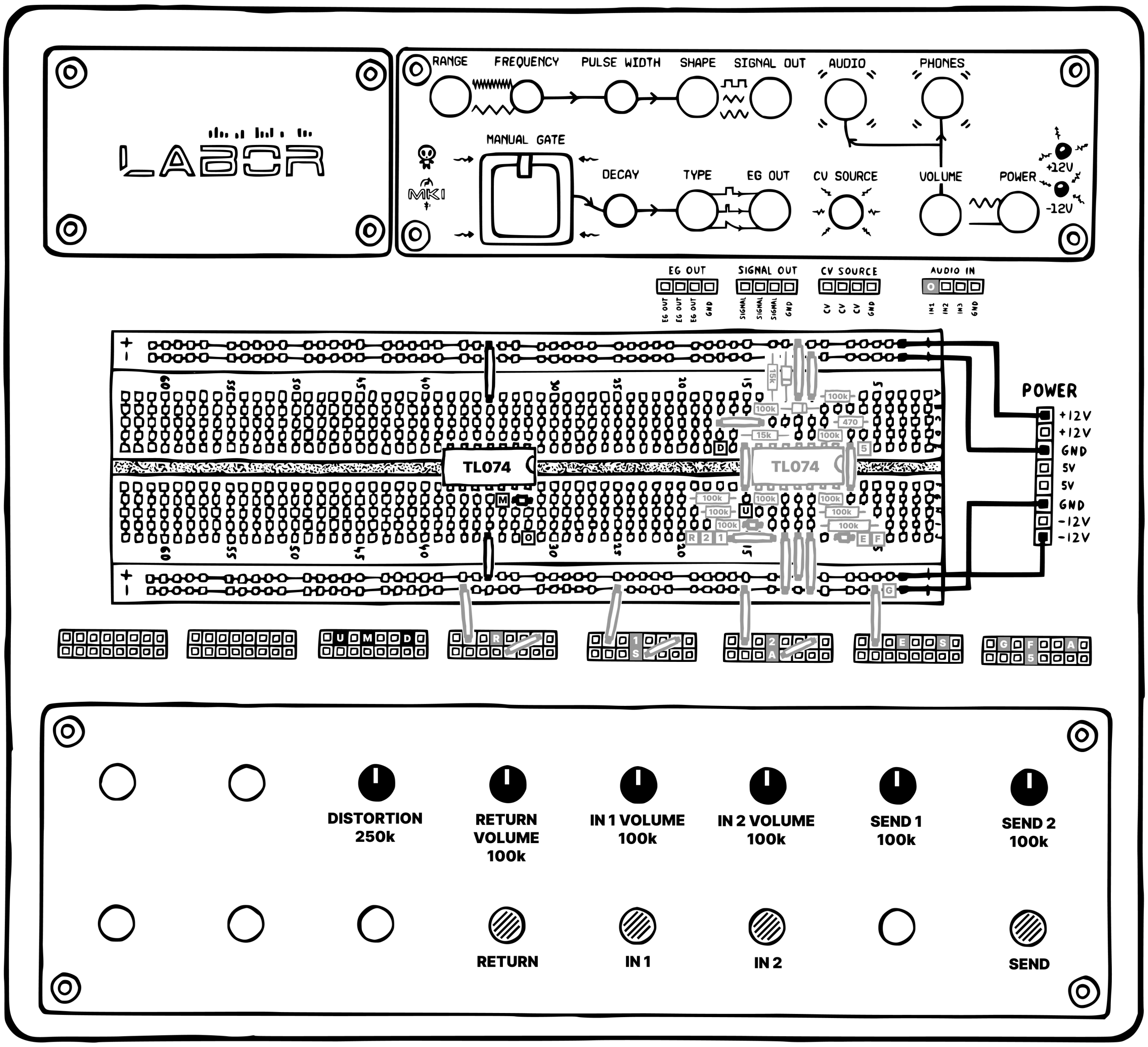
This means that as you lower the input level, the signal doesn't just get less distorted – it also drops in amplitude. So if we simply attenuated the entire mix before sending it into the distortion stage, changing the distortion amount would always change the overall volume, too. What we really want is to control the distortion amount independently of the output volume. **To do that, we'll take a different approach: instead of changing what we feed into the distortion circuit, we'll mix the distorted and the clean versions of the signal after the clipping stage**. Both signals are roughly the same amplitude, so blending between them won't affect the output level – only the texture.



Here's how it works. We take both the dry and the distorted signals and connect them to opposite sides of a 100k potentiometer. **The wiper in the middle gives us a mix of the**

[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/22j3yd6b – you can change all values by double clicking on components.

two: turned one way, it connects entirely to the clean signal; turned the other, to the distorted one. Anywhere in-between, and you get a blend of both. To make sure the circuit can drive whatever comes next, we'll buffer the wiper's output with a non-inverting op amp buffer – this keeps the output impedance low and prevents any loading effects. One small detail: because your Labor kit only comes with five 100k potentiometers, and we already use all of them, we'll have to fall back to a 250k pot for our distortion blend control.

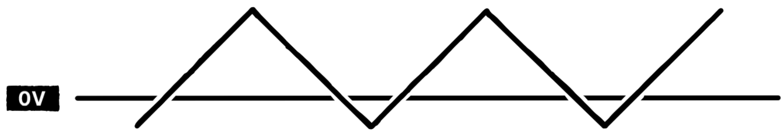


After setting this up, try dialing in different distortion levels. You should hear the character of the distortion change smoothly from subtle warmth to full-on grit – while the overall volume stays the same.

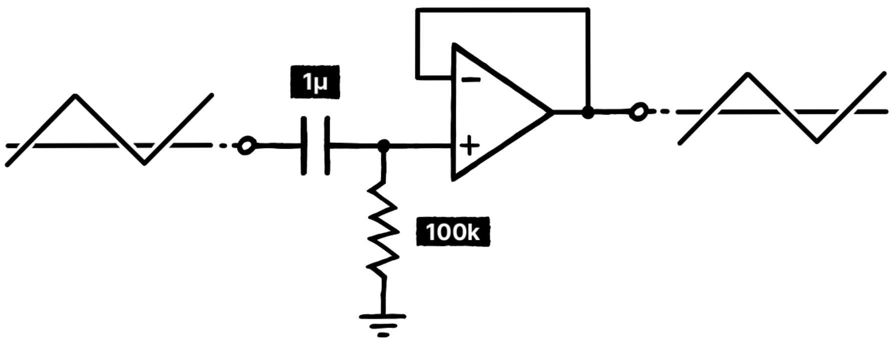
OUTPUT STAGE

While the op amp at the distortion circuit's output can drive small loads like an audio interface just fine, it can't handle headphones without introducing unwanted distortion. That's not ideal – I want the DIY Drums System to be a self-contained instrument. You should be able to plug in your headphones and play without any extra gear. **So let's add an output stage that can handle heavier loads.**

Before we boost anything, though, we should make sure our signal is clean. When combining all of our drum circuits and effects, a DC offset might have crept in – meaning that the waveform might not be centered around 0 V anymore.



That's a problem because feeding a signal with DC offset into a power amplifier wastes energy, stresses the output transistors, reduces headroom, can damage headphone drivers, and causes loud thumps when plugging cables in. To remove it, we'll use AC coupling: a 1 µF capacitor in series with the signal, followed by a 100k resistor to ground.[9]

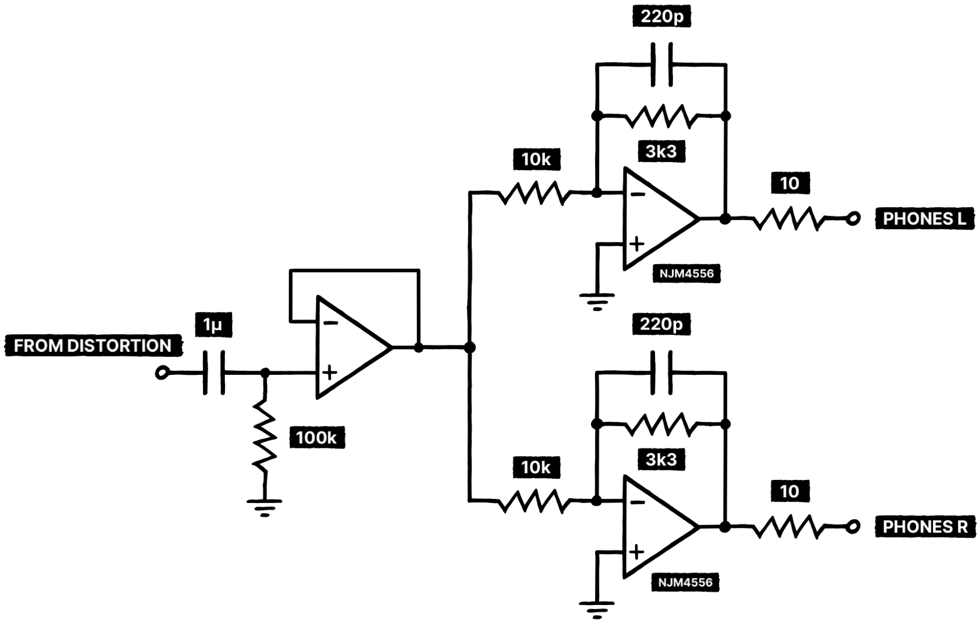


Together, these form a high-pass filter with a cutoff around 1.6 Hz – low enough to remove DC and sub-audio content without affecting the audible range. To prevent loading effects, we buffer this stage with another op amp before processing the signal further.

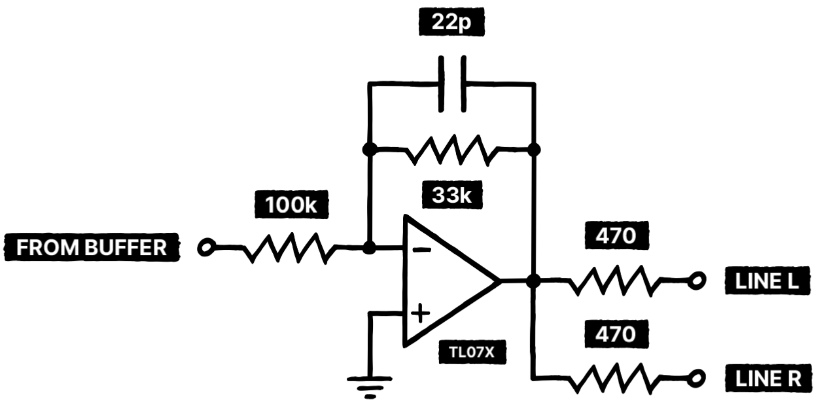
Now for the actual headphone driver. There are two common ways to do this: you can either build a discrete push-pull amplifier using power transistors (like in our original EDU DIY Output Mixer) or use a power op amp that's designed to drive headphones directly. For simplicity, we'll go with the latter. **I've picked the NJM4556, which packs two powerful op amps into a single chip.** Even though our signal is mono, it makes sense to use both amplifiers – one for the left channel and one for the right – so we get sound on

[9] You can try this chapter's circuit in a circuit simulator. I've already set it up for you right here: https://tinyurl.com/28nvt7ky – you can change all values by double clicking on components.

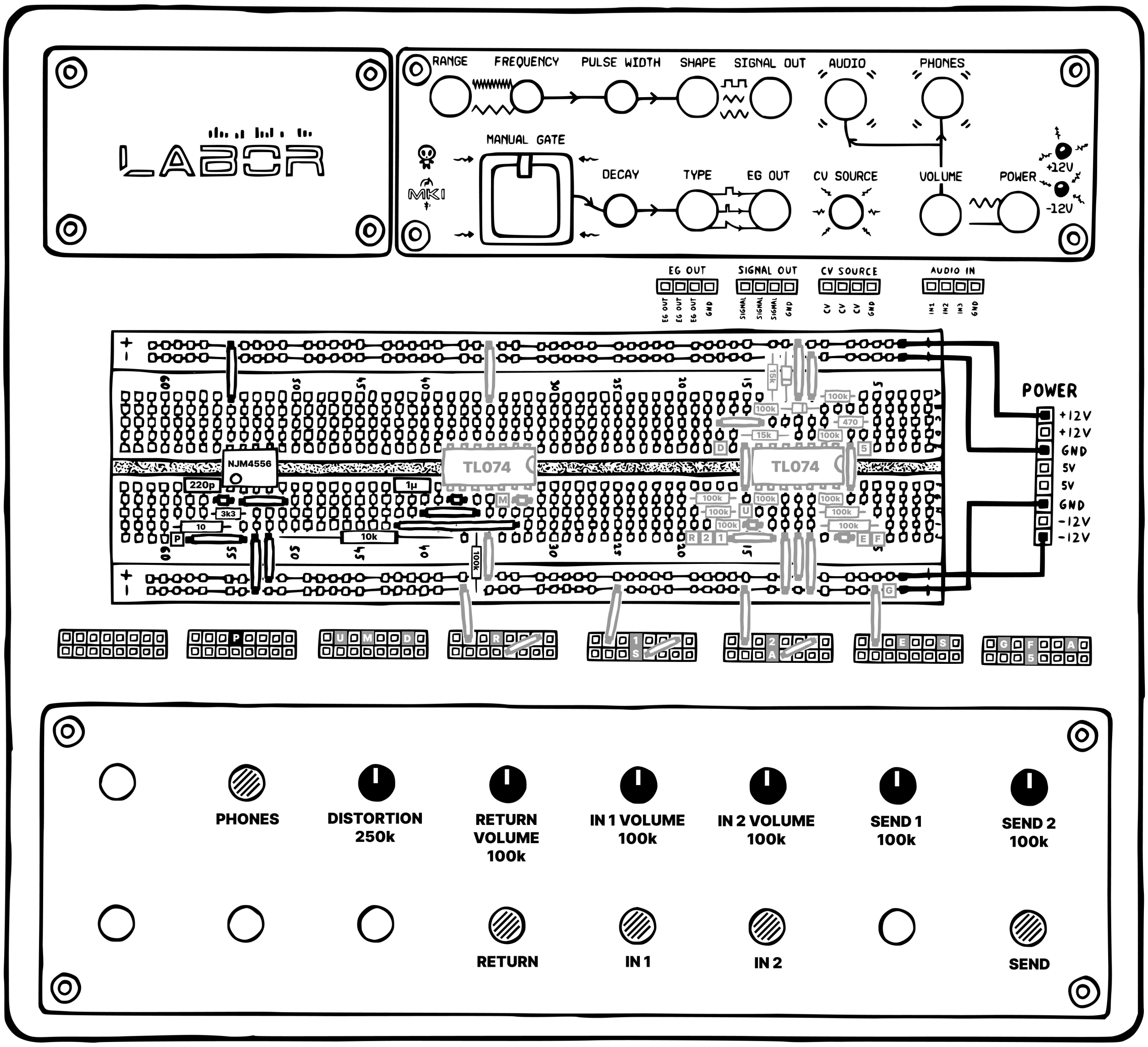
both ears. We'll set both op amps up as inverting amplifiers with a gain of about 0.33, bringing our ±5 V eurorack-level signal (10 V peak-to-peak) down to roughly professional line level at 3.5 V peak-to-peak.



Notice how we're using 10k and 3k3 resistors in the feedback loop instead of higher values (e.g. 100k and 33k)? **That's because power op amps like the NJM4556 require low-impedance feedback networks to remain stable and prevent high frequency ringing**. To further improve stability, we'll add small 220 pF capacitors in parallel with the feedback resistors – creating very gentle low-pass filters that dampen unwanted oscillations without affecting the audio itself. Finally, we'll add 10 ohm resistors in series with each output for the finishing touch. (These are smaller than the 470 ohm resistor we used before, because we need to be able to deliver enough current to drive our load.)
Since you might want to connect the DIY Drums System to an audio interface or external mixer while simultaneously listening through headphones, we'll also include a line output. For this, we'll use a regular op amp configured similarly, but with a 100k/33k feedback combination to reduce current consumption.



A small 22 pF capacitor across the feedback resistor keeps it stable, and both sides of the stereo output jack are driven from this single op amp through 470 ohm resistors.
To avoid cluttering the breadboard unnecessarily, we'll only set up one headphone output stage – you've already seen firsthand how a regular op amp drives low impedance loads.



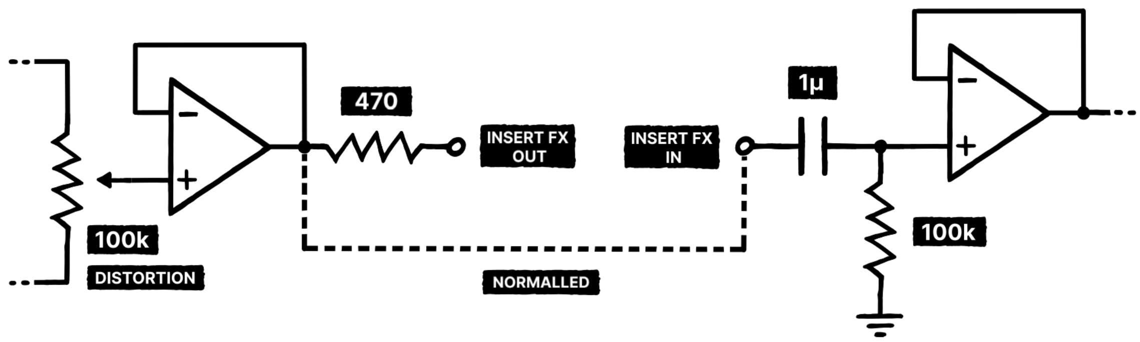
Try connecting a pair of power-hungry headphones to the new output (bottom left) – you should get a clean, solid signal at proper listening level. With the output stage complete, the signal path is now done – and we can turn our attention to one final feature: insert effects.

INSERT EFFECTS

Earlier, we built a send/return loop so we could process individual voices separately – maybe a touch of delay on the snare and hi-hat. But what if we actually *do* want to affect all voices equally? For example, what if we wanted to compress the entire mix? The most straightforward way to do that would be to route the line output into the effect and take its output back into your audio interface.

That works fine if you're recording or monitoring through external gear – but not if you want to plug in headphones directly. Most effects modules, including our compressor, aren't designed to drive low-impedance loads like headphones. So if you connected them as shown above, you'd lose the ability to monitor directly from the DIY Drums System.

To fix this, we'll borrow a trick from studio mixing consoles: the insert point. **An insert point lets you route the full mix out of the circuit, through an external effect, and back in again – without breaking the signal path when nothing is plugged in.** We'll do this by exposing two new sockets: INSERT FX OUT (post-distortion) and INSERT FX IN (pre-output stage).



Here's how it works. The mixed and distorted signal is sent to the INSERT FX OUT jack, where it can be patched into an effect like the compressor or BBD. The processed signal then returns through the INSERT FX IN, which feeds directly into the high-pass filter at the start of our output stage.

To make sure the circuit still works when no effect is connected, we'll normal the INSERT FX IN to the INSERT FX OUT. **This means the two sockets are internally linked – the signal passes straight through as if the jacks weren't there.** As soon as you plug in a cable, that connection is broken, and the signal is rerouted through your external effect.

Since we are already using all output sockets from your Labor kit, we can't try this concept in practice without removing other functionality. If you want, try repurposing the send/return sockets and set up the insert point on your own!

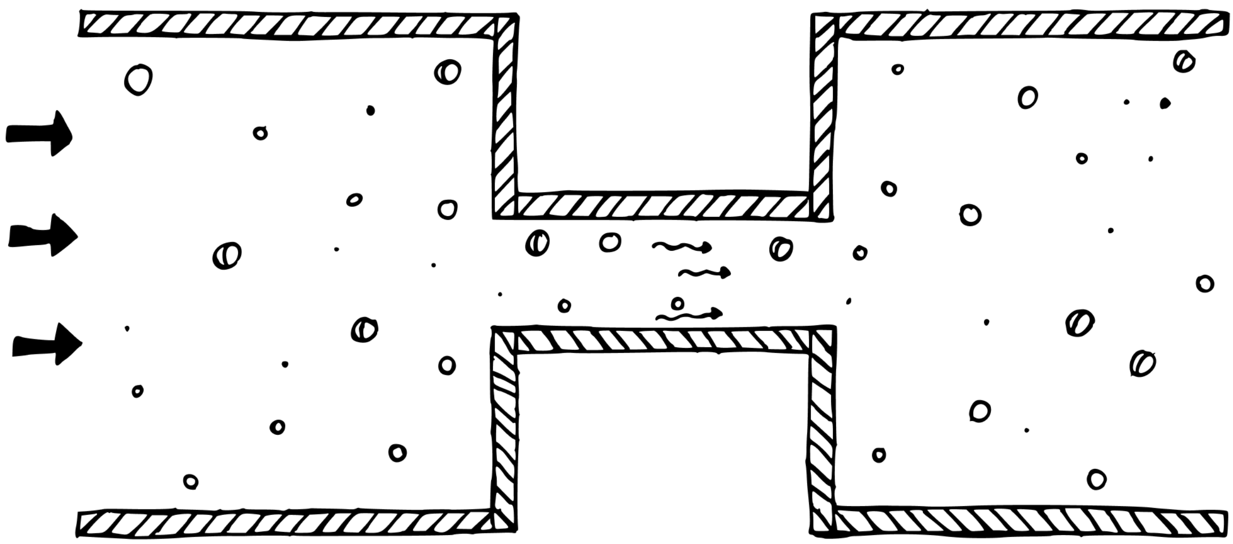
With that, our Output Mixer is complete. Once you're done experimenting, grab the panel and PCB from the kit, heat up your soldering iron, and start building. You'll find detailed instructions for populating the board and soldering in the appendix.

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

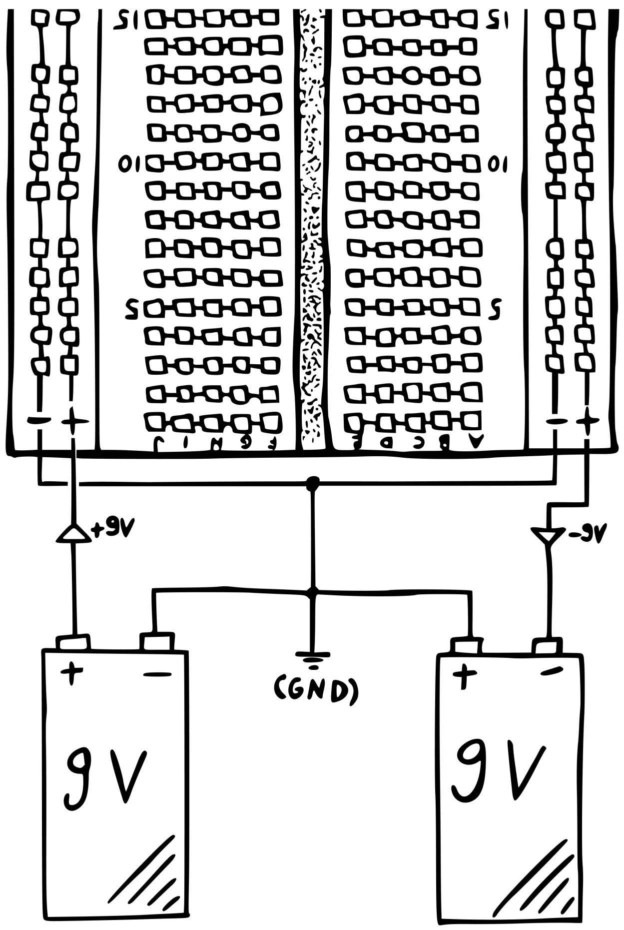
There are three main properties we're interested in when talking about electronic circuits: resistance, voltage and current. To make these less abstract, we can use a common beginner's metaphor and compare the flow of electrons to the flow of water through a pipe.



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.[10] **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.

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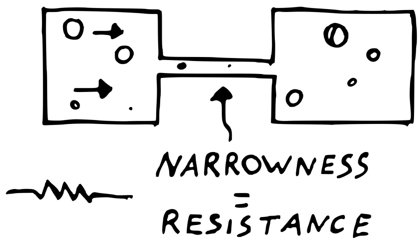
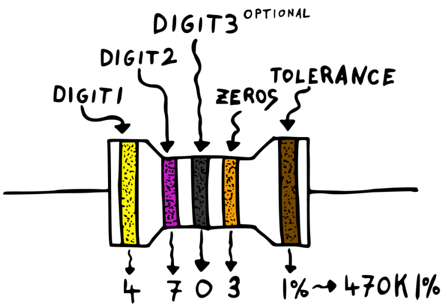
[10] If you're struggling with setting this up, you can watch me do it here: https://youtu.be/XpMZoR3fgd0?t=742

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.



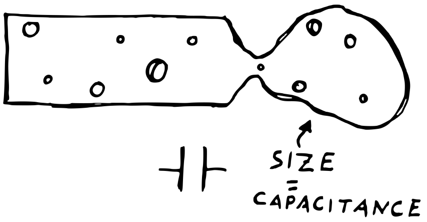


Resistors have two distinctive properties: linearity and symmetry. Linearity, in this context, means that for a doubling in voltage, the current flowing will double as well. Symmetry means that the direction of flow doesn't matter – resistors work the same either way.
On a real-life resistor, you'll notice that its value is not printed on the outside – like it is with other components. Instead, it is indicated by colored stripes[11] – 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.

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

CAPACITORS

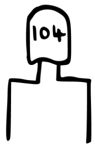
A capacitor is a bit like a balloon that you can attach to the open end of a pipe. If there's some pressure in the pipe, the balloon will fill up with water until the pressure equalizes. (Since the balloon needs some space to expand into, both of the capacitor's legs need to be connected to points in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

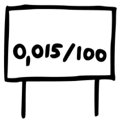


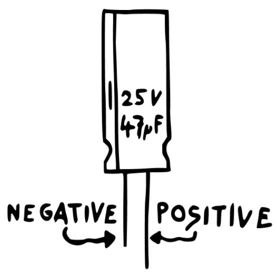
Then, should the pressure in the pipe drop, the balloon releases the water it stored into the pipe. The maximum size of the balloon is determined by the capacitor's capacitance, which we measure in farad (F). There are quite a few different types of capacitors: electrolytic, foil, ceramic, tantalum etc. 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.[12] 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.[13]

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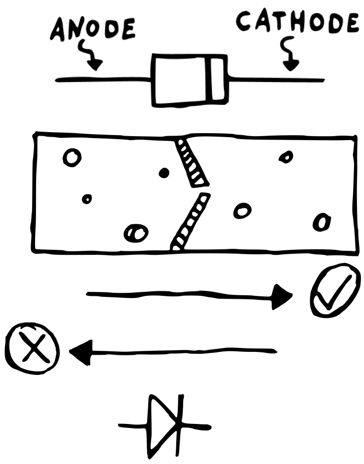
[12] For a detailed breakdown, look up ceramic capacitor value code. There are also calculation tools available.

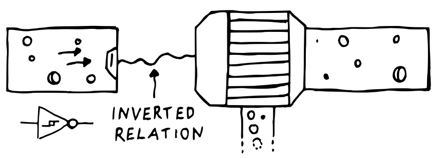
[13] If yours do encode their values, same idea applies here – look up film capacitor value code.

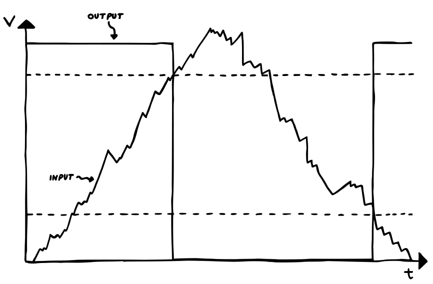
DIODES



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.

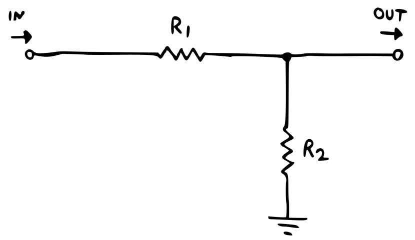
SCHMITT TRIGGER INVERTERS



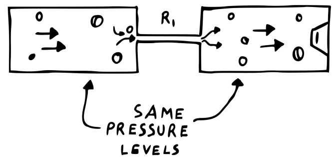


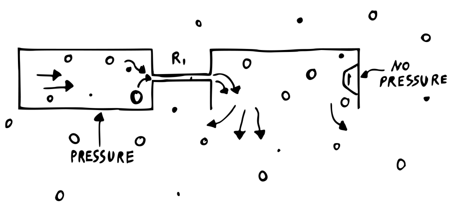
You can think of a Schmitt trigger inverter as two separate things. On the left, there's a sensor that measures the pressure inside an attached pipe. On the right, there is a water pump. This pump's operation is controlled by the sensor. Whenever the pressure probed by this sensor is below a certain 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.

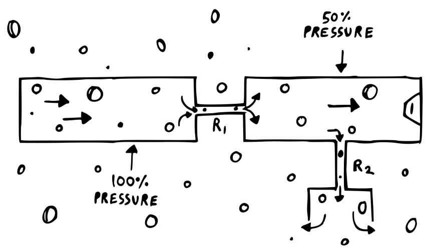
VOLTAGE DIVIDERS



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**.

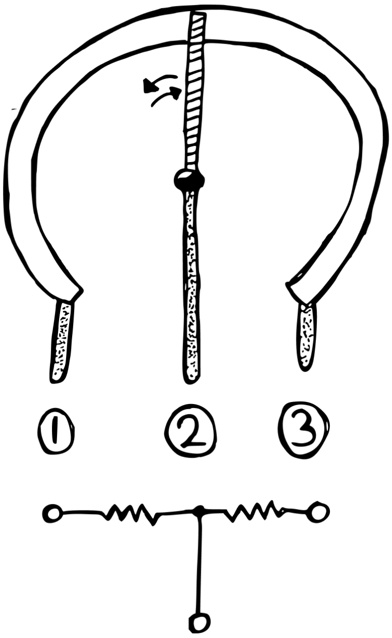






POTENTIOMETERS

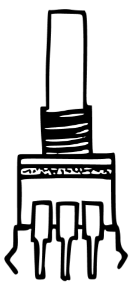
Potentiometers can be used as variable resistors that you control by turning a knob. But, and that's the handy part, they can also be set up as variable voltage dividers. To see how that works, let's imagine we open one up.

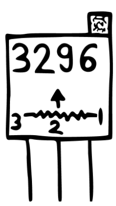


Inside, we would find two things: a round track of resistive material with connectors on both ends plus what's called a wiper. This wiper makes contact with the track and also has a connector. It can be moved to any position on the track. Now, the resistance value between the two track connectors is 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.





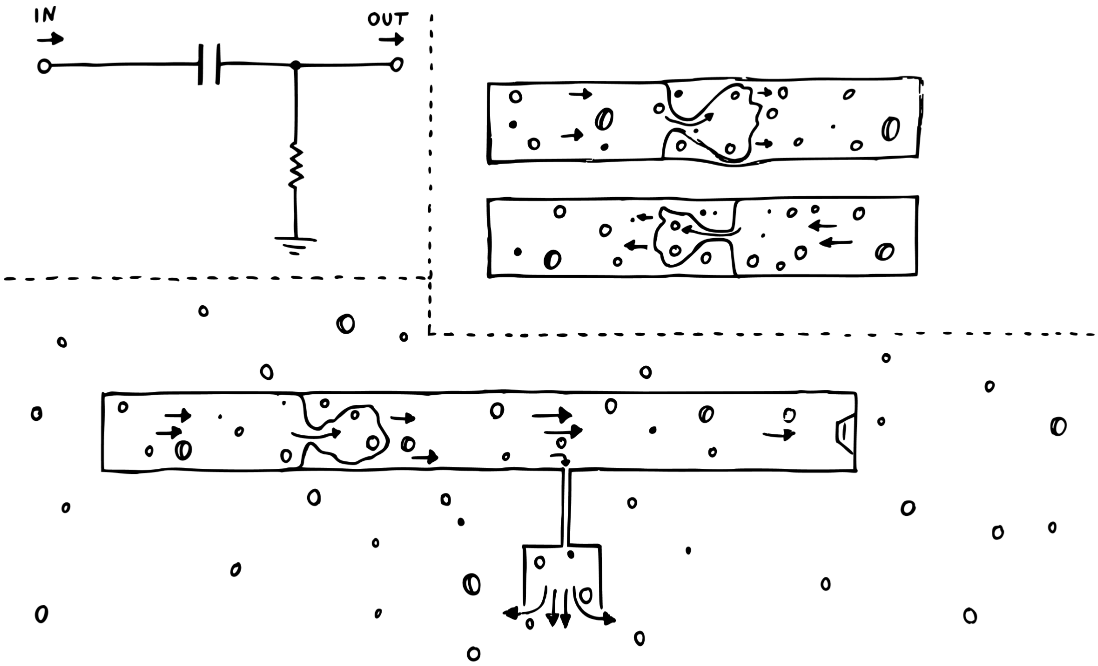
In these kits, you will encounter different types of potentiometers. First, there's the regular, full-size variant with a long shaft on top. These are used to implement user-facing controls on the module's panel and they usually – but not always – indicate their value directly on their casing. Sometimes, they'll use a similar encoding strategy as capacitors, though.[14]

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.

[14] 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**.

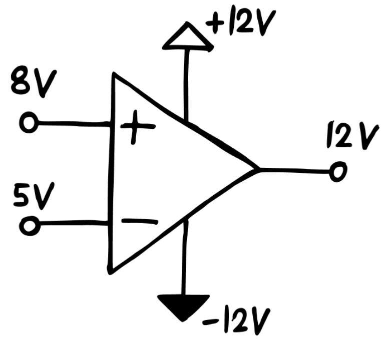


Next, we'll bring in a resistor after the coupling point, going straight to ground. **This acts like a kind of equalizing valve**. Now imagine we apply a steady 5 V from one side. Then on the other side, we'll read 0 V after a short amount of time. Why? Because we're pushing water into the balloon with 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?



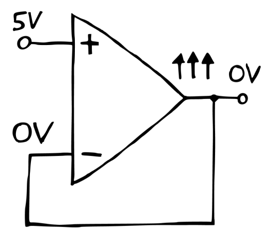
The key here is that op amps are essentially differential amplifiers. This means that they only amplify the difference between their two inputs – not each of them individually. If that sounds confusing, let's check out a quick example. So we'll imagine that one sensor – called the non-inverting input – 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.

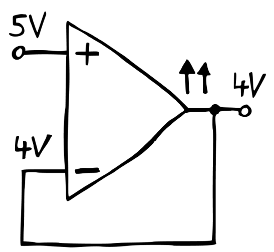
The handy thing though about op amp outputs is that they draw their power directly from the power source. This means that they can supply lots of current while keeping the voltage stable. **That's why we say an op amp has a very low output impedance**.

OP AMP BUFFERS/AMPLIFIERS

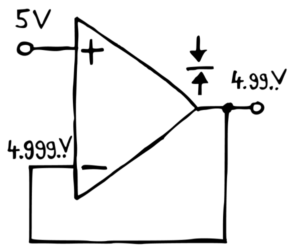
Buffering, in the world of electronics, means that we provide a perfect copy of a voltage without interfering with that voltage in the process. With an op amp-based buffer, the buffering process itself works like this. We use the non-inverting input to probe a voltage, while the inverting input connects 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.



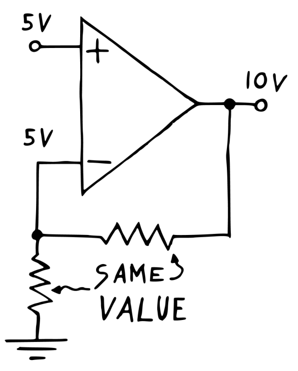
Before the op amp starts processing the voltages at its inputs, the output will be switched off. This means that **output and inverting input sit at 0 V at first**. So then, the op amp will subtract 0 from 5 and multiply the result by its gain. Finally, it will try and increase its output voltage to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



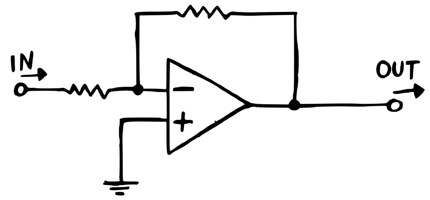
But as it's pushing up that output voltage, the **voltage at the inverting input will be raised simultaneously**. So the difference between the two inputs is shrinking down. Initially, this doesn't matter much because the gain is so large. As the voltage at the inverting input gets closer to 5 V though,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

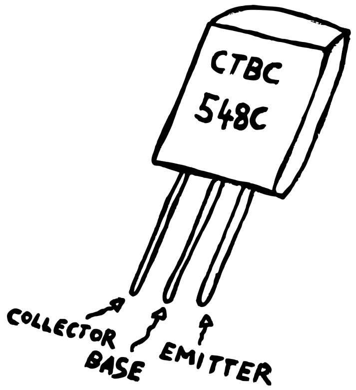
How does this work? Well, let's assume that we're applying a steady voltage of 5 V on the left. Then, as we already know, the op amp will subtract the inverting input's voltage from the non-inverting input's voltage, leaving us with a result of –5 V. Multiply that by the huge internal gain, and the op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

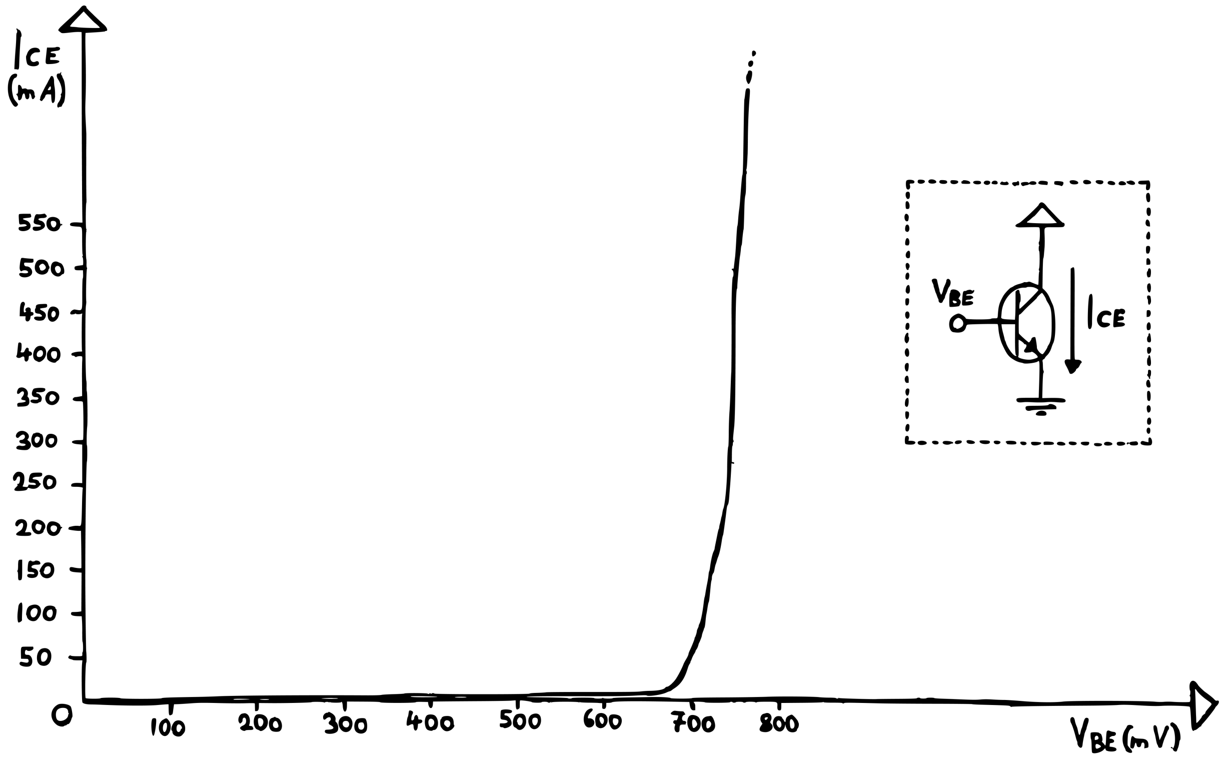
And that means that the voltage at the inverting input will be lowered to about 0 V, allowing our op-amp to settle on the current output voltage level. So while we read 5 V on the left, we'll now read a stable –5 V at the op amp's output. Congrats – we've built an inverting buffer! **If we want to turn 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.**[15] 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[16] 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.



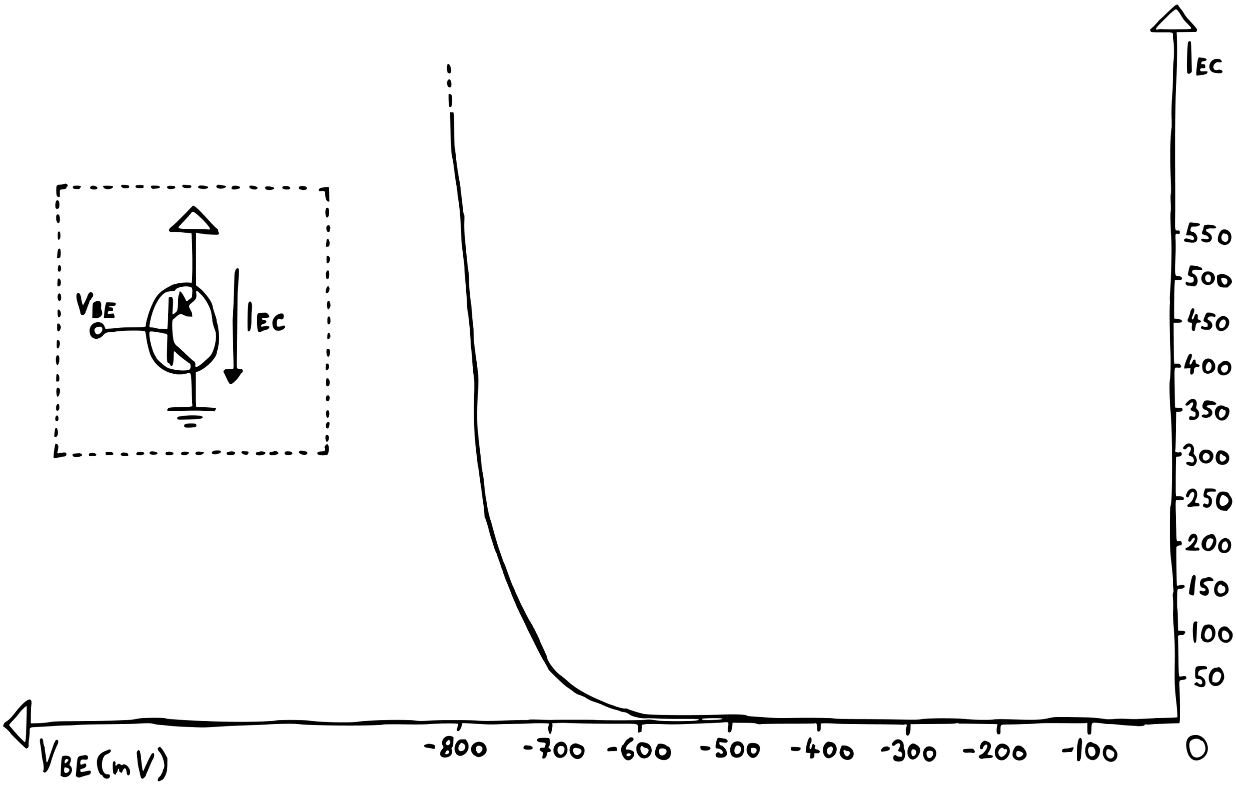
First, the relation between base voltage and collector current is exponential. Second, unlike a resistor, a BJT is not symmetrical – so we can't really reverse the direction of the

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

[16] 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**[17]. 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!

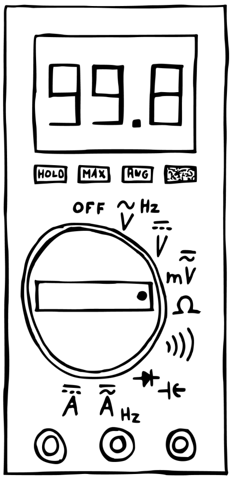


[17] Again, the voltage is measured between base and emitter.

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.

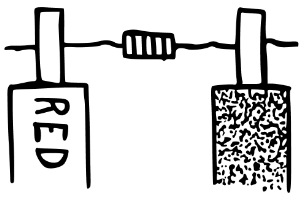
MULTIMETERS

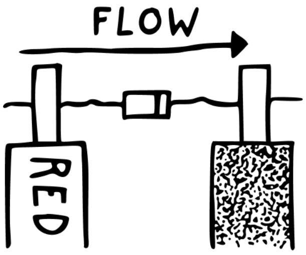


Multimeters come in different shapes and sizes, but the most common type is probably the hand-held, battery powered variant. It can measure a bunch of different things: voltage, current, resistance, continuity. Some have additional capabilities, allowing you to check capacitance, oscillation frequency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



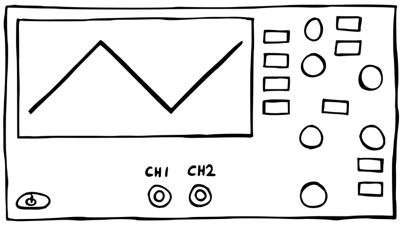


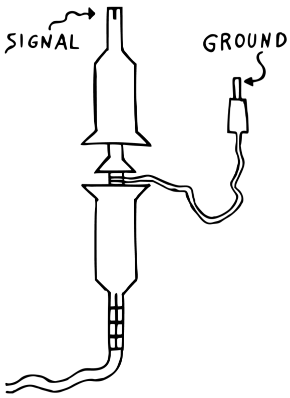
In some cases, it doesn't matter which probe you connect to which component leg or point in your circuit. This is true for testing resistors, non-polarized capacitors (foil/film, ceramic, teflon, glass etc.), continuity[18] or AC voltage.

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.

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

OSCILLOSCOPES



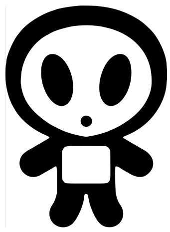


While multimeters are fairly cheap and compact, oscilloscopes are usually somewhat pricey and bulky. **If you're willing to make the investment, they are a huge help with the troubleshooting process, though**. Using one is, again, surprisingly straightforward – if you manage to work your way through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

Before we start building the module, let's take a look at the complete **mki x es.edu Output Mixer 2** 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**.

XS are inputs and outputs of various signals - 3.5mm jack sockets – 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).

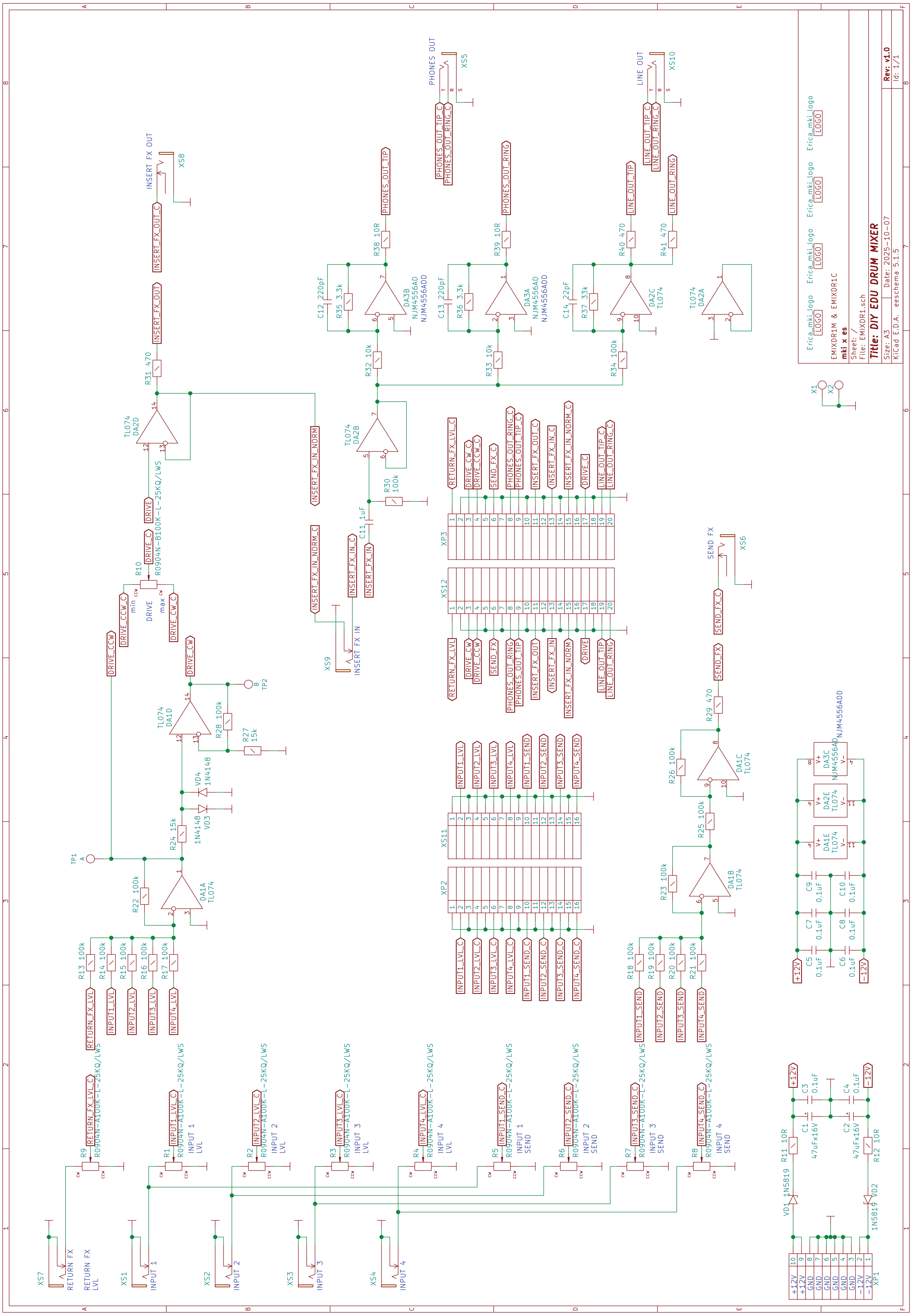
XP1 is a standard eurorack **power connector**. It's a 2x5 male pin header with a key (the black plastic shroud around the pins) to prevent accidental reverse polarity power supply connection. This is necessary because connecting the power incorrectly will permanently damage the module.

VD1 and **VD2** are **schottky diodes** that double-secure the reverse polarity power supply protection. Diodes pass current only in one direction. Because the anode of VD4 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 VD4, it closes, and no current passes through. The same goes for VD5, 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** (**R11** and **R12**) on the + and – 12 V rails, with **decoupling** (or **bypass-**) capacitors **C3**, **C4**. 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 (C1 and C2)** 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.

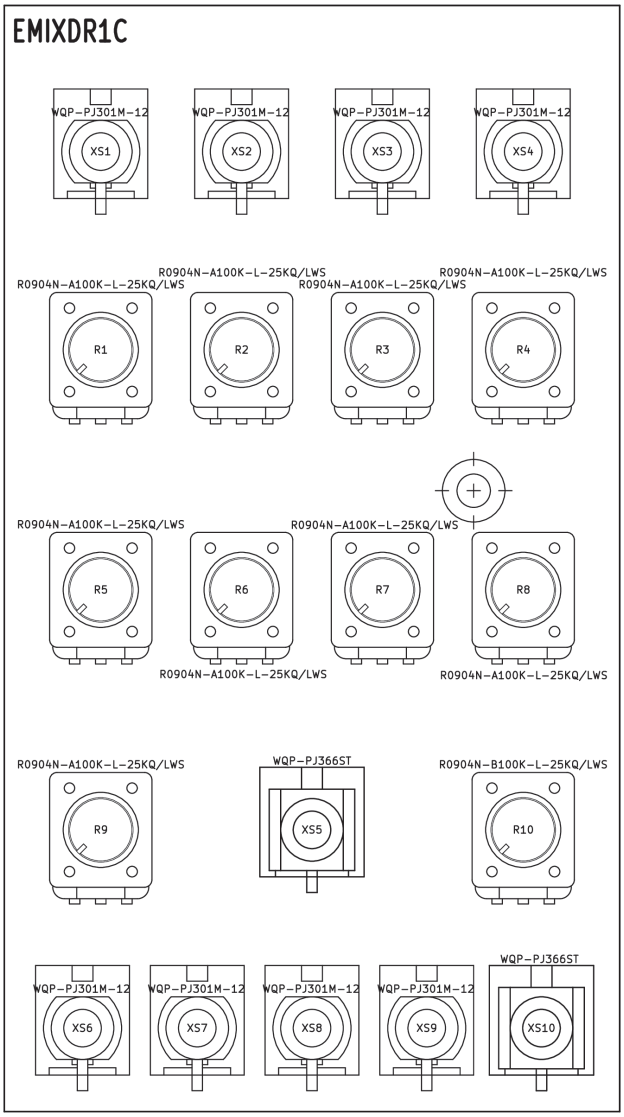
Another advantage of 10 Ohm resistors is that they will act like **slow “fuses”** in case there’s an accidental short circuit somewhere on the PCB, or an integrated circuit (IC) is inserted backwards into a DIP socket. The resistor will get hot, begin smoking and finally break the connection. Even though 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**.

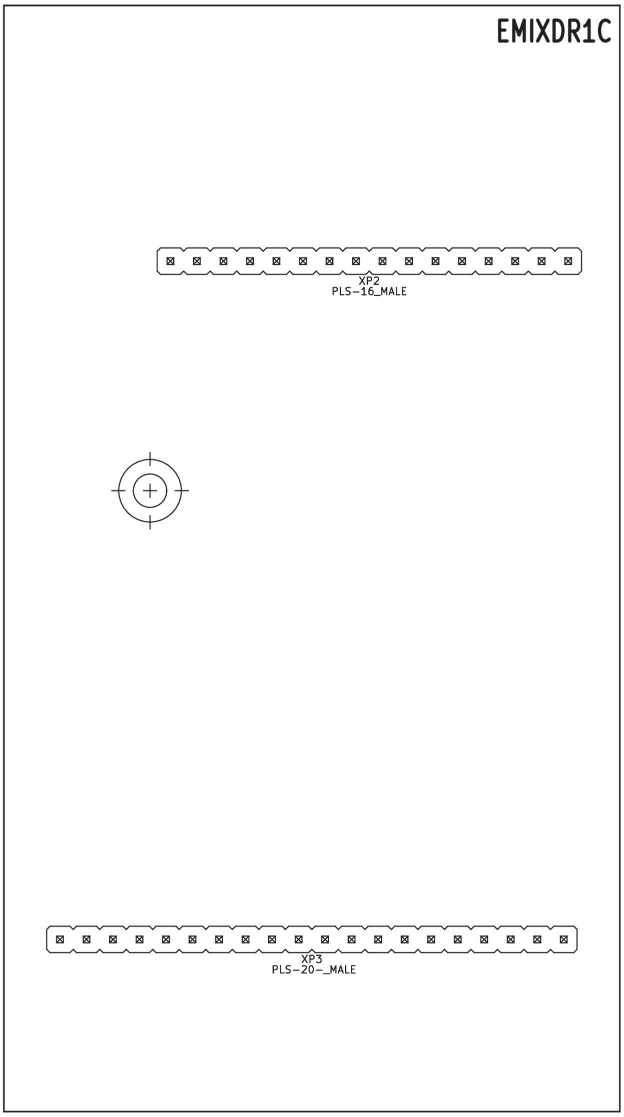
Capacitors **C5 – C9** are additional decoupling capacitors. If you inspect the PCB, you’ll see that these are placed as close to the power supply pins of the ICs as possible. For well-designed, larger PCBs you will find decoupling capacitors next to each IC. Like the others, their job is to simply compensate 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.

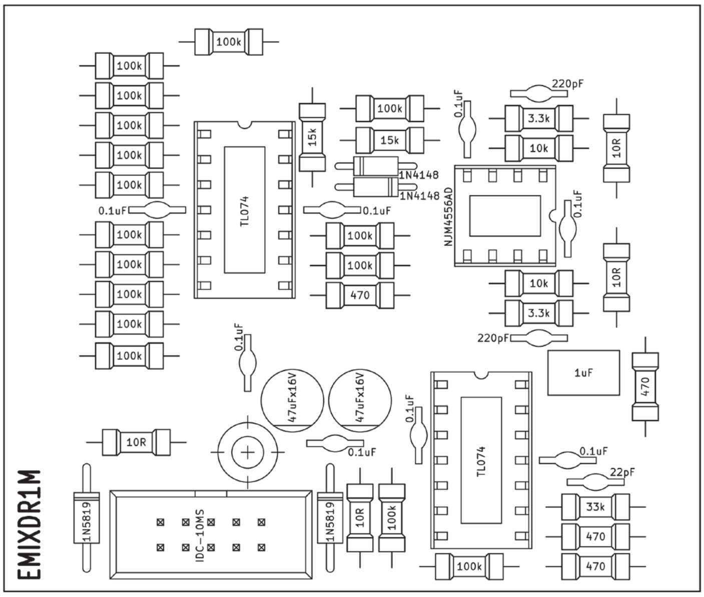

Title: DIY EDU DRUM MIXER
Rev: v1.0
EMIXDR1M & EMIXDR1C
mki x es
Sheet: /
File: EMIXDR1.sch
Size: A3
Date: 2025-10-07
KiCad E.D.A. eeschema 5.1.5
Id: 1/1

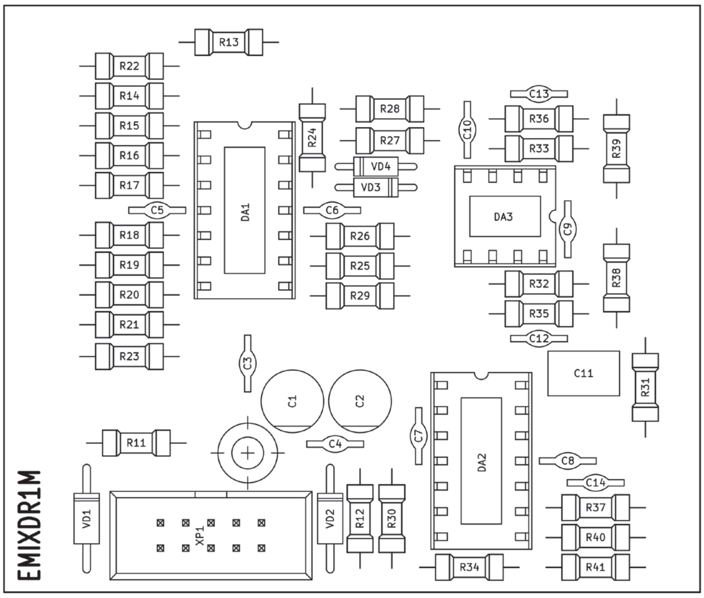
Before you start soldering, we highly recommend printing out the part placement diagrams with designators and values and follow step-by step instructions below, because this will help you to avoid mistakes in the build process. The user interface of the module is very densely populated with jack sockets and potentiometers; therefore we had to split the build to two PCBs – Control PCB with all components that are visible on the front panel and the Main PCB with all other components. Both PCBs are connected via long one row connectors, and the schematics indicate exact connector pinout.

NB! During assembly, please, follow **the silkscreen on the PCB and part placement diagram!** Photos of the assembly process below may be slightly different from the production version of the module.






EMIXDR1M
100k
TL074
15k
1N4148
NJM4556AD
0.1uF
220pF
3.3k
10k
10R
470
1uF
47uFx16V
22pF
33k
1N5819
IDC-10MS


EMIXDR1M
R13
R22
R14
R15
R16
R17
C5
R18
R19
R20
R21
R23
DA1
R24
R28
R27
VD4
VD3
C6
R26
R25
R29
C10
C13
R36
R33
R39
DA3
C9
R32
R35
R38
C12
C11
R31
C3
C1
C2
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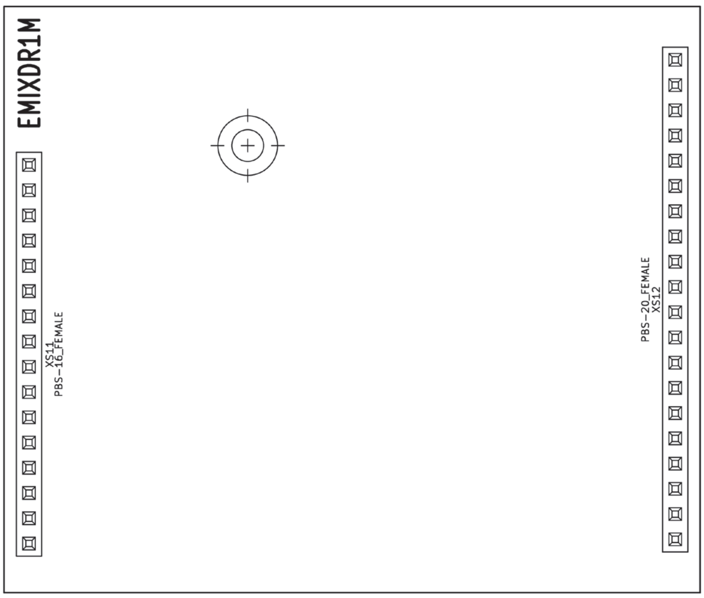
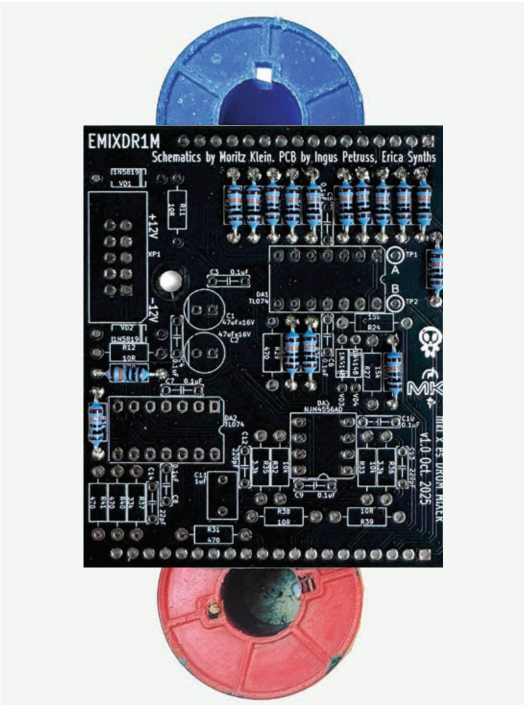
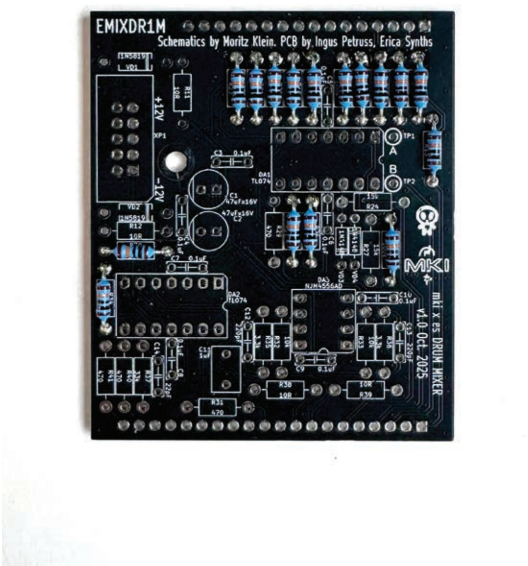
DA2
C8
C14
R37
R40
R41
VD1
XP1
VD2
R12
R30
R34


EMIXDR1M
XS11
PBS-16_FEMALE
PBS-20_FEMALE
XS12

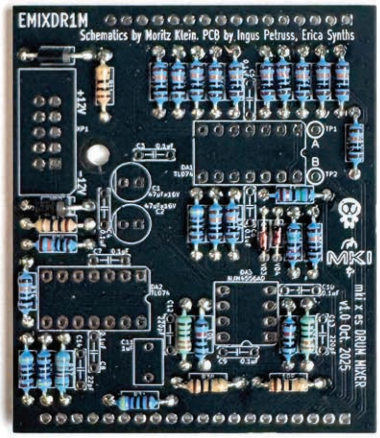


Let’s start assembly with the Main PCB! Place the **Output Mixer 2 PCB in a PCB holder for soldering** or simply on top of some spacers (I use two empty solder wire coils here).

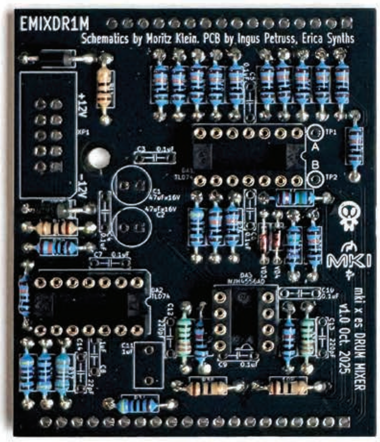


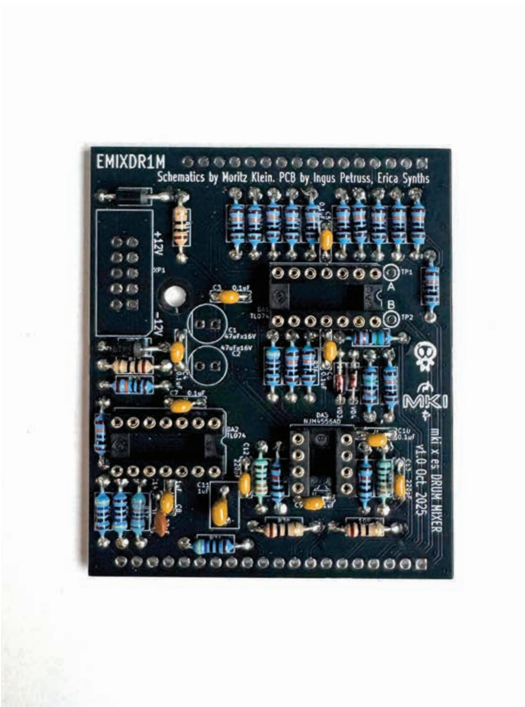
I usually start populating PCBs with lower, horizontally placed components, resistors specifically. There are lot of 100k resistors, so let’s start with those! Bend the resistor leads and insert them in the relevant places according to the part placement diagram above. Flip the PCB over and solder all components. Then, use pliers to cut off the excess leads.

Now, proceed with other **resistors, 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.



Next, insert the first **DIP socket**, hold it in place and solder one of the pins. Continue with other DIP sockets. 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 socket placement with the one on the picture below.

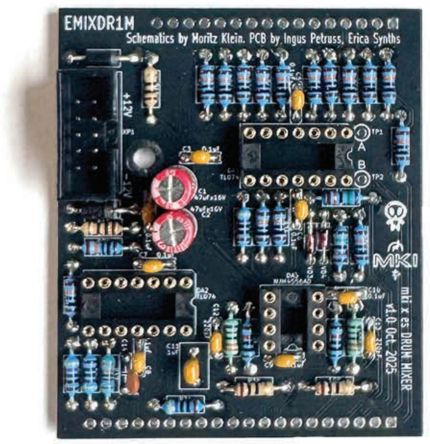




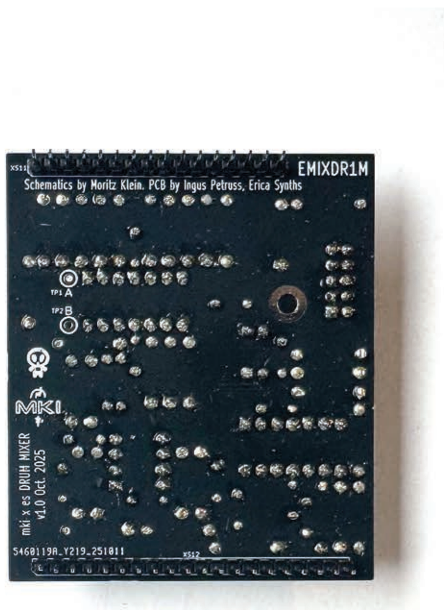
Then proceed with the ceramic capacitors. Start with soldering 0,1uF capacitors - place the PCB in your PCB holder or on spacers, insert the capacitors and solder them like you did with the resistors & diodes before then proceed with other ceramic and film capacitors. When completed, your PCB should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

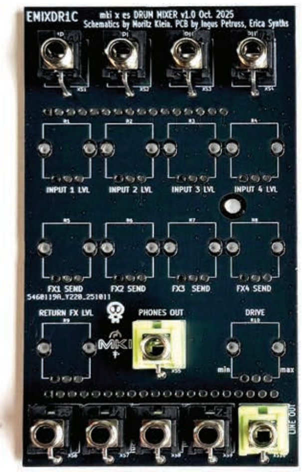
Now, **insert & solder the electrolytic capacitors**. Electrolytic capacitors are bipolar, and you need to mind their orientation. The positive lead of each electrolytic capacitor is longer, and there is a minus stripe on the side of the capacitor’s body to indicate the negative lead. On our PCBs, the 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.

Then populate the **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 inwards the PCB. Now your PCB should look like this :

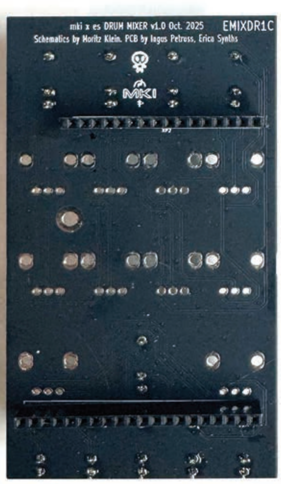


Now, turn the PCB around and inspect your solder joints. **Make sure all components are soldered properly and there are no cold solder joints or accidental shorts.** Clean the PCB to remove extra flux, if necessary. Then insert the male connectors and solder those. The completed Main PCB has to look like this:



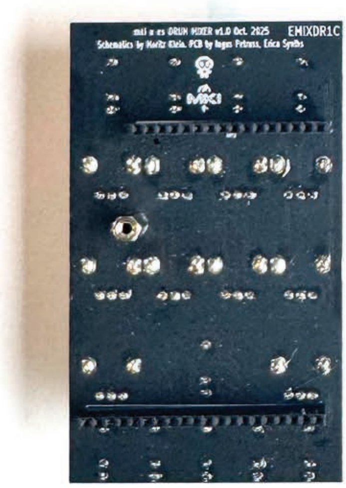


Next, proceed with the Controls PCB. Insert all **jack sockets** and solder them. Please, note that two jack sockets are **stereo**. They are **green**, and their pinout is identical to mono sockets, so make sure you solder them in correct places.



Turn the Connectors PCB around and solder the **female connectors!**

Now, insert the potentiometers in relevant places, but **do not solder** them, yet! There are 9 A100k potentiometers, and one B100k potentiometer (DRIVE). Make sure, you don't mix the values.



Next, fit the front panel, screw some nuts on the 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 and then remove the front panel. Then use the M3 screw to fix the 11mm spacer – it will fix both PCBs together and secure integrity of the module.

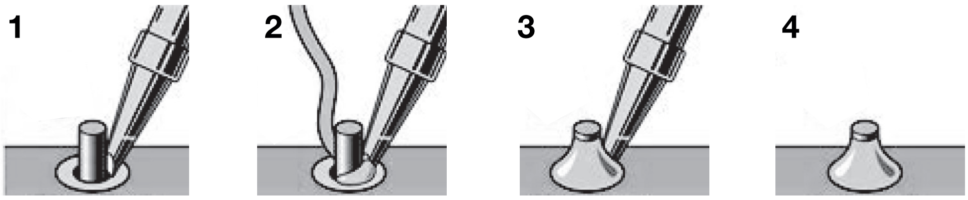
Fit the front panel and fix it with the nuts on the jack sockets. Finally, connect the Main PCB, fix it with another M3 screw and **insert the ICs into their respective DIP sockets**. Mind the orientation of the ICs – match the notch on each IC with the one on its socket.

Congratulations!
You have completed the assembly of the mki x es.edu Output Mixer 2 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 does not need calibration and, if your assembly is correct, it has to work straight away. Connect some sound sources to the input and FX module (mki x es DIY.EDU BBD is good choice) to FX Send/Return jack sockets and give it a try! **Enjoy!**

SOLDERING APPENDIX

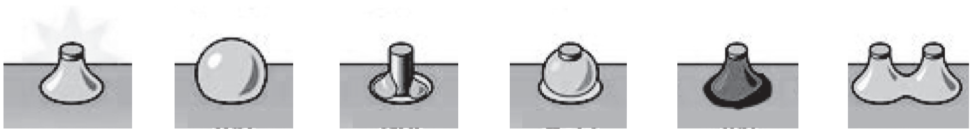
If you've never soldered before – or if your skills have become rusty – it's probably wise to check out some **THT** (through-hole technology) **soldering tutorials on YouTube**. The main thing you have to remember while soldering is that melted solder will flow towards higher temperature areas. So you 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:



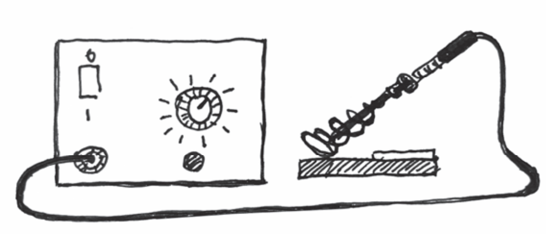
Heat part and pad 2 - 3 sec | Add solder | Continue heating 1 -2 sec. | Let cool

After you have completed soldering, inspect the solder joint:

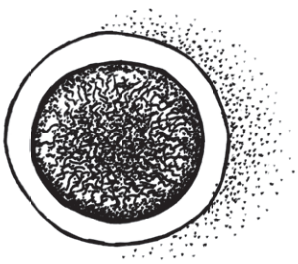


Perfect | Too much solder | Not enough solder | Cold joint | Too much heat | Short

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:

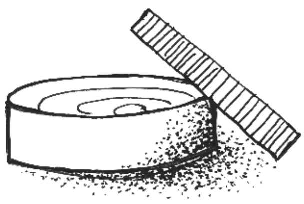


A decent soldering station. Top-of-the-line soldering stations (brands like Weller) will cost 200€ and above, but cheaper alternatives around 50€ are often good enough. Make sure your soldering station of choice comes with multiple differently-sized soldering iron tips. The most useful ones for DIY electronics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



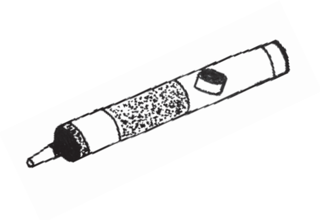
When heated up, the tips of soldering irons tend to oxidize. As a result, solder won't stick to them, so you'll need to clean your tip frequently. Most soldering stations come with a **damp sponge for cleaning the iron tips** – but there are also professional solder tip cleaners with **golden curls** (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.

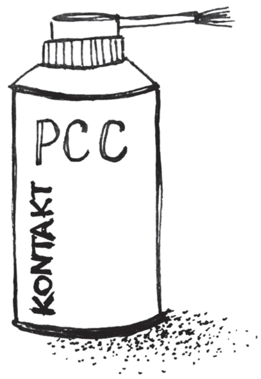


Some **soldering flux** paste or pen will be useful as well.

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.

"We keep moving forward, opening new doors, and doing new things, because we're curious and curiosity keeps leading us down new paths."

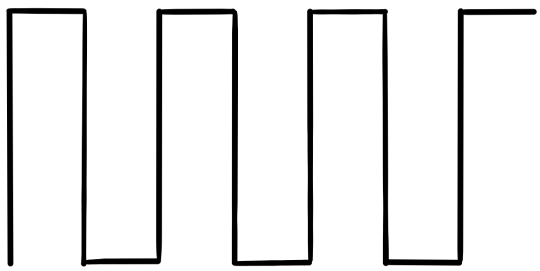
– Walt Disney

TROUBLESHOOTING

Your assembled module might not immediately work as expected. This could be due to cold solder joints, faulty, missing, or misplaced components – or you might've skipped over some solder joints entirely. **To help you diagnose issues, we've exposed multiple test points on the PCB**. Checking these with your oscilloscope can give you an indication as to what went wrong with your build.

TP1 (A)

This test point exposes the the **main mixer's output**. Connect your oscilloscope probe here, and then send a simple signal (e.g. a square wave) into one of the 4 inputs. Turn up the input channel's volume, and you should see the input signal on your scope. (Be aware that the output will be inverted – this is expected.) If you turn down the input volume, the signal on your scope should shrink. Repeat this with all 4 input channels plus the **return FX** input.

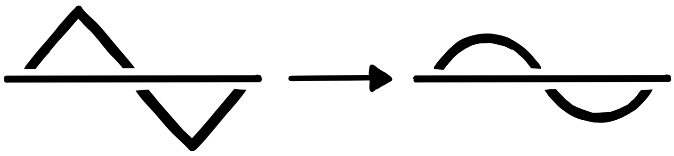


- If the output is stuck at 0 V, check if the input channel's 100k input resistor (**R13** – **R17**) is properly soldered to the PCB. Also check that its volume knob (**R1** – **R4**, **R9**) and input socket (**XS1** – **XS4**, **XS7**) are properly installed.
- If the output volume is not changing when turning down the input knob (at least not until it's **fully** turned down), check if **R22** is properly soldered to the PCB.
- If all channels seem dead, check that **DA1** is in its socket and correctly oriented, and that the socket's pins are properly soldered to the PCB.

TP2 (B)

Test this after you've ensured that the main mixer (TP1) works!
This test point exposes the **fully distorted version** of the main mix. Connect your oscilloscope probe here, and then send a triangle wave signal into one of the 4 inputs. Turn up the input channel's volume, and you should see a heavily distorted semi-square

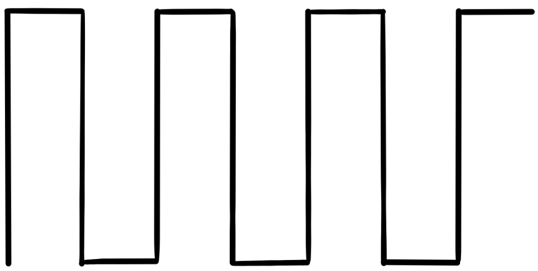
wave on your scope. If you turn down the input volume, the signal should get less distorted.



- If the output is stuck at 0 V, check if the resistor **R24** is properly soldered to the PCB.
- If the output is distorted, but very quiet compared to the input, check if the resistor **R27** is properly soldered to the PCB.
- If the output is extremely loud and distorted (swinging between -12V and +12V), check if the resistor **R28** and the diodes **VD3** and **VD4** are properly soldered to the PCB.
- If the output looks asymmetrical, check if the diodes **VD3** and **VD4** are **both** properly soldered to the PCB.

SEND OUTPUT

The **Send Out** is driven by a second mixer basically identical to the main mixer. Connect your oscilloscope probe to it, and then send a simple signal (e.g. a square wave) into one of the 4 inputs. Turn up the input channel's send knob, and you should see the input signal on your scope. (Be aware that the output should be in phase with the input!) If you turn down the send volume, the signal on your scope should shrink. Repeat this with all 4 input channels.



- If the output is stuck at 0 V, check if the input channel's 100k send resistor (**R18 – R21**) is properly soldered to the PCB. Also check that its send potentiometer (**R5 – R8**) is properly installed.
- If the send volume is not changing when turning down the send knob (at least not until it's **fully** turned down), check if **R23** and **R26** are properly soldered to the PCB.
- If all send channels seem dead, check if **DA1** is in its socket and correctly oriented, and whether the socket's pins are properly soldered to the PCB. Also check if the resistors **R25** and **R29** are properly installed.

INSERT FX OUTPUT

The **Insert FX Out** carries the (optionally distorted) main mix. Connect your oscilloscope probe here, send a triangle wave signal into one of the 4 inputs, and turn the volume up fully. Now slowly rotate the **Drive** knob. The output signal should morph between the clean input and a clipped version.

- If you get no output at all, check if **DA2** is in its socket and correctly oriented, and whether the socket's pins are properly soldered to the PCB. Also check if the resistor **R31,** the potentiometer **R10**, and the socket **XS8** are properly installed.
- If the drive knob does not affect the output signal, make sure all legs of the potentiometer **R10** are properly soldered to the PCB.

PHONES OUT

The **Phones Out** socket is used to drive headphones. Connect your headphones here, send a triangle wave signal into one of the 4 inputs, and turn the volume up half way. You should hear the tone on both sides.

- If you get no output at all, check if **DA3** and **DA2** are in their sockets and correctly oriented, and whether the sockets' pins are properly soldered to the PCB. Also check if the resistors **R32**, **R33**, **R38**, **R39**, the capacitor **C11**, and the sockets **XS5** and **XS9** are properly installed.
- If you only get sound on the left side, check if **R33** and **R39** are properly soldered to the PCB.
- If you only get sound on the right side, check if **R32** and **R38** are properly soldered to the PCB.
- If the sound on the left ear is distorted and very loud, check if **R35** is properly soldered to the PCB. On the right side, check **R36**.

LINE OUT

The **Line Out** socket is used to connect your mix to an audio interface or external sound system. Connect it to one of those, send a triangle wave signal into one of the 4 inputs, and turn the volume up half way. You should hear the tone on both the left and right channels.

- If you get no output at all, check if **DA2** is in its socket and correctly oriented, and whether the socket's pins are properly soldered to the PCB. Also check if the resistors **R34**, **R40**, **R41**, the capacitor **C11**, and the socket **XS10** are properly installed.
- If you only get sound on the left side, check if **R41** is properly soldered to the PCB.
- If you only get sound on the right side, check if **R40** is properly soldered to the PCB.
- If the sound is distorted and very loud, check if **R37** is properly soldered to the PCB.

INSERT FX IN

The **Insert FX In** socket is used in combination with the **Insert FX Out** to route your entire post-distortion mix through an external effect. Connect the **Insert FX Out** to a filter, the filter to the **Insert FX In**, send a square wave signal into one of the 4 inputs, and turn the volume up half way. Connect your headphones to the **Phones Out**. You should hear how the filter affects the signal when you change the cutoff.

- If you get no output, check if the socket **XS9** is properly soldered to the PCB.