mki x es.edu Output Mixer

Thank you for building **mki x es.edu Output Mixer module**!
This is slightly different user manual, because the mki x es.edu Output Mixer was developed in collaboration with Maris Zeltins, a teacher in Riga Technical University. Besides Output Mixer, Maris did a major contribution to SYNTRX and SYNTRX II design and several Black Series modules designs. So, instead of Moritz's' hand drawn breadboard images, we'll split the schematics of the module in parts and take an academic look on each part and how it contributes to the functionality of the entire module. The mki x es.edu Output Mixer essentially is a panning mixer – it takes **two mono inputs** with adjustable gain and **pans them left or right** to a stereo output, and an integrated **amplifier boosts the signal** for use for headphones.

Let's take a look at the **mki x es.edu Output Mixer** schematics (see next page)! 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**. The **Input stages** are built around opamps DA1A and DA1B followed by the **Panning stage** around potentiometers R3 and R4 and finally signal passes to the **Output amplifier**. **XS1 and XS2** are input **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 jack sockets (part number WQP-PJ301M-12). These sockets come with three lugs – a connector lug to which the tip of the patch cable is connected (and respectively, the audio or CV signal is applied), a ground lug which connects the patch cable to circuit ground and a **switching lug**. The switching lug is normally connected to the connector lug, but as soon you insert a patch cable into the socket, it gets disconnected from the connector lug. This is very handy for grounding inputs when nothing is patched into them. If the input is not grounded, some tiny current may bleed into the module, which can result in audible noise or CV fluctuations.

XS3 and **XS4** are **output jack sockets** and they are TRS (Tip-Ring-Sleeve) stereo sockets for use with TRS jacks.

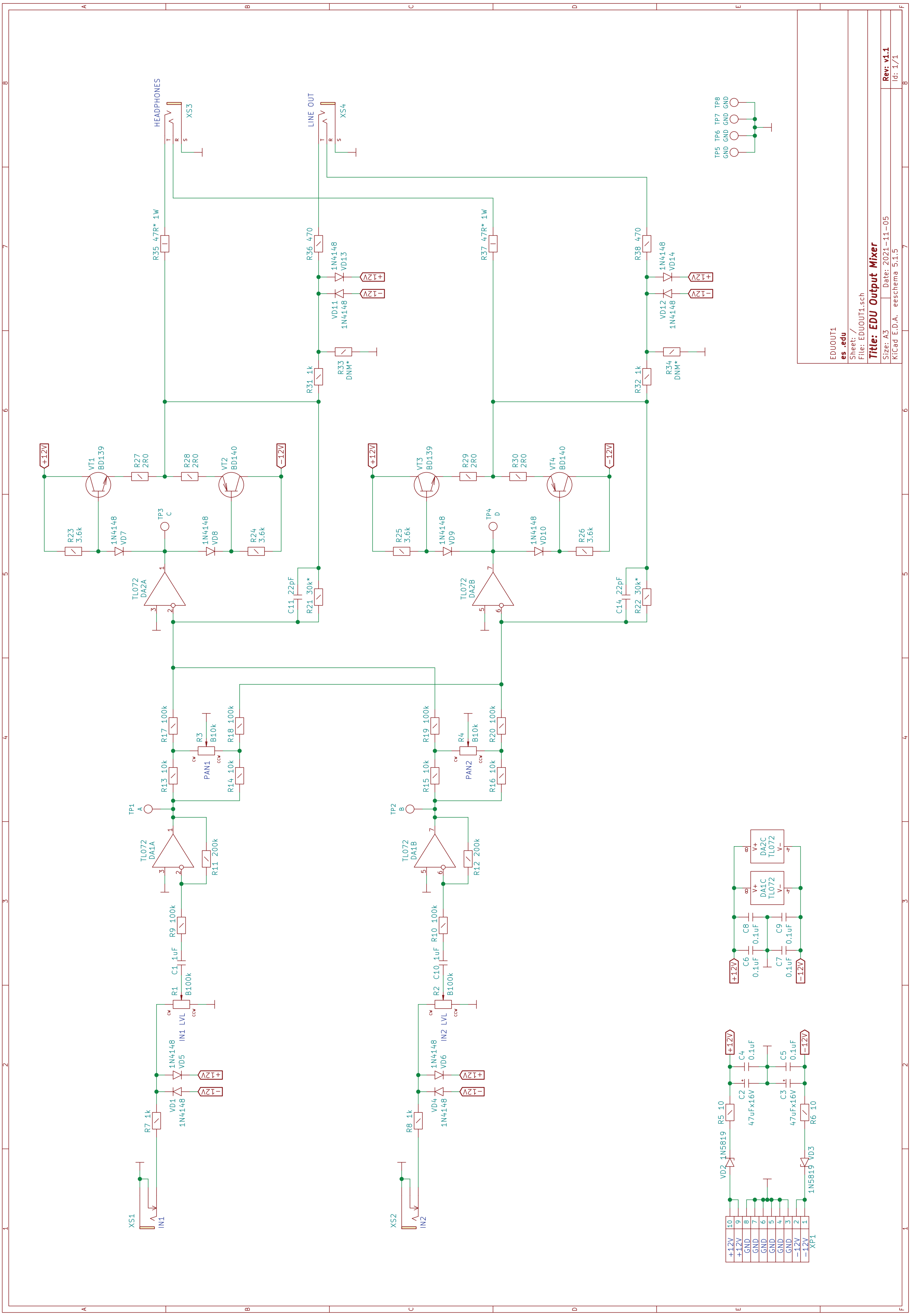
XP1 is a standard eurorack **power connector**. It's a 2x5 male pin header with a key (the black plastic shroud around the pins) to prevent accidental reverse polarity power supply connection. This is necessary because connecting the power incorrectly will permanently damage the module.

VD2 and **VD3** are **schottky diodes** that double-secure the reverse polarity power supply protection. Diodes pass current only in one direction. Because the anode of VD1 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 VD1, it closes, and no current passes through. The same goes for VD2, 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** (**R5** and **R6**) on the + and – 12 V rails, with **decoupling** (or **bypass**-) capacitors **C2 – C5**. 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 R5 and R6, the large 47 microfarads pair (C2 and C3) compensates for low frequency fluctuations, while C4 and C5 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, smoke, 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 **C6 – 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.


EDUOUT1
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Size: A3
Date: 2021-11-05
Rev: v1.1
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HEADPHONES
LINE OUT

Input stage

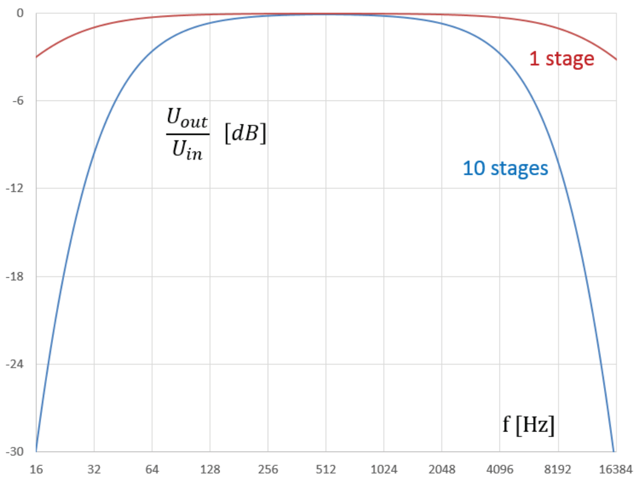
Sometimes a signal source (another module, for example, connected to the input of this module) has some DC voltage at the output. This voltage, additionally amplified, can induce undesired clipping even at low signal amplitudes. Thanks to capacitor C1, this stage only amplifies AC voltage.

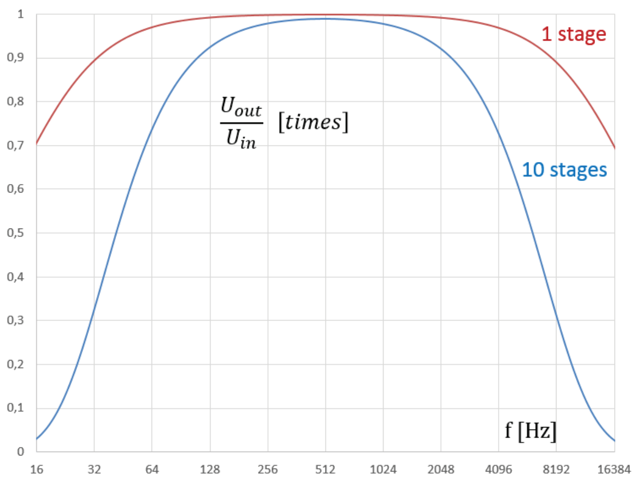
Capacitor impedance should be low compared to R9 in the audio frequency range. The frequency at which both impedance ($R9$ and $\frac{1}{2\pi f C1}$) modules are equal is the corner frequency. At this frequency, gain decreases $\sqrt{2} = 1{,}414$ times (or -3dB) compared to the middle band. In our case the corner frequency is $f_C = \frac{1}{2\pi C1 R9} = 1{,}59Hz.$ This seems very low compared to the lowest audible frequencies, but in the total signal chain there can be several such modules and the attenuation at the lowest frequency multiplies.

For example – we wish to pass through the lowest frequency of 8Hz.

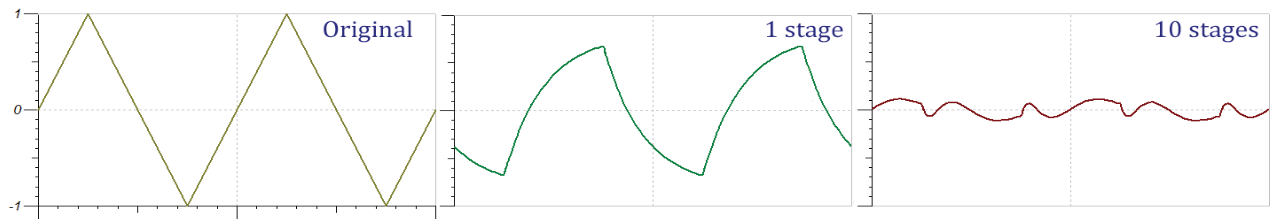
One given circuit at 8Hz creates attenuation 0,9808 times (-0,169dB). But 10 such circuits in series – already 0,823 times (-1,69dB).

In the next two graphs, each stage has corner frequencies: for High-Pass $f_C = 16Hz$ and for Low-Pass $f_C = 16k$:

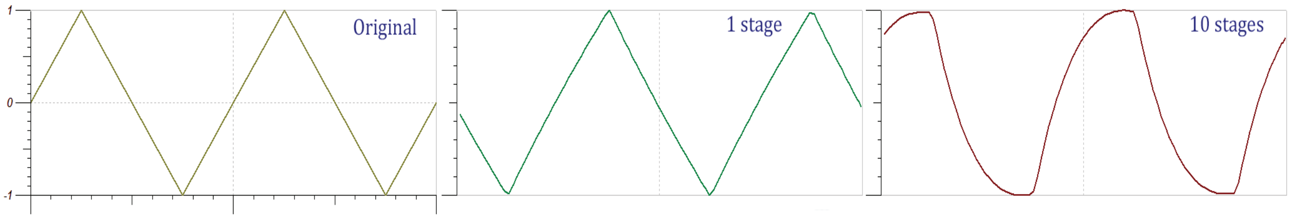




The next graphs shows an original triangle signal (20Hz) and what happens to it when passing through 1 and 10 stages with the previously indicated corner frequencies (16Hz and 16kHz).

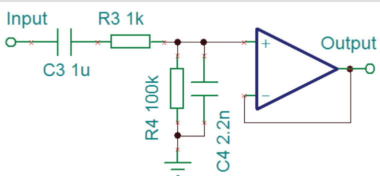
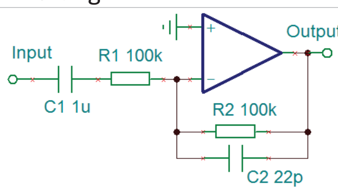


And in the next graphs, the same but with a High-Pass corner frequency that is 10x lower – 1,6Hz:

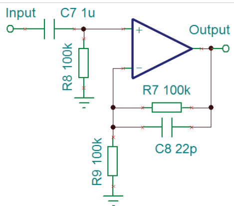
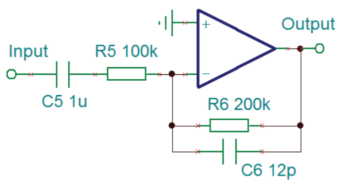


Why use capacitors to create additional Low-Pass?
Some modules can oscillate at higher frequencies (this usually is due to bad design). Also, cables connecting modules or traces on the PCB between stages can receive high-frequency noise from other equipment. Even if not audible, this noise can interact with audio-frequency signals, creating distortions or even audible noise. To avoid this, Low-Pass filters should be used.

Examples of buffers with unity gain and both filters:

	Non-inverting	Inverting
Circuit		
Gain at middle freq.	$\frac{R_4}{R_3+R_4} = \frac{100k}{1k+100k} = 0{,}99$	$-\frac{R_2}{R_1} = -\frac{100k}{100k} = -1$
High-Pass elements	R4, C3	R1, C1
HP corner freq.	$f_{HP} = \frac{1}{2\pi R_4 C_3} \approx 1{,}6Hz$	$f_{HP} = \frac{1}{2\pi R_1 C_1} \approx 1{,}6Hz$
Low-Pass elements	R3, C4	R2, C2
LP corner freq.	$f_{LP} = \frac{1}{2\pi R_3 C_4} \approx 72kHz$	$f_{LP} = \frac{1}{2\pi R_2 C_2} \approx 72kHz$

Examples of amplifiers:

	Non-inverting	Inverting
Circuit		
Gain at middle freq.	$1+\frac{R_7}{R_9} = 1+\frac{100k}{100k} = 2$	$-\frac{R_6}{R_5} = -\frac{200k}{100k} = -2$
High-Pass elements	R8, C7	R5, C5
HP corner freq.	$f_{HP} = \frac{1}{2\pi R_8 C_7} \approx 1{,}6Hz$	$f_{HP} = \frac{1}{2\pi R_5 C_5} \approx 1{,}6Hz$
Low-Pass elements	R7, C8	R6, C6
LP corner freq.	$f_{LP} = \frac{1}{2\pi R_7 C_8} \approx 72kHz$	$f_{LP} = \frac{1}{2\pi R_6 C_6} \approx 66kHz$

How panning works

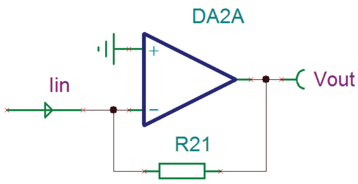
We have two signal sources and two outputs. This circuit allows you to freely manage the amount of each input signal going into one or the other output. The first stages are simple amplifiers - DA1A (gain is R11/R9=2) to compensate for losses in the following panning network. After the panning circuit, there is an op-amp - DA2A - which works as a current-to-voltage converter.

Let’s go through some rules for ideal op-amp operation (in most cases it is enough with these to understand how a real circuit works):

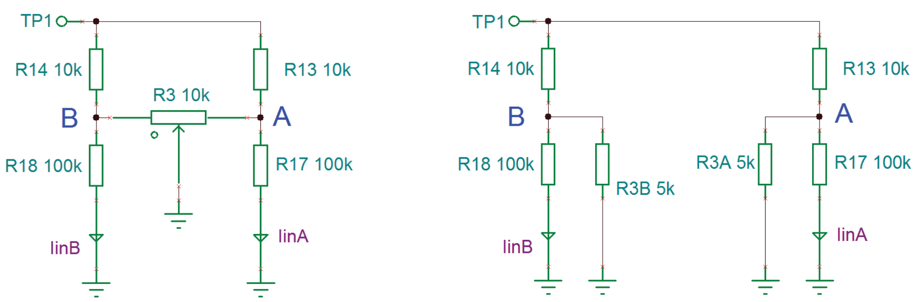
1. Voltage on both inputs is equal.
2. Differential gain is infinitely large (that means – only the voltage difference between inputs is amplified).
3. Input impedance for both inputs is infinitely large (that means – there is no input current).
4. Output impedance is = 0 (the amp works like an ideal voltage source).

So, for the circuit in the picture:

- Voltage at the inverting input is =0V because the non-inverting input is grounded.
- If some circuit (not shown in picture) provides a current I_{IN} (let’s name it “incoming”), the same current should flow through the feedback resistor R1 as “outgoing”.
- Therefore, the output voltage will be $V_{OUT} = -I_{IN} \cdot R_1$.



In our case, the circuit which provides the input current I_{IN} is the panning circuit. Because the inverting input of DA2A is “grounded”, we can assume that the currents trough R18 and R17 go to ground. Let’s calculate them.



The input voltage V_{TP1} for this circuit is the output voltage of DA1A (we can measure it at test point TP1). The potentiometer R3 slider is grounded. So actually there are two resistances (R3B and R3A), the values of which are dependant on the slider position. Let's characterise the slider positions with values ranging from $= 0 \dots 1$.
In the left position $k = 0$ and $R_{3B} = 10k\Omega \cdot 0 = 0\Omega$, $R_{3A} = 10k\Omega \cdot (1-0) = 10k\Omega$.
In the middle position $k = 0{,}5$ and $R_{3B} = R_{3A} = 10k\Omega \cdot 0{,}5 = 5k\Omega$ (shown in picture).
In the right position $k = 1$ and $R_{3B} = 10k\Omega \cdot 1 = 10k\Omega$, $R_{3A} = 10k\Omega \cdot (1-1) = 0\Omega$.

Already in this moment it is clear: in the left position $I_{INB} = 0$ and in the right position $I_{INA} = 0$ because in each case, point A or B is grounded.

Before we get current values, let's calculate the voltages at points A and B. To make this easier, let's replace parallel-connected resistors with singles:
$R_B = R_{18}||R_{3B} = \frac{R_{18} \cdot R_3 \cdot k}{R_{18}+R_3 \cdot k}$ and $R_A = R_{17}||R_{3A} = \frac{R_{17} \cdot R_3 \cdot (1-k)}{R_{17}+R_3 \cdot (1-k)}$.

Now voltages at A and B are: $V_B = V_{TP1}\frac{R_{14}}{R_{14}+R_B}$ and $V_A = V_{TP1}\frac{R_{13}}{R_{13}+R_A}$.

And the currents:

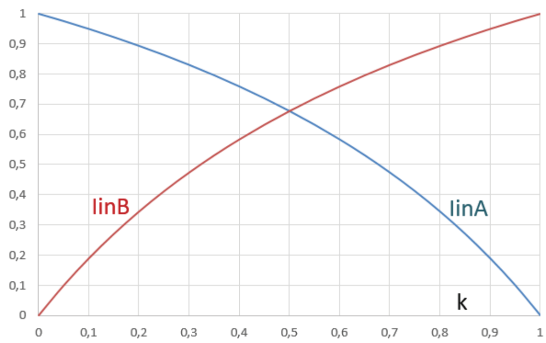
$$I_{INB} = \frac{V_B}{R_{18}} = \frac{V_{TP1}}{R_{18}} \cdot \frac{R_B}{R_{14}+R_B} = \frac{V_{TP1}}{R_{18}} \cdot \frac{\frac{R_{18} \cdot R_3 \cdot k}{R_{18}+R_3 \cdot k}}{R_{14}+\frac{R_{18} \cdot R_3 \cdot k}{R_{18}+R_3 \cdot k}} = V_{TP1}\frac{R_3 \cdot k}{R_{14}(R_{18}+R_3 \cdot k)+R_{18} \cdot R_3 \cdot k} = \frac{V_{TP1}}{\frac{R_{14}R_{18}}{R_3 \cdot k}+R_{14}+R_{18}}$$

Thus:

Thus: $$I_{INA} = \frac{V_A}{R_{17}} = \frac{V_{TP1}}{R_{17}} \cdot \frac{R_A}{R_{13}+R_A} = \frac{V_{TP1}}{R_{17}} \cdot \frac{\frac{R_{17} \cdot R_3 \cdot (1-k)}{R_{17}+R_3 \cdot (1-k)}}{R_{13}+\frac{R_{17} \cdot R_3 \cdot (1-k)}{R_{17}+R_3 \cdot (1-k)}} = V_{TP1}\frac{R_3 \cdot (1-k)}{R_{13}(R_{17}+R_3 \cdot (1-k))+R_{17} \cdot R_3 \cdot (1-k)} = \frac{V_{TP1}}{\frac{R_{13}R_{17}}{R_3 \cdot (1-k)}+R_{13}+R_{17}}$$

The denominator in each equation is equivalent to the resistance of the circuit. For current I_{INB} it becomes $= \infty$ at $k = 0$ (that means – current → 0). And for current I_{INA} it becomes $= \infty$ at $k = 1$.

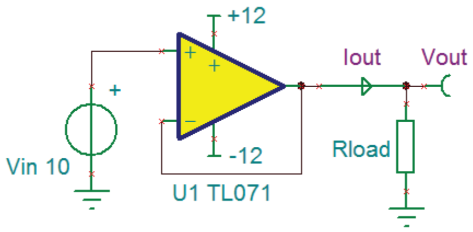
The next figure shows the relative values of currents depending on slider position k:

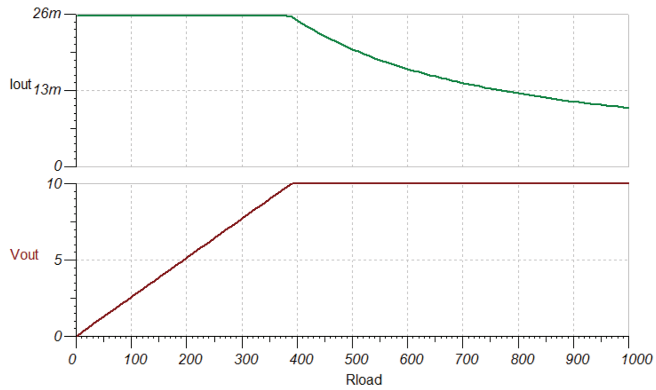


Output amplifier

This module can drive headphones with a variety of impedances – from 16Ω to some tens or even hundreds of ohms. The recommended load impedance of most op-amps, however, is 2kΩ and higher. We therefore need to improve the capability to source/sink higher currents.

The graph to the right shows the capability of the popular op-amp TL071 for driving different loads. The input voltage is +10V and the follower circuit should be set at the same +10V at the output. The given op-amp, however, can only source about 26mA. Correct operation therefore happens only when $R_{LOAD} > 400\Omega$.

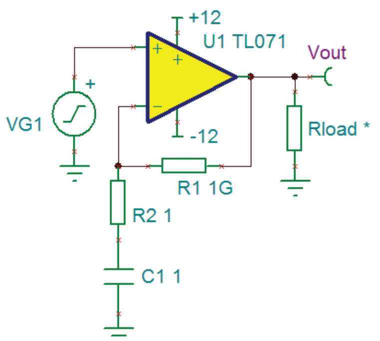


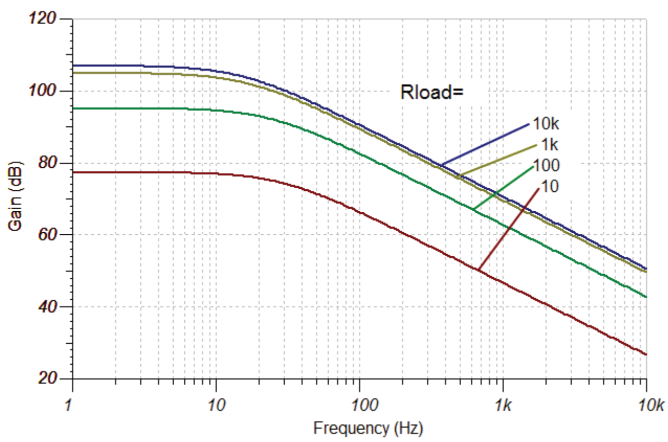


Of course, real examples of the TL071 can have different curves – this is to illustrate the trend.

There is another aspect to how a low impedance load degrades the op-amp's characteristics: heavy (low impedance) load decreases the open-loop gain. That applies both to a single transistor amplifier and to complex circuits like an op-amp.

In the next example, the gain is set to an unreal value - 1.000.000.000 times (180dB), therefore, the real gain in set by the op-amp itself:



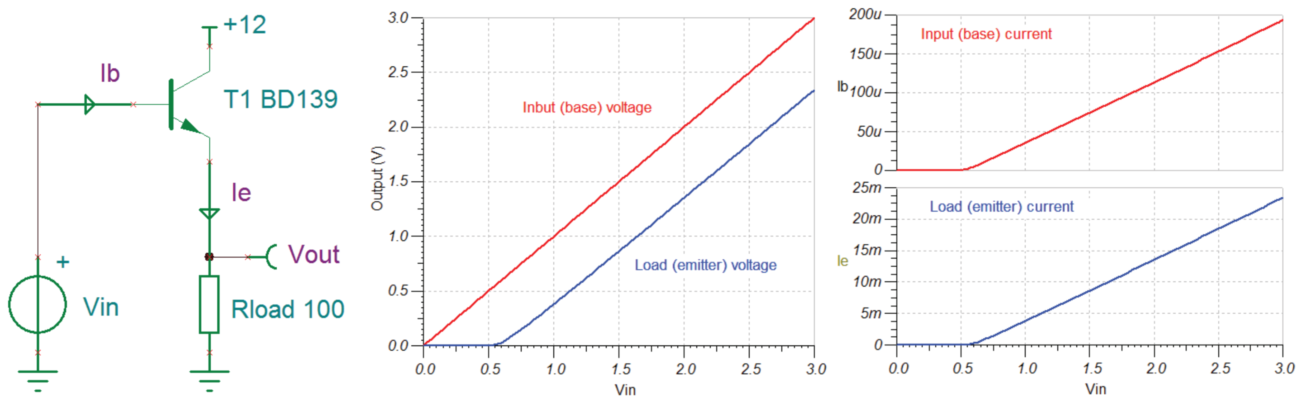


Every op-amp feels fine with a 10kΩ load. For TL071, also 1kΩ is good. But with 100Ω and 10Ω loads, the open loop gain lowers for 4dB and 30dB (4 and 32 times). But lower open loop gain means:

- The actual gain is lower like set by the feedback circuit,
- non-linear distortions (caused by the op-amp itself) are not so deeply pressed-down (compensated) by the feedback.

The simplest solution – to add a complementary emitter follower. “Complementary” means – two transistors are used (bipolar or field-effect) with similar parameters but with different structures (in this case – npn and pnp).

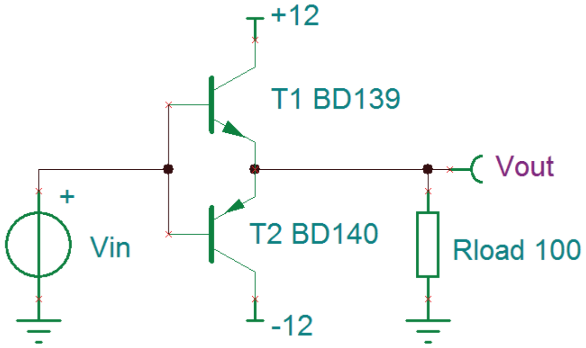
But first let’s analyse a single transistor emitter follower:

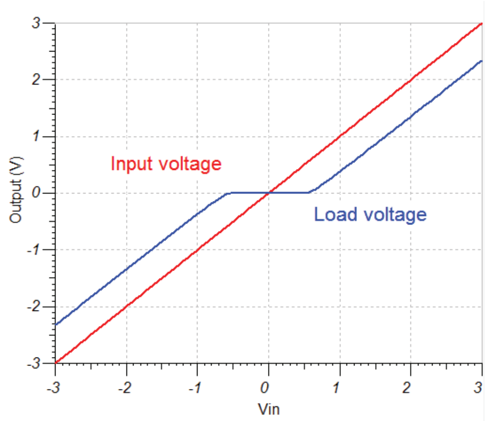


At the input, a voltage source V_{in} is connected, at the output – a load resistor R_{load}.
What we see:

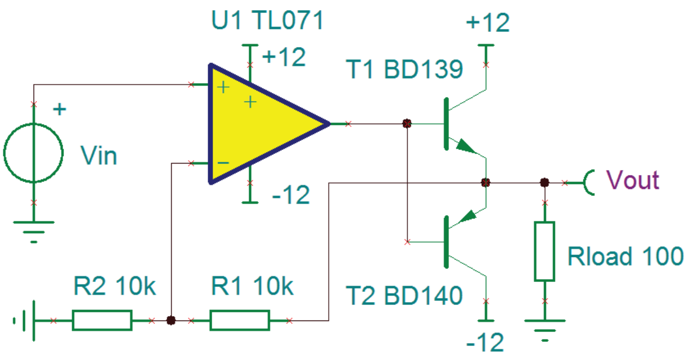
- Load voltage is zero until the input voltage reaches 0,7 volts. This value (in practice 0,5…1,0V at different current values) is the voltage necessary to open the base-emitter junction and to start the base current flowing.
- Also when the input voltage is larger $V_{load} = V_{in} - V_{BE} = V_{in} - 0{,}7$ - it is not very good. At we get $V_{in} = 3{,}0V$ we get $V_{load} = 2{,}34V$.
- But let’s compare currents! At the same input voltage $I_{load} = 23{,}4mA$ and $I_{in} = 0{,}194mA$ – about 120 times lower. So we can say – a signal source instead of a 100Ω load feels 12kΩ! The value 120 responds to the so-called transistor “current gain” or simply “β”. It is different for various types and is dependent on current values. Typically, for low-power transistors – some hundreds, for medium and high power – some tens.

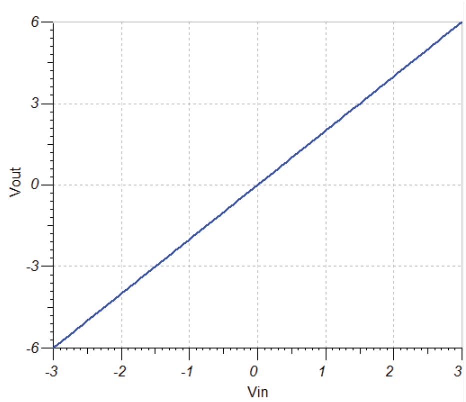
Bad things for the given circuit: it works only with positive voltages, starting from 0,7V. And complementary pair already have a -0,7…+0,7V “dead” zone:





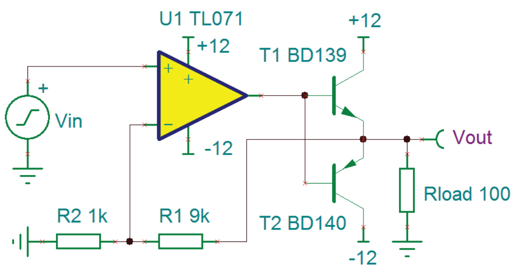
Now let’s add such a pair to the op-amp output. Note – the feedback signal comes from the emitters, not from the op-amp output.



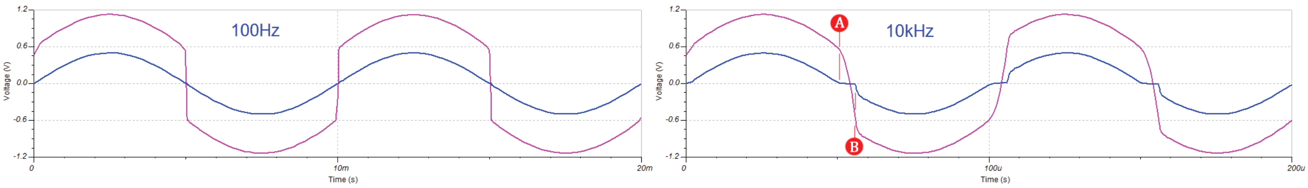


Now the voltage transfer curve is straight, without a “dead zone”. But this is a DC transfer curve

Let’s look at what happens with an audio signal (sine with 100Hz and 10kHz frequency). And in this case the circuit has non-inverting gain= $1 + \frac{R1}{R2} = 10$.



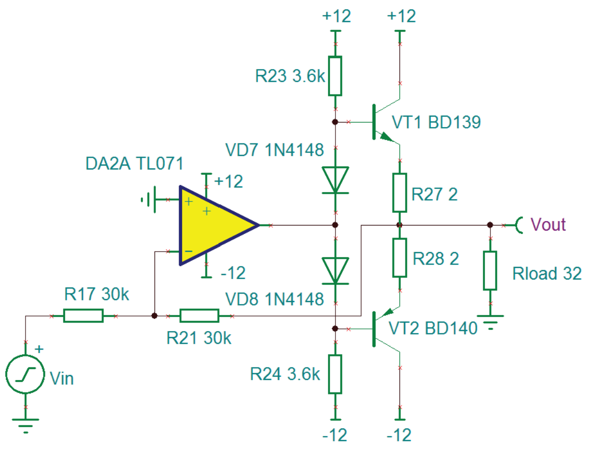
The op-amp output voltage V_{OP} is represented by the pink curve, load voltage V_{LOAD} – by the blue one.
Note that V_{OP} is always more positive or negative than V_{LOAD} – up to the value of V_{BE}.
When an op-amp voltage goes through the “dead zone” (between points A and B), the amplifier loses feedback signal – it quickly changes V_{OP} to the opposite value. How quickly and how it affects load voltage – depends on the “slew rate” parameter of the given type. For TL071 $SR = 13\frac{V}{\mu s}$. At 100Hz the switchover happens quickly enough – we can't see any distortions. But at 10kHz, these so called “crossover distortions” are clearly visible.

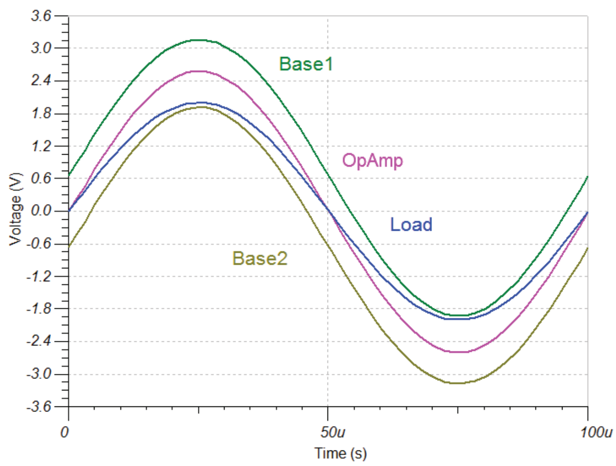


A method to minimise this type of distortion used in this module is only one of many known. Diodes VD7,VD8 keep both transistor BE junctions close to an open state (approx. 0,65V on each junction).

$$V_{VD1} + V_{VD2} = V_{BE1} + I_Q(R_{27} + R_{28}) + V_{BE2}$$

With given values of resistors the quiescent current (it flows from +12 through both transistors to -12V). It is small enough to avoid overheating and large enough to keep the transistors ready to work. Try to shorten the diodes and hear the result on the output!





The graph shows voltages at different amplifier points at 2Vpp SINE 10kHz at the input and very low - 10Ω load resistance. Voltage on the load is exactly 2Vpp and without visible crossover distortions. Difference between op-amp output and load voltages is the voltage drop on (when positive) or on (when negative). In a real circuit there is an additional series resistor used to limit headphone power and protect your ears.
How large a voltage on load without distortions can this amp provide?

R_{LOAD} [Ω]	16	32	64	80	120	250	300	470
V_{MAX} [Vpp]	3,5	6,0	9,4	10,3	12,6	15,8	17,0	18,2
P_{MAX} [mW]	96	140	172	166	165	125	120	88

There are many different signal levels defined. In our case two are significant:

- Professional audio line level $\sim 3{,}5Vpp$
- Many Eurorack modules working with a larger output level around $\sim 10Vpp$. This also applies to the EDU series modules.

Vpp or “peak-to-peak voltage” is easier to measure using an oscilloscope, it is applicable to different signal forms and is especially important when caring about dynamic range (for example – ability to process signal without clipping).

Total gain of a given module to the Line output (at Level=max, Panning=max and input impedance of connected next unit is 47kΩ) can be calculated as $G = 0{,}00442\frac{R_{11}R_{21}}{R_9}$, where the resistances unit is kΩ.
So at $R_9 = 200k$ $R_{11} = 100k$ $R_{21} = 39k$ $G = 0{,}344$ and $V_{IN} = 10Vpp$ and we get $V_{LINE} = 3{,}44Vpp$.

If you wish to change the gain:

- To a higher value, better increase R21 because the input amplifier DA1A is operating already near clipping level;
- To a lower value – decrease R11.

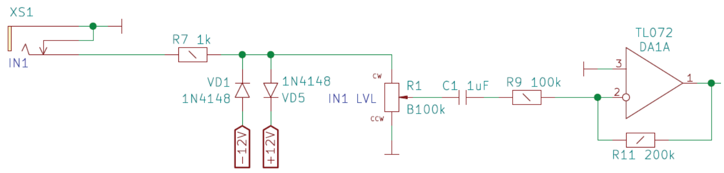
In both cases you maybe need to change R35 to adapt to the loudness of headphones used.

Module input/output protection

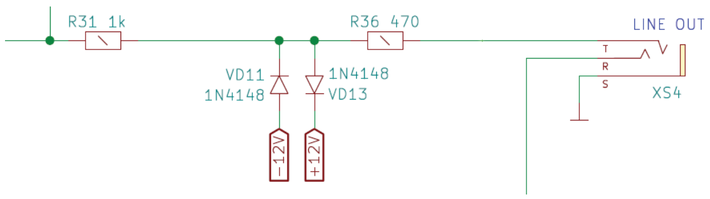
There are some elements (diodes and resistors) at the input and output of module which don't impact signal during normal operation. But they are responsible for a very important thing – protection. More specifically, protection against overvoltage created by electrostatic discharges (ESD), against wrong connections etc. In everyday life we touch different materials. During this mechanical contact (friction) we collect or give free electrons from/to this surface. So we collect some charge and actually get positive or negative potential (relative to earth or zero potential).

To test the ability of different devices to withstand ESD different models are used. Among others – Human Body Model (HBM) which consists of 100pF capacitance and 1500Ω series resistance. During the test this capacitor is charged up to some voltage (in the range of kV) and after that – connected to the device under test. Let's suppose we collect positive 20kV potential, take a jumper cable and connect it in to IN1 socket.

Some part of our charge leaks directly to the case which usually is grounded. The rest of the charge finds a way through the resistor R7 and diode VD5 (because we are positive). So, the voltage on the level potentiometer (and op-amp input) for only some microseconds reaches +12,7V without causing any further damage.



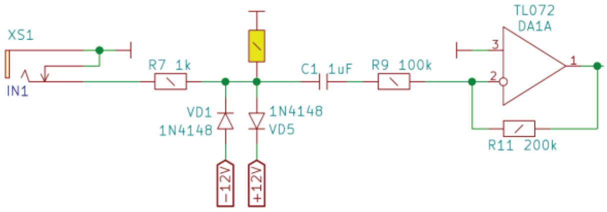
The same happens at the output.



But there R36 has one more feature. Most op-amps don't like a capacitive load. For example – you connect a module's output to your power amplifier using a 10m long screened audio cable. They have quite high capacitance – up to 200pF/m. So you attach 2000pF capacitance to module output. Because of the op-amp's limited current drive capability (or in other words – because of non-zero output impedance), voltage on load gets an additional phase shift which finally can cause an overshoot or "ringing" in the step response or even oscillations in the worst case. In the given module, R31 and R36 effectively prevent this from happening.

One more thing relating input/output capacitors – the side which is connected to the "outer world" should be grounded. In the example input circuit this happens twice:

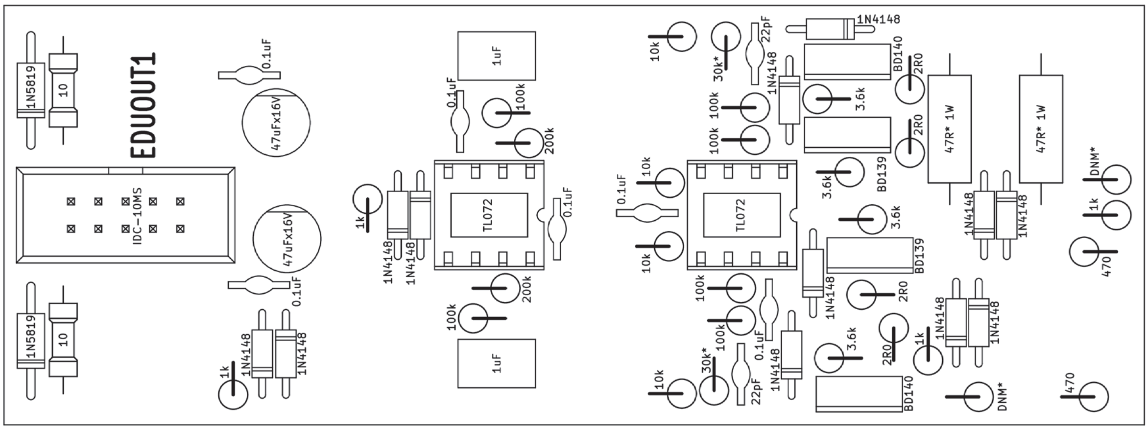
- Through the normally grounded input socket contact
- Through the yellow resistor (it resistance can be 1MΩ).

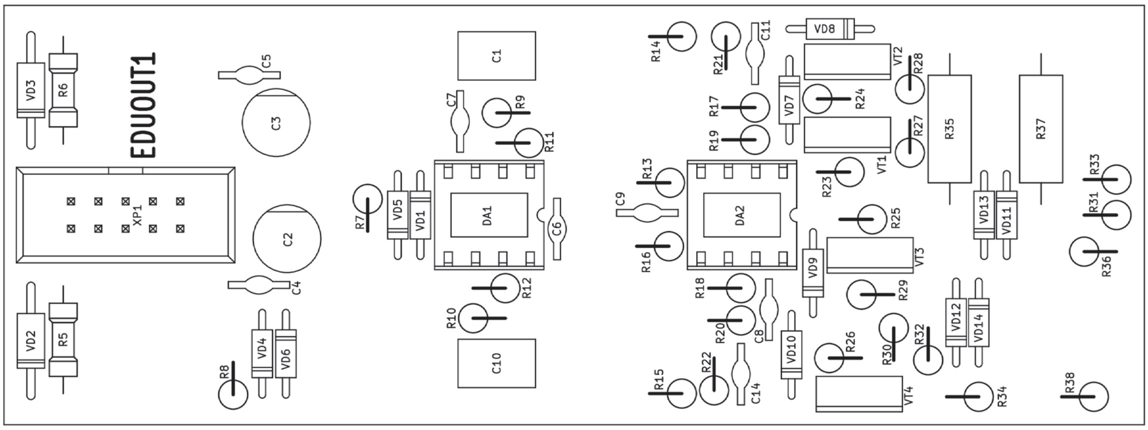


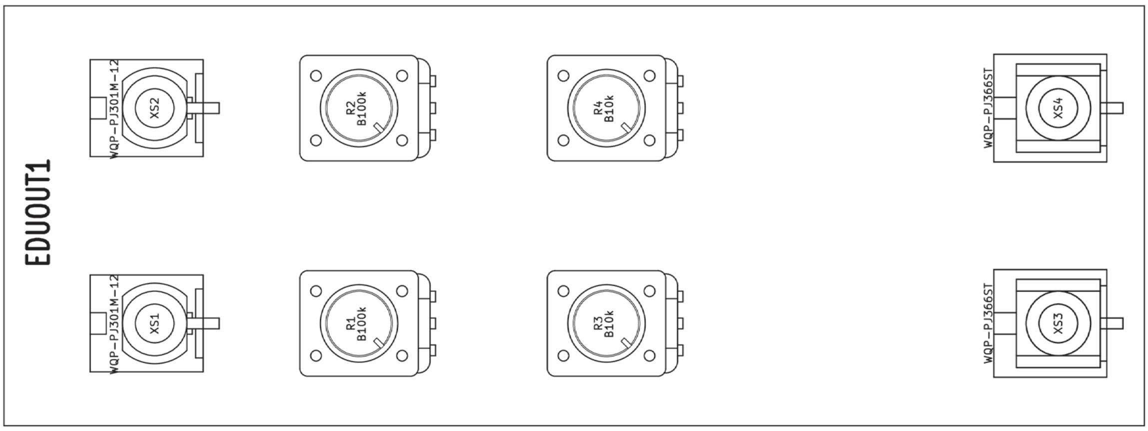
Without such measures, the following happens: Protecting diodes have some leakage current of around 1nA and both currents are somewhat different. Let's suppose they are 0,9nA and 1,1nA. This difference (0,2nA) will slowly charge capacitor C1. After a few minutes C1 will reach a positive or negative supply voltage.

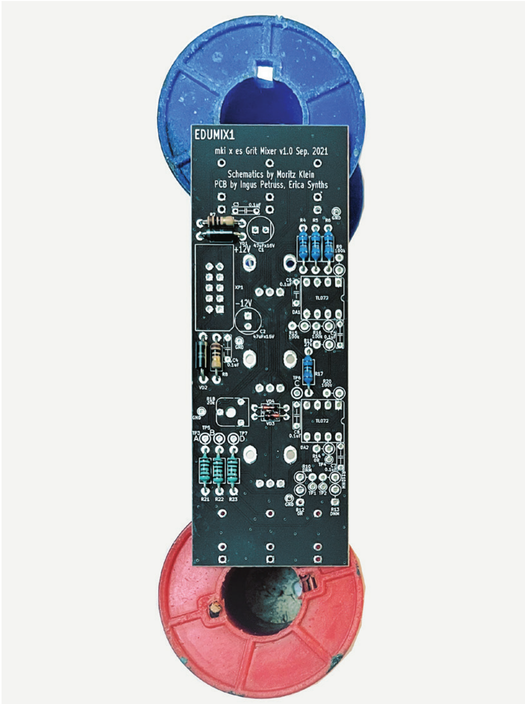
When the input is connected to a low impedance output of another module, the left side of C1 is grounded, but the right side gives a 12V impulse to following op-amp. Consequently, we get a loud, audible click from the speakers.

Before you start soldering, we highly recommend printing out these part placement diagrams with designators and values. Because some of our PCBs are rather densely populated, this will help you to avoid mistakes in the build process.

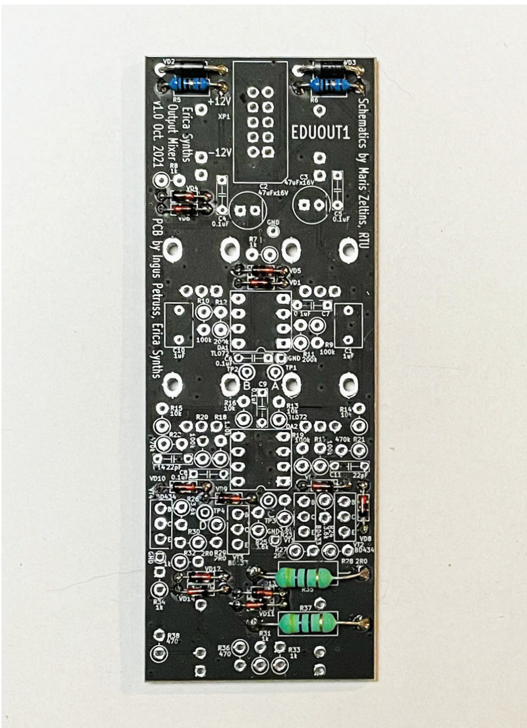




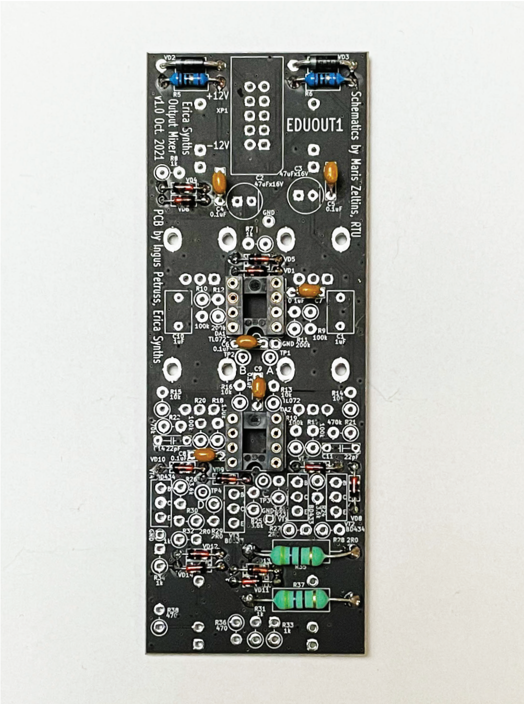




Place the Output Mixer PCB in a PCB holder for soldering or simply on top of some spacers (I use two empty solder wire coils here).

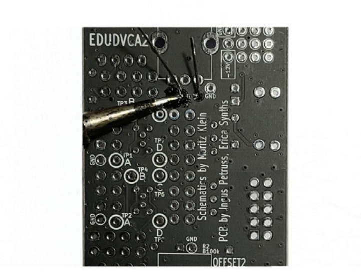


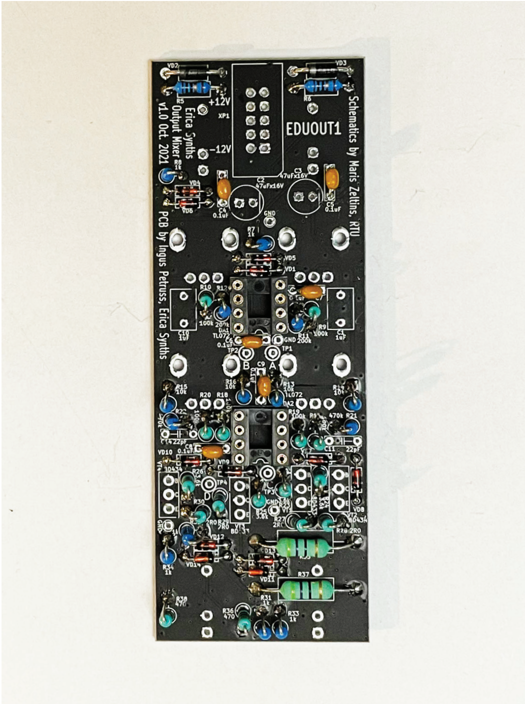
I usually start populating PCBs with lower, horizontally placed components. In this case, these are **R5**, **R6**, the **power protection diodes**, the **1N4148** diodes and two **1W 47ohm resistors.** Bend the component leads and insert them in the relevant places according to the part placement diagram above. All components on the PCB have both their value and denomination printed onto the silkscreen. If you are not sure about a resistor's value, use a multimeter to double check. Next, insert the 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 a stripe on the component. Flip the PCB over and solder all components. Then, use pliers to cut off the excess leads.



Next, 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. Then proceed with the ceramic 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, your PCB should look like this.

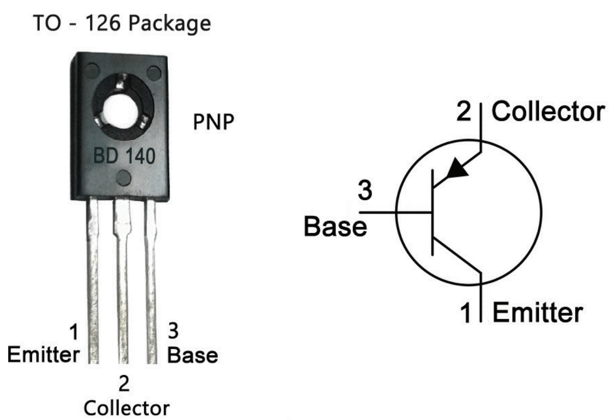
In order to save space on the PCB, some of our projects, including the Output Mixer, have **vertically placed resistors**. So the next step is to place & solder those. Bend a resistor's legs so that its body is aligned with both legs and insert it in its designated spot. Then solder the longer lead from the top side of the PCB to secure it in place, turn the PCB around and solder the other lead from the bottom. You can insert several resistors at once. Once done with soldering, use pliers to cut off excess leads.

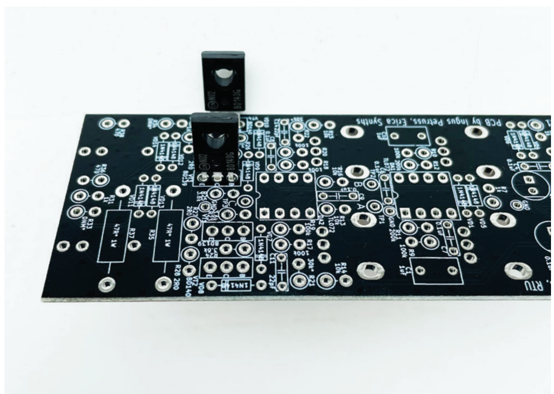


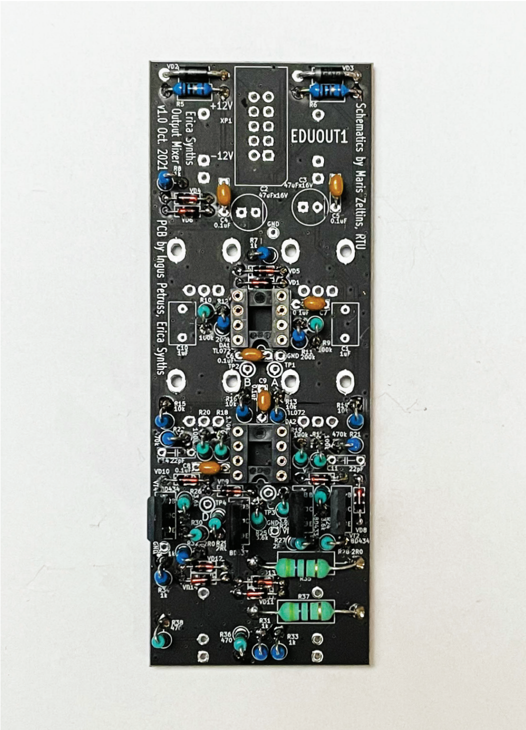


Once you are done with soldering all resistors, your PCB should look like this.

Next up: inserting & soldering the transistors. In total we have 4 transistors in TO-126 package – 2x NPN BD139 and 2x PNP BD140. These transistors are commonly used in power amplifiers, and they have a hole in a middle to fix them to heatsinks because they tend to overheat, when operating at critical conditions. Our headphone amplifier does not push transistors to extremes, so we do not use heatsinks. Make sure you place the transistors in their designated spots according to a pinout – **orientation is critically important here.**

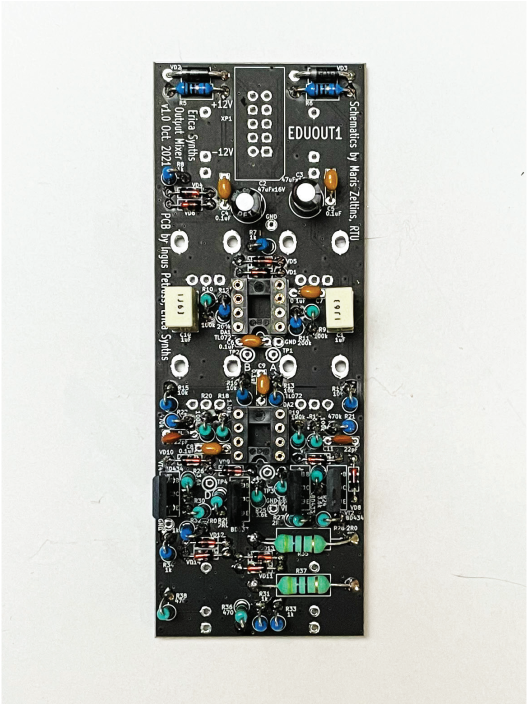




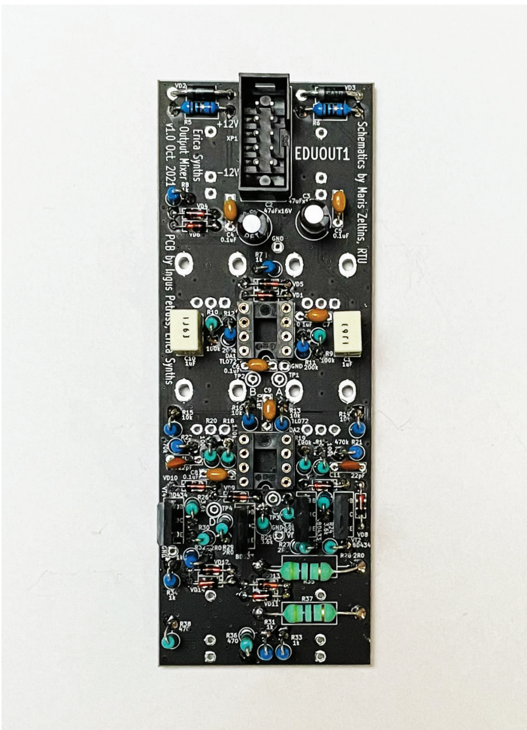


Once done, your PCB should look like this

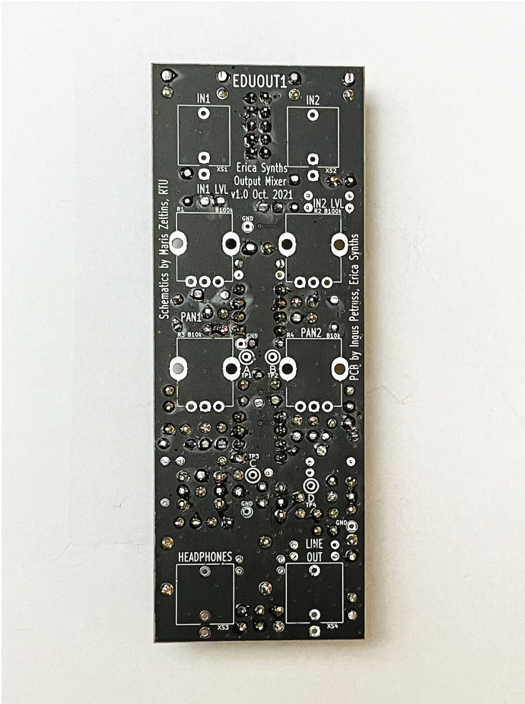
Next, insert & solder the electrolytic capacitors. Electrolytic capacitors are bipolar, and you need to mind their orientation. The positive lead of each electrolytic capacitor is longer, and there's a minus stripe on the 5 side of the capacitor's body to indicate the negative lead. On our PCBs, the positive pad for a capacitor has a square shape, and the negative lead should go into the pad next to the notch on the silkscreen.



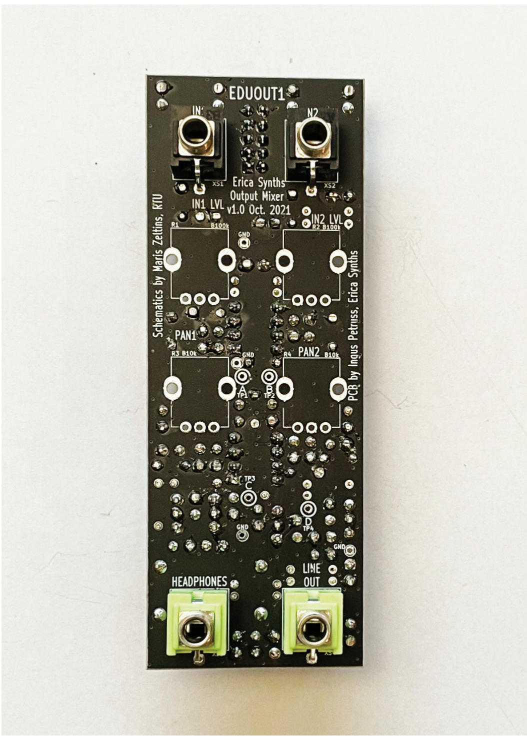
Also, solder two large **film capacitors**. They are not polarized, so orientation is not critical. Now your PCB should look like this.



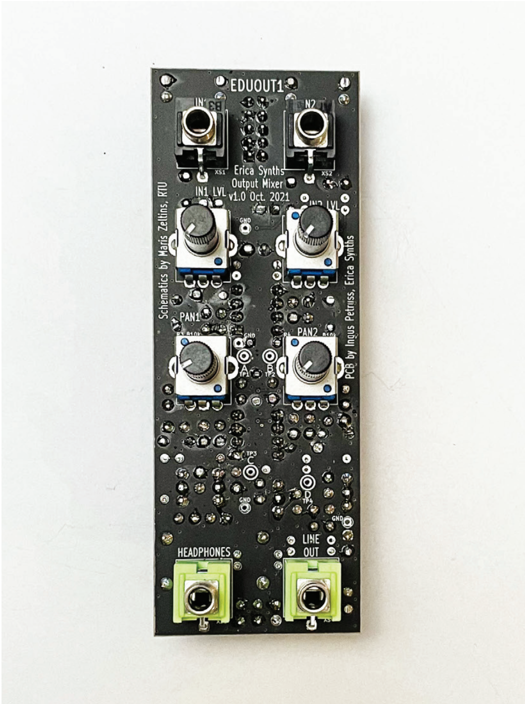
Next, solder the power supply socket. Make sure the orientation of the socket is as shown in the picture below – the arrow pointing to the first pin is aligned with a notch on the silkscreen. The key on the socket will be facing to the right.



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.

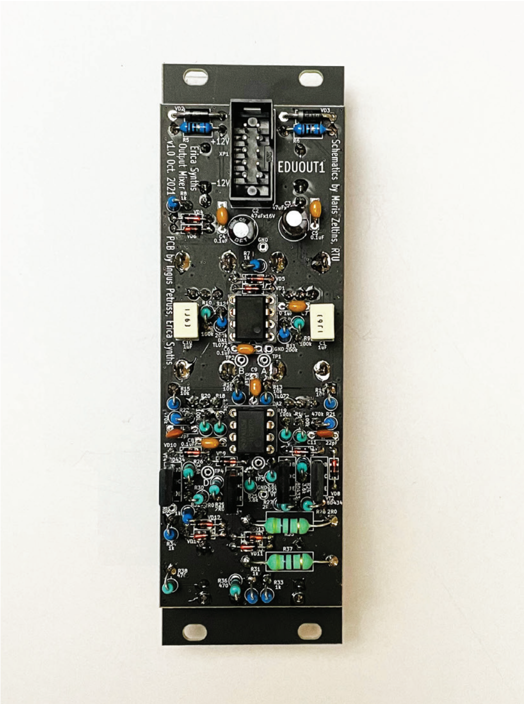


Insert the jack sockets and solder them. Please note that two lower jack sockets are green ones – they are TRS (Tip – Ring – Sleeve) stereo sockets and instead of switching lug, which is found on the black jack sockets, green ones have ring lug, which makes them suited for use with headphones and stereo jacks.



Insert the potentiometers, but don't solder them yet! Fit the front panel and make sure that the potentiometers' shafts are aligned with the holes in the panel – and that they're able to rotate freely. Now, go ahead and solder the potentiometers.

Install the front panel and fix it in place with the 4 hex nuts on the jack sockets. We are almost done!



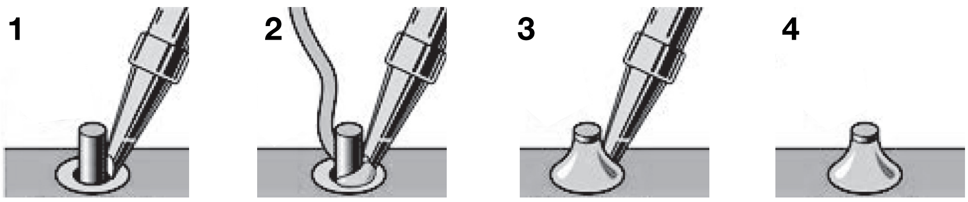
Now, **insert the ICs into their respective DIP sockets**. Mind the orientation of the ICs – match the notch on each IC with the one on its socket.

Congratulations! **You have completed the assembly of the mki x es.edu Output Mixer module!** Now 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. Connect a VCO's SAW output to IN1, and PULSE output to IN2. Plug the headphones in the relevant output, increase LEVEL and monitor sound. Rotate the PAN knobs and check is sound moves from the left to the right. If it does, your build is successful. The module needs no calibration and will work straight away.

SOLDERING APPENDIX

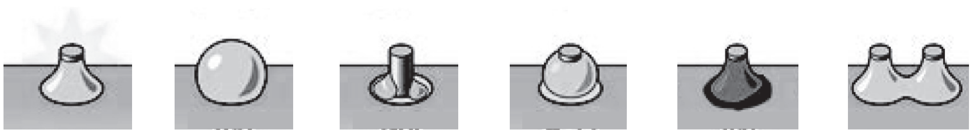
If you've never soldered before – or if your skills have become rusty – it's probably wise to check out some **THT** (through-hole technology) **soldering tutorials on YouTube**. The main thing you have to remember while soldering is that melted solder will flow towards higher temperature areas. So you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



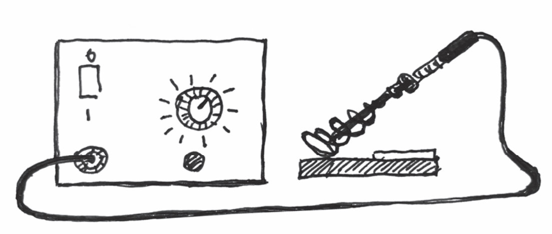
Heat part and pad 2 - 3 sec | Add solder | Continue heating 1 -2 sec. | Let cool

After you have completed soldering, inspect the solder joint:

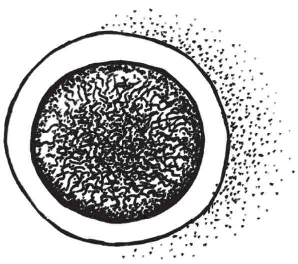


Perfect | Too much solder | Not enough solder | Cold joint | Too much heat | Short

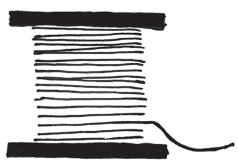
DIY electronics is a great (and quite addictive) hobby, therefore we highly recommend you invest in good tools. In order to really enjoy soldering, you'll need:



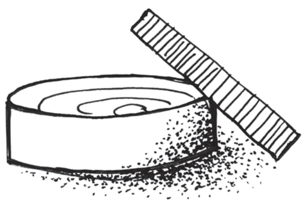
A decent soldering station. Top-of-the-line soldering stations (brands like Weller) will cost 200€ and above, but cheaper alternatives around 50€ are often good enough. Make sure your soldering station of choice comes with multiple differently-sized soldering iron tips. The most useful ones for DIY electronics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

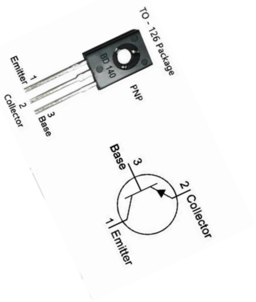


Solder wire with flux. I find 0,7mm solder wire works best for DIY projects.

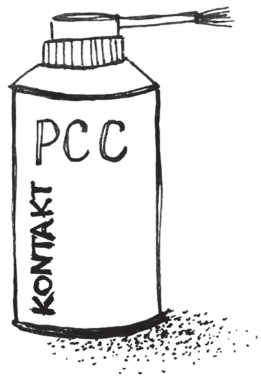


Some **soldering flux** paste or pen will be useful as well.

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.

"Music touches us emotionally, where words alone can't."

– Johnny Depp