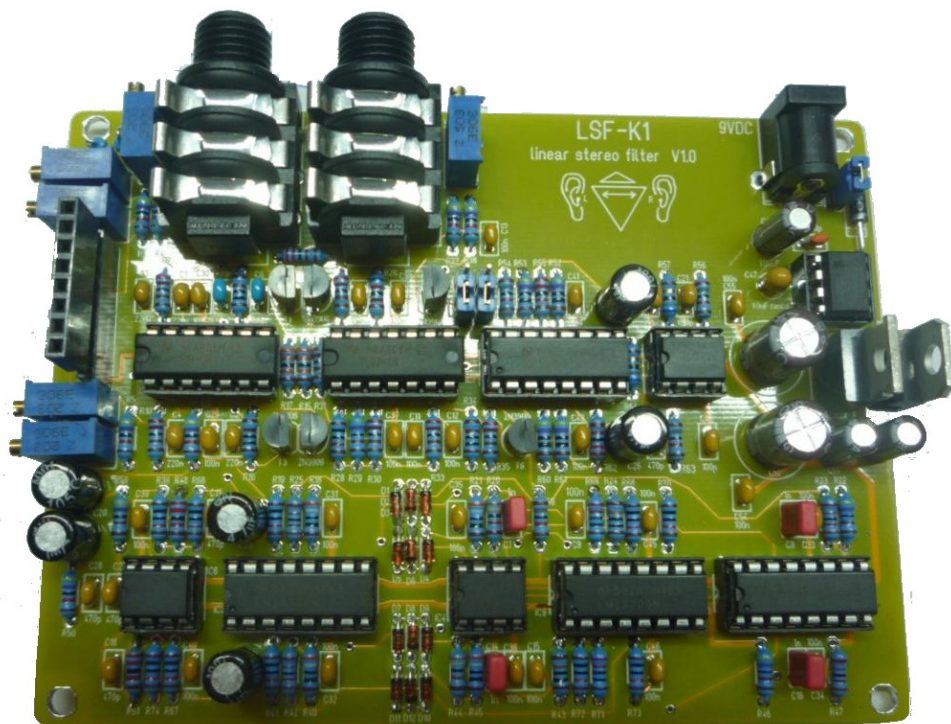


**TubeOhm
Shruthi/Phoenix Stereo Filter LSF-K1
Version V 1.0 und V 1.01
LSF-K1 Filter DIY Manual English**



FOREWARD: I would like give an immense 'Thank You' to Olivier of Mutable Instruments for the provision and continuing development of the Shruthi, which may only be marketed under the name 'Phoenix' of TubeOhm.

FEATURES: LSF-K1 stereo filter
FREQUENCY RANGE: 20Hz-30kHz 2 Filter isolated stereo
FILTER TYPE: State Variable 2x 12 dB low pass, resonance stabilization
OPERATING VOLTAGE: 9VDC 600mA
CONNECTIONS: Stereo audio input / Stereo audio output
INPUTS: Cutoff, CV1, CV2, resonance, VCA,
EXT: 1V ss ** the external audio input.
OUTPUT AMPLITUDE: max 3 V ss
REQUIRED HARDWARE: Shruthi/Phoenix motherboard firmware version 0.97 or later
DIY SKILL : Middle/High

Filter DIY kit LSF-K1 for the Shruthi/Phoenix

The LSF-K1 kit is a linear stereo filter designed to be used in conjunction with a Shruthi/Phoenix Motherboard with firmware version 0.95 or later installed. Older versions should run fine but have not been tested with this release.

The LSF-K1 operates under the normal 'LP' (SMR4MK-II) selected mode.

What is the LFS-K1?

The LFS-K1 is a stereo filter board for the Shruthi/Phoenix. It includes 2 identical 12dB LP filters running simultaneously for the left and right channels. Both filters are controlled together via the Filter menu, but can also be controlled separately in the Modulation(Mod) Matrix.

Therefore, it is possible that the LFO 1 modulates the left filter, while the LFO 2 modulates the right filter.

Another variation could assign ADSR 1 exclusively to the right filter, and LFO 1 exclusively to the left filter.

All sources of the sound control can be assigned to the left and/or right filter using modulation matrix.

With this unrestricted assignability the LFS-K1 creates a completely new sound dimension!!!

The sound effects capable with the LFS-K1 filterboard range from a wide panorama to varying floating sounds, to hard 'left-right' stereo panning.

The built-in stereo audio input allows the user to manipulate incoming signals and design a personalized filter set, in which the parameters can be controlled via MIDI and also be saved.

Like all of the TubeOhm DIY projects, the LFS-K1 kit is for moderate to advanced builders. Due to the tight component density this kit is not recommended for beginners.

To build the LFS-K1 kit you should have soldering experience and technical knowledge in handling and understanding of electronic components.

The final setup of the LFS-K1 is a little more extensive because both filters have 2 trim pots to adjust which must be calibrated individually so as to function together. The calibration process is not difficult in itself but adjusting the filters to the desired character is

a matter of personal individualism.

Enough introduction!!! - Now Butter By the Fishes!!!

I will explain the basic functions of the Phoenix, so that you get an impression of how the LFS-K1 stereo filter was implemented along with its functions.

PWM control

The Phoenix motherboard gives out a Pulse-Width-Modulation (PWM) with 39 KHz to control the Cut, Res, VCA and CV 1/VC2. All of the signals and the output of the oscillator are PWM outputs. The pulse-pause ratio of the 39 KHz square wave signal, also called the carrier signal, is modulated with the oscillator signal or the LFO/ADSR signal, which is calculated by the CPU. This PWM signal contains the modulated audio or LFO/ADSR signal.

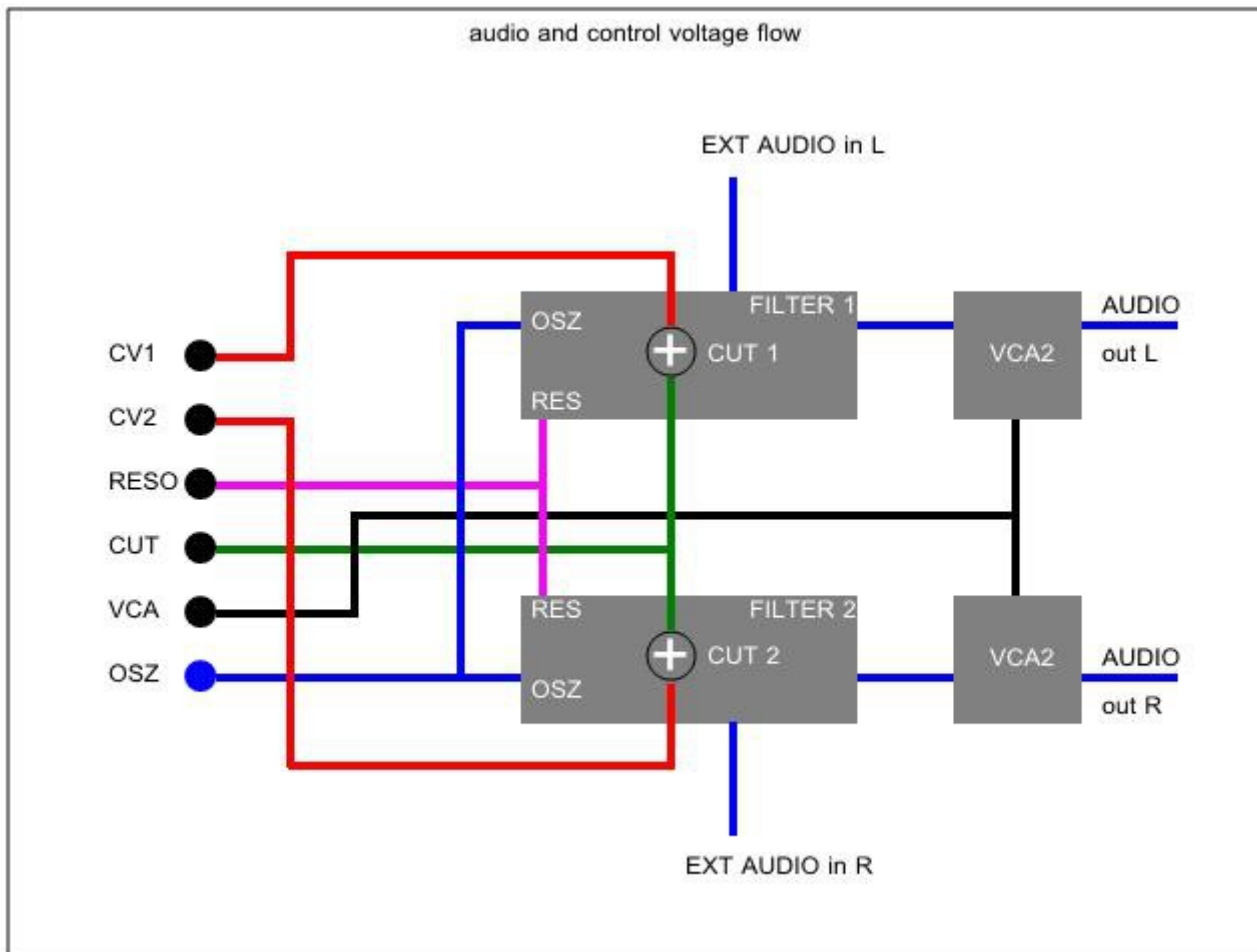
Since the Phoenix motherboard only generates digital signals, and was not designed as a digital to analog converter, the encoded PWM signal (OSC/LFO/ADSR) therefore must be converted to an analog signal. To do this, the digital PWM signal requires an analog filter. The oscillator's signal is reconstructed by the main filter through the ADSR/LFO signals which are significantly low frequency and are supplied with a 6dB filter, $F_0 = 600\text{-}1\text{Kz}$ to smooth the digital signal.

OSC	= PWM output of the oscillator signal
VCO	= PWM output of the VCA envelope (ADSR)
CUT	= PWM output for Cutoff, Filter ENV, filter modulation (LFO)
RESO	= PWM output for controlling the filter resonance
CV2	= PWM output - Cutoff control of the second filter
CV1	= PWM output - Cutoff for controlling the first filter

*** CV 1 and CV 2 are freely programmable outputs and can be selected directly in the modulation matrix of Phoenix as targets.

If, for example, the ADSR is routed to CV 1, the cutoff of the first filter is triggered.

THE LFS-K1 FUNCTION BLOCK DIAGRAM



(Important : Take a look to this function diagram here.)
Let's see how it works....

Preliminary an INFO. All PWM control signals give only positive voltage values. Except for the oscillator, this provides the subsequent circuit in an AC voltage.

The LFS-K1 From the Bottom Up...

First we see the oscillator signal (blue) will be routed as a mono signal to both filters 1&2 simultaneously. The external stereo input is routed in the first and the the second filter. The filter can therefore also process an external stereo signal.

Also, both VCA's (black) are driven by one signal. The cutoff (green) and resonance (purple) values are simultaneously controlled both filters.

Then we come to the CV 1 and CV2 signals. As we can see in the diagram, CV 1 directly controls the cutoff in filter 1. Respectively the cutoff in filter 2 is directly controlled by the CV 2 signal. Internally, the CV1/2 controls are added to the control signal on the common cutoff control voltage.

How am I to understand this???

The Phoenix first gives out a combined CUT/LFO/ADSR control signal via the CUT control on both filters.

This may be unusual but it is not detrimental to the overall sound. It just means both filters are driven simultaneously by the same control signals. If you don't use the Mod Matrix, the

LFS-K1's stereo filter will work like a mono filter. In the Filter menu of the Phoenix the LFO adjust option acts on both filters - but can be switched 'OFF' and instead re-routed to the LFO 1 in the Mod Matrix to CV 1.

!!!AHA you say??? Thus, the LFO 1 controls only the first filter. Then we can route in the Mod Matrix LFO 2 up to CV 2 and... again !!!AHA!!!, both filters are controlled with different LFO's!!! Got It???

This same adjustment works with the ADSR controls. We simply set the ADSR values in the Phoenix's Filter menu to '0'. and instead, route the ADSR1 in the Mod Matrix on CV 1 and because of the new LFS-K1 stereo design, the second ADSR to CV 2.

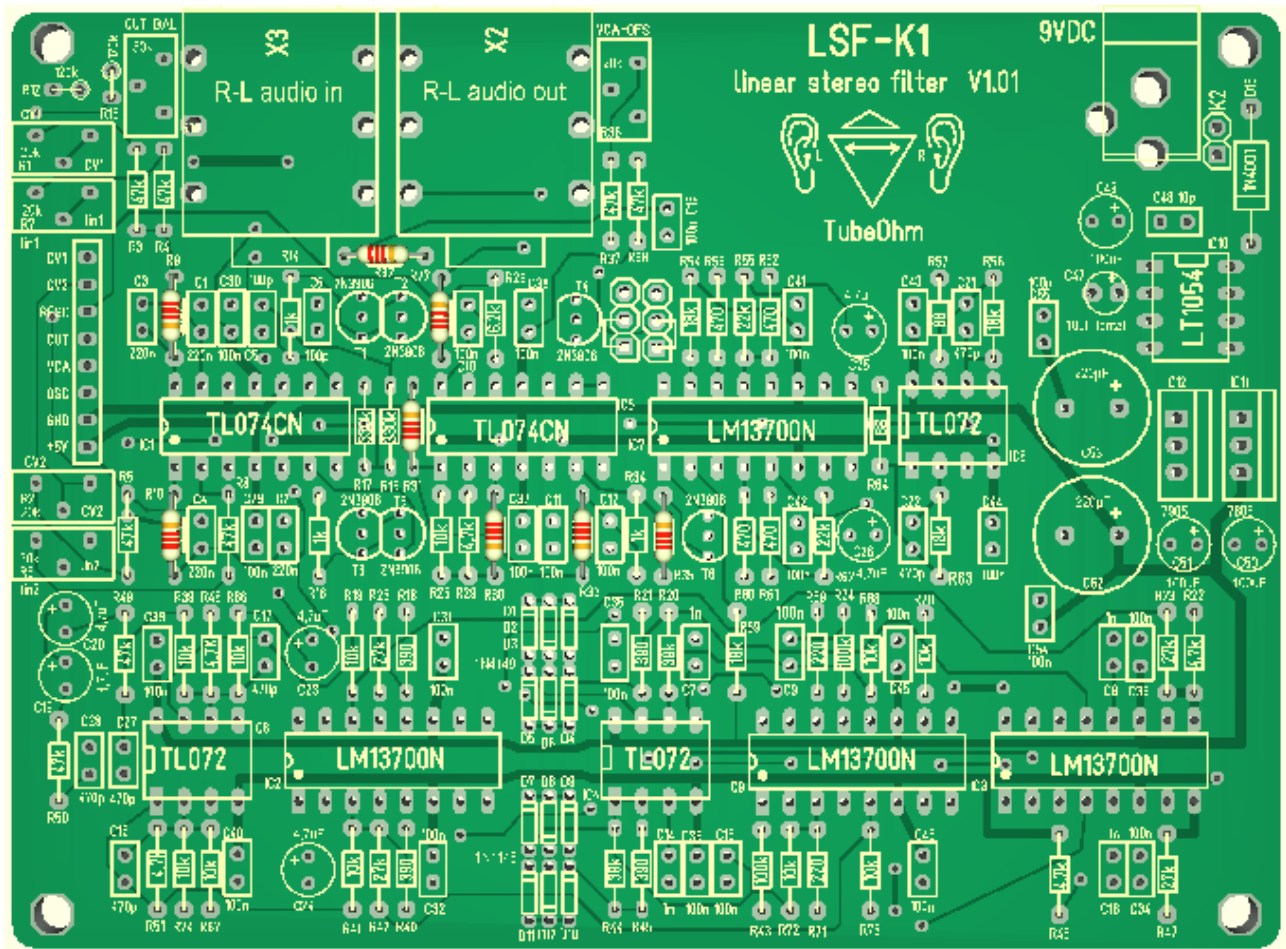
Another possibility could be LFO 1 to CV1 and ADSR 1 to CV2 or, or, or. Of course you can also route multiple sources to the CV outputs. The possibilities go on and on and on....

I hope you now understand how flexible having a stereo filter can be especially with manipulation via the Mod Matrix.

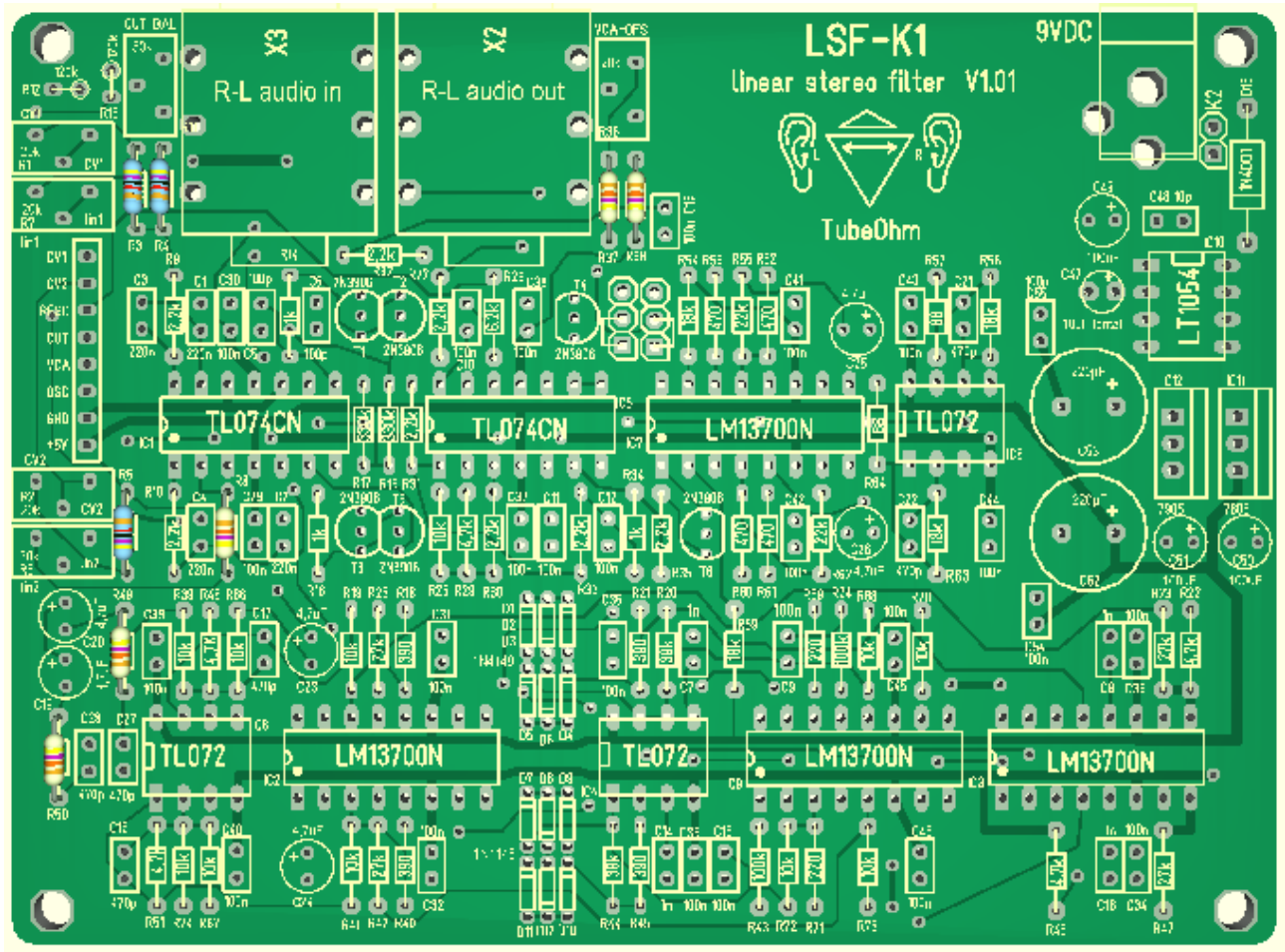
As we can see numerous combinations can be achieved, and with the global parameters (CUT, ADSR, LFO in the filter menu) you can still manipulate both filters directly together. That is all the time we have for the basics of the LFS-K1 filter board.

We begin with the resistors R9,10,30,31,32,33,27,35

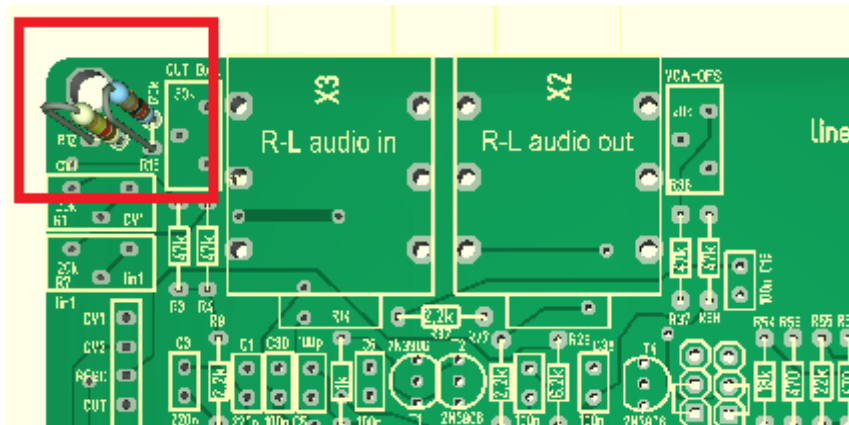
8 x 2,2Kohm, color code: red,red,black,brown,brown



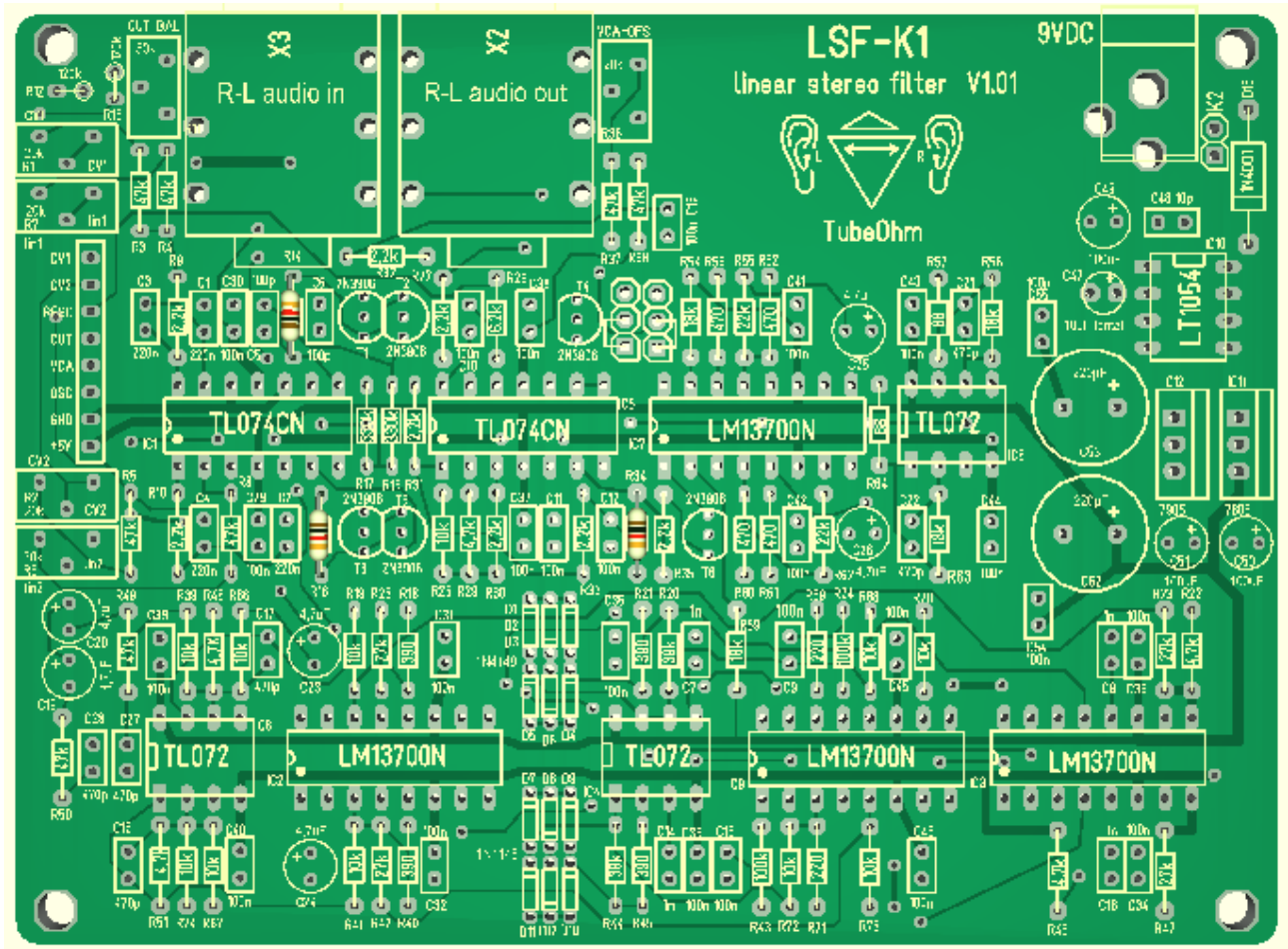
8 x 47 KOhm resistors R3,4,5,8,37,38,49,50 color code: yellow,purple, black, red, brown



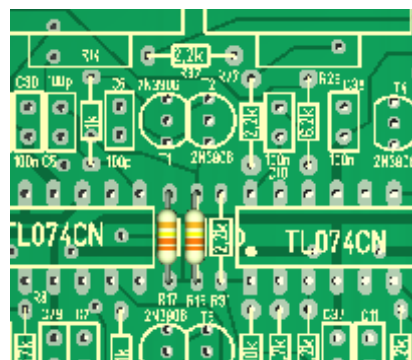
2x 120 kOhm, standing , R11,12 color code : brown,red,black, orange,brown



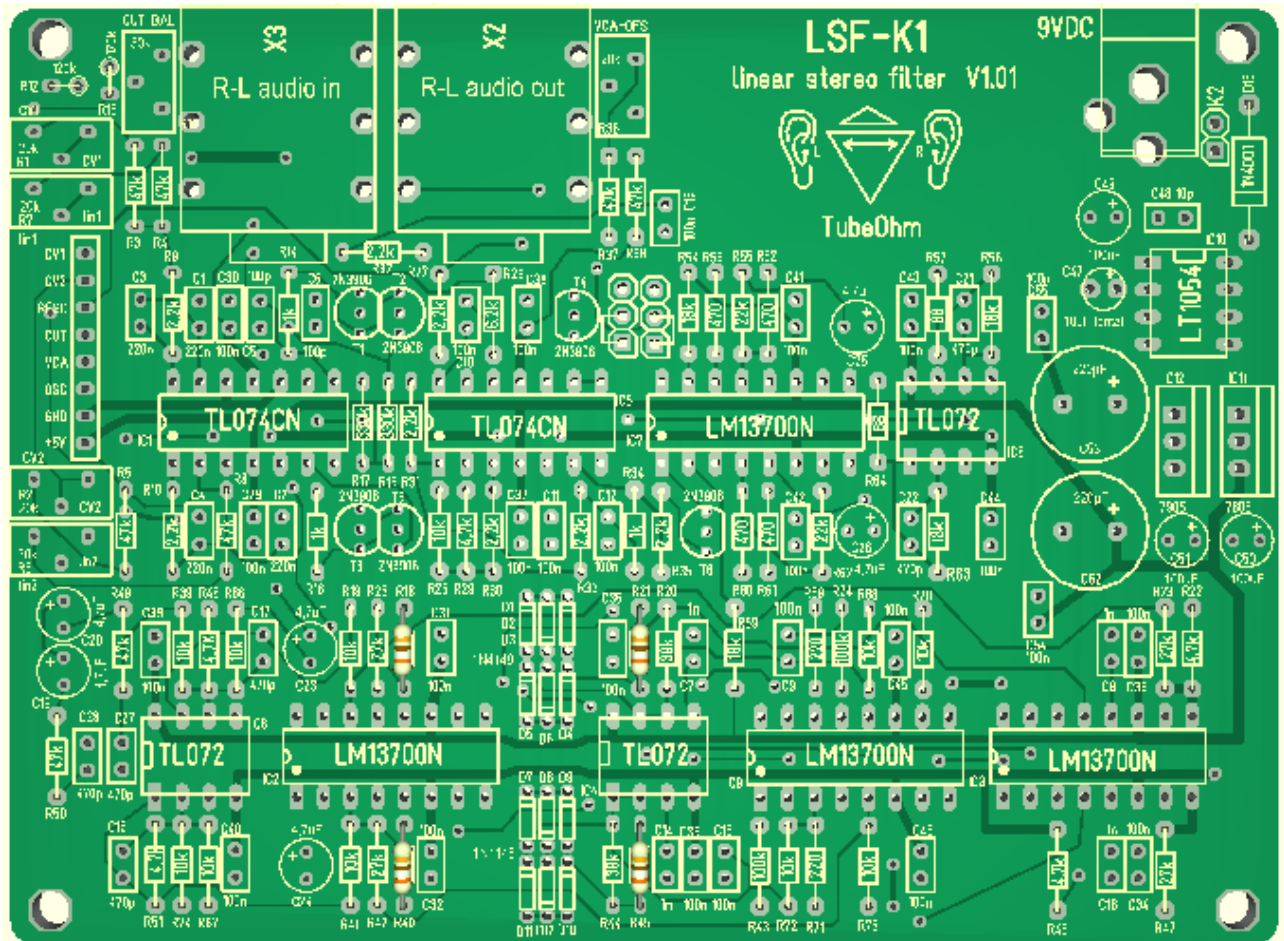
3x 1 Kohm R14,16,34 color code : brown, black, black,brown,brown



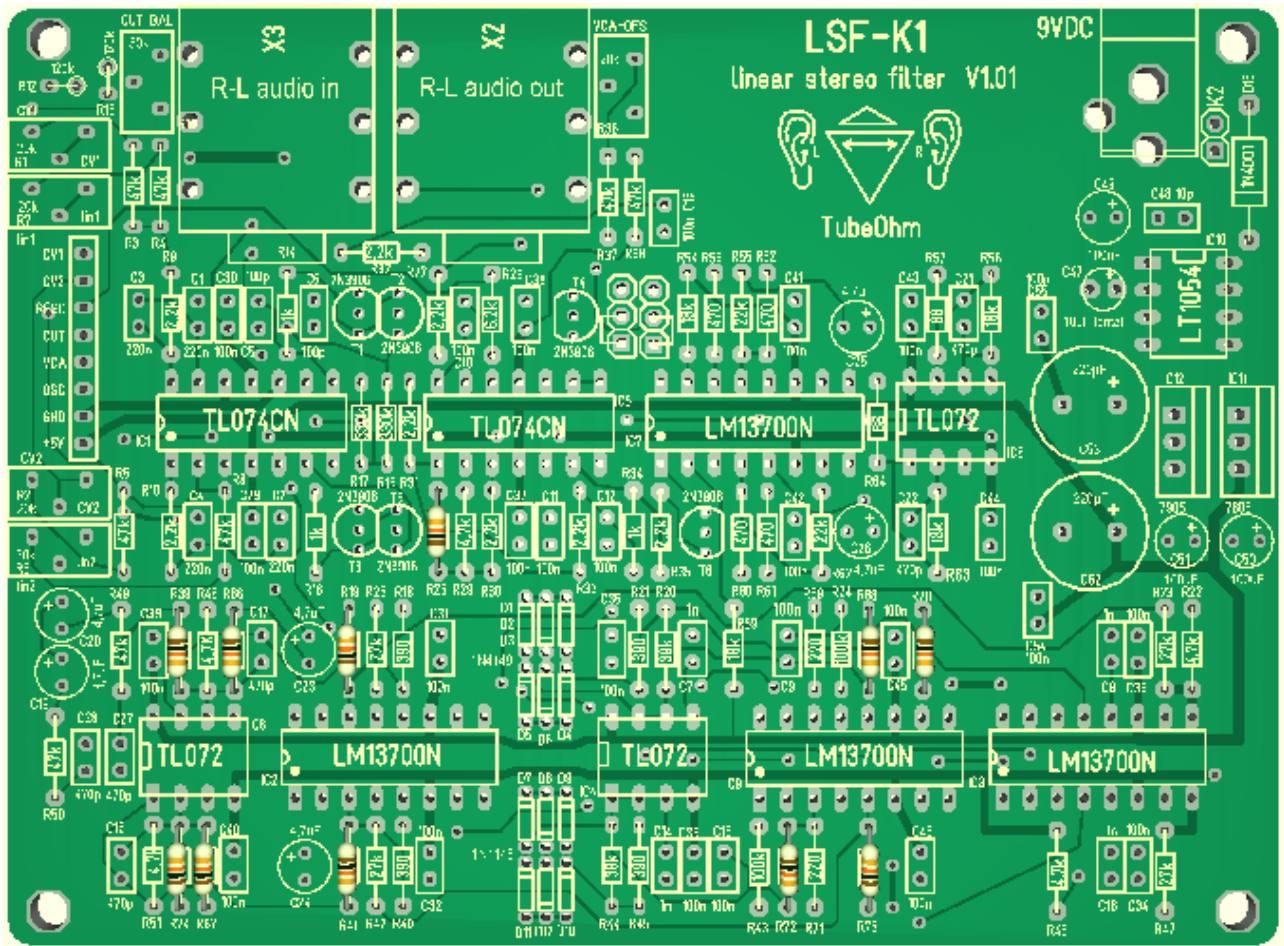
2 x 330 Kohm R15,17 color code : orange,orange,black,orange,brown



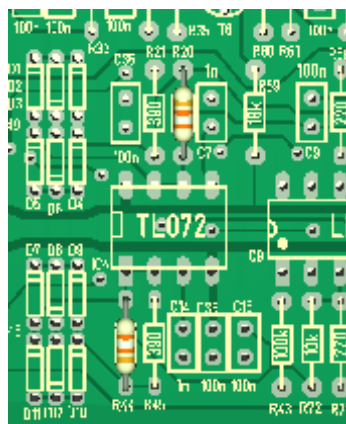
4x 390 Ohm R18,21,40,45 color code : orange,white,black,black,brown



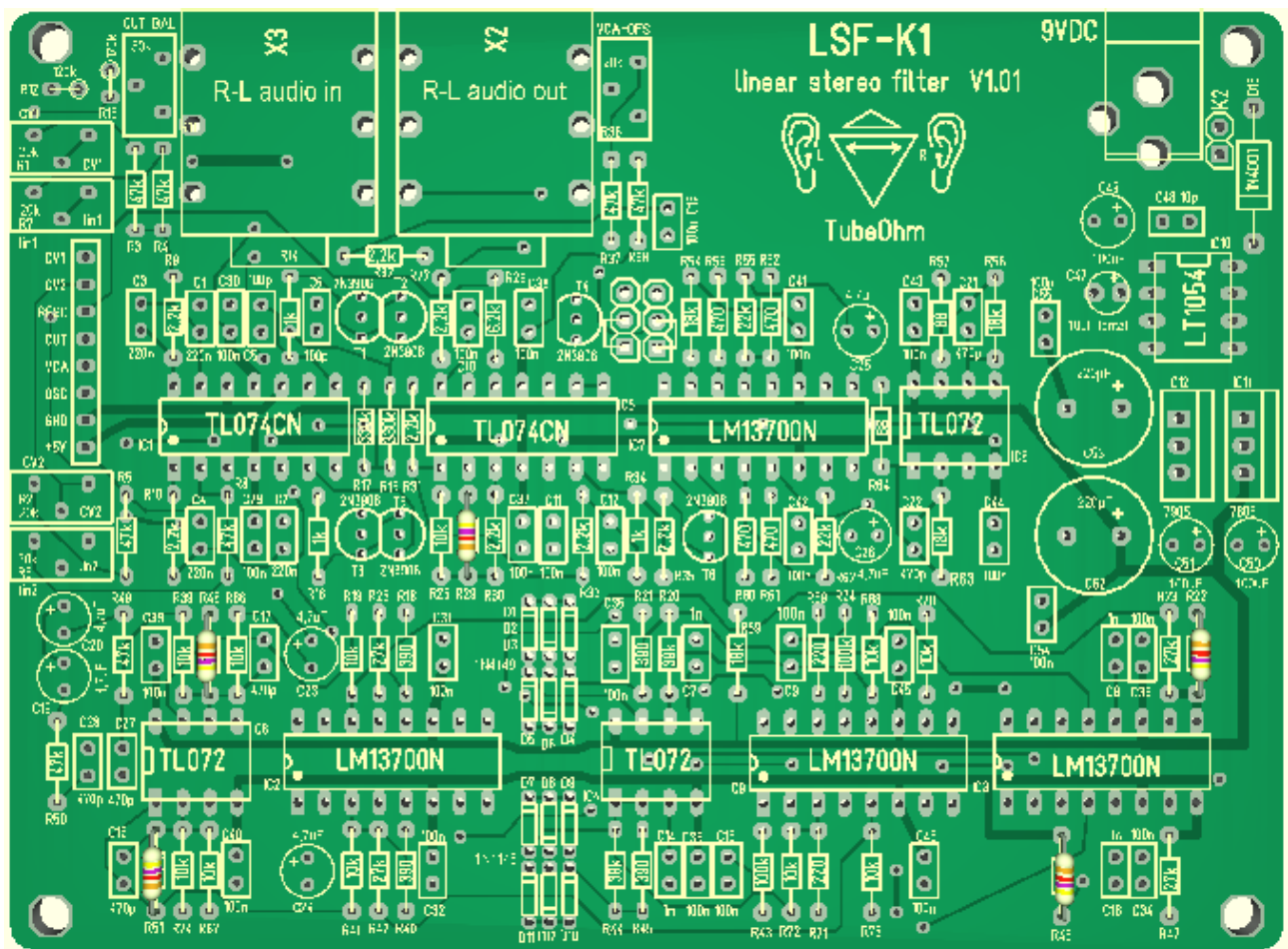
11 x 10 Kohm R19,28,39,41,66,67,68,70,72,73,74
 color code: brown, black,black,red,brown



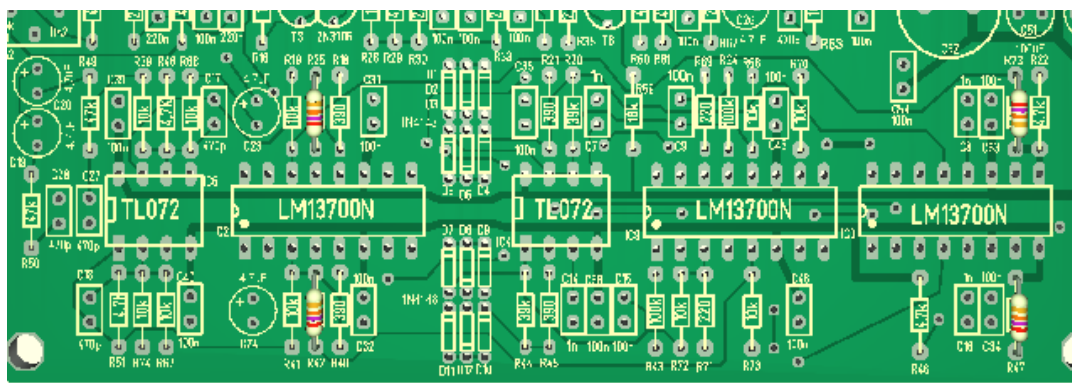
2x 39k Ohm, R20,44 color code:orange,white,black,red,brown



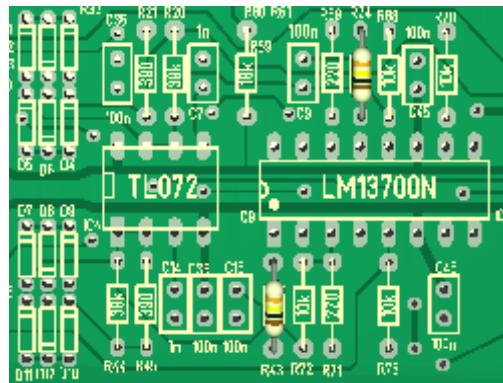
5x 4,7 Kohm R22,29,46,48,51 color code: yellow,purple,black,brown,brown



4x 27k Ohm , R 23,25,42,47 color code: red, purple, black,red,brown

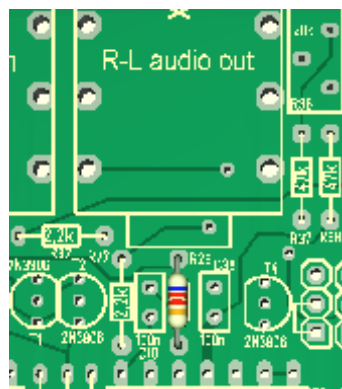


2 x 100 K ohm , R 24,43 color code: brown,black,black, orange,brown



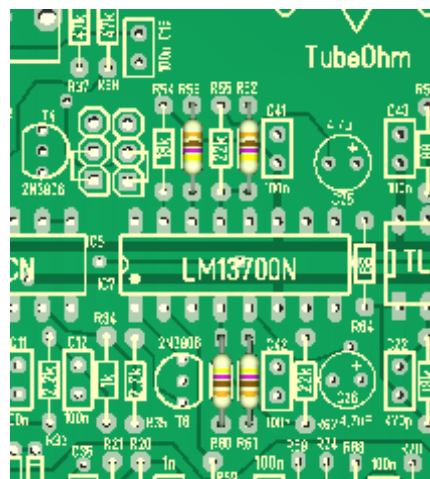
1x 6,2 Kohm , R 26

color code: blue,red,black,brown,brown



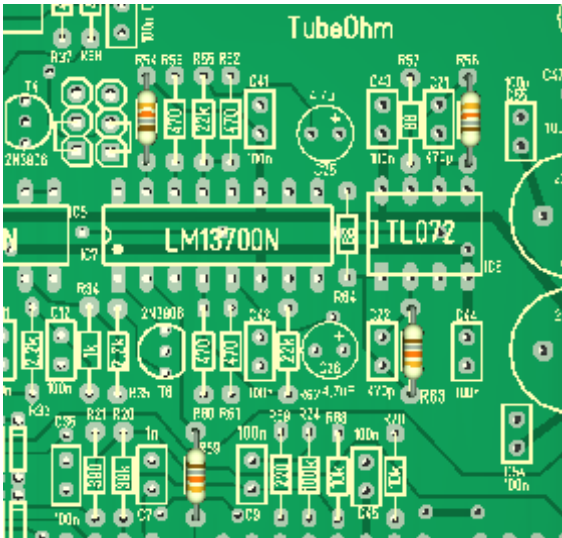
4x 470 ohm, R52,53,60,61

color code:yellow,purple,black,black,brown



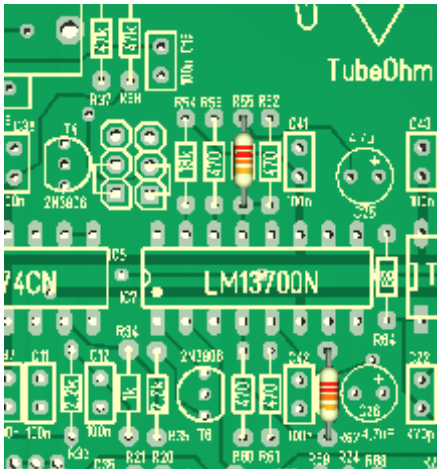
4 x 18 Kohm, R 54,56,59,63

color code: brown,gray,black,red,brown



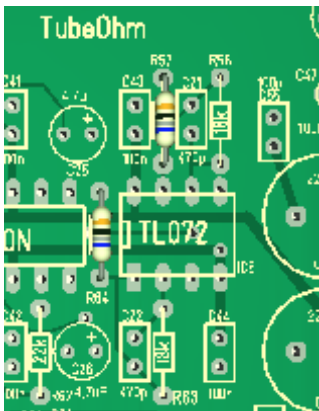
2x 22 Kohm, R 55,62

color code:red,red,black,red,brown

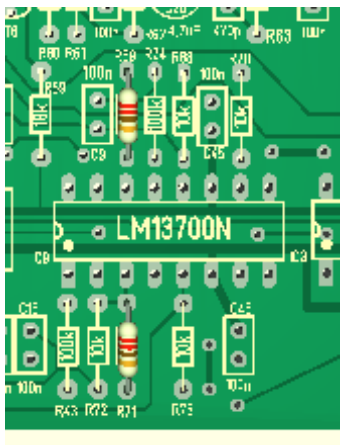


2 x 68 ohm R57,64

color code:blue,gray,black,gold,brown



2x 220 ohm ,R69,71

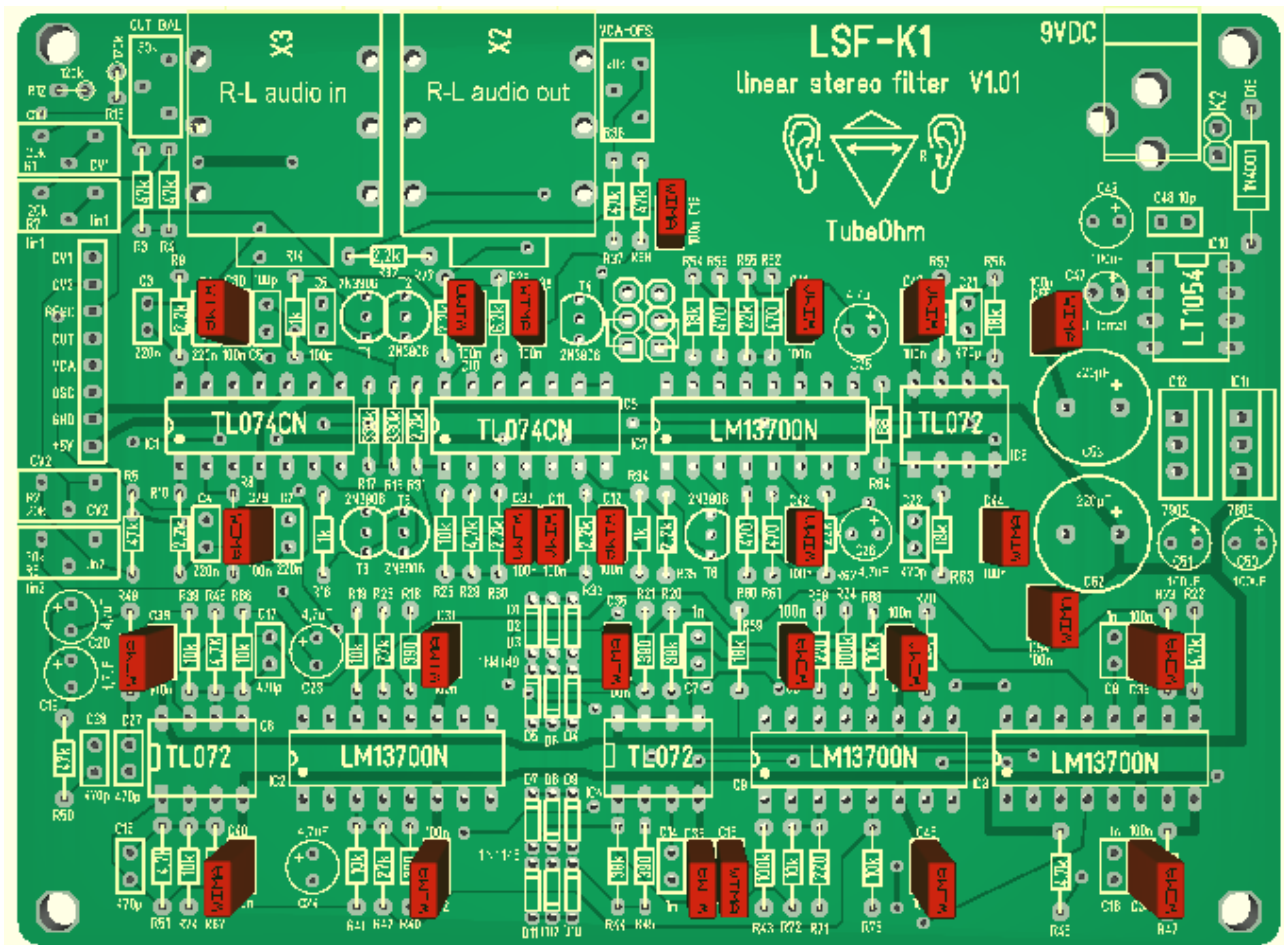


These were the resistors.

Now the capacitors

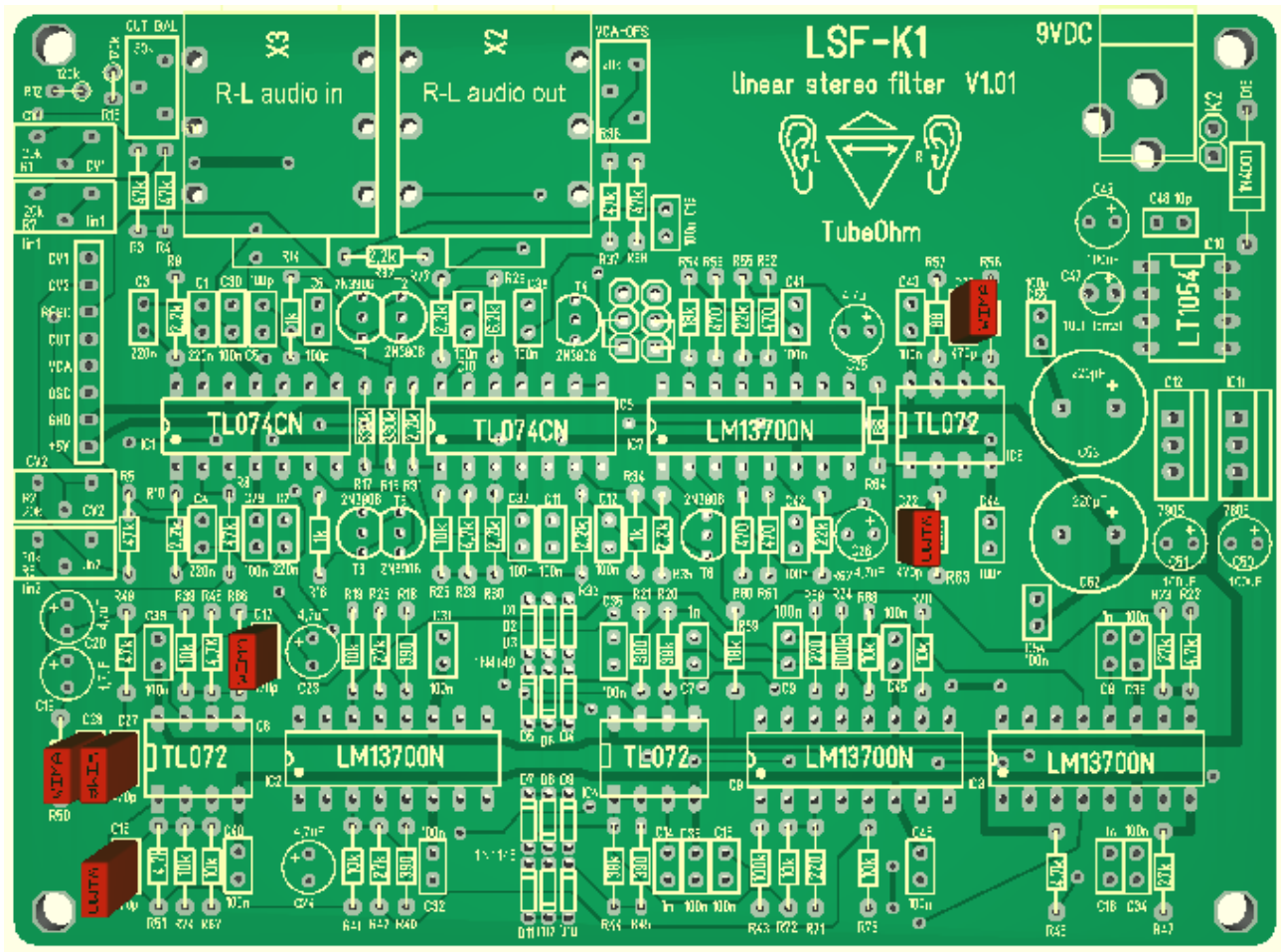
26 x 100 nF **Label 104**

C 9,10,11,12,13,15,29,30,31,32,33,34,35,36,37,38,39,40,41,42,43,44,45,46,54,55



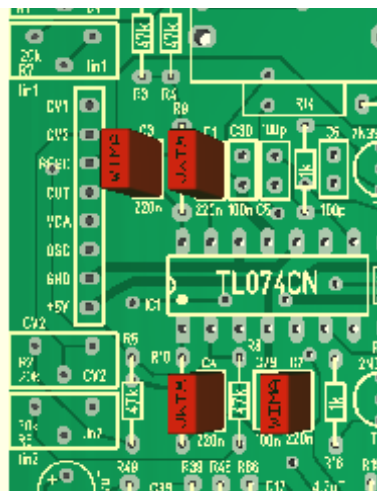
6 x 470 pF, Label 471

C17,18,21,22,27,28

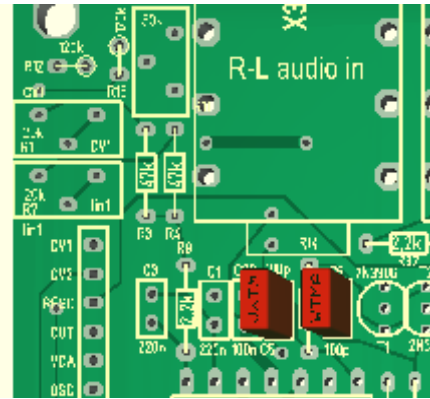


4x 220nF Label 224

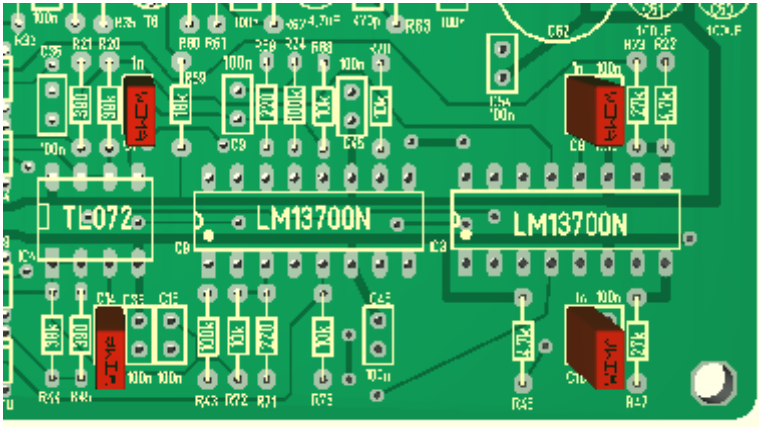
C1,2,3,4



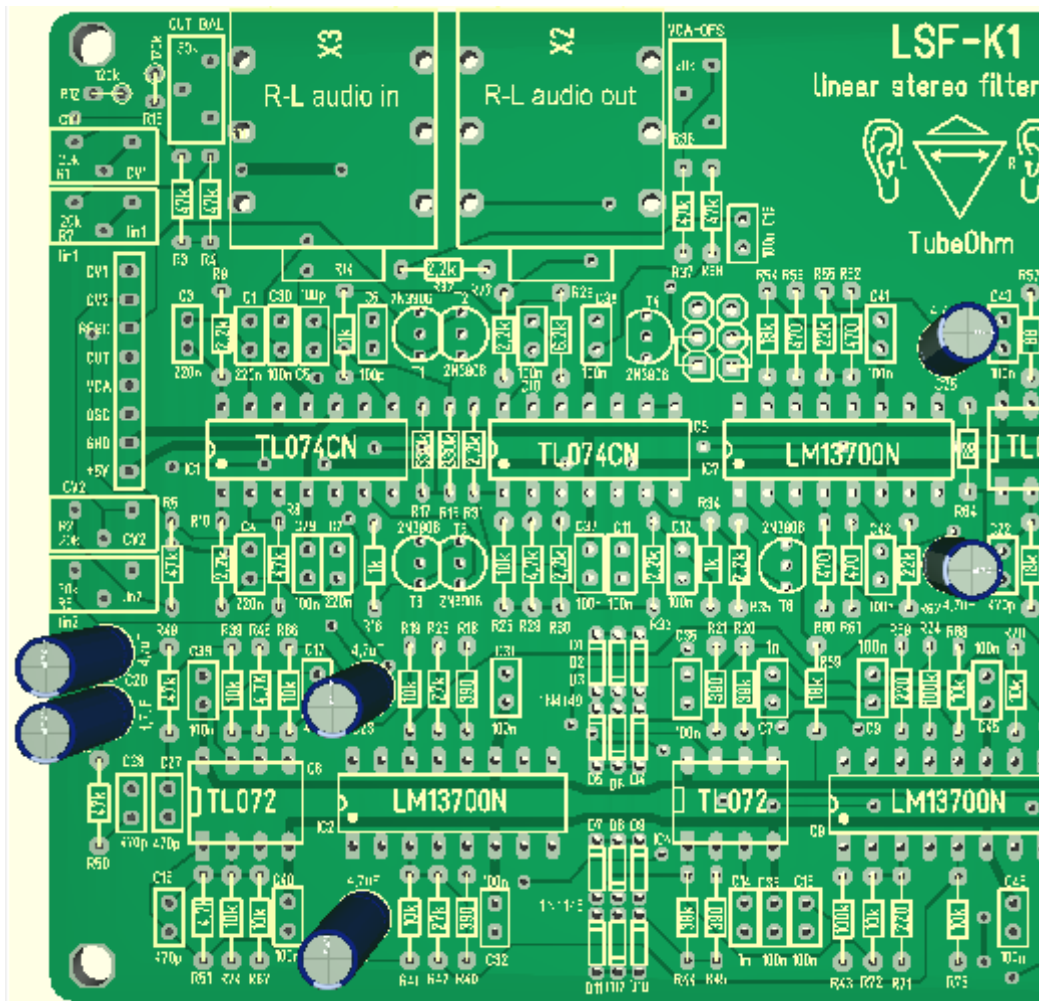
2 x 100 pF Label 101
C 5,6



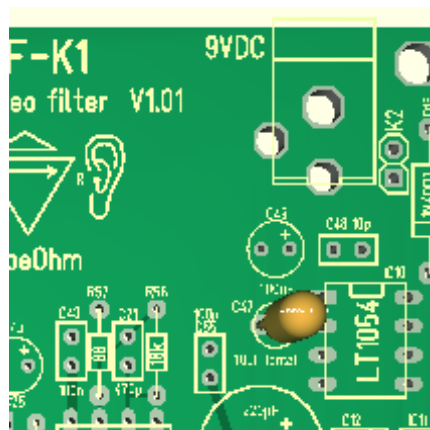
Filter-capacitors 1 N Label 1000 (Folie-WIMA)
C 7,8,14,16



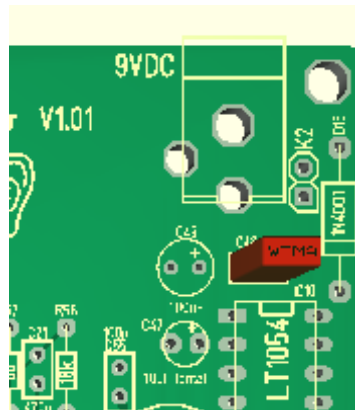
6 x 4,7 NP Elko, non polarized/keine Polarität
 C19,20,23,24,25,26



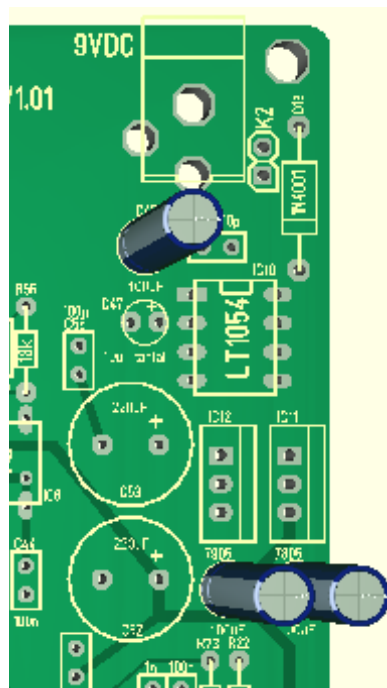
Tantal 10 uF polarized for the LT1054 / auf die Polarität achten !!
 C47



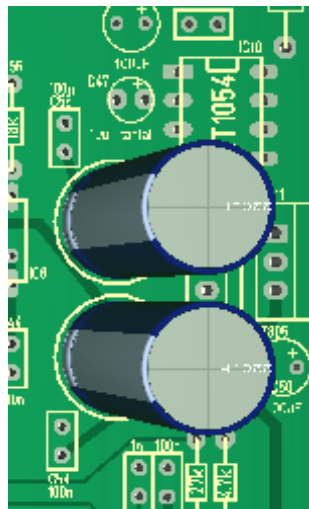
1x 10 pF Label 10pF
für den TL 1054
C 48



Elko **polarized** 100uF for the power supply / **auf die Polarität achten**
Attention , -negative marked
C49,50,51

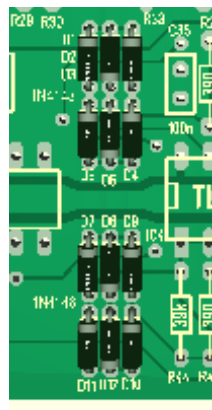


2 x 220 uF Elko , **polarized – negative marked /auf die Polarität achten**
C 52,53



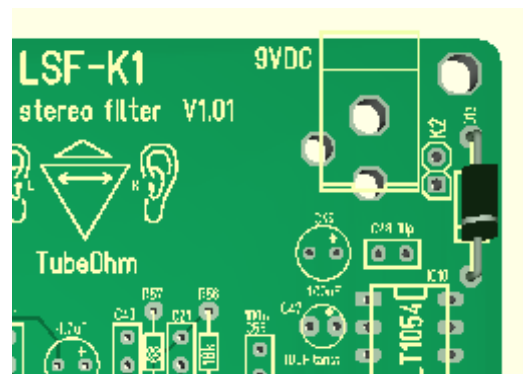
those were the capacitors.
Now the diodes.

12 x Diode 1N4148 -polarized. D 1....12/**auf die Polarität achten !!!**



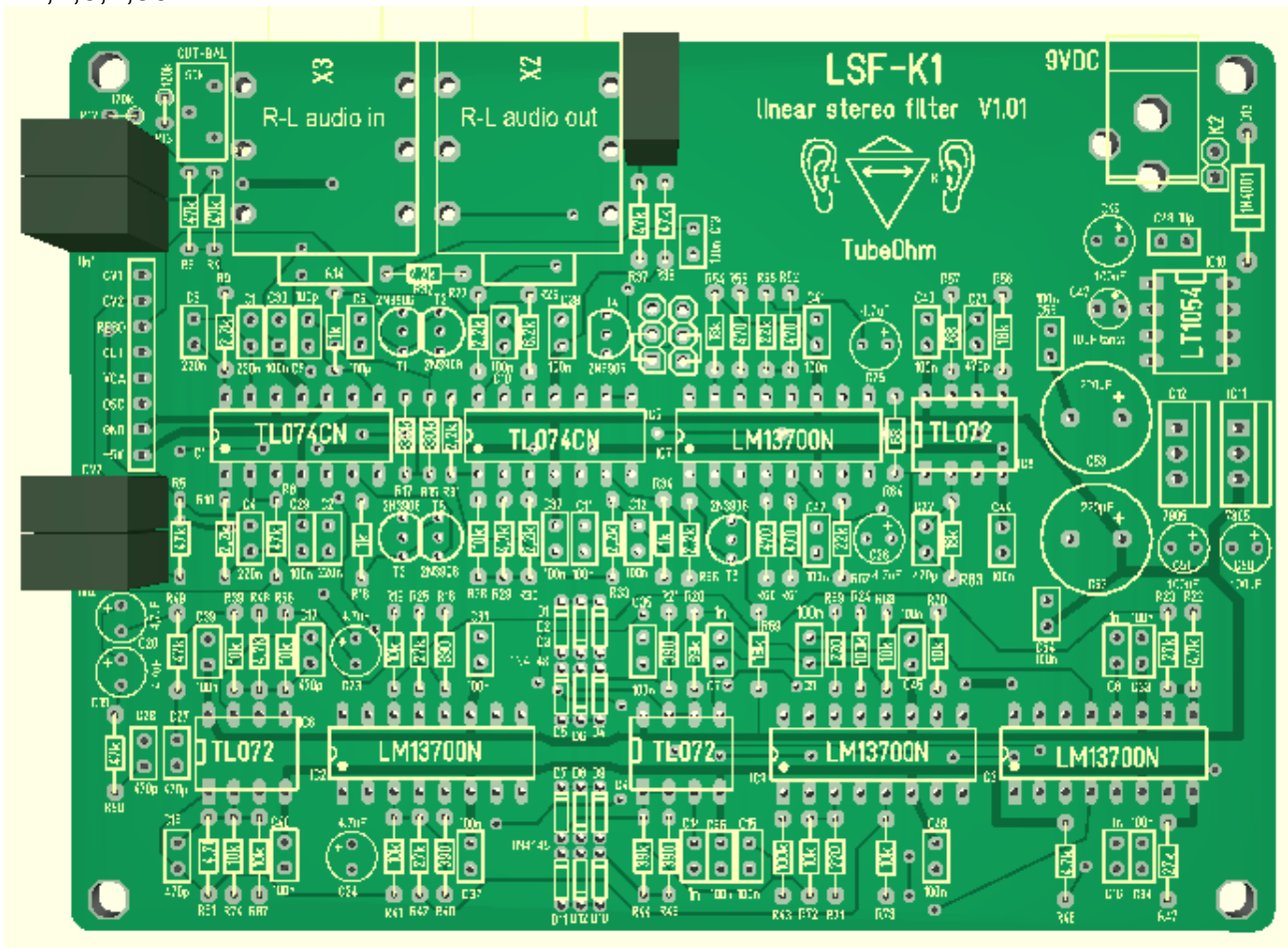
1x 1N4001 -polarized/ **Verpolarungsschutz/auf die Polarität achten**

D13



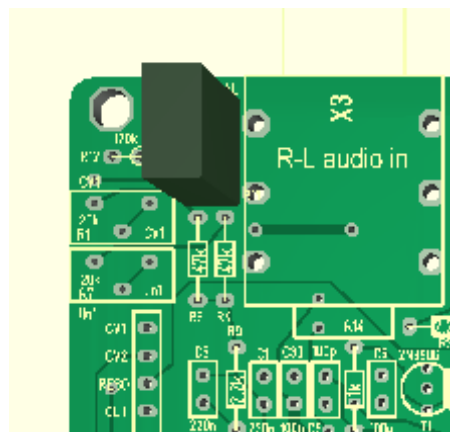
5 x 20Kohm potis/pots

R1,2,6,7,36

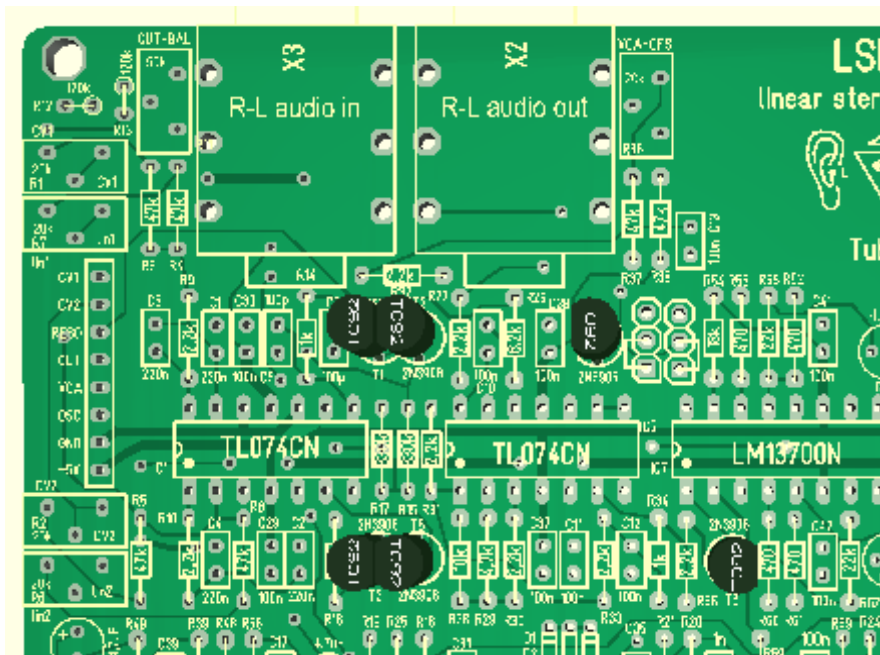


1x 50 Kk ohm poti /pots

R11



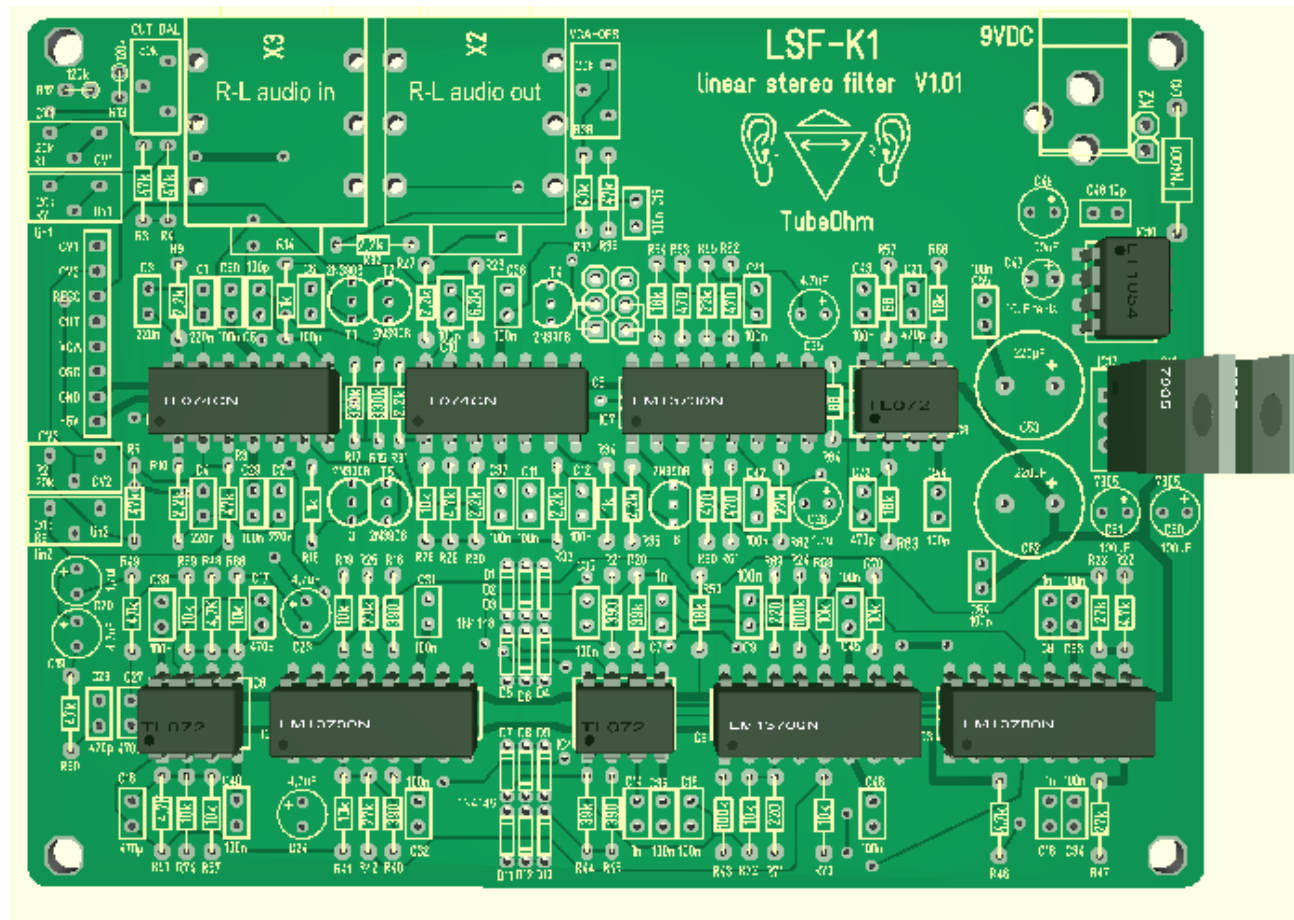
Now, the transistors
6 x 2N 3906, T1,2,3,4,5,6



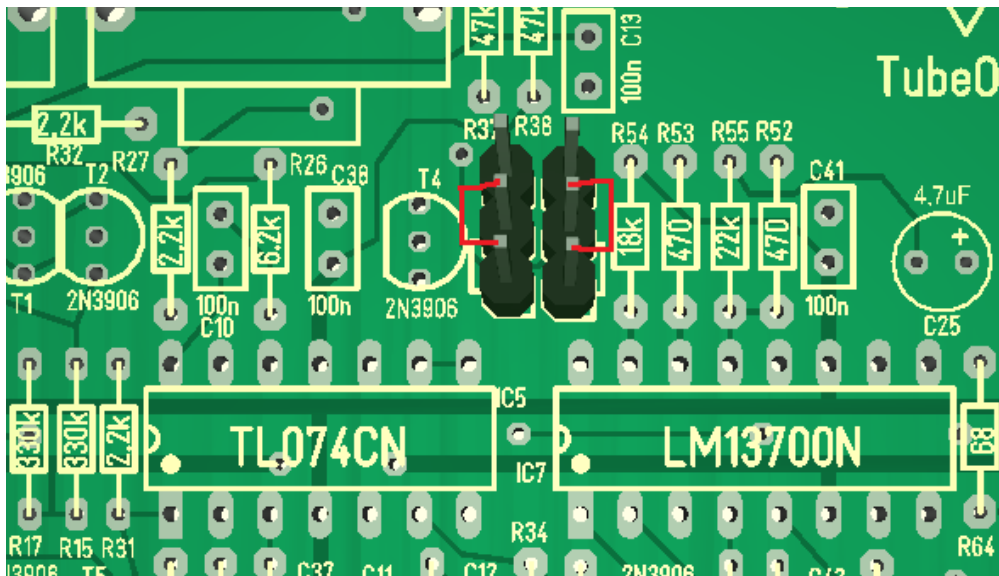
Now the sockets and the voltage regulator can be soldered.

!!!Attention!!!

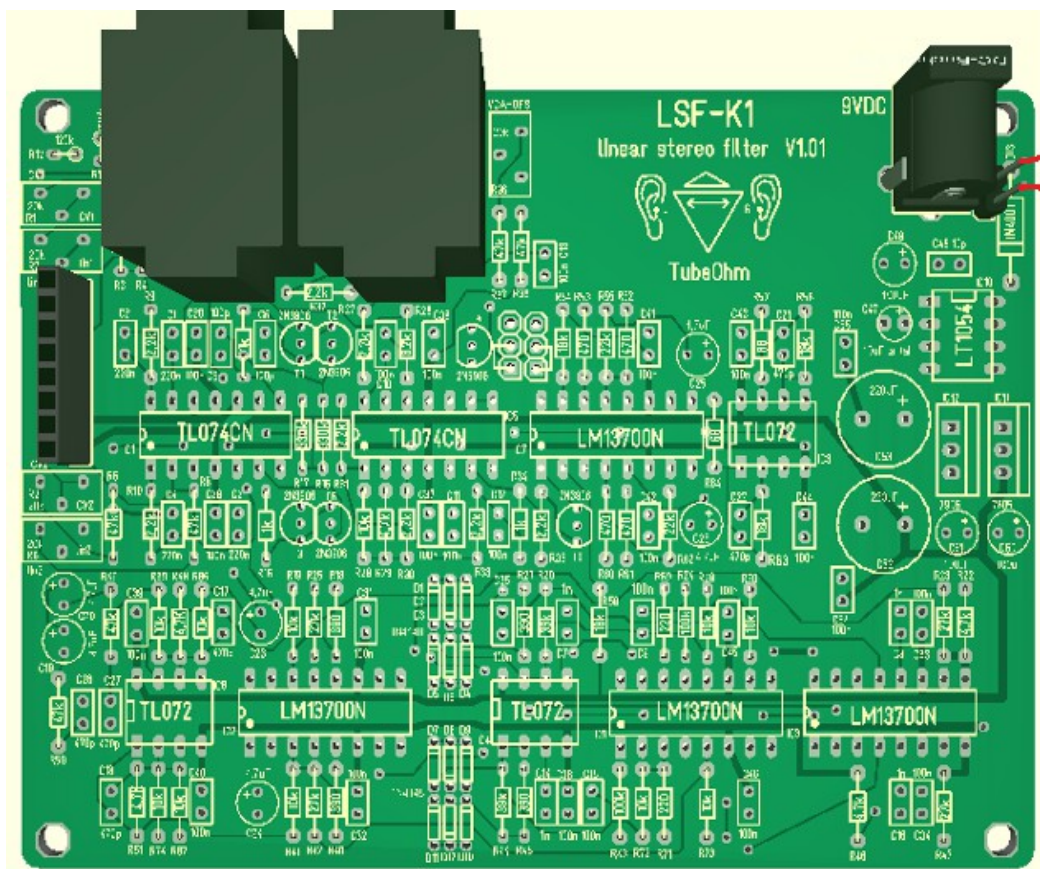
All IC's are provided with a directional marking. The sockets are also marked directionally. Be sure that you solder it into the right orientation.



On the Filter Board a stereo 2x10Kohm Pot can be connected to regulate the stereo output volume. Since the LFS-K1 is a stereo filter this would be a practical install. If you do not have a stereo potentiometer to install, then a 10k mono audio taper pot can be used and the stereo contacts need to be be bridged. To do this make a short between the two points.



Mechanical elements - Stereo audio jacks, Arduino header, 9 volt socket and K2. At K2, the on / off switch can be connected - or the two contacts can be easily bridged.



That should complete the soldering. Once you feel you have completed populating the LFS-K1 it is time to inspect the finished

board...

Are all the resistors and capacitors installed?

Are all the polarities in the right direction?

Are all the IC sockets in the right direction?

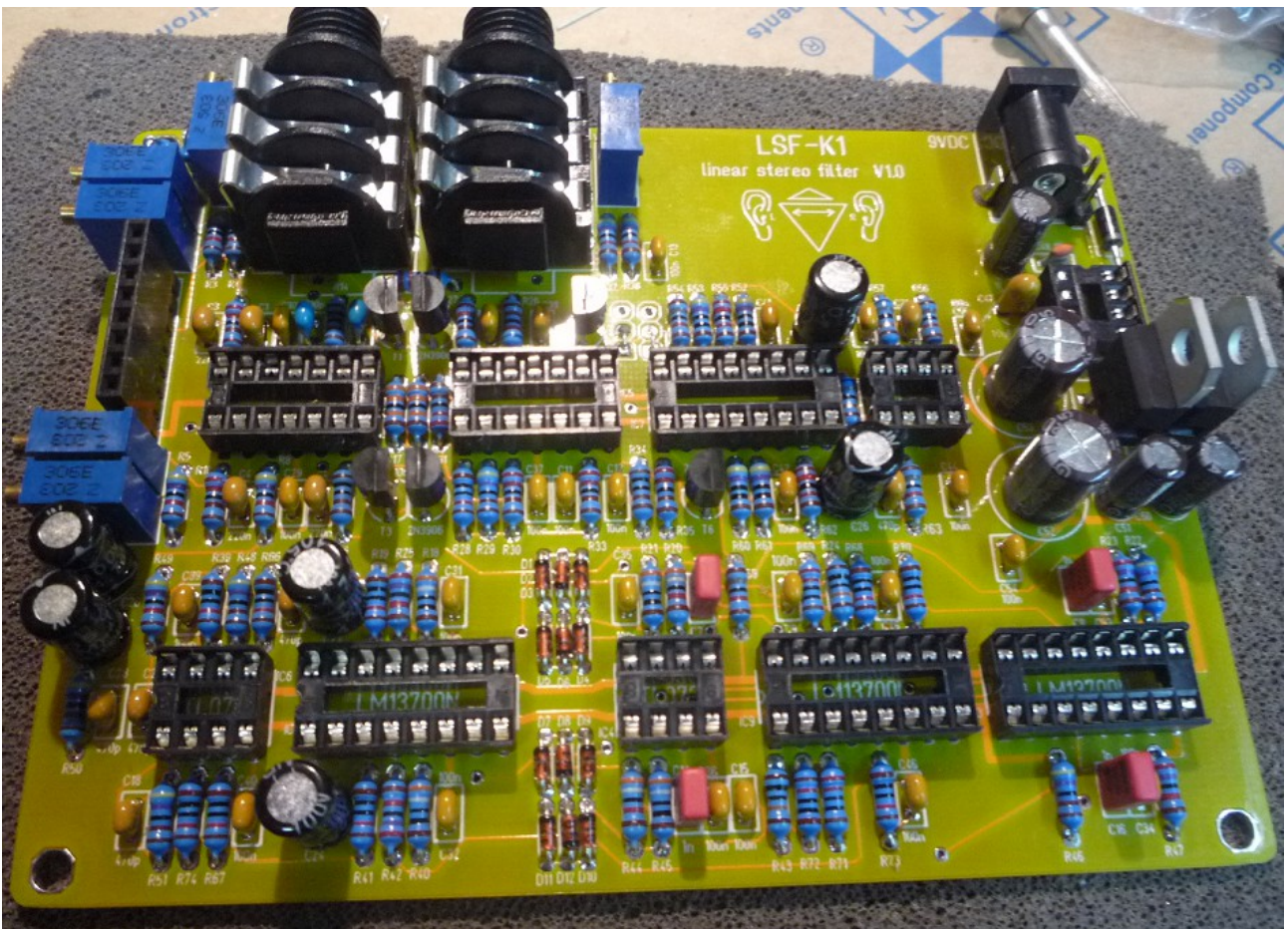
Is the contact K2 bridged – shorted?

Are the contacts for the volume pot connected and/or shorted? - Otherwise no sound comes out!!!

Inspect once again...

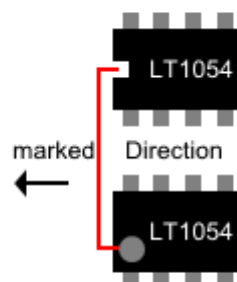
If everything is ok the main work is done.

The board should now look like this:



Now we can insert the LT 1054 into the circuit board through its corresponding IC socket. Pay attention to the orientation. Depending on the manufacturer, the IC may not be marked but there is a circle on one end.

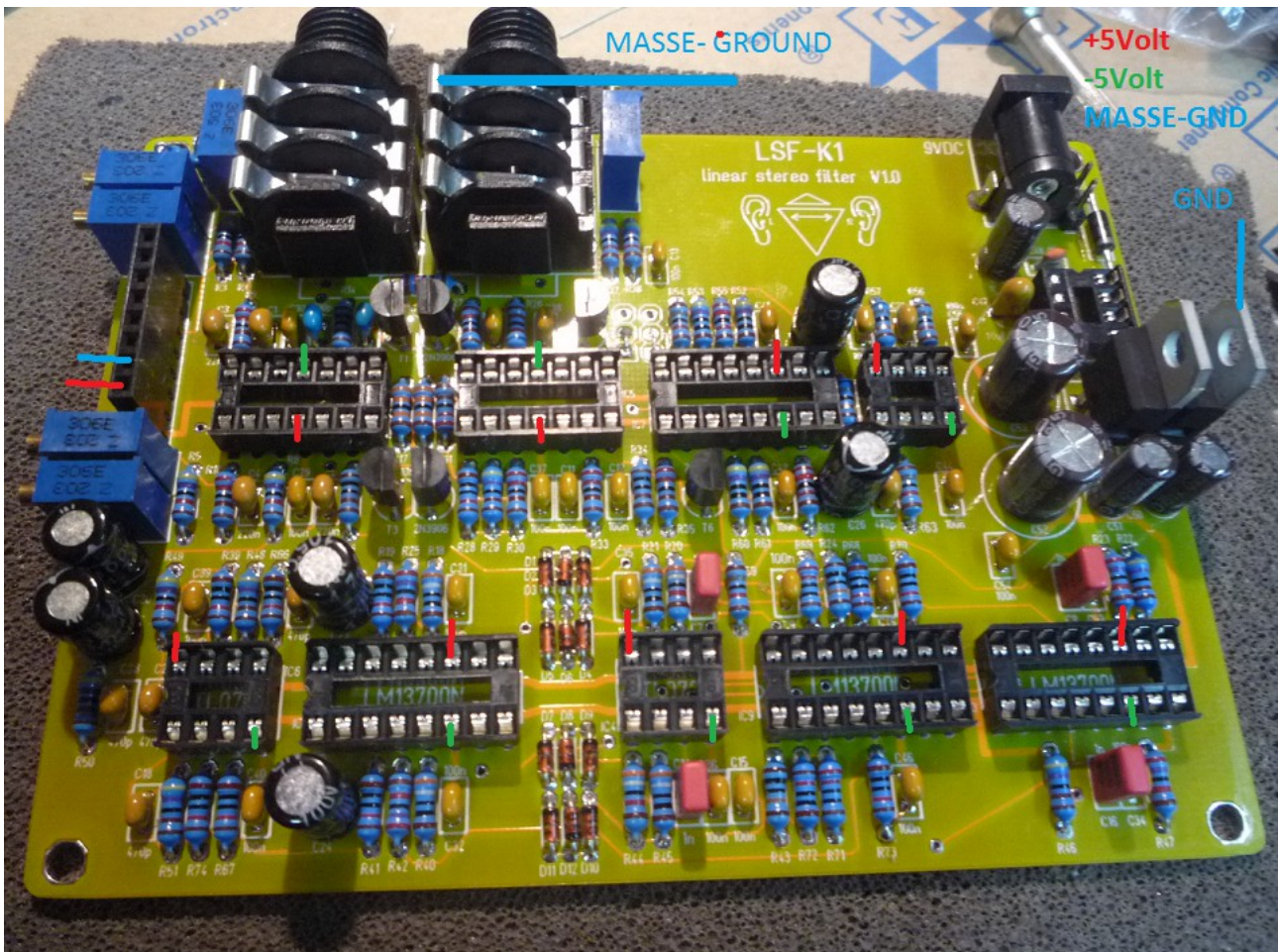
This will determine the correct orientation on the IC LT 1054.



Once installed, we can now check whether the voltages are active.
It is time to plug the 9V 600mA DC power supply into the 9-volt socket of the filter board.
Anything smoking!?!?! (joke!)

OK....

Now we need our multimeter to measure the voltage + 5V and - 5V .
For this you will only need one GROUND. I always use the GROUND off an audio jack.
Alternatively, you can take the ground off the cooling fin from the 7805.



red = +5Volt, (4,9..5,1)
green = -5Volt, (-4,9...-5,1)
blue = Masse-GND

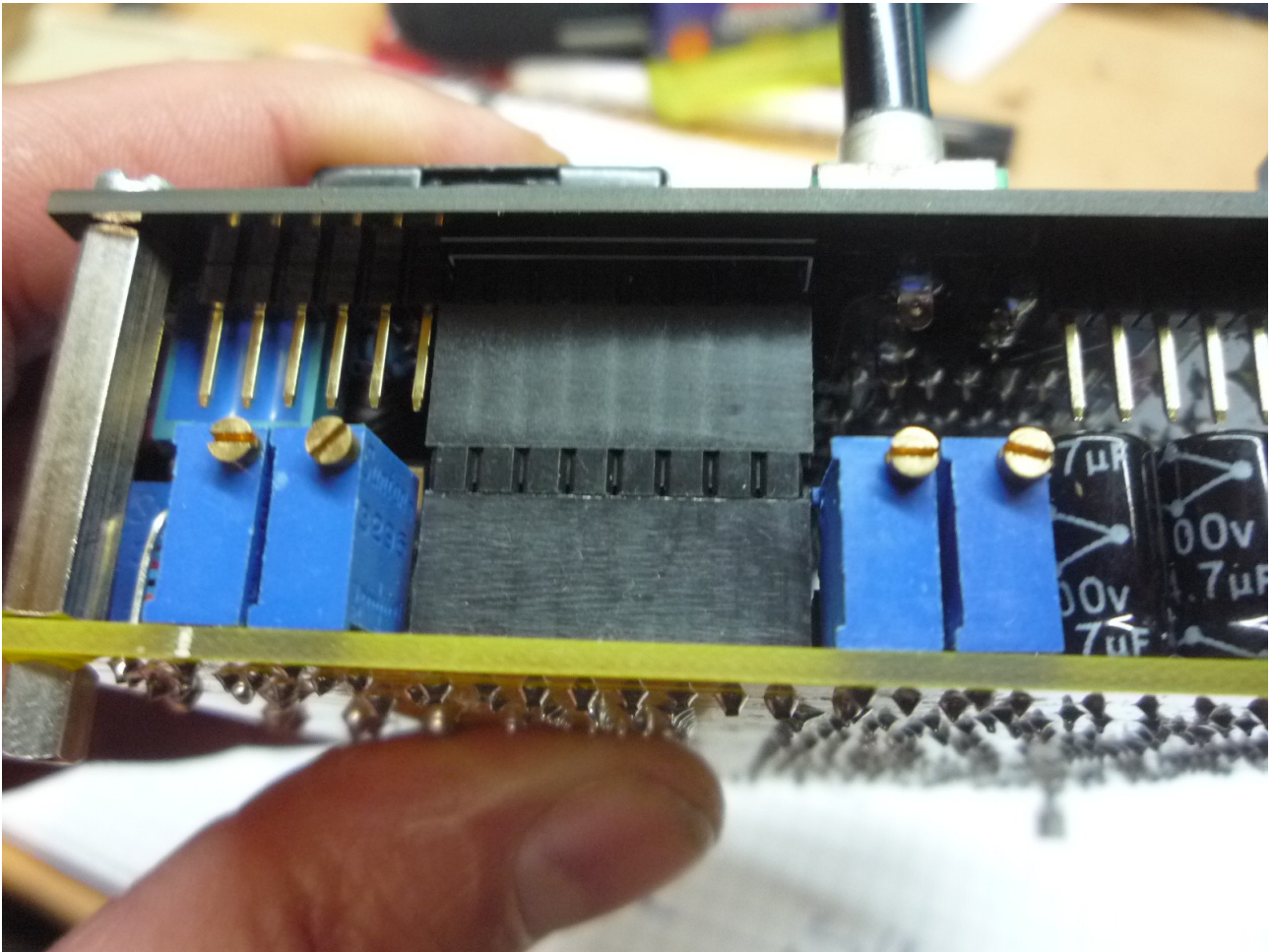
Are all voltages showing now? The voltages should be displaying in the range of about +/- 4.9 to +/- 5.1 volts.

So now comes the exciting moment!!!

Disconnect the power supply from the filter board in order to place all IC'S into their sockets.

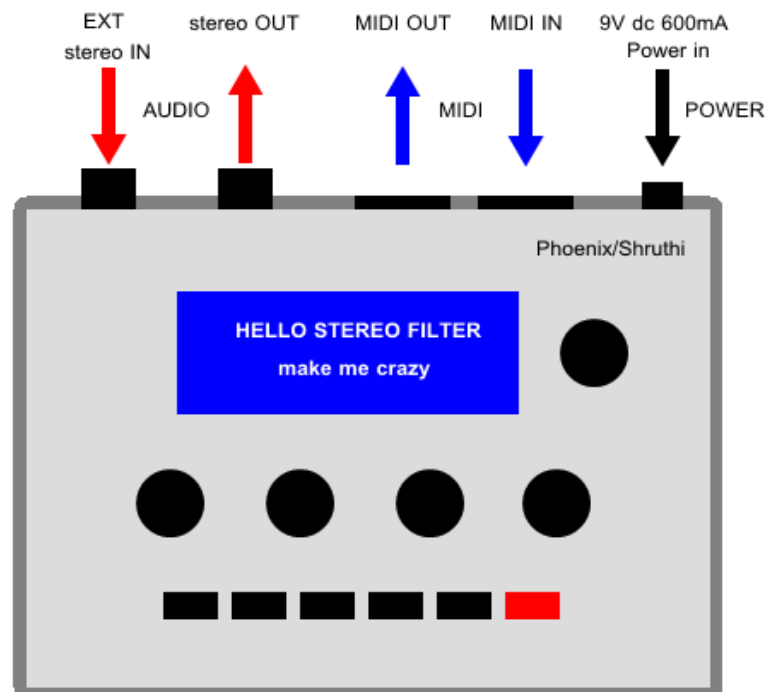
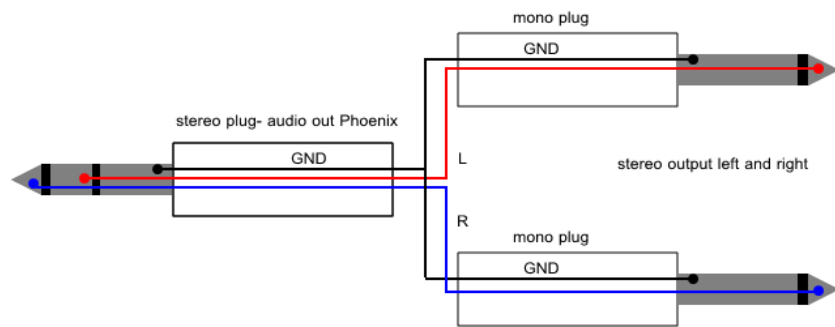
!!!Pay careful attention to the identity and orientation of each IC and check 2X or even better 3X before inserting them!!!

Now we connect the filter board with the accompanying Arduino header to the Phoenix motherboard. The Arduino header included can still be cut approx. 2mm.



On the LFS-K1 filter board there are 2x 6mm stereo jacks. The left audio jack is for the external stereo input signal and the right stereo jack is now for stereo output!!!

In order to hear the stereo signal you need a 6.3mm stereo jack. A headset does well. Make sure all the connections are tight and sound and plug in your power adapter.



When you are ready, turn on your Phoenix/Shruthi.
Does the display turn on? Do the switches and knobs respond?
If so switch the Phoenix software to 'LP' Mode.
Connect a MIDI keyboard via a MIDI cable to the MIDI IN of the Phoenix LFS-k1 filterboard.
Press a key on the keyboard once to hear if sound comes out.

If not... people!!! have you bridged the connector for the stereo volume potentiometer???
Aha!!! Now it should work.
Congratulations - it's done. The LFS-K1 filter works!!! Juhuuuuuu

!!!And now the **HARDCORE** part – the **FILTER** calibration!!! But ... don't panic!!!

Basic Settings:

Plug in the Phoenix, hook up a MIDI controller and connect to a mixer/amplifier.

Turn everything on and test that everything is working properly.

A key must be played and held on the keyboard.

If you don't hold the key, the VCA closes and you hear nothing.

---Don't laugh!!! I've received emails with questions such as, 'there is no sound - how should I calibrate?' I ask >> 'have you also pressed a key on the MIDI keyboard?' >> 'Ahh No?!'---

Well folks, a synthesizer **MUST** play a sound so you can hear something - even during calibration!!!

For calibration I recommend using our Filterkalibrator because it is easy to work with. But you can also use a guitar tuner or a frequency counter to calibrate.

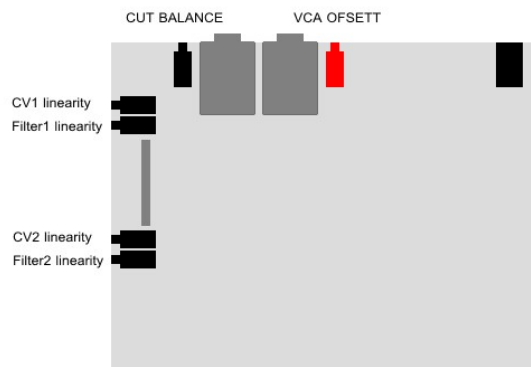
Play 3 octaves on the keyboard, for example C3, C4, C5.

It is important that the resonant frequency always doubles at a one octave jump.

Cutoff 32 is a – round about - chosen value .

You will get good results if you tune the cutoff at a lowest frequency of C3 =100Hz , then C4 must be 200Hz and C5 is then 400 Hz.

First, the **VCA CALIBRATION**.



The VCA must be calibrated in this way... When closed completely it will receive the resonance signal from the filter board.

First of all, the resonance of the filter needs to be turned up to 64 on the Phoenix (don't play a key at this time).

The resonant frequency should be at an audible range at about 1 kHz.

If there is a small whistle, the VCA is not completely closed.

Now you can adjust the potentiometer R36 (VCA OFFSET) so that there is no whistling.

Due to component tolerances, it can occur that one channel is completely silent, but the

other channel transmits a low whistle from the feedback of the filter. In this case, exchange the LM 13700 (IC7) with another LM 13700 from another LM 13700 socket. (For example, switches IC 2 or IC 9 with IC 7)

If both channels are silent, the VCA calibration is ready.

Filter BALANCE CALIBRATION

Here first the basic steps.

- a :) filter 1 normal linearity volt per octave =Poti R7
- b :) filter 2 normal linearity vold per octave =Poti R6
- c :) tune both filters to the same freuency = Poti Cut-Bal
- d :) filter 1 CV1 linearity -Poti R1
- e :) filter 2 CV2 linearity -Poti R2

Why is this so?

Both filters should run at the same frequency while the same octave is being played. Therefore both filters can be played with CV1/2 so that they run at the same volts per octave. Furthermore, it is preferable that the two filters have the same cutoff and resonant frequency.

V/Octave calibration with the potentiometers lin 1/2 (linearity).This output is the MAIN cutoff out and drives both filter with the same Value. But we must calibrate both filters in it's liniarity.

It is better to isolate the output from the filter you want to calibrate.

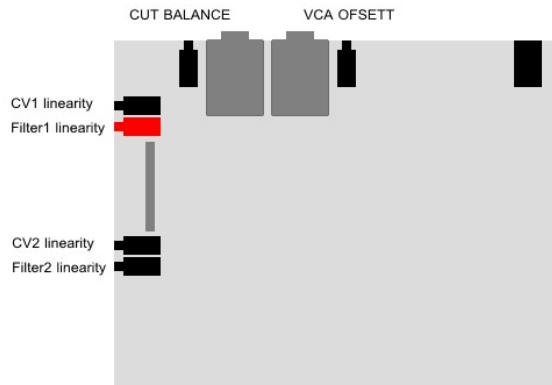
So, if you first calibrate Filter 1, be sure that you have only the output from Filter 1 audible. When Filter 1 is calibrated, switch to the output of Filter 2 and repeat the calibration for filter 2.

Here the steps:

- 1 :) all oscillators in Phoenix must be switched off
- 2 :) Cutoff in Phoenix on 32, resonancy to 64, LFO and ENV to 0.
(be sure the the velocity for Cutoff is switched off in the modulation matrix))

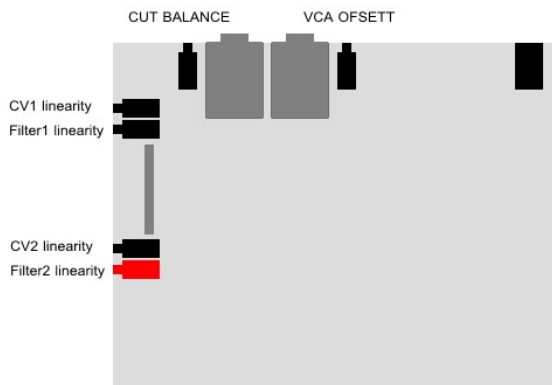
You will now be able to clearly hear the resonance signal.

3 :) Connect the TubeOhm Filtercalibrator to the Phoenix via an audio/MIDI interface or use a guitar tuner /frequency counter. Set the filter calibrator to automatic, now 3 octaves will be scanned. Or simply play C3, C4, C5 one after another.



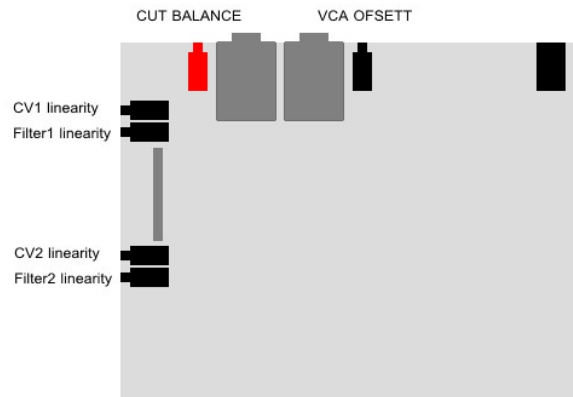
4 :) Screw the potentiometer R7 (lin1) so that the frequency is doubling pro octave.

Then remove the first channel from the mixer / sound card and connect the 2nd channel. The cutoff and resonance values in Phoenix should not be changed.



5 :) Now we can adjust the potentiometer R6 (lin 2) and the second channel so that we get a doubling of frequency per octave as in channel one.

Now both filter channels aligned and adjusted with CUT-BAL to zero beat.



With this we just completed matching both filters to a doubling of its frequency per octave, but not on a common base frequency.

For example, Channel 1 can have 100,200,400 Hz, but channel 2 is 120.240 and 480Hz. With the potentiometer Cut-balance the two filters are now so adjusted that both resonance frequencies have the same frequency.

The easiest way is to adjust the potentiometer as long as (and thereby a key on the keyboard Press and hold) until both filters have the same resonant frequency. This can be heard through the phasing.> Set to zero phasing.

If the filters are detuned a phasing arise in the signal. With cut-Bal now the frequencies are set so that the phasing is minimal.

Caution, slight phasing is normal because the filters calibration are not compensating for the temperature.

Now we adust the Filter CV1 and CV 2 in the modulation matrix.

Note: Cutoff and resonance values on the Phoenix mobo should not be changed !
First, we must stop the run along the cutoff / resonance frequency in the Phoenix filter menue.

By default the filter always receives a V / oct control signal from the CUT-PWM. This we turn in the MOD matrix to OFF

For this we go into the modulation matrix and make the following adjustments.

1 :) modulation matrix : **source = note destination= cutoff, Amount is -64 (minus 64)**

*** Now the cutoff frequency does not follow the note played. All notes almost have the same frequency output.

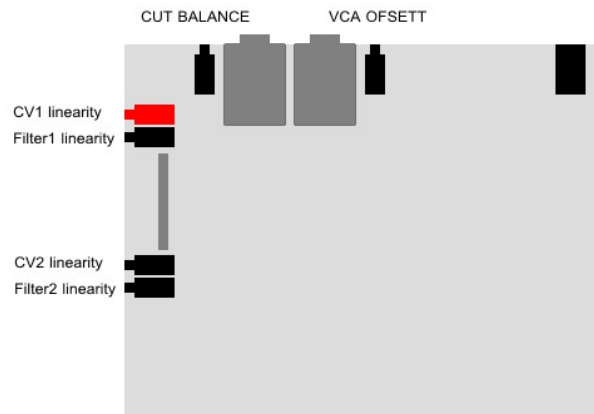
2:) Modulation Matrix 2: **Source = OFS, destination = CV1, Amount = 10**

The offset must be 10 so that the correct voltage for CV Phoenix are starting out.

3 :) Modulation Matrix = 3: **Source = note, destination = CV1, Amount = 64**

NOTE this controls now filter1 via CV1 in V / oct

*** Only now we can adjust R1-CV1 frequency per octave with the potentiometer.



Now we calibrate R1-CV1 as long as is set to a frequency doubling per octave.

The Filter Calibrator drives C3,4,5

**** or ,play C3,4,5 on your MIDI keyboard .

Now we set the second filter. This are the same adjustment , except we now change CV1 for CV 2. CV 2 for the second filter !

1 :) modulation matrix 1, **source = note destination= cutoff, Amount is -64 (minus 64)**

*** Now the cutoff does not follow frequency

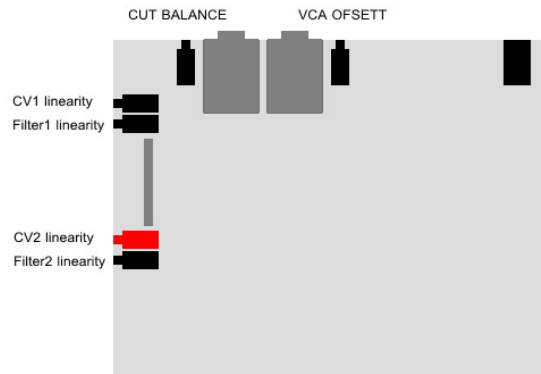
2:) Modulation Matrix 2, **Source = OFS,destination = CV2, Amount = 10**

The offset must be 10, that is the correct voltage for CV2 in the Phoenix's initializing.

3 :) Modulation Matrix = 3, **Source = note, destination = CV2, Amount = 64**

NOTE this controls the filter via CV2 in V / oct

*** Only now we can adjust R1-CV2 frequency per octave with the potentiometer.



Now R2-CV2 is set to the 2nd channel with a doubling of frequency per octave calibration.

The Filter Calibrator drives C3,4,5

****** or ,play C3,4,5 on your MIDI keyboard .**

Congratulations, now the filters are tuned .

Why so? Functionality.

Both filters get the same V/OCT signal from CUT (PWM) from the MOBO.
This is the first calibration step. Both filters must double it's frequency per OCT.
Than both filter must have the same basic frequency . This we adjust with the filter-balance.

But we also need that both filters drive with CV 1 and 2 with V/OCT.

For this we must disable the normal CUT-Volt pro OCT .

This happens in Mod Matrix 1 NOTE > CUT -(minus)64. This disables the CUT PWM and set it to 0.Now both filters doesn't double it's octave per NOTE.

But now we had to calibrate both filters in V/OCT with the NOTE >to CV1/2

This happens in MOD matrix 3>note>CV1/2 64.

*** MOD Matrix 2 OFFSET CV1 >> 10 is a dummy . This is need to get the correct basic Value on CV 1/2 .** it is only need for adjustment .

Now a few tips on how to program a sound in stereo.

Example 1: The Phoenix, now with the LFS-K1 filter, has 2 ADSR's, I want to drive the first Filter with ADSR 1 , the second Filter with ADSR 2...

In the Filter menu of Phoenix / Shruthi the ADSR Envelope is set to 0.
Cutoff is also set in this way , that the filters are nearly closed.

Then we go into the modulation matrix make the following adjustments.

1:) Modulation Matrix 1: **source = ADSR 1, destination= CV1, amount is 10..64.**

Now ADSR1 is routed to the first filter.

2:) Now we go to the modulation matrix 2: **source ADSR2, destination= CV2, amount = 10..64.**

Now we have route ADSR 2 to filter. 2

Depending on the setting of both ADSR we now get different filter characteristics for the first and second filter.

Example 2: the LFO 1 should control filter 1 and also filter 2 but for filter 2 with changing the LFO wave phase to 180 degrees. The sound should move from right to left.

Note: Offset is added to the subsequent signal in the Mod Matrix . Therefore you must first set the offset and then change the LFO in the MOD MATRIX . Conversely, Changing the LFO first and then the Offset doesn't work!!!

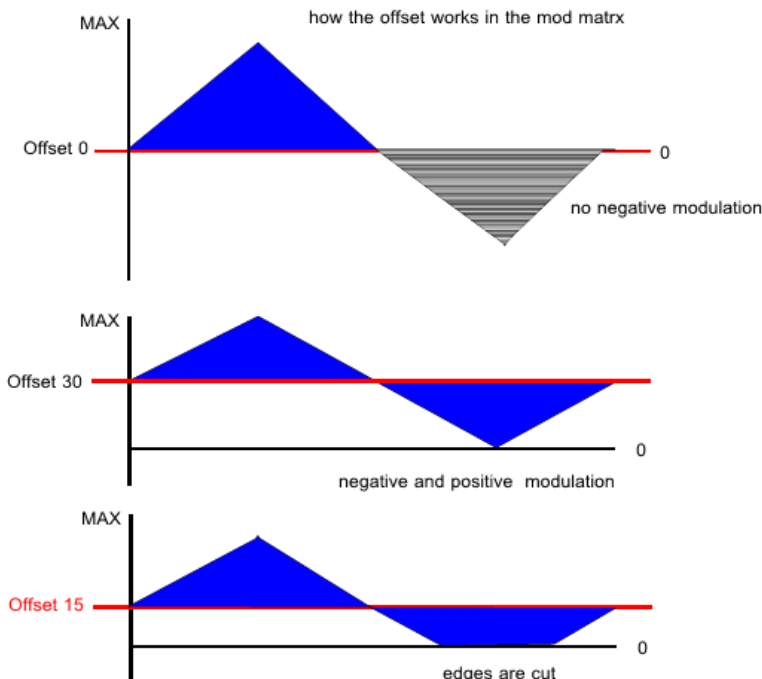
Modulation Matrix 1: source = offset, destination = CV1, amount = 30

Modulation Matrix 2: Source = LFO1, destination= CV1, amount = 30

Modulation Matrix 3: source = offset, destination = CV2, amount = 30

Modulation Matrix 4: source = LFO 1, destination = CV2, amount = -30 (minus 30)

This seems more difficult... Why shift the offset ?



Well, the CV outputs only send positive values. The LFO has both positive and negative values.

This means that the negative values can modulate the filter if it is shifted via the offset into positive territory.

For example, an LFO signal has a value range of '-32...0...+32'. So only the values '0...+32', which is a positive half-wave would modulate the filter.

By offsetting the entire LFO signal, it is now shifted completely into the positive range, we now have a value range of '0 ...+64', and therefore I can modulate the filter with a full sinusoidal oscillation.

If the offset is too small, the LFO signal is simply clipped in the lower region - which can also be quite sexy.

If the offset is too high, it clips the positive signal.

Let's go again to the Mod Matrix 4. This is where we change the phase rotation of 180 degrees by the amount value '-30'

Giving a negative value so the LFO signal rotates 180 degrees, we get a wandering sound because the positive LFO waveform opens one filter and (because of the phase degree) the second filter is closed.

Note , positive amount values gives a phase rotation from 0 Degree while negative values gives a phase rotation of 180 degree.

In addition, CV1 and CV2 can additional modulate by all other sources in the modulation matrix to give a rich stereo sound.

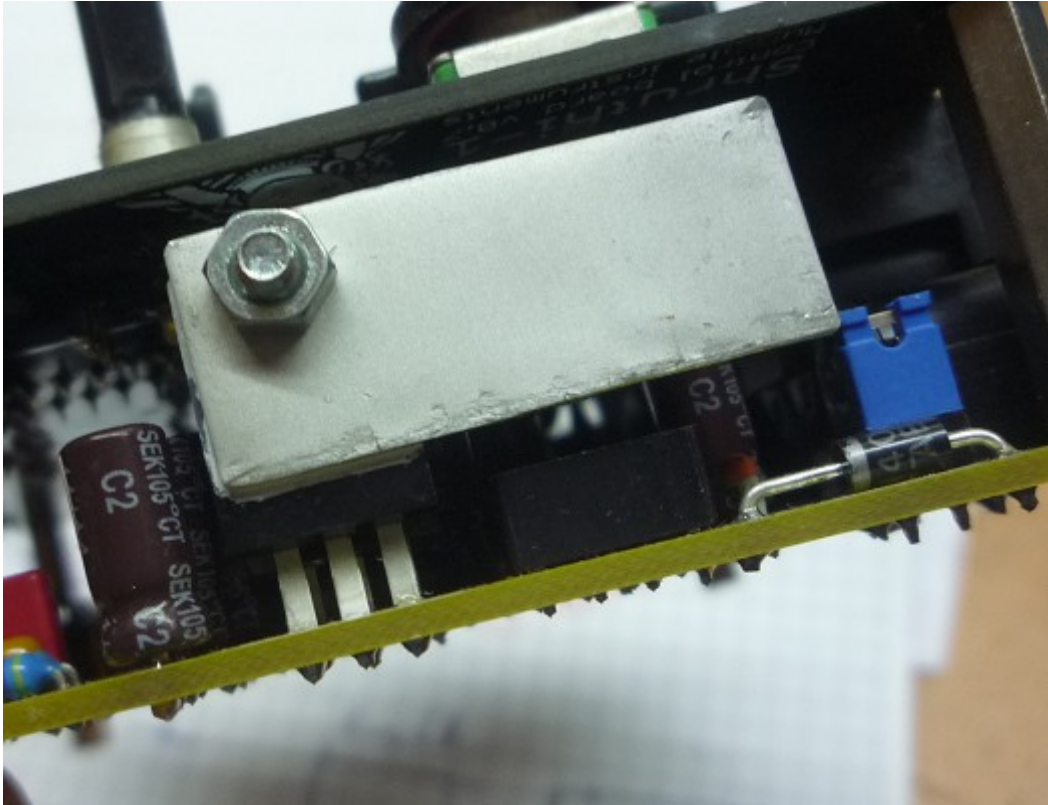
Have fun with your new TubeOhm LFS-K1 stereo filter!!!

TubeOhm 01/14/2016

Notes and modding .

Depending on the LCD you use in your Phoenix, the LM7805 can become very hot. If this is the case, simply screw a small piece of aluminum plate as an additional heat sink on to the LM 7805's fin. You will need a 6mm M2 screw, a nut and a piece of aluminum 1, 3x2 cm.

Be sure that you don't make a short between the 7805 and the 7905 regulators!!!



BOM

Stereo Filter Shruthi				Mouser	Farnell	Reichelt
Date-29.12.2015	Tube	Ohm				
	Value	description				
Arduino stackable 8 Pin header	2		Small Board header	872-920-0088-01	xxx	xxxx
C1,2,3,4	4	220n	2.54x5.08 RM2.54Ceramic2,	810-FK18Y5V1H224Z	1141778	Z5U-2,5 220N
C5,6	2	100p	2.54x5.08 RM2.54Ceramic2,	594-K101K15C0GF53L2	1694179	KERKO 100P
C7,8,14,16	4	1n	Filter Caps. WIMA 1NFilter	505-FKP0C011000BK100	xxx	FKP-02 1,0N
C9,10,11,12,13,15,30-	26	100n	2.54x5.08 RM2.54Ceramic2,	75-1C10Z5U104M050B	1694337	X7R-2,5 100N
C17,18,21,22,27,28	6	470p	2.54x5.08 RM2.54Ceramic2,	75-1C10C0G471J050B	1827826	KERKO 470P
C19,20,23,24,25,26	6	4,7uF NP	D5R2.54 ELKOD5R2.54 E	647-UVP1E4R7MDD	1236689	xxxx
C47	1	10uF tantal	D4R2.54 ELKOD4R2.54 E	581-TAP106K016SCS	1753975	TANTAL 10/16
C48	1	10p10p	2.54x5.08 RM2.54Ceramic2,	594-K100K15C0GF53L2	1694176	KERKO 10P
C49,50,51	3	100uF	D5R2.54 ELKOD5R2.54 E	667-ECA-1VM101	1600568	RAD 100/16
C52,53	2	220uF	D11R5.08 ELKOD11R5.08	667-ECA-1HM221	9451099	RAD 220/35
D1,2,3,4,5,6,7,8,9,10,11,12	12	1N41481	DiodeDiode	512-1N4148	9565124	1N4148
D13	1	1N40011	DiodeDiode	512-1N4001	1651084	1N4001
IC1,IC5	2	TL074CN	DIL14	595-TL074CN	9755934	TL 074 DIL
IC2,IC3,IC7,IC9	4	LM13700N	DIL16	926-LM13700N/NOPB	1851866	LM 13700 DIL
IC8,IC6,IC4	3	TL072	DIL8	595-TL072IPE4	1103005	TL 072 DIP
IC10	1	LT1054	DIL8	595-LT1054CP	1648667	LT 1054 CN8
IC11	1	7805	TO220	511-L7805CV	9666095	µA 7805
IC12	1	7905	TO220	863-MC7905ACTG	9666141	µA 7905
9Volt connector	1	9VDC	9V DC plug	163-7620-E	1737246	HEBW 21
Connector Stereo Volume pot	2	optional	Bridge it			
R1,R2,R6,R7,R36,	5	20k	Pot 20Kohm	594-64Z203	1141378	64Z-20K
R11	1	50k	Pot 50Kohm	594-64Z503	1141392	64Z-50K
IC socket 8 pol	4	IC socket		571-1-2199298-2	1654374	GS 8
IC socket 14 pol	2	IC socket		571-1-2199298-3	1183573	GS 14
IC socket 16 pol	4	IC socket		571-1-2199298-4	1183574	GS 16
R3,4,5,8,37,38,49,50	8	47k	Metall	594-5073NW47K00J	9341960	Metall 47.0 k
R9,10,30,31,32,33,27,35	8	2,2k	MetallMetall	594-5073NW2K200J	9341536	Metall 2.2 k
R12,13	2	120k	MetallMetall	594-5073NW120K0J	9341242	Metall 120 k
R14,16,34	3	1k1k	MetallMetall	660-MOS1CT52R102J	9341102	Metall 1.0 k
R15,17	2	330k	MetallMetall	594-5073NW330K0J	9341765	Metall 330 k
R18,21,40,45	4	390	MetallMetall	660-MOS1CT52R391J	9341846	Metall 390
R19,28,39,41,66,67,68,70,72,73,74	11	10k	MetallMetall	660-MOS1CT52A103J	9341110	Metall 10.0 k
R20,44	2	39k	MetallMetall	660-MO1CT52R393J	9341862	Metall 39 k
R22,29,46,48,51	5	4,7k	MetallMetall	660-MOS1CT52R472J	9341951	Metall 4.7 k
R24,43	2	100k	MetallMetall	594-5073NW100K0J	9341129	Metall 100 k
R23,47,R25,R22	4	27k	MetallMetall	660-MOS1CT52R273J	9341650	Metall 27 k
R26	1	6,2k	MetallMetall	660-MO1CT52R622J	9342117	Metall 6.2 k
R52,53,60,61	4	470	MetallMetall	660-MOS1CT52R471J	9341943	Metall 470
R54,63,56,59	4	18k18k	MetallMetall	660-MOS1CT52R183J	9341447	Metall 18.0 k
R55,62	2	22k22k	MetallMetall	660-MOS1CT52R223J	9341544	Metall 22.0 k
R57,64	2	68	MetallMetall	660-MOS1CT52R680J	9342176	Metall 68
R69,71	2	220	Metall	660-MOS1CT52R221J	9501398	Metall 220
T1,2,3,4,5,6	6	2N3906	TO92(1)	512-2N3906BU	1574372	2N 3906
X2	2	Neutrik stereo	EBS63P	550-20301	152063	xxxx
STEREO PCB	1	LSF-K1	TubeOhm			