EL84 Push-Pull Amplifier

Joost Breed – TubeSociety 2026

This amplifier is a redesign of one I've made a few years ago. I've made several improvements, including adding local feedback in the preamplifier stage and an auto-bias system to minimize the THD in the low frequency range due to transformer saturation.

Main features:

- 6CG7 preamplifier with local feedback and bootstrapping
- 6CG7 phase inverter with constant current source
- 10 W EL84 push-pull power amplifier with local feedback
- Digital potentiometer
- ESP32 controlled OLED display with touchscreen
- Rotary encoder with push button
- Digital auto-bias system
- Uses only local negative feedback, no global feedback
- Damping factor 9 with 8 Ohm load
- Regulated high voltage power supply

Inhoud

SPICE simulation

When designing a new circuit, I always start by simulating it in LTspice to verify its behaviour and performance. Both the amplifier and the regulated power supply were first designed using LTspice.

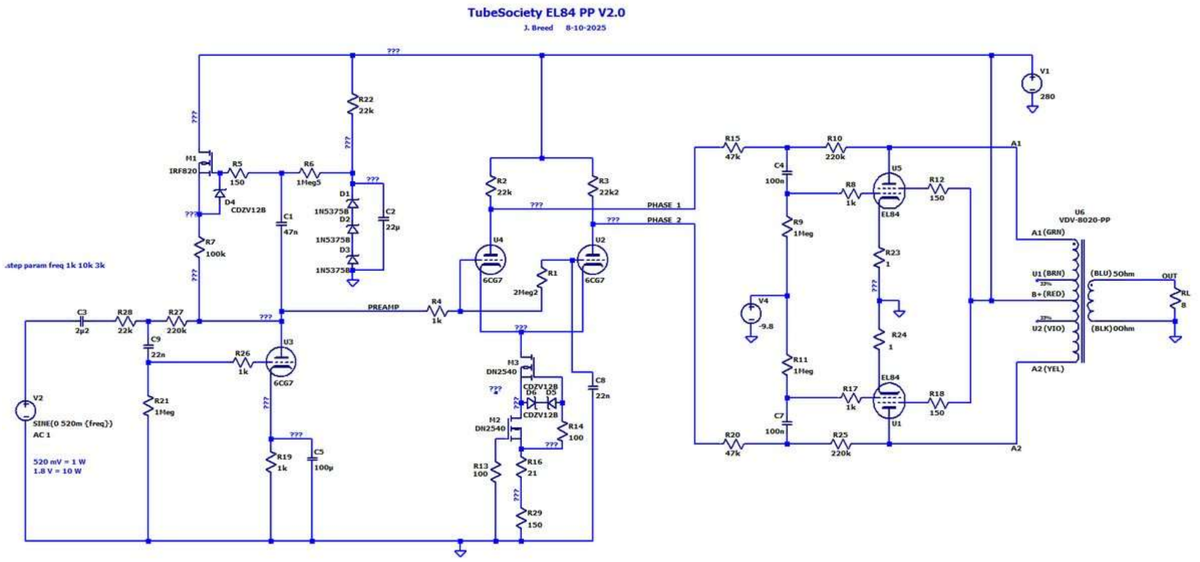


Figure 1 - The amplifier in LTspice

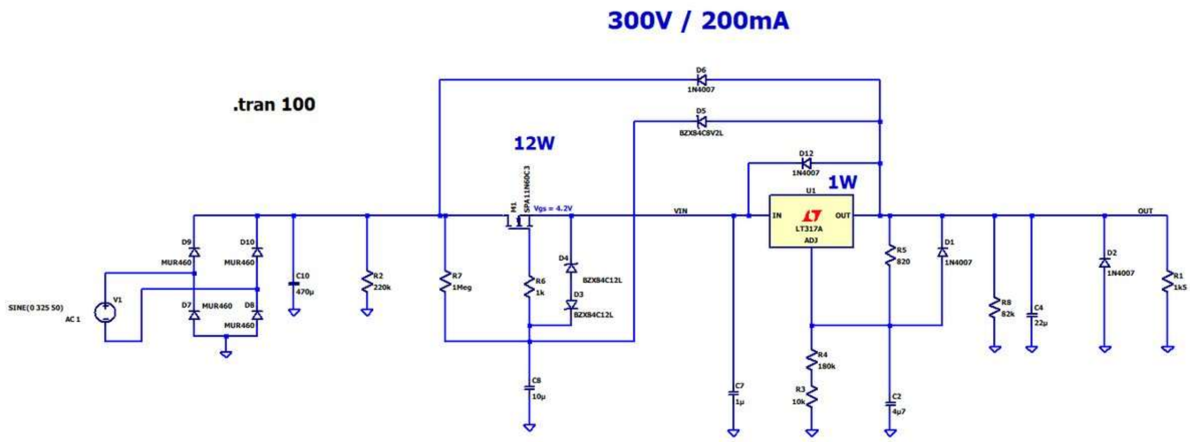


Figure 2 - The regulated power supply in LTspice

Testing the design on a breadboard

After completing the simulations, I built the amplifier on a breadboard and compared my measurements with the simulations using an oscilloscope.

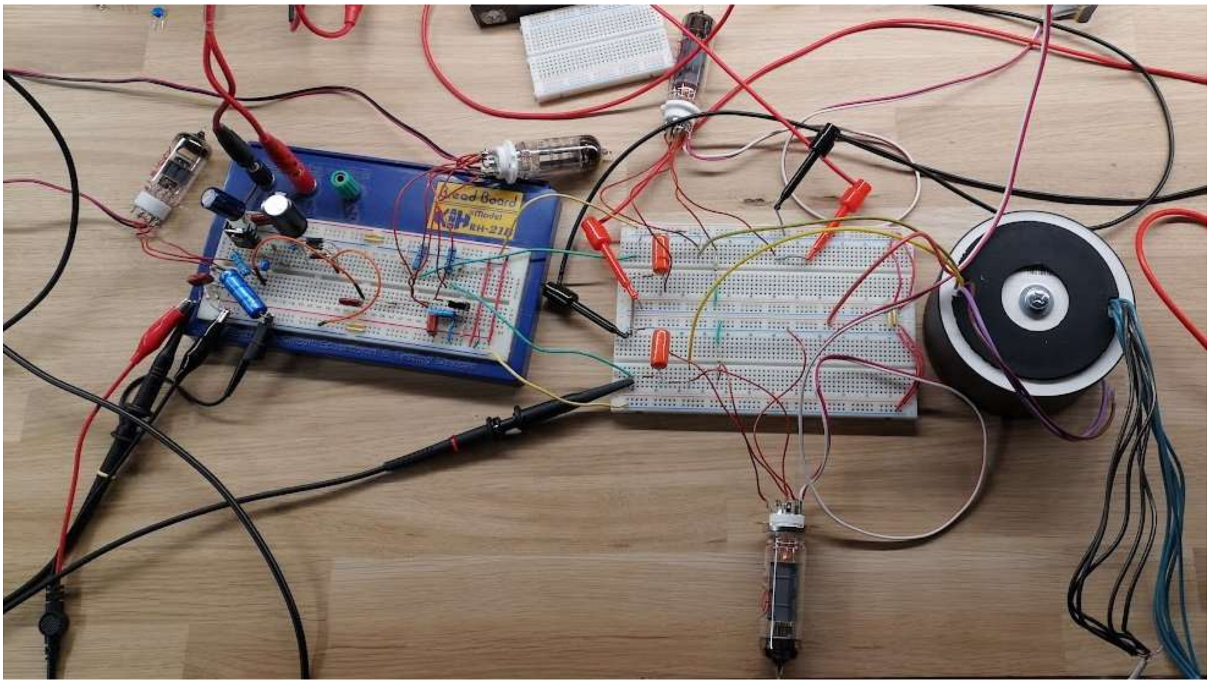


Figure 4 – The complete amplifier on breadboards

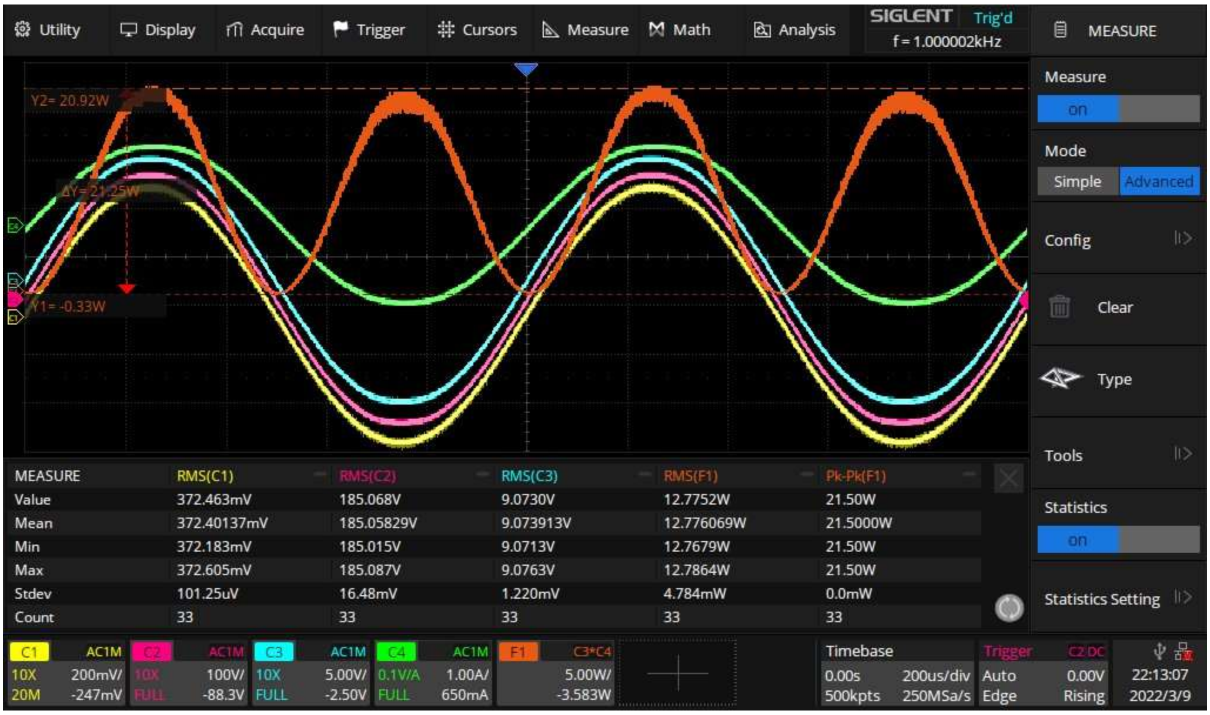


Figure 13 - Oscilloscope power measurement of the breadboard circuit

Amplifier design

The amplifier consists of five PCBs designed in KiCad. The photo below shows the inside of the amplifier seen from the bottom.

In the upper left corner sits the power supply. In the upper right corner the audio input board. Below that the auto-bias controller. At the bottom the large amplifier pcb. And on the display an adapter board to connect the display to the audio input board and auto-bias controller.

A lot of smd components are used. Smd components have shorter or no wires so have less inductance. Newer components are only available in smd packages as well. I use smd as much as I can, but not for the high voltage stuff because of the high voltage clearance.

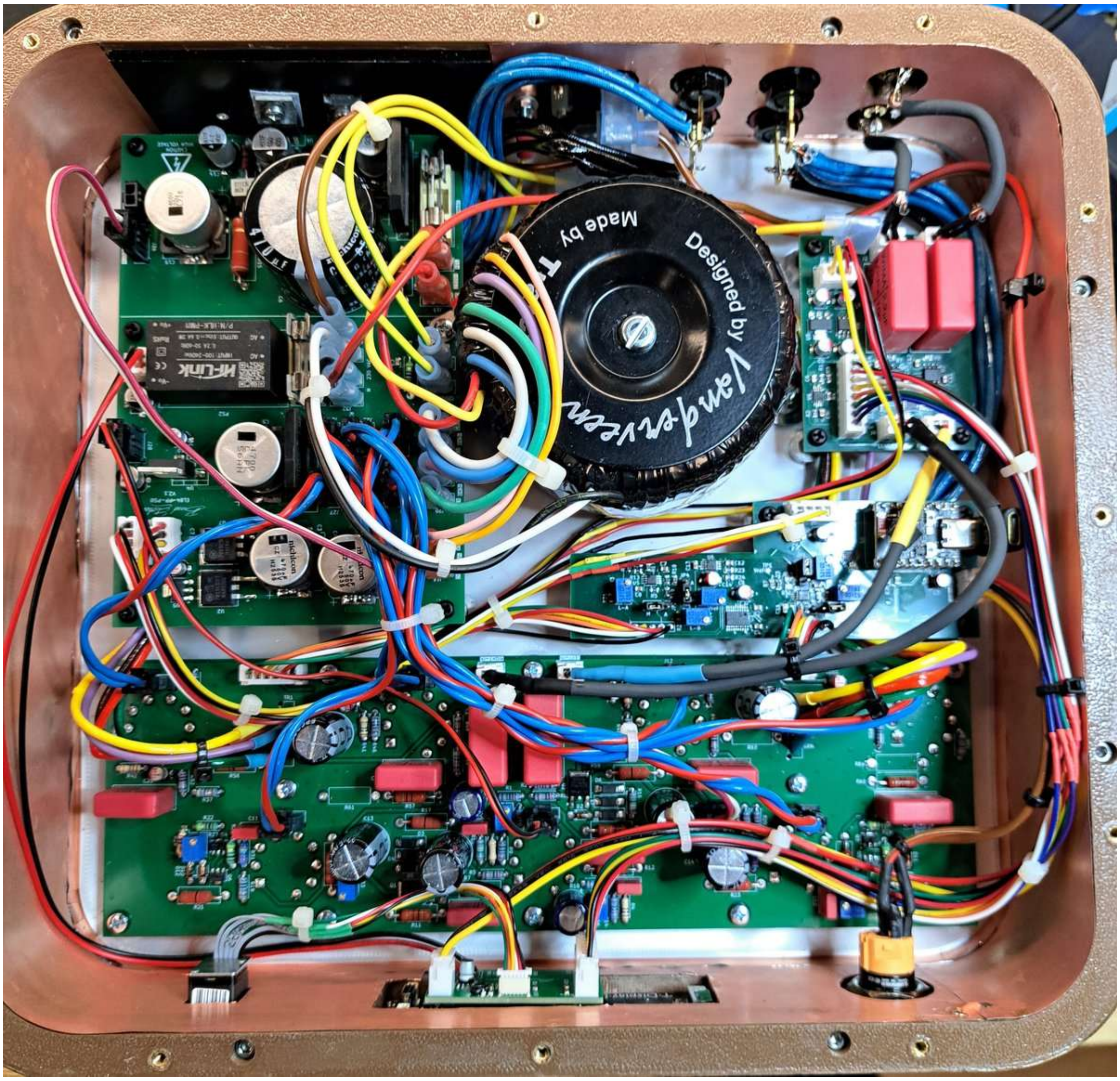


Figure 9 – The inner works

Power supply

The transformer used for this power supply is a VDV-MAINS-D. It has a 230VAC/300mA primary, a 25VAC/100mA secondary, and two 6.3V/3A heater windings. A Maida type regulator is used to get a noise free regulated dc-voltage of 282 V.

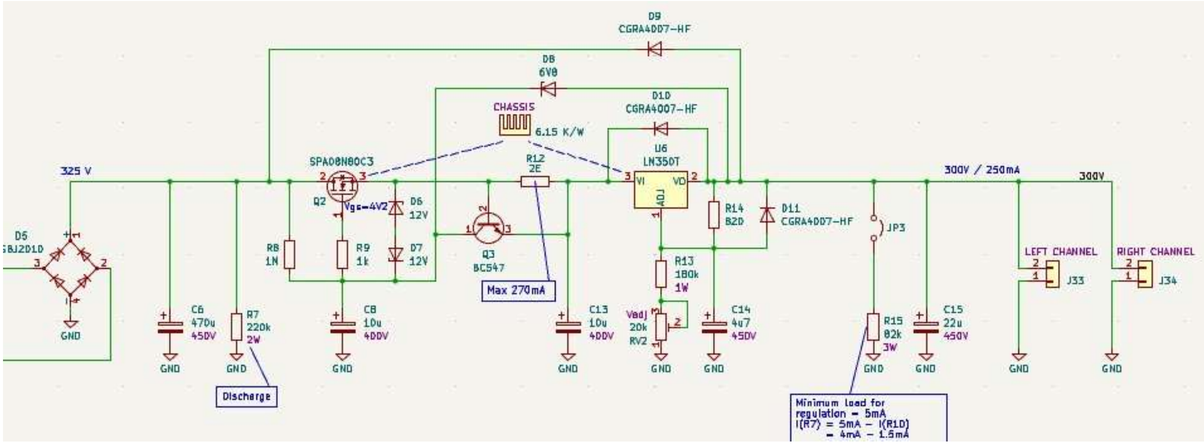


Figure 13 - The power supply circuit

The +5V, +15V, and -15V rails are derived from the 25VAC voltage and two linear regulators. These voltages are used to power the op amps on the audio input board and the auto bias controller. The +5V rail is used to power the auto bias microcontroller.

The -15V rail is also used to generate the negative bias voltage of the EL84 tubes.

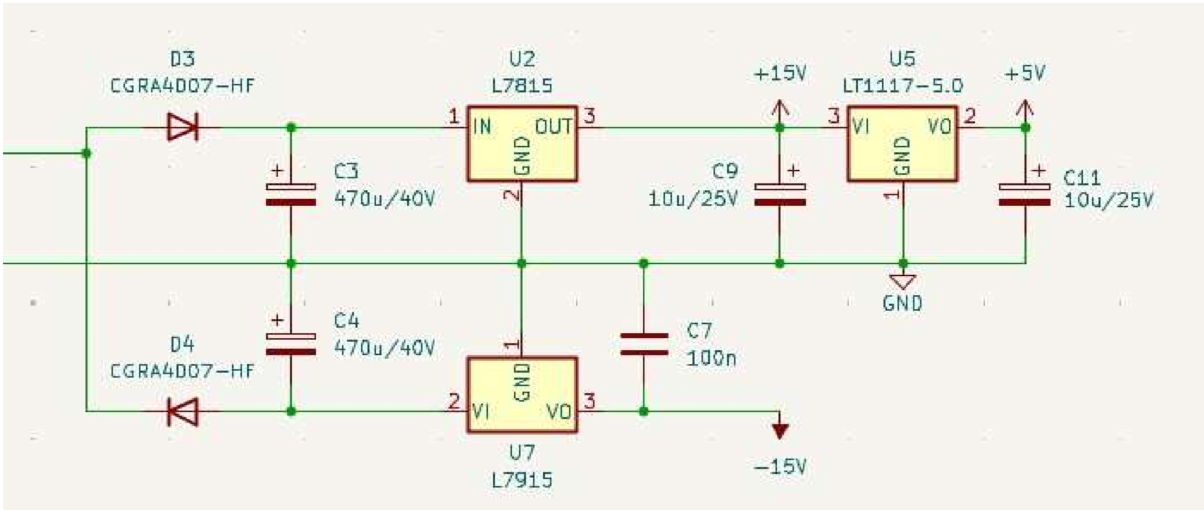


Figure 4 - Op amp power supply

The filament supply outputs 6.4 Vdc for the preamp tube. The other tubes are heated with ac-voltages.

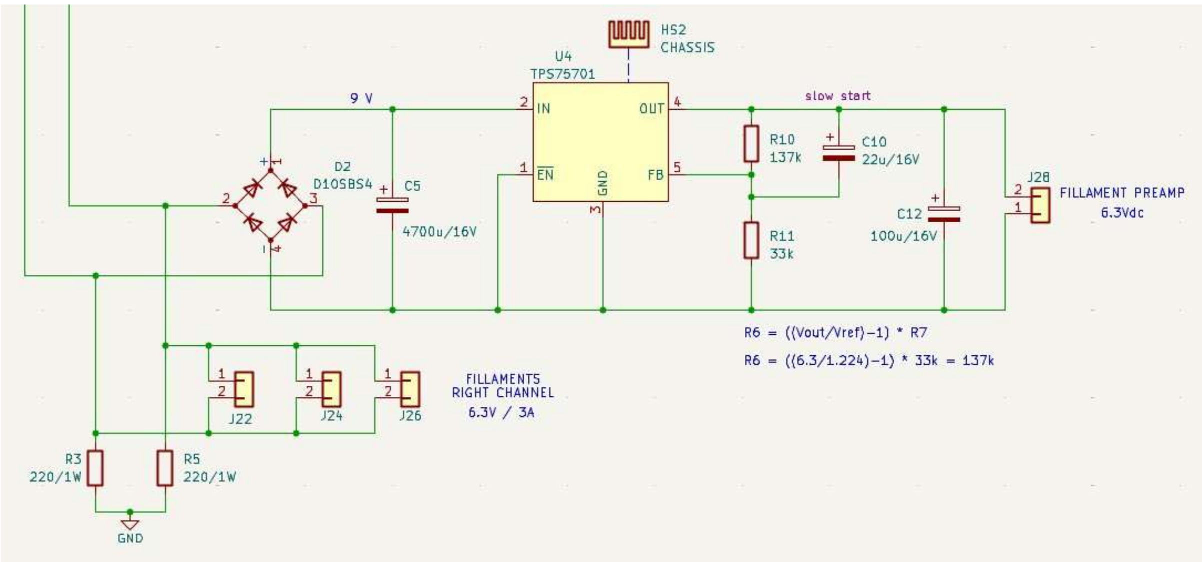


Figure 5 - 6.3 Vdc power supply

A separate power supply generates 5 Vdc directly from 230 Vac. This 5V is used by the OLED display and its microcontroller.

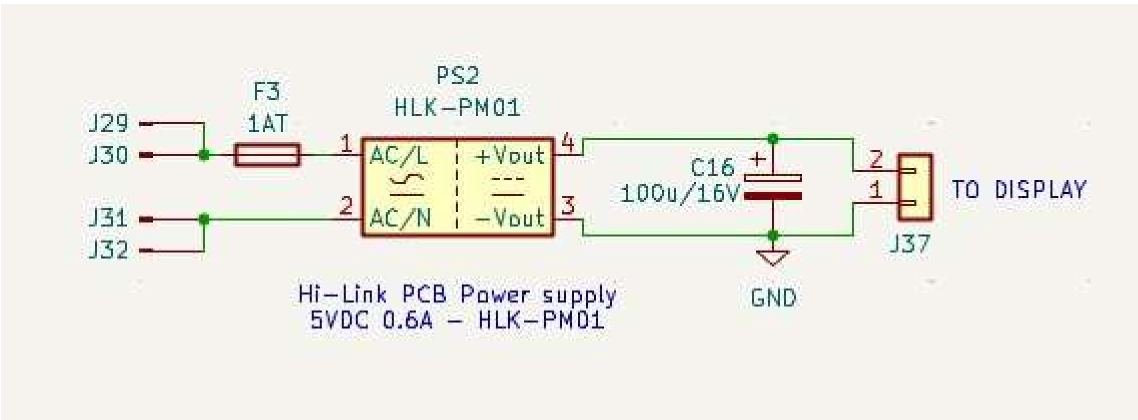


Figure 6 - 5V microcontroller supply

Audio Input

The audio input consists of a digital potentiometer which is buffered on both sides by low-THD op amps. The buffers make sure the volume control is not loaded and it does not see any impedance changes at the amplifier inputs.

The gain of the potentiometer can be changed between 0 and -62 dB and has a -80 dB mute function which is used during biasing at startup.

The microcontroller controls the DS1882 potentiometer via an I2C interface. The microcontroller uses 3.3 V signals but the potentiometer ic uses 5 V signals. Two MOSFETs (Q1 and Q2) are used to do bi-directional level shifting between the two voltages.

The THD of the digital potentiometers is 0.005 % and it has -120 dB crosstalk. It changes volume during zero-crossing of the audio signal.

The volume can be changed by turning the encoder at the front of the amplifier. The volume is changed in 1 dB steps.

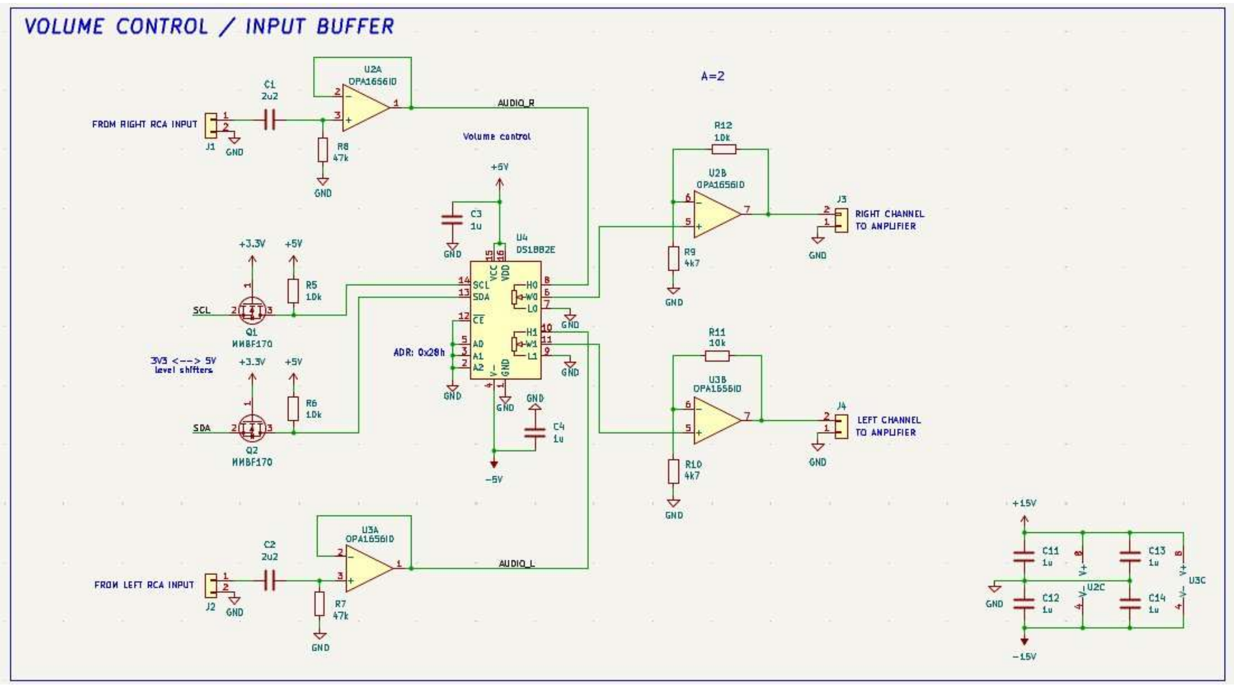


Figure 7 - Volume control circuit

The audio signal is also fed to the microcontroller which controls the OLED display. This signal is taken before the potentiometer and is buffered/isolated with LM2904 op amps to minimize the noise injection of the microcontroller. This signal is used for displaying the audio level on the display. The display also controls the digital potentiometer so it takes the volume into account via software.

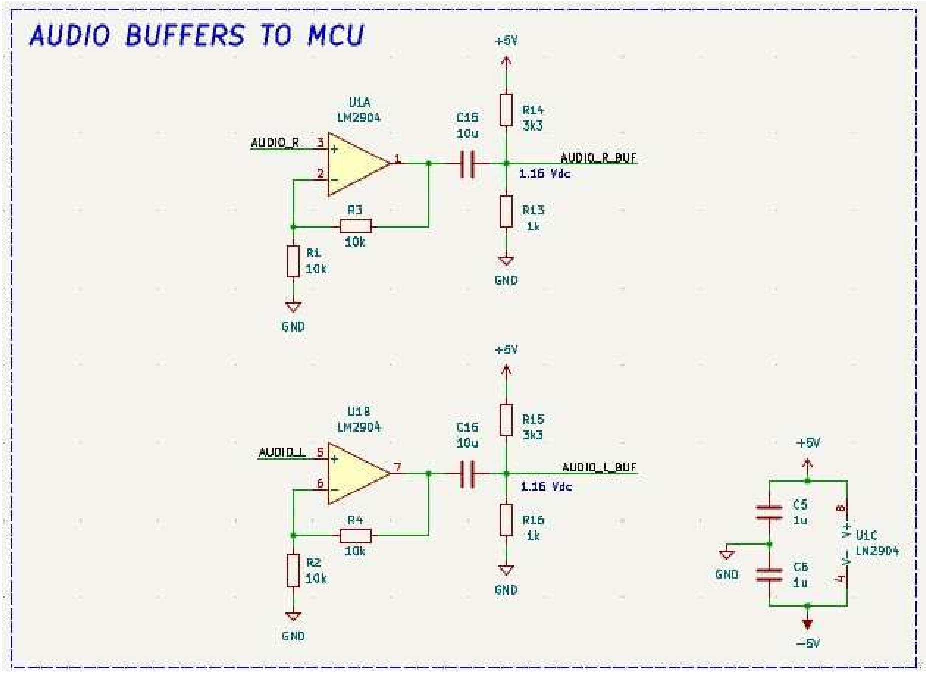


Figure 8 - Audio buffers for microcontroller isolation

The +5V and -5V rails for the potentiometer are generated from the +15V and -15V with some voltage regulators.

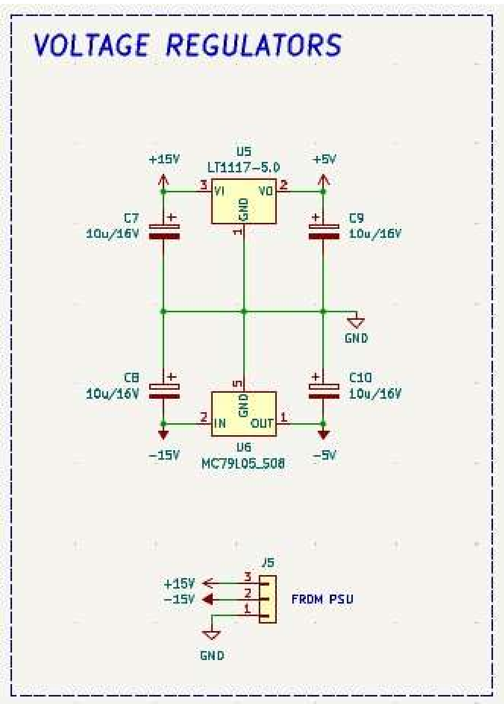


Figure 9 - Voltage regulators

Preamp

The preamp is built around a 6CG7 tube. It uses local feedback via R5 and C3 to make the preamplifier cleaner by itself, rather than relying on global negative feedback. The amplification factor is set by R5 and R1 to 10x.

Q1 acts like a current source. It is biased by a voltage set by zener diodes. Any audio signal at the anode of the tube is fed through C7 to the gate of Q1. Q1 acts like a source follower so the audio signal is fed back through the MOSFET to the top of R7. The voltage on both sides of R7 are therefore the same for ac signals. If the anode voltage rises then the source voltage of the MOSFET also rises with the same amount. This results in the tube seeing R7 as an infinite resistance/load for ac-signals. That way the amplification factor will be the same as (µ) and the THD will stay low. The THD for this preamplifier is measured to be 0.1 %.

R65 is used to set the anode voltage to 60 V. This voltage is used to bias the phase inverter because there is no coupling capacitor between the preamp and the phase inverter.

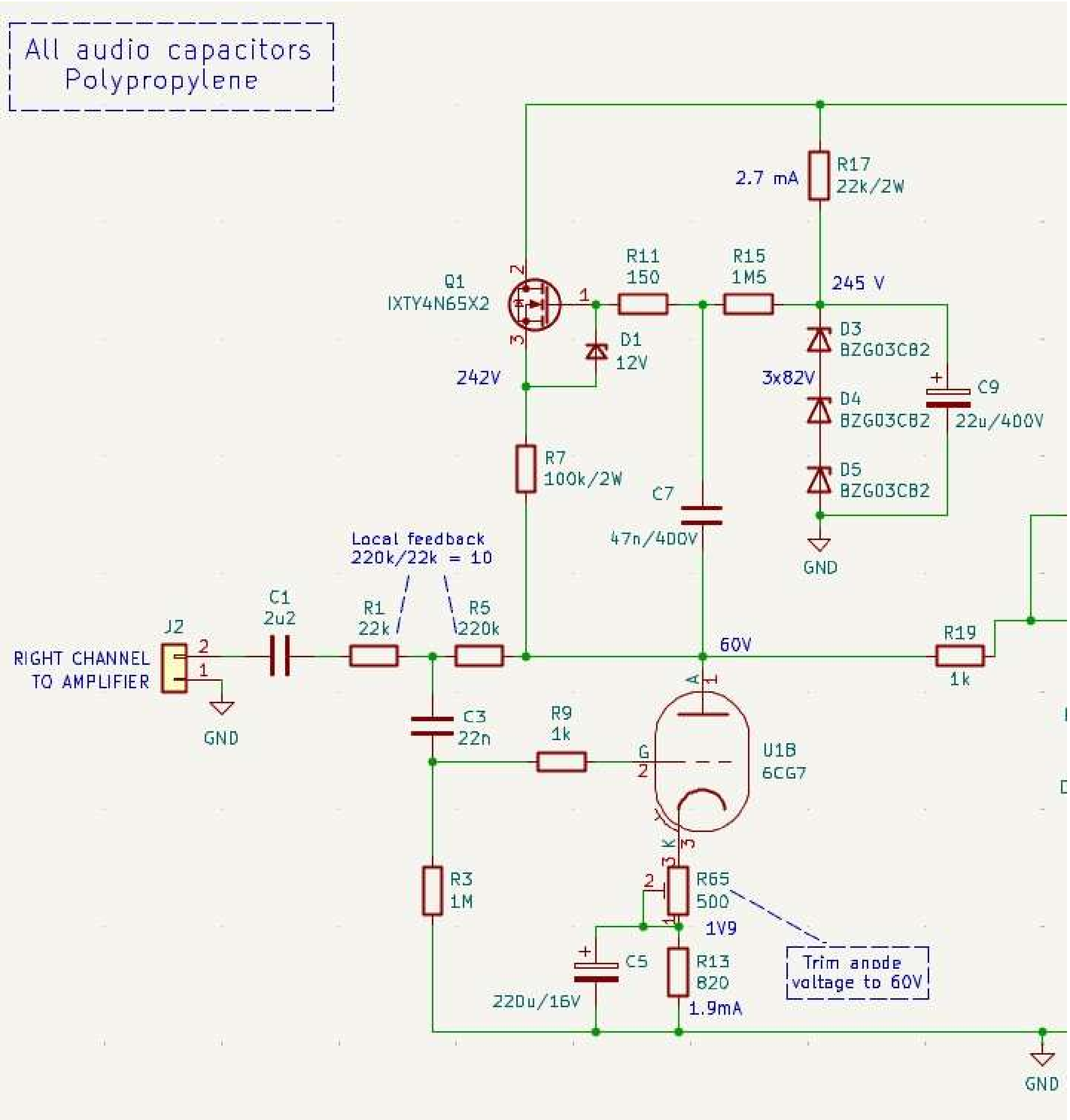


Figure 10 - The preamplifier

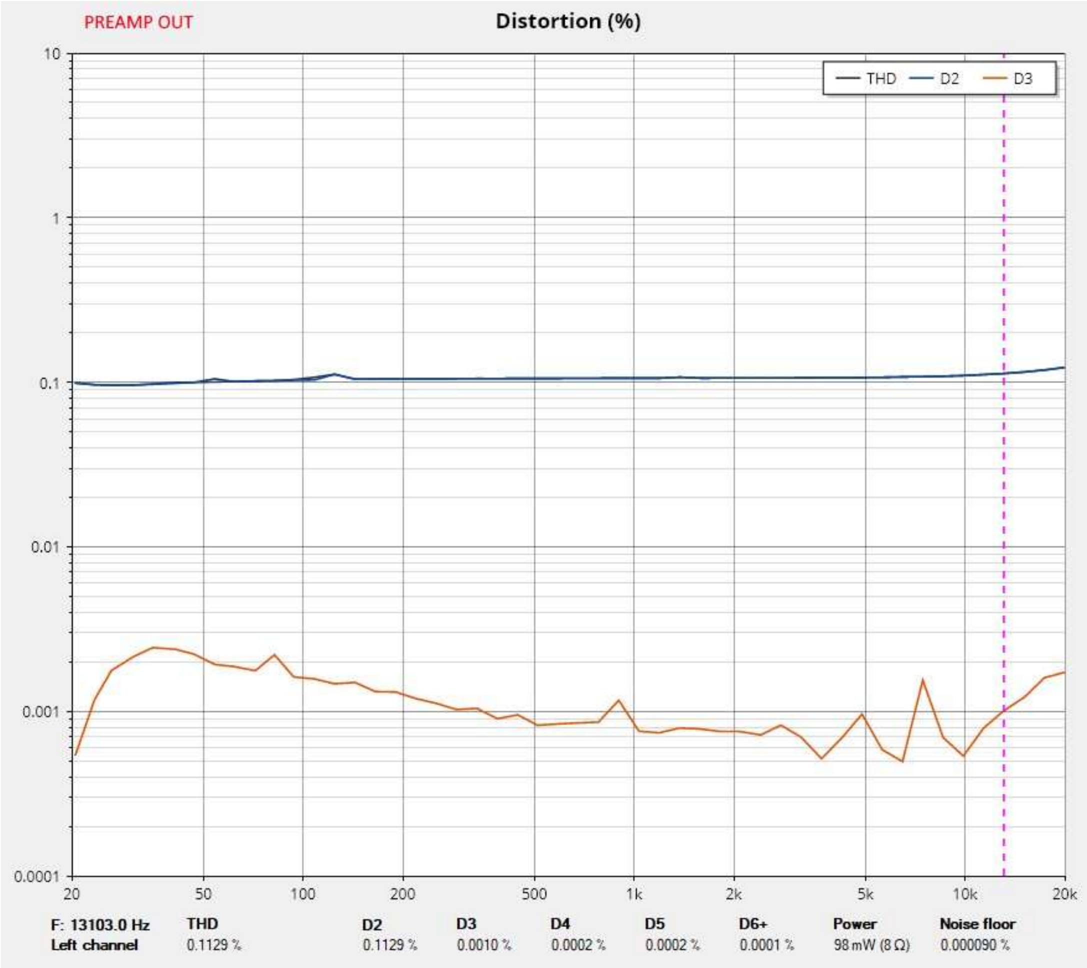


Figure 11 - Distortion measurement of the preamplifier. It contains a lot of 2nd harmonics.

Phase inverter

The phase inverter also uses a 6CG7 tube. The tail uses a current source built around two MOSFETs. This topology maximizes the tail impedance, which minimizes the THD of the circuit. The right tube is shorted to GND via C11 for ac-signals. The anode of U2B contains a signal which is out of phase of the input signal, the anode of U2A contains a signal which is in phase. These two signals are used to drive the push-pull stage.

Trimpot R33 is used to balance the two output signals and minimize 2nd harmonic distortion.

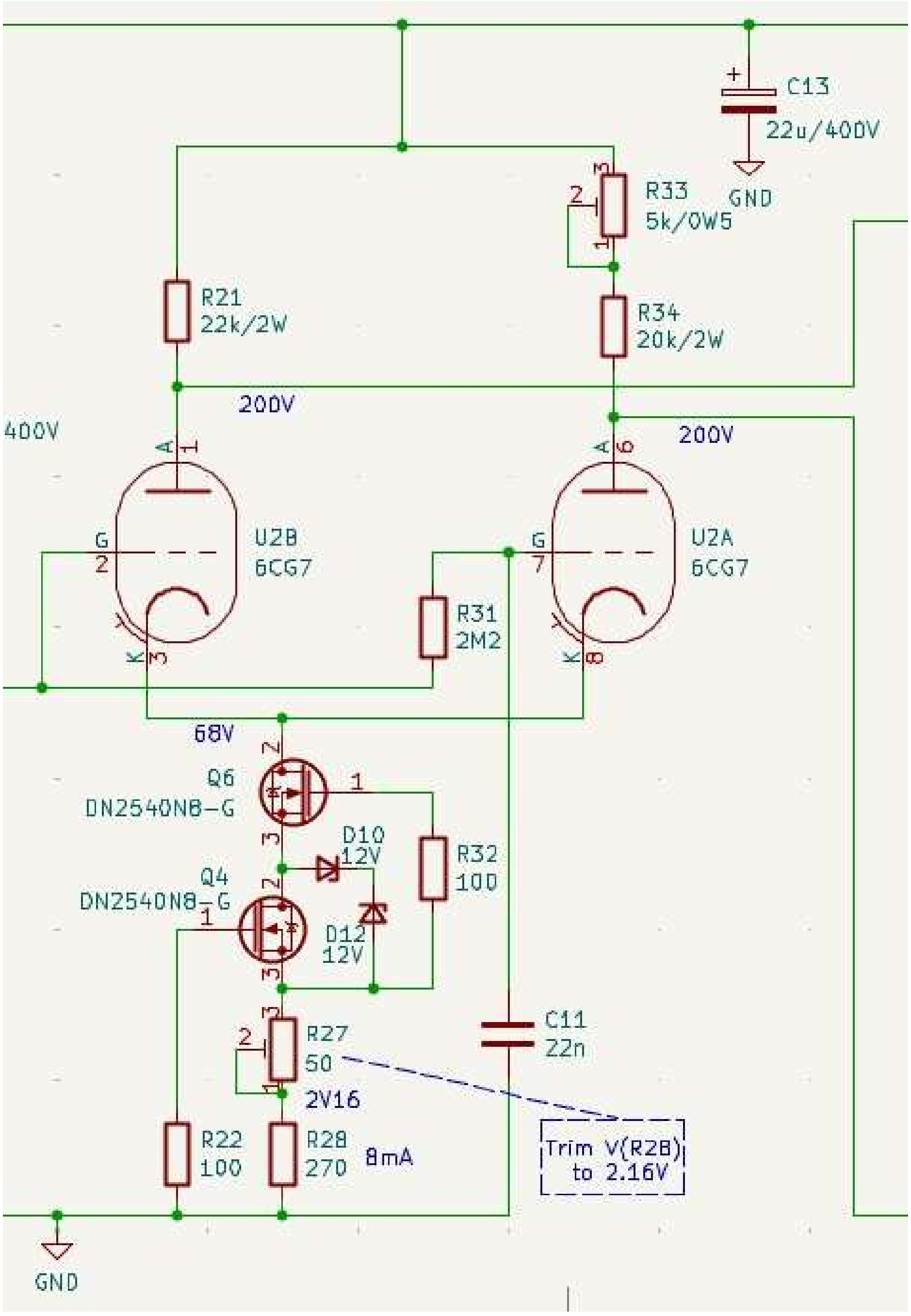


Figure 12 - Phase inverter

The distortion measured with 25 Vpp at the output of the phase inverter is about 0.066 %. The input signal was fed by the preamplifier during that measurement.

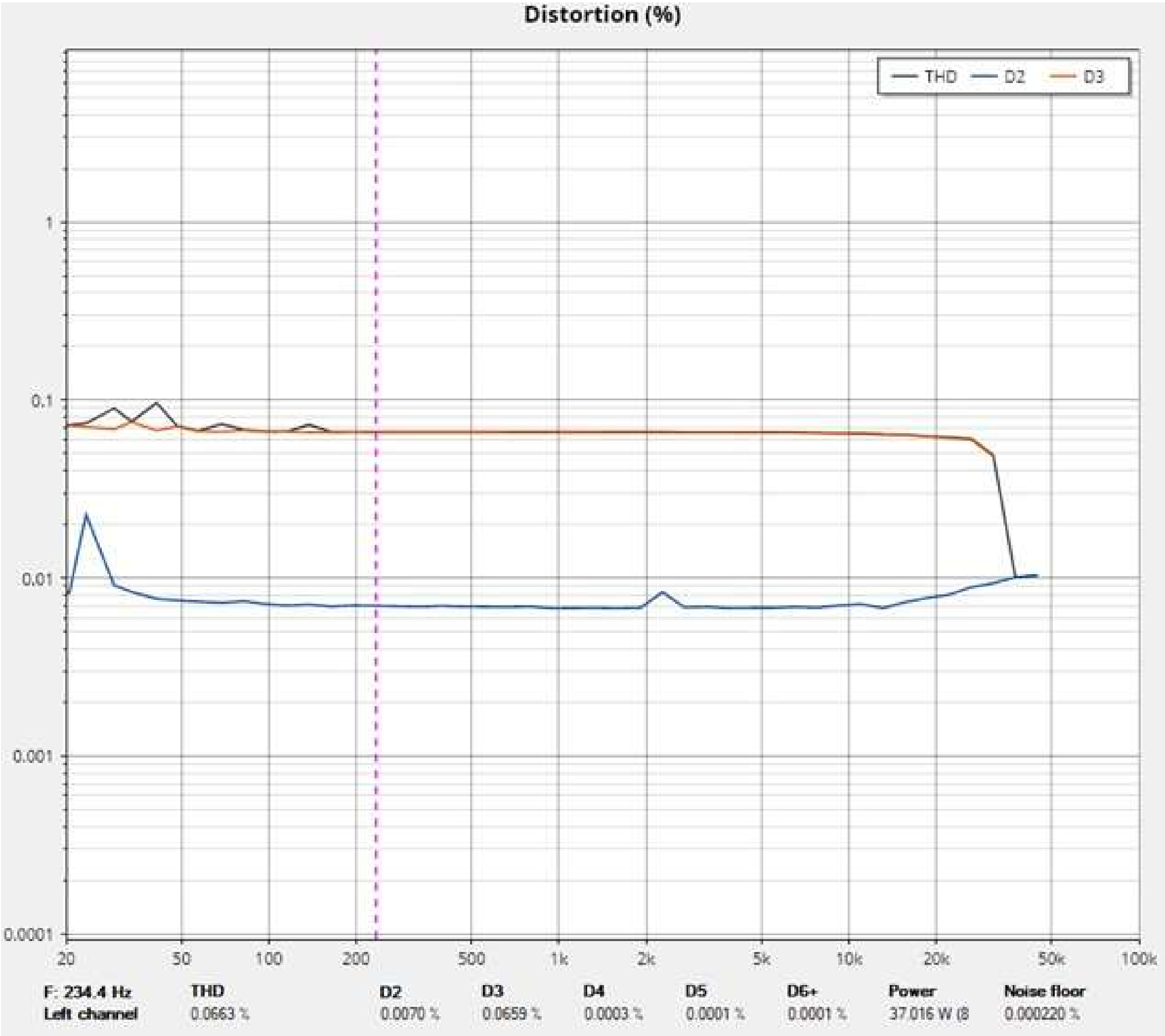


Figure 13 - Vout = 25 V differential using a 6CG7 of General Electric

Push-Pull stage

The push-pull stage uses two EL84 tubes biases around 38 mA.

Local negative feedback is used via R45/C15 and R52/C16. The ratio between R45/R39 and R52/R37 define the amplification factor to be around 4.7x.

The UL resistors are not installed. The tubes are biased by the auto bias controller.

The 1 Ohm cathode resistors R55 and R56 are used to measure the cathode current for the auto-bias controller.

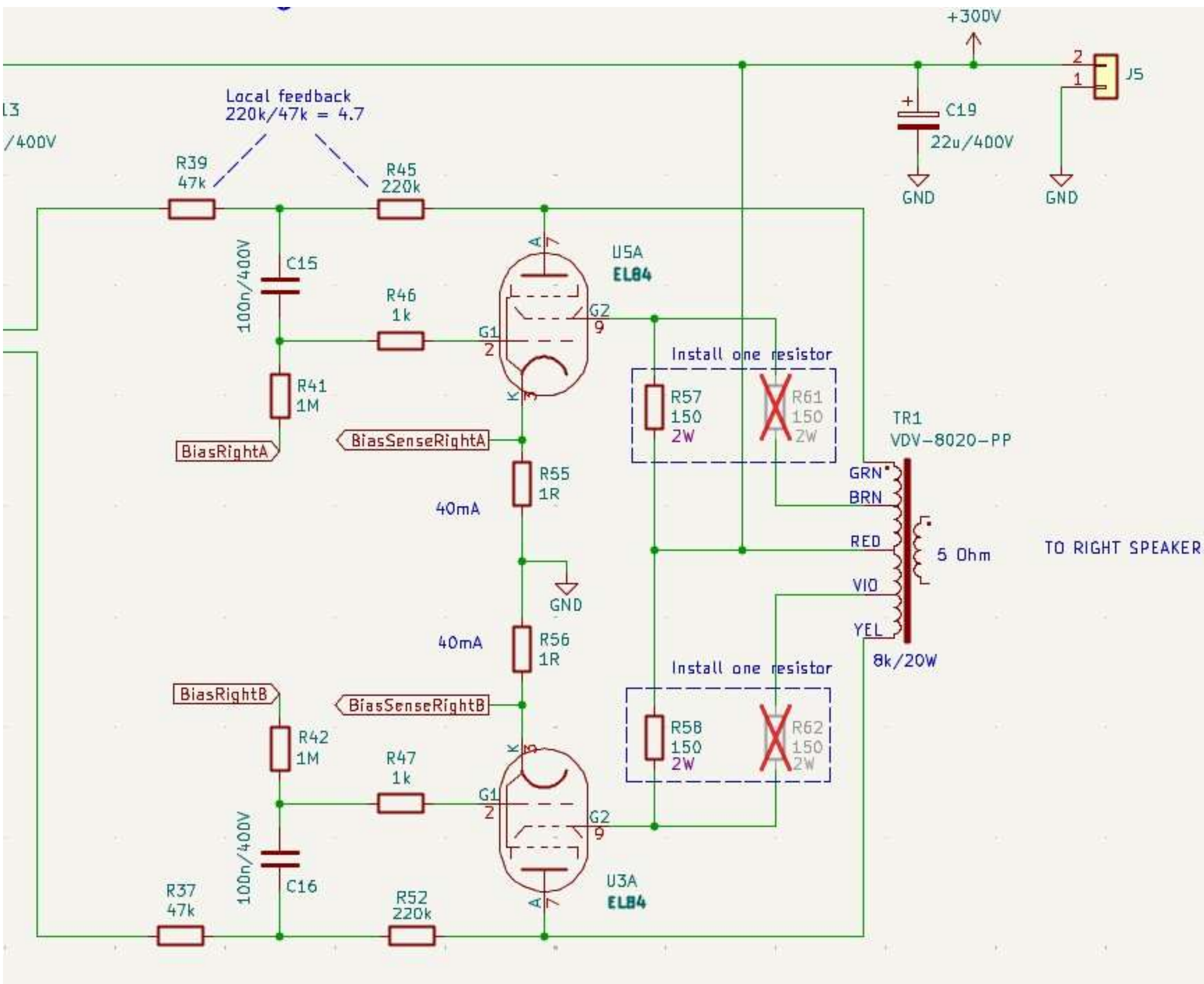


Figure 14 - Push-pull amplifier

The current sense resistors R55 and R56 need to be as small as possible so they do not influence the audio signal. Current sense amplifiers (INA180) are used to isolate the audio signal from the adc which does the measurement. They also amplify the voltage with a factor of 20. Currents up to 120 mA can be measured with this setup. The current is measured with a 24 bit DAC. So the theoretical resolution is in the nA range.

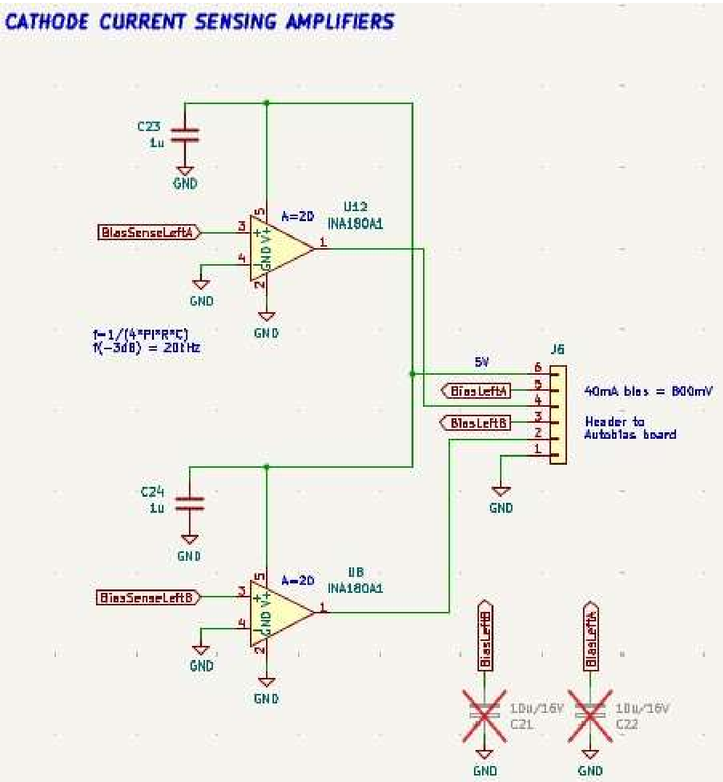


Figure 15 - Current sense amplifiers

Display

A Lilygo T-display long is used for the display. It consists of an ESP32 with an OLED display and also has a touch screen.

Figure 16 - Lilygo - T-display - Long

An optical rotary encoder with push button is connected to the display board to control the amplifier volume and bias settings. The microcontroller communicates with the auto-bias controller via a serial interface at 155.2 kBaud. It can receive measurements from the auto-bias controller of send new setpoint to the auto-bias controller.

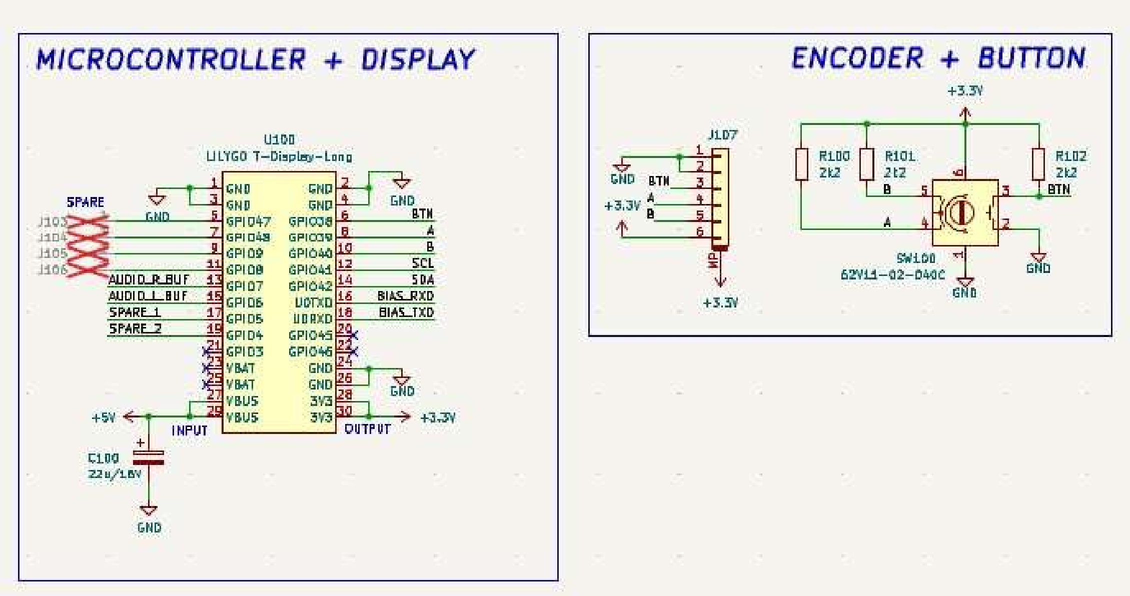


Figure 17 - Display connector board

Auto-bias system

The amplifier output transformers are toroidal transformers. I use these because these transformers have much better specifications than EI-transformers. They offer higher bandwidth and produce almost no stray field. So orientation is not an issue.

The downside of toroidal transformers is that they are very sensitive to dc-electric fields; this is because they lack an air gap. If the dc-currents running through the two primary halves of the output transformer is too large the transformer will saturate which will result in higher distortion at low frequencies. I found out that the sum of the bias currents running through those transformer halves must be less than 50 uA. This is difficult to achieve if we use trimmer resistors because these resistors and the tubes age and are temperature sensitive. So the bias currents will constantly change.

So I designed my own auto-bias system to actively monitor and change these currents to meet this 50 uA requirement all the time. The diagram below shows how that works.

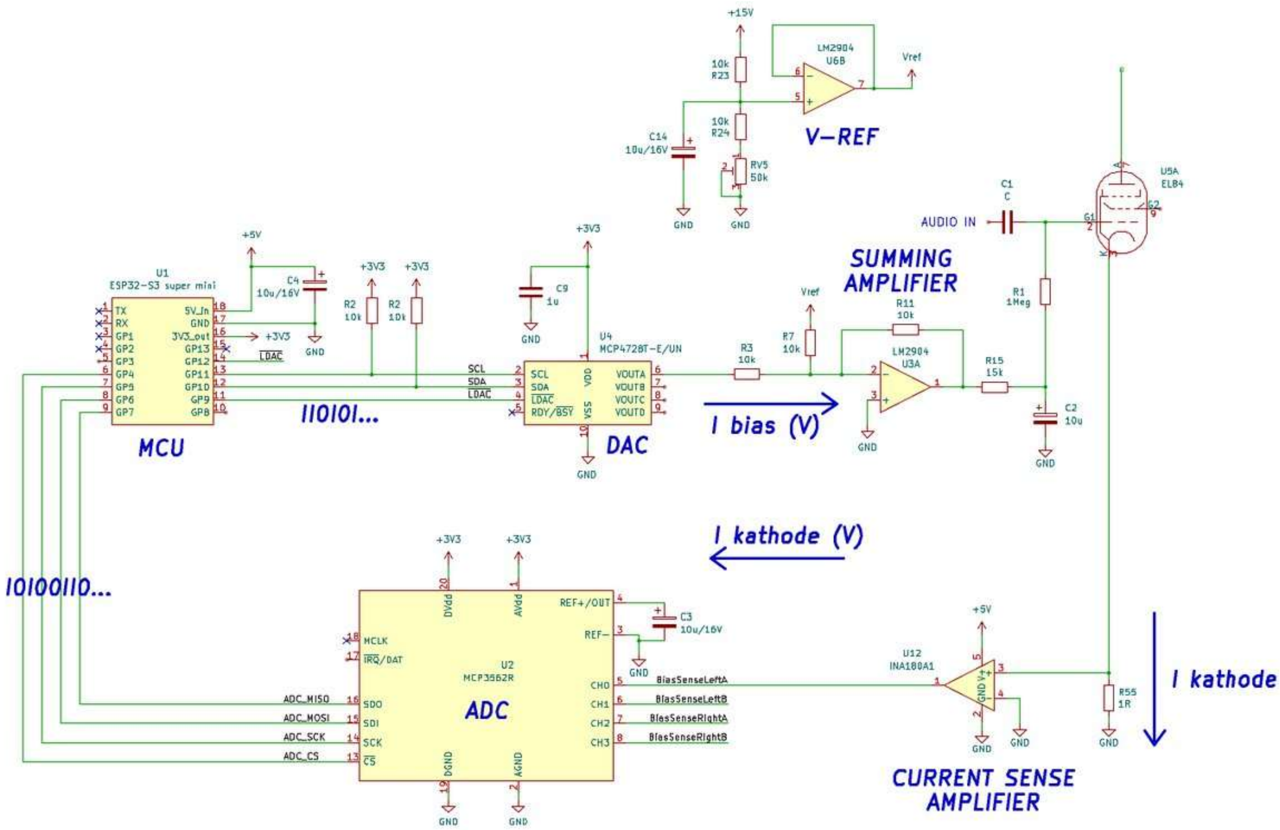


Figure 18 - Auto bias working principle

- The cathode current is measured as a voltage over a 1 Ohm cathode resistor (R55).
- This voltage is measured and amplified by 20 by a current sense amplifier and sampled by a 24 bit ADC with a sample rate of 153.6 ksps.
- A microcontroller (ESP32) captures these samples, which are already 256x oversampled by the ADC. The microcontroller receives about 600 samples per second.

- The microcontroller uses the bias current measurements only when there is no audio signal. That way we are sure the bias current is measured and not the music. If audio is detected the measurements are discarded.
- When no audio is detected the microcontroller takes 300 samples and calculates the average current. This current is the bias current.
- It then calculates the error between the measured current and the setpoint, and adjusts the 12-bit DAC accordingly.
- The DAC voltage is added to a reference voltage of 9V by a summing amplifier. The summing amplifier inverts the voltage to a negative grid bias voltage for the tube. So by changing the DAC the bias current is changed.

Below is a measurement example. The blue signal is the current caused by the audio signal. The green signal is high when the bias is being measured and low when the measurements are considered invalid. The measurements are valid when the difference between the highest and lowest current is less than 1 mA. That normally happens between the songs. The red signal is the average bias current.

It takes about 1.5 s for the system to get to the new setting after the DAC has been changed. So the system will wait for 2s before the next measurement and correction is done. On average about 2 to 4 corrections between songs can be made when the audio volume is low enough.

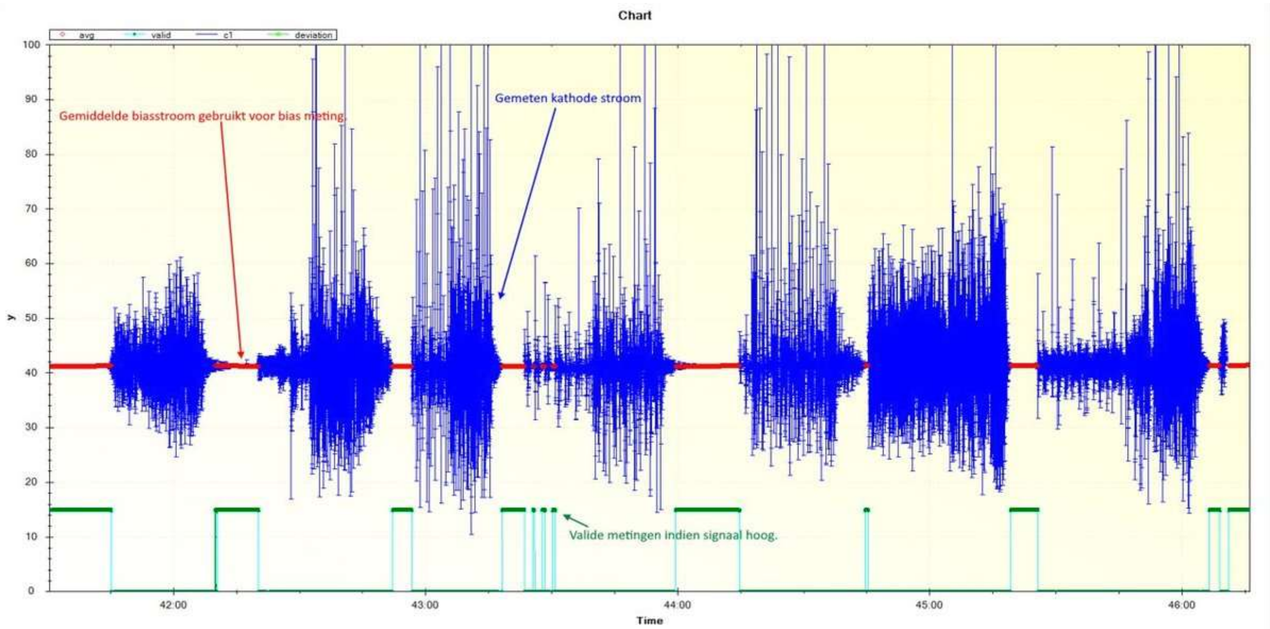


Figure 19 - Biasing data

Below the startup sequence of the amplifier measured by the auto-bias. The cathode current (blue) is rising as the filament heats up and the power supply ramps up to 280 V. The bias system waits until the signal is stable enough before it starts biasing. It normally takes about 1 to 2 corrections to get the current close enough to the setpoint (red). During this process, the volume is muted to prevent any audio signal from interfering with the measurements.

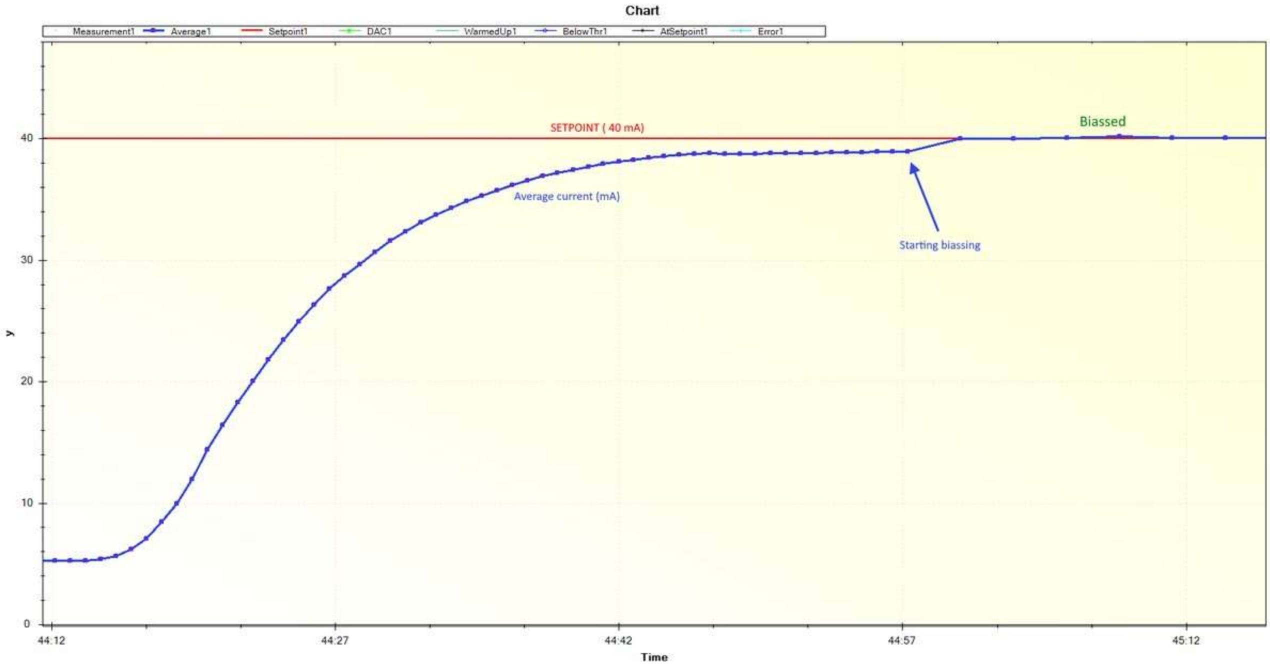


Figure 20 - Startup of the amplifier

Below another biasing example. This time during the playing of music. The grey dots are the cathode current samples measured by the ADC while music is being played. If the samples are below the threshold of, in this example, 1.5 mA then the measurements are valid and the system start to do corrections. The red dots show the corrections. Later the threshold was changed to 1 mA to get more accurate measurements.

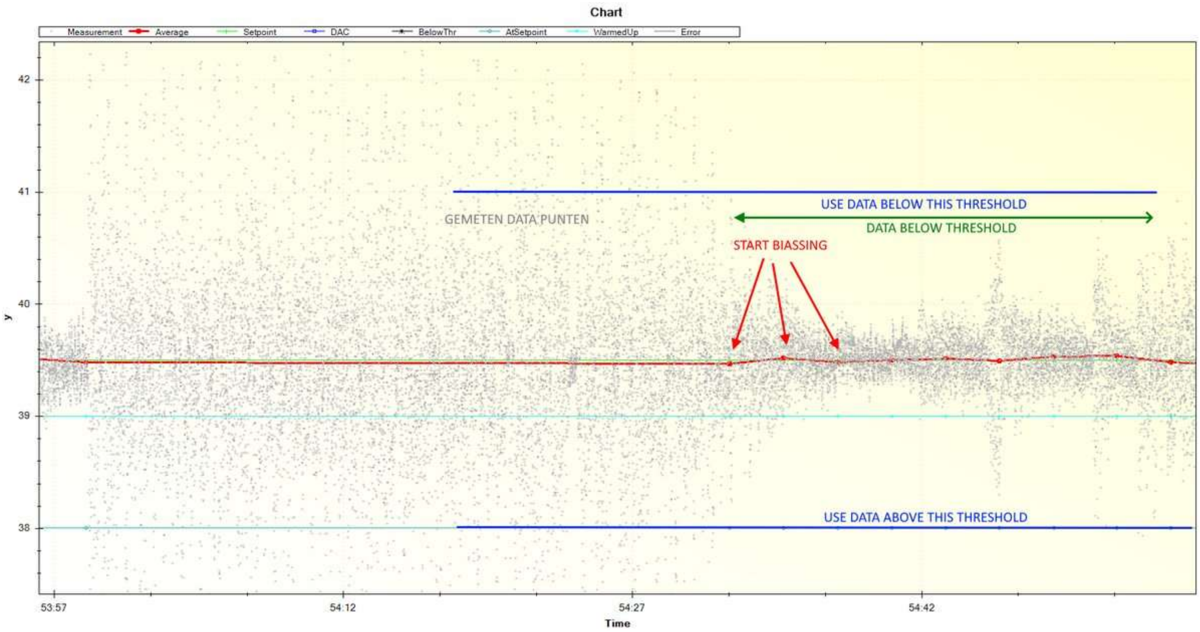


Figure 21 - Biasing while playing music

Here a photo of the display with the four cathode currents in a chart. This photo is taken when the biasing was disabled. You can clearly see the four separate currents. The currents all drift at the same rate. The scale of the y-axis is 2.5 mA.

Figure 22 - Drifting currents

I noticed that the drift is correlated to the mains voltage. This is because when the mains-voltage changes, the ac-heater voltage and therefore the current also changes. You can clearly see the correlation in the image below where the mains voltage and the DAC corrections of the enabled auto-bias system are both displayed. The system keeps the current the same. When the voltage drops the DAC voltage also drops which decreases the bias voltage and therefore increases the current.

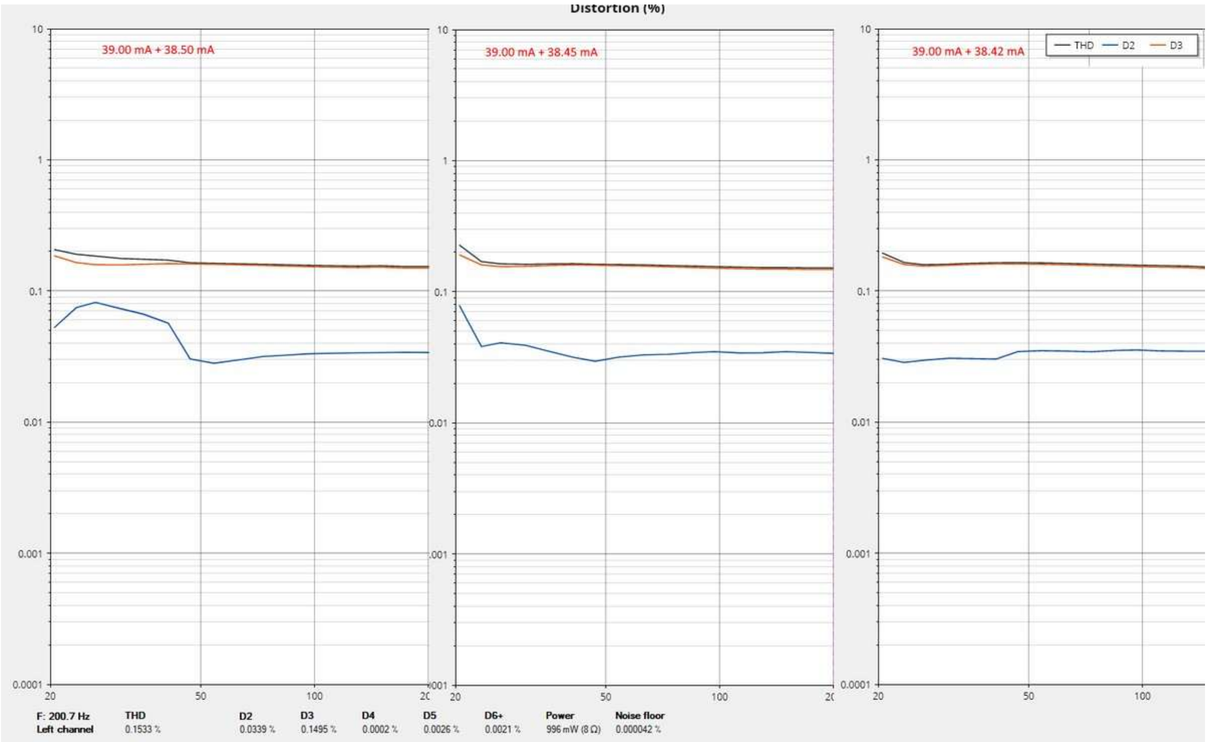
Figure 23 - Correlation between mains voltage and DAC value when biassing

Below again the cathode currents, but this time with auto-bias enabled. Notice the scale and the different setpoints of the currents. The red and green currents are the same (39 mA) and plotted on top of each other. The scale of the y-axis is 1.4 mA.

Figure 24 - Constant currents when biassed

The next image shows three separate measurements with different biasing settings. The right measurement has the least distortion. During that measurement the currents of the tubes were set to 39.00 mA and 38.42 mA. Although these settings give the best results the cathode currents running through the tubes are not both the same because the tubes are not the same. The anode currents are the same, but the screen grid currents are not. Therefore the cathode currents are also different because these are the anode and the grid currents added together.

When the current of one tube is changed by 80 uA (most left measurement) you clearly see the distortion at low frequencies caused by the transformer saturation.



So my initial thought of keeping the cathode currents through the tubes exactly the same is not possible. Doing that will cause a large distortion which you can see in the measurement below. The THD rises to 0.6 % instead of 0.16 % at 20 Hz.

I found out that after the best setting is found the difference between the two currents of a push-pull pair must not change more than 50 uA. Otherwise the distortion will rise and H2 will become higher than H3.

Setting the correct current unfortunately can only be done using THD-measurements. I first set the currents through both tubes the same and measure the THD between 20 Hz and 100 Hz. Then I change the bias current of one tube one way or the other until I get the least amount of distortion. In both channels the difference of the cathode current between the tubes was around 600 uA for the best result.

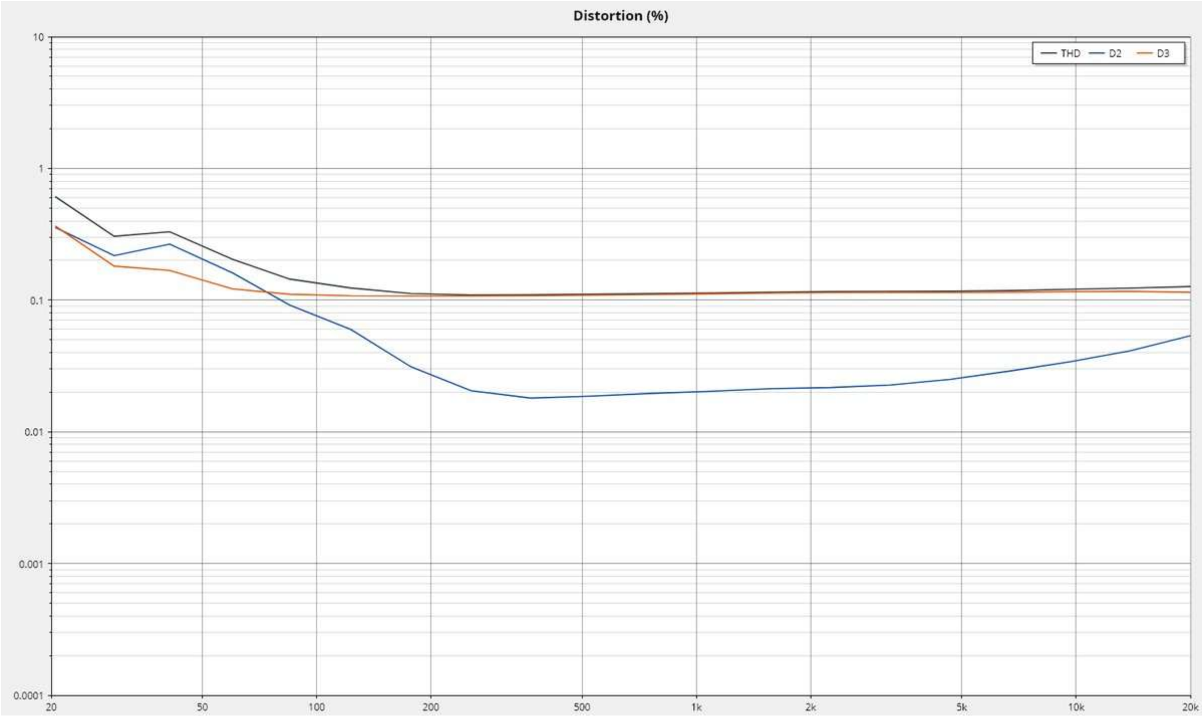


Figure 25 - THD when the cathode current of one tube changed 600 uA

Measurement results

To measure the performance of the amplifier I did some measurements. Most measurements have been made with the Quant Asylum Audio Analyzer QA403 and some custom software I built myself for this purpose. This software is free to use and the source code can be downloaded here: https://github.com/breedj/qa40x-audio-analyser

The first measurement shows the FFT between 20 Hz and 80 kHz.

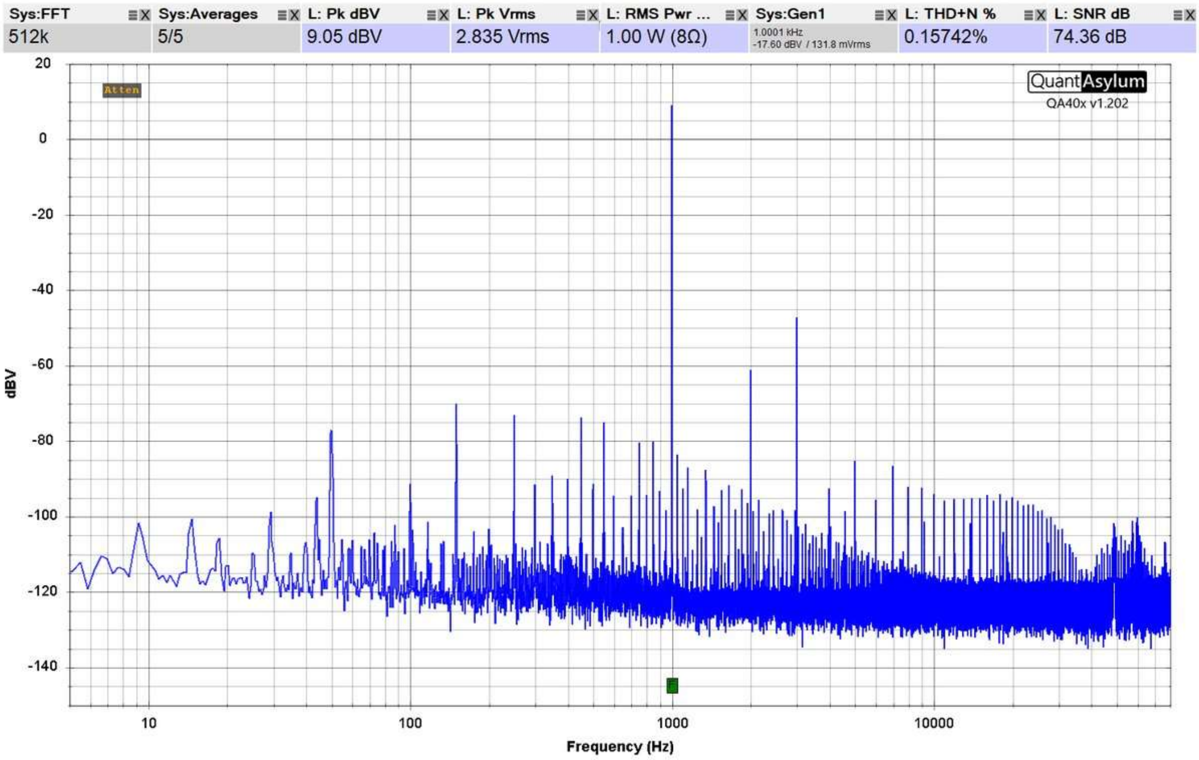


Figure 26 - FFT THD at 1 W / 8 Ohm

The next picture shows the same picture from the LTspice simulation. They look similar apart from the 50 Hz hum and its harmonics picked up from the surroundings and power supply.

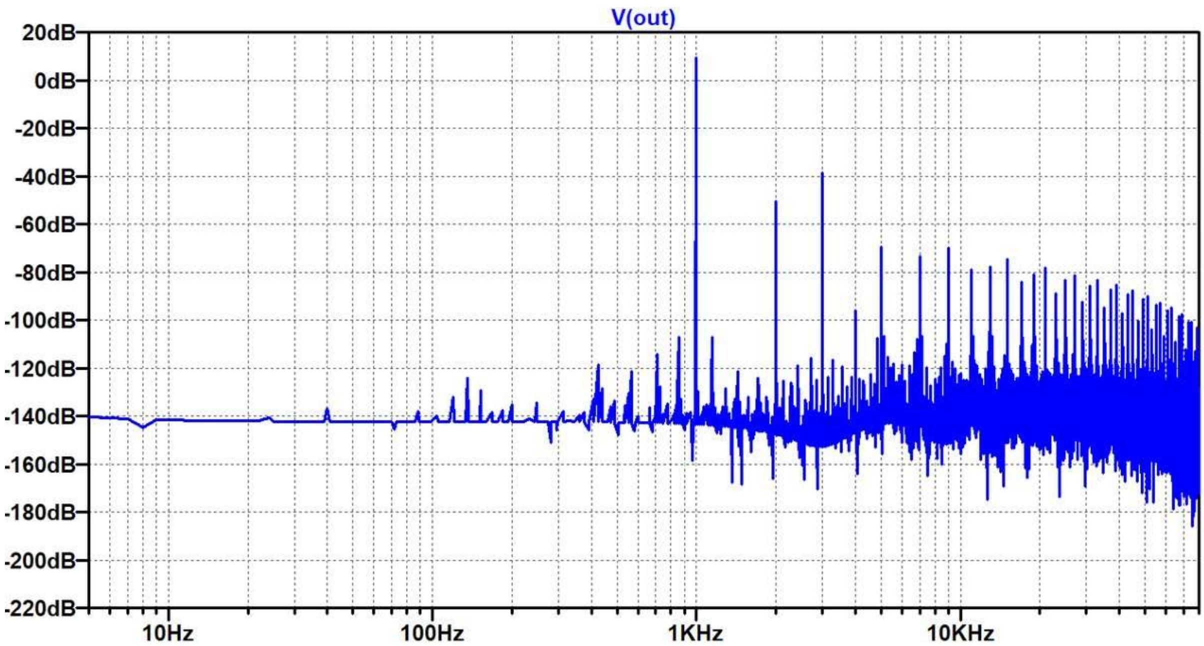


Figure 27 - Simulation at 1W / 8 Ohm

The distortion of the complete amplifier, including the output stage, at +9dBV output signal was measured to be THD= 0.16%. The noise floor is around 75dB.

The 1 dB bandwidth of the amplifier turned out to be 13 Hz – 80 kHz and the amplifier gain 26.6 dB.

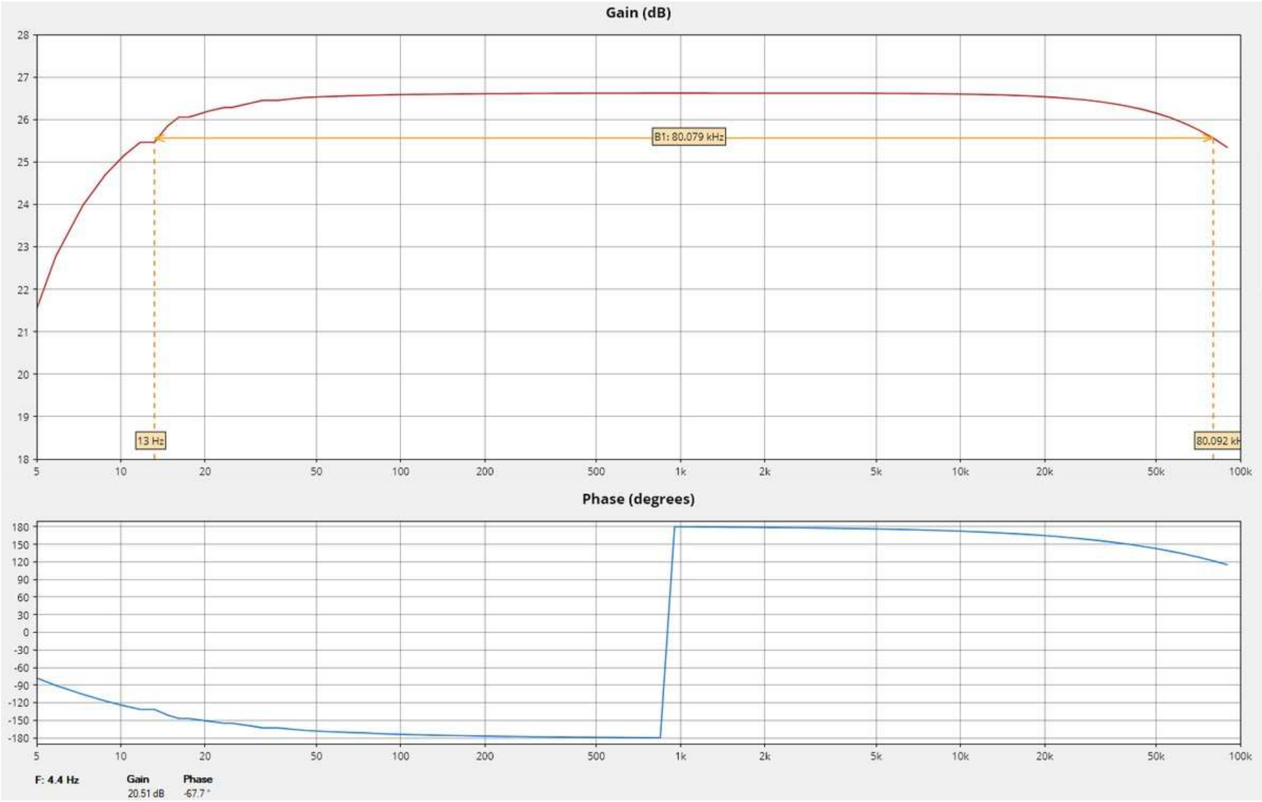


Figure 16 - Bandwidth measurement

The distortion between 20 Hz and 20 kHz is about 0.16 %. The 2nd harmonic is significantly lower than the 3rd harmonic. So my goal is achieved. The amplifier sound very clear and has a sharp base.

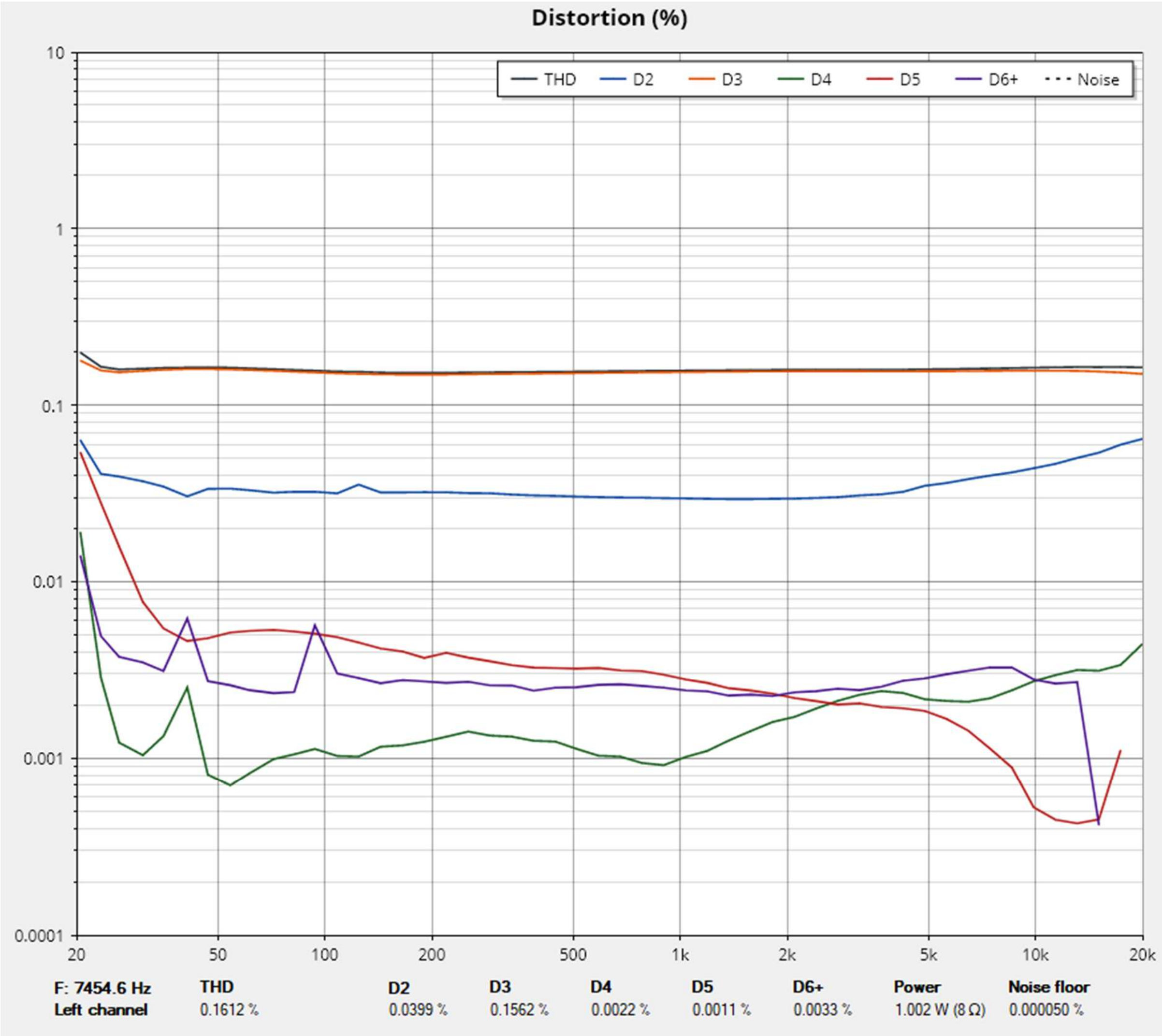


Figure 17 - Distortion measurement complete amplifier @ +9 dBV output level (105 mVrms input signal)

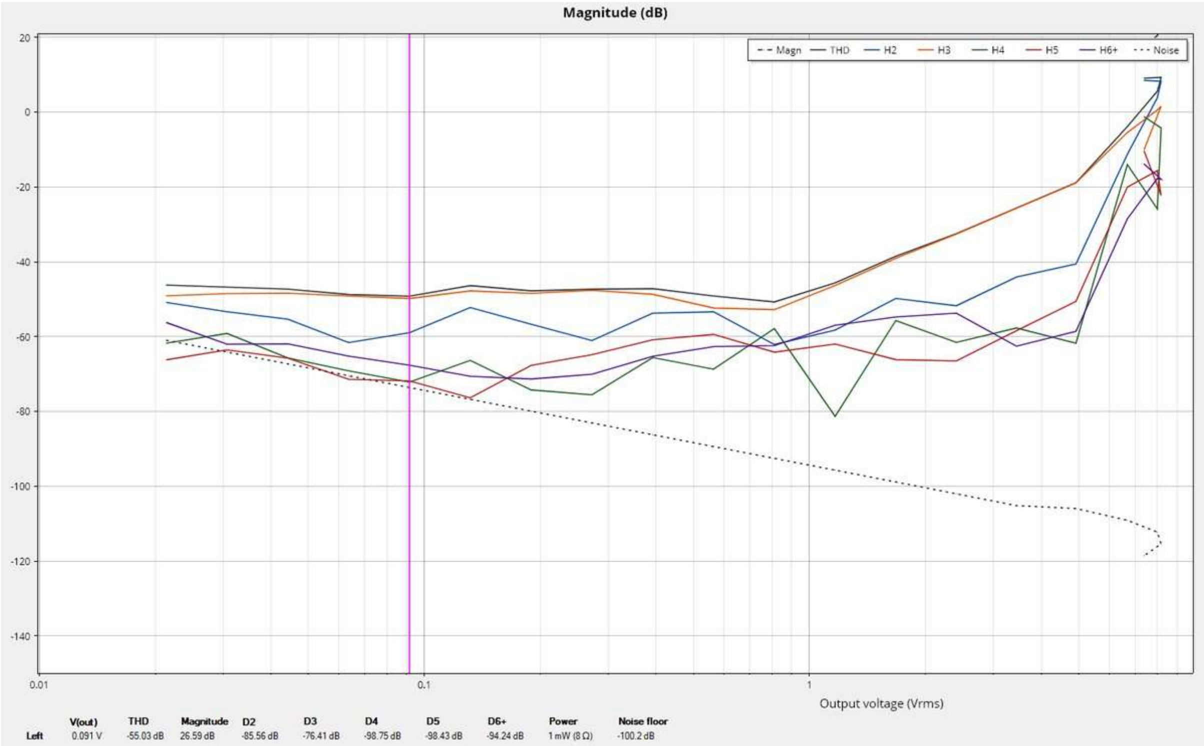


Figure 18 – Distortion against output voltage at 1 kHz

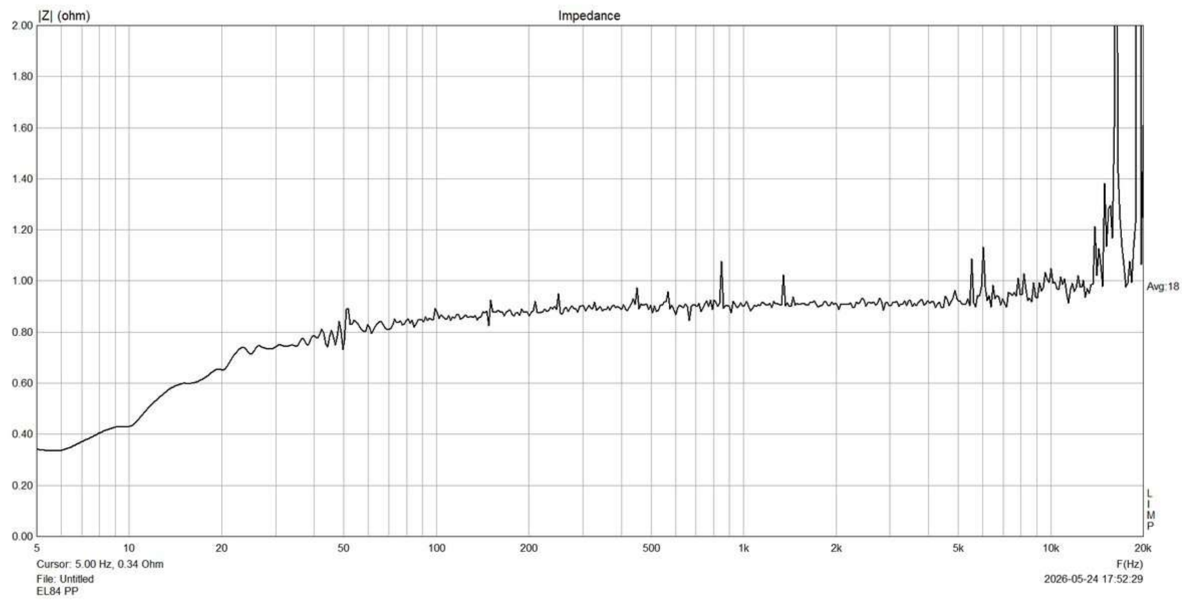


Figure 21 - Output impedance measurement of the amplifier is around 0.9 Ohm

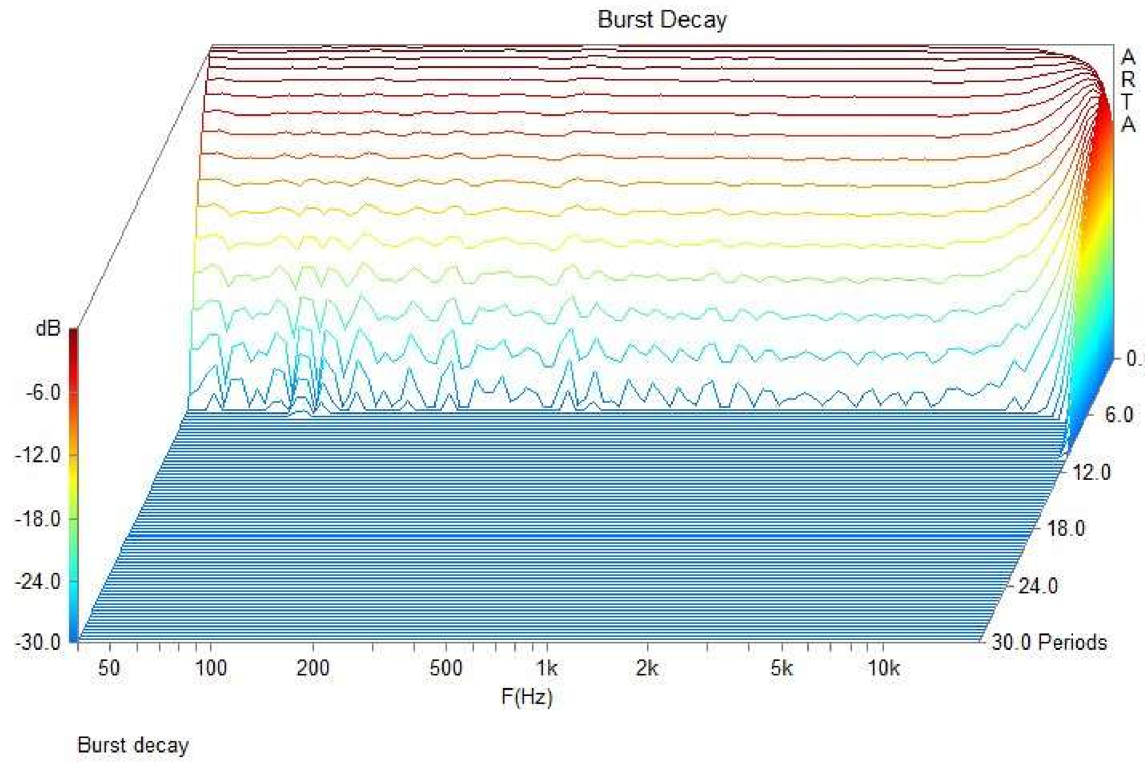


Figure 28 - Burst decay

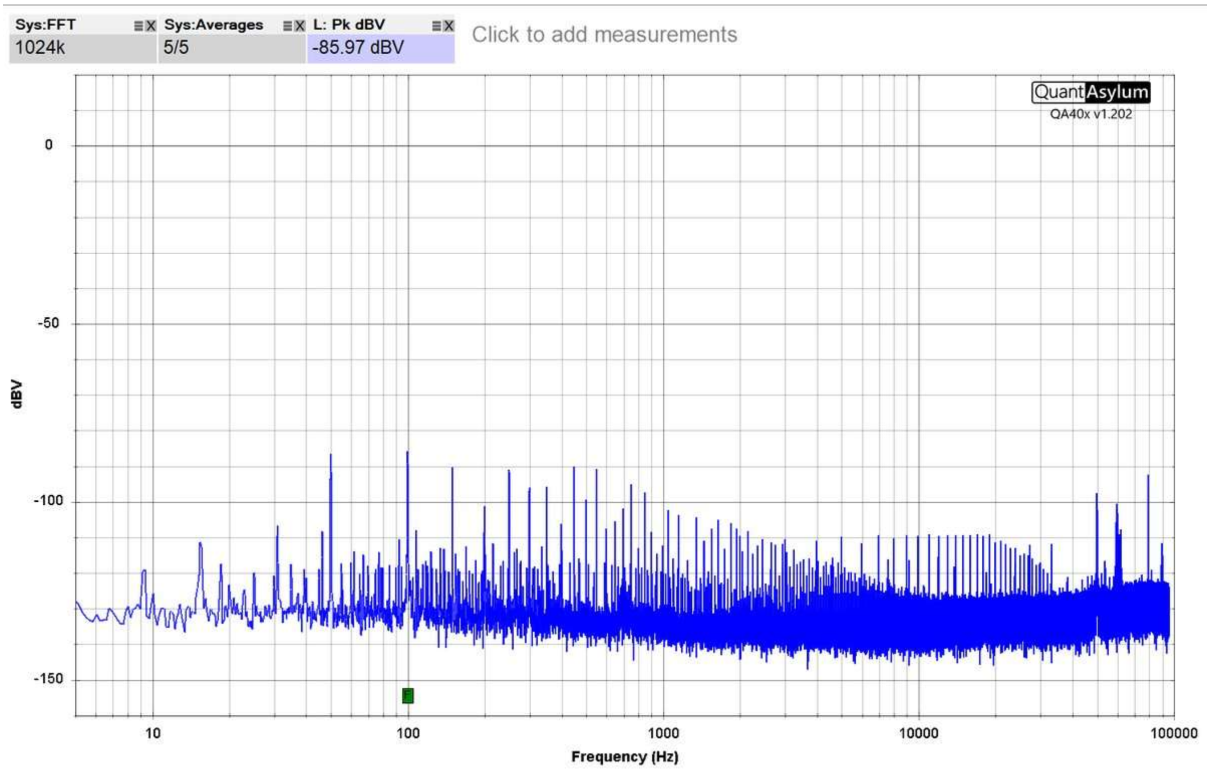


Figure 29 - Noise floor measurement. Noise including 50 Hz and 100 Hz hum is below -85 dBV.

PCB renders

Here are some renders of the pcb. Most of the pcbs are populated with smd components.

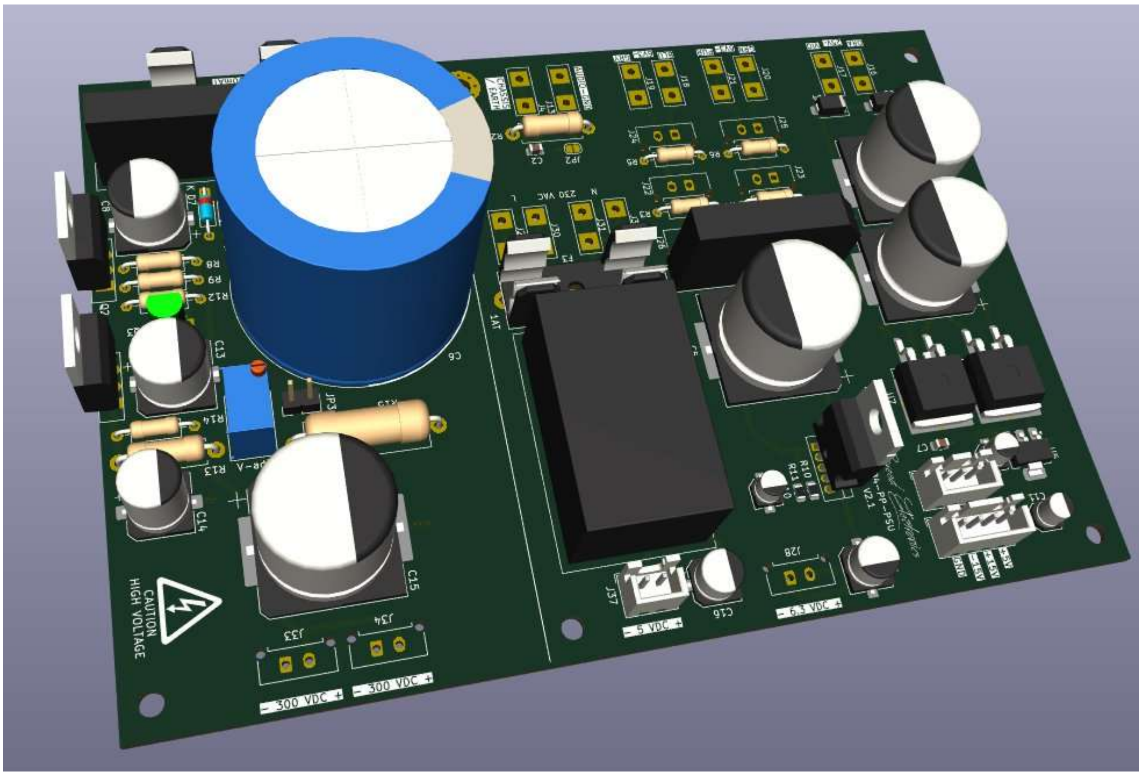


Figure 30 - Power supply pcb

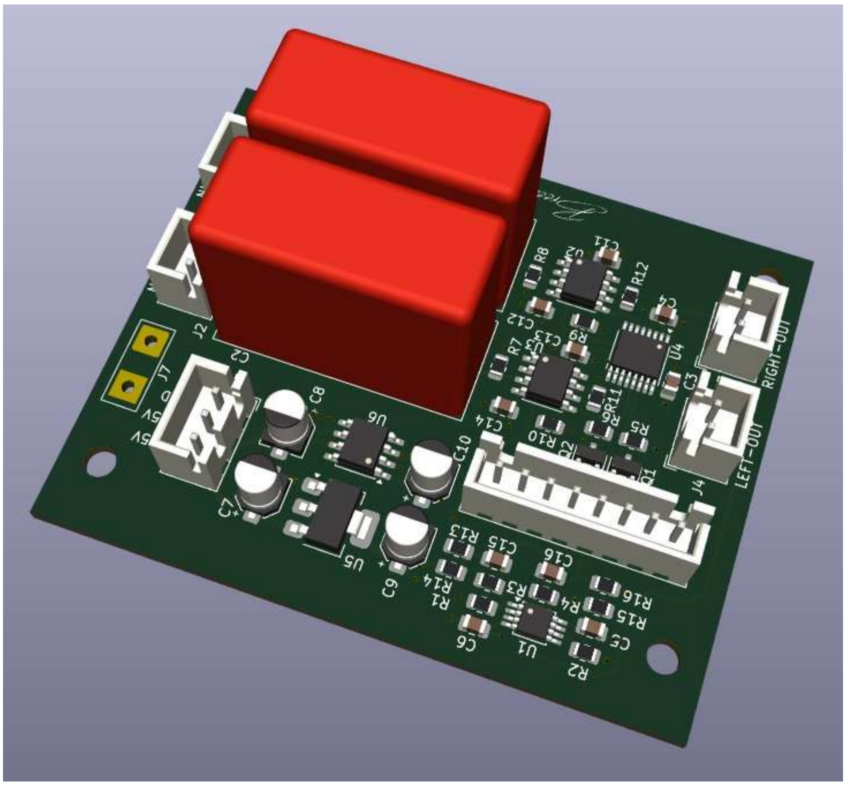


Figure 31 - Audio input pcb, mostly smd

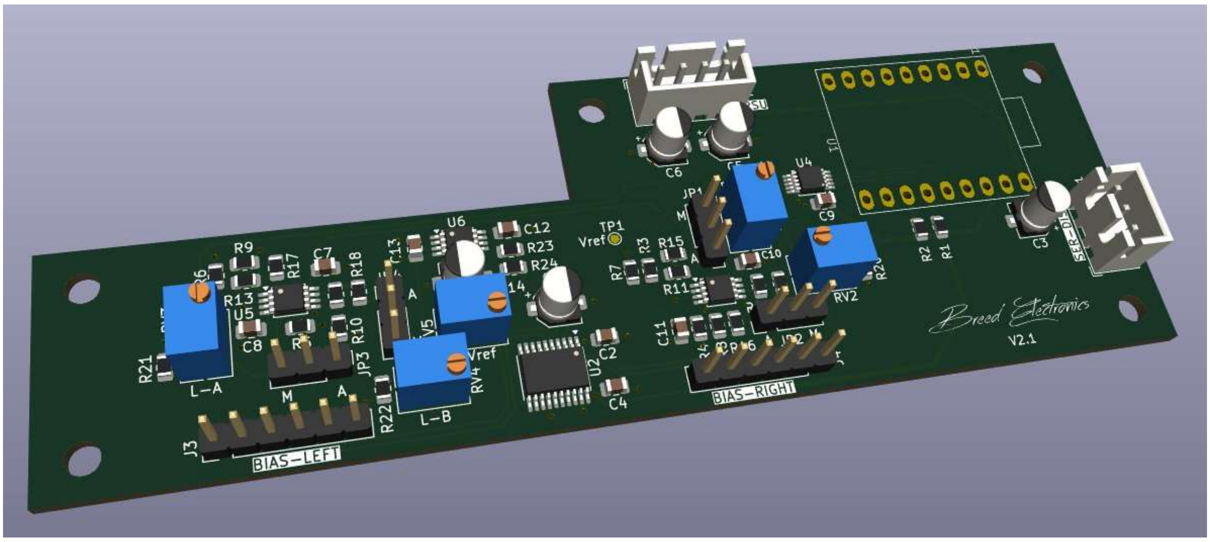


Figure 32 - Auto-bias pcb. It also includes trimpots for manual biassing. Jumpers can be used to switch between auto- and manual biassing.

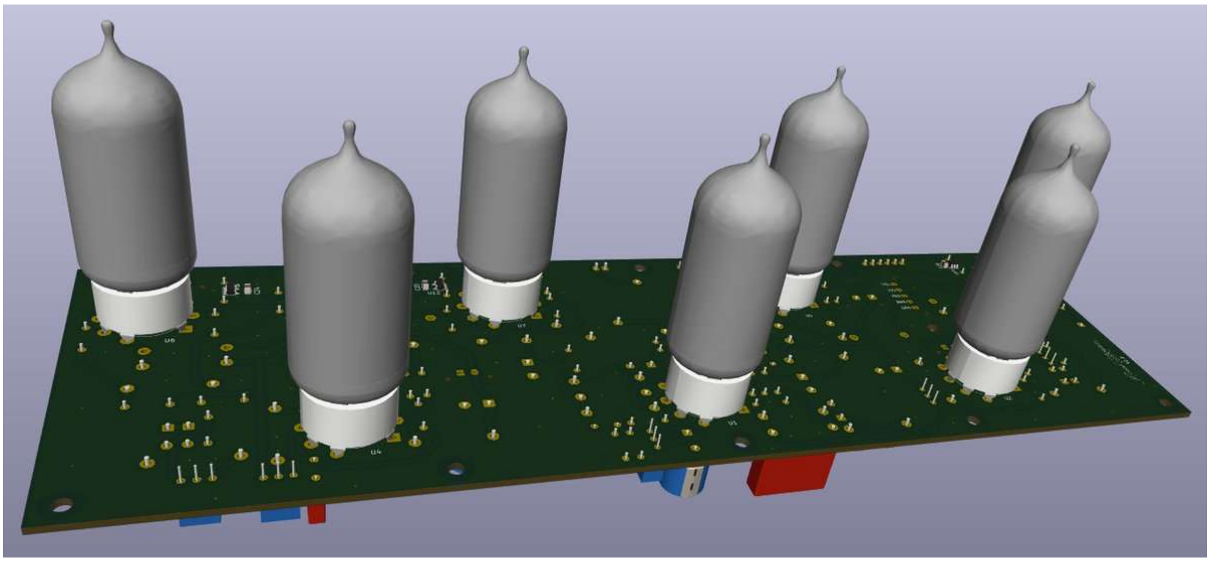


Figure 33 - Top side of the amplifier pcb

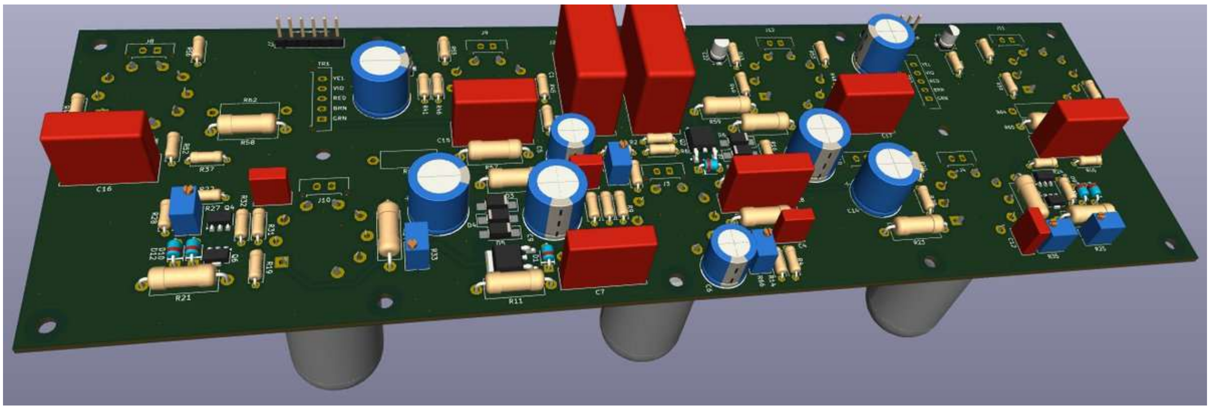


Figure 34 - Bottom side of the amplifier pcb

Enclosure

I designed the enclosure with a free online CAD program called OnShape. The parts are exported to STL files and 3D-printed in PLA using a BambuLab H2S 3D-printer. The inside of the enclosure is covered with copper foil which is connected to mains-earth for electric shielding. The sides of the amplifier are covered with walnut veneer. The veneer is glued to the PLA with epoxy.

All bolts are on the underside of the enclosure. So no bolts can be seen when looking at the amplifier from above. The two cylinders at the back are the output transformers. M3 inserts are sunken into the plastic with a soldering iron.

All pcbs are attached to the top. When the bottom cover is taken of all components can be accessed easily. One downside is that of the amplifier is powered on when laying on its top the pla tends to warp. Also the mains transformer gets very hot, around 50 ºC which can get problematic. So I might swap the pla top and cover for milled aluminium top and cover in the future.

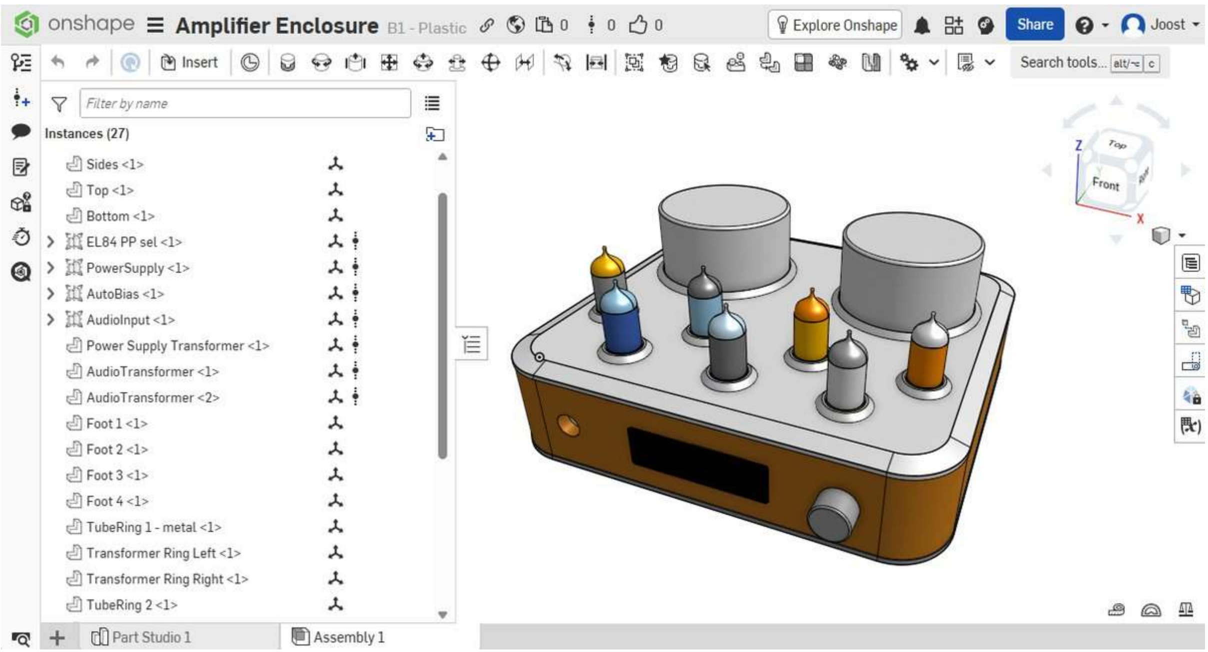


Figure 35 - Designing the enclosure in OnShape, which is free to use.

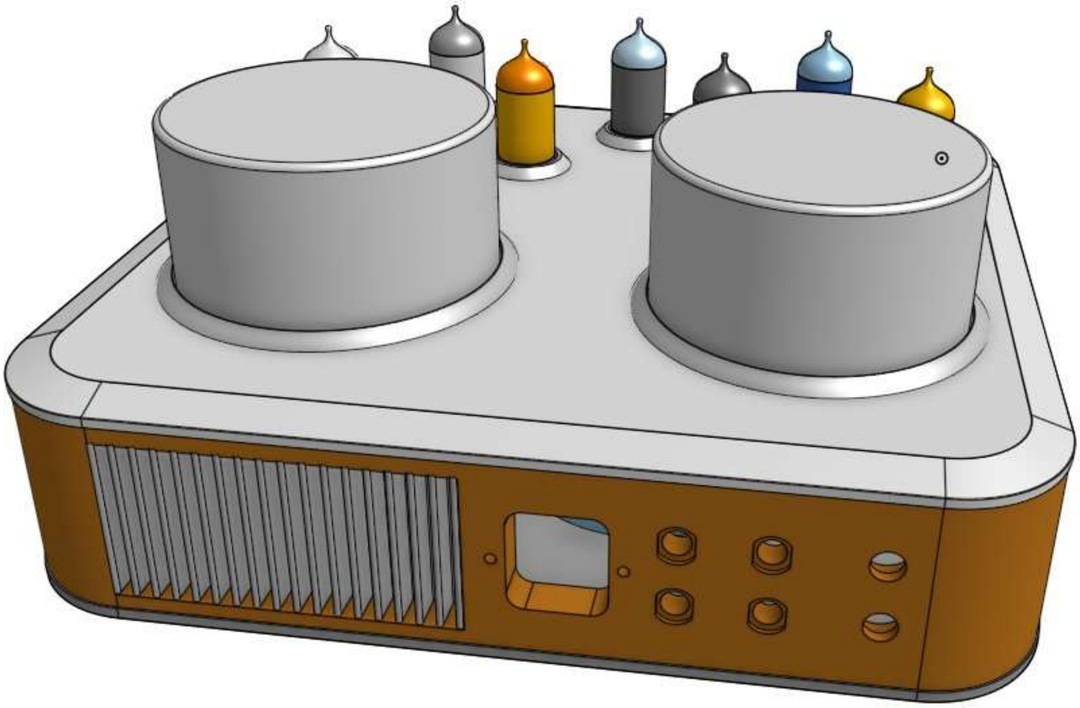


Figure 36 - Render of the back side. Notice the heatsink for the power supply. It only reaches 25 C.

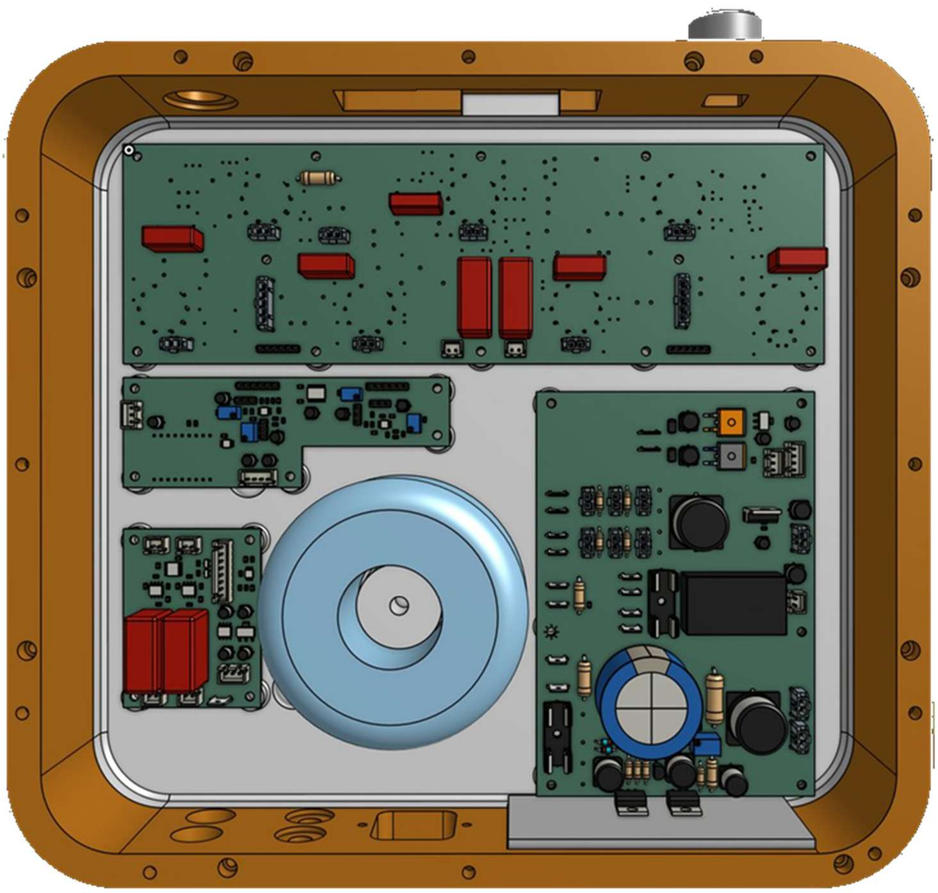


Figure 37 - The PCBs were exported from KiCad and used in OnShape for the correct measurements.

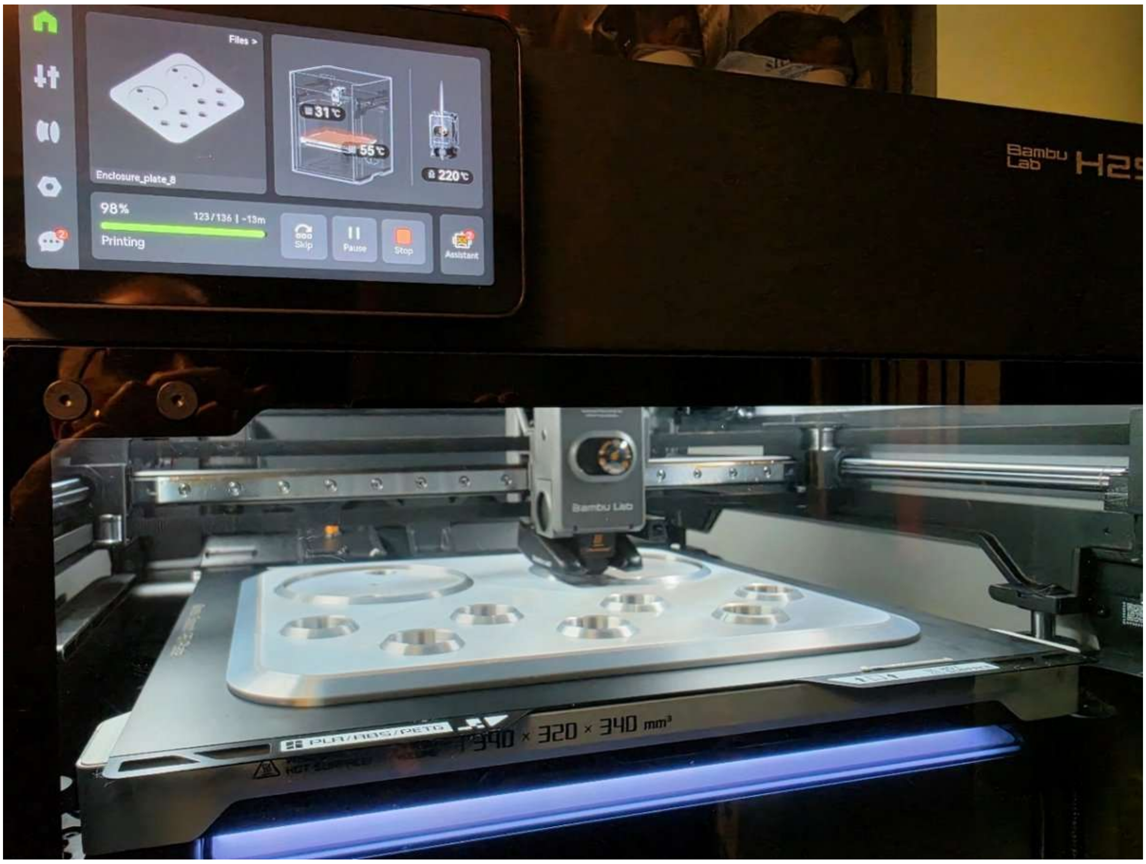


Figure 38 - The top side being printed by the 3D printer.

Figure 39 - Copper on the inside and veneer on the outside. All copper is connected together via wires.

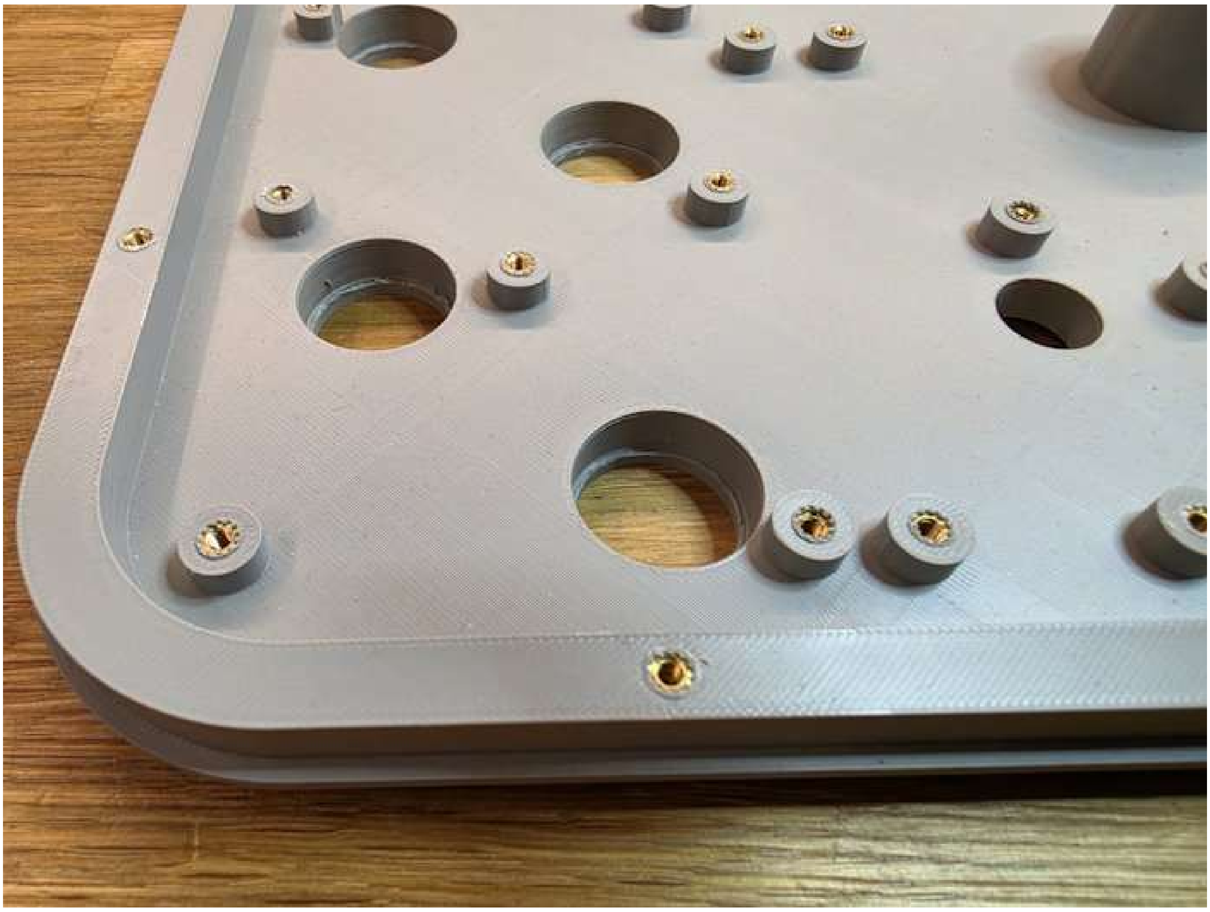


Figure 40 - M3 inserts are used. The topside of the enclosure is shown here upside down.

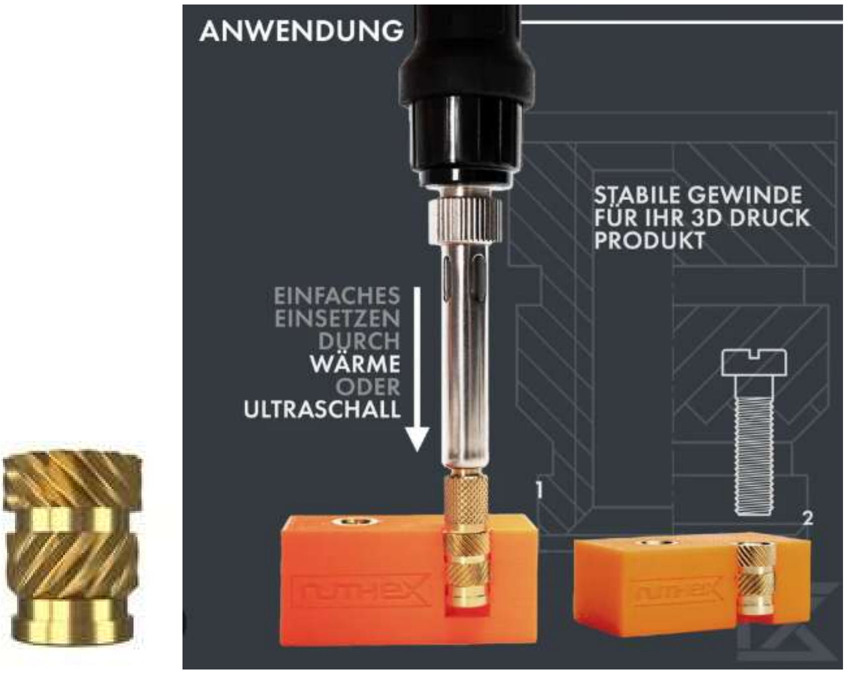


Figure 41 - Inserts I used are made by Ruthex. Specially designed for 3d printing.

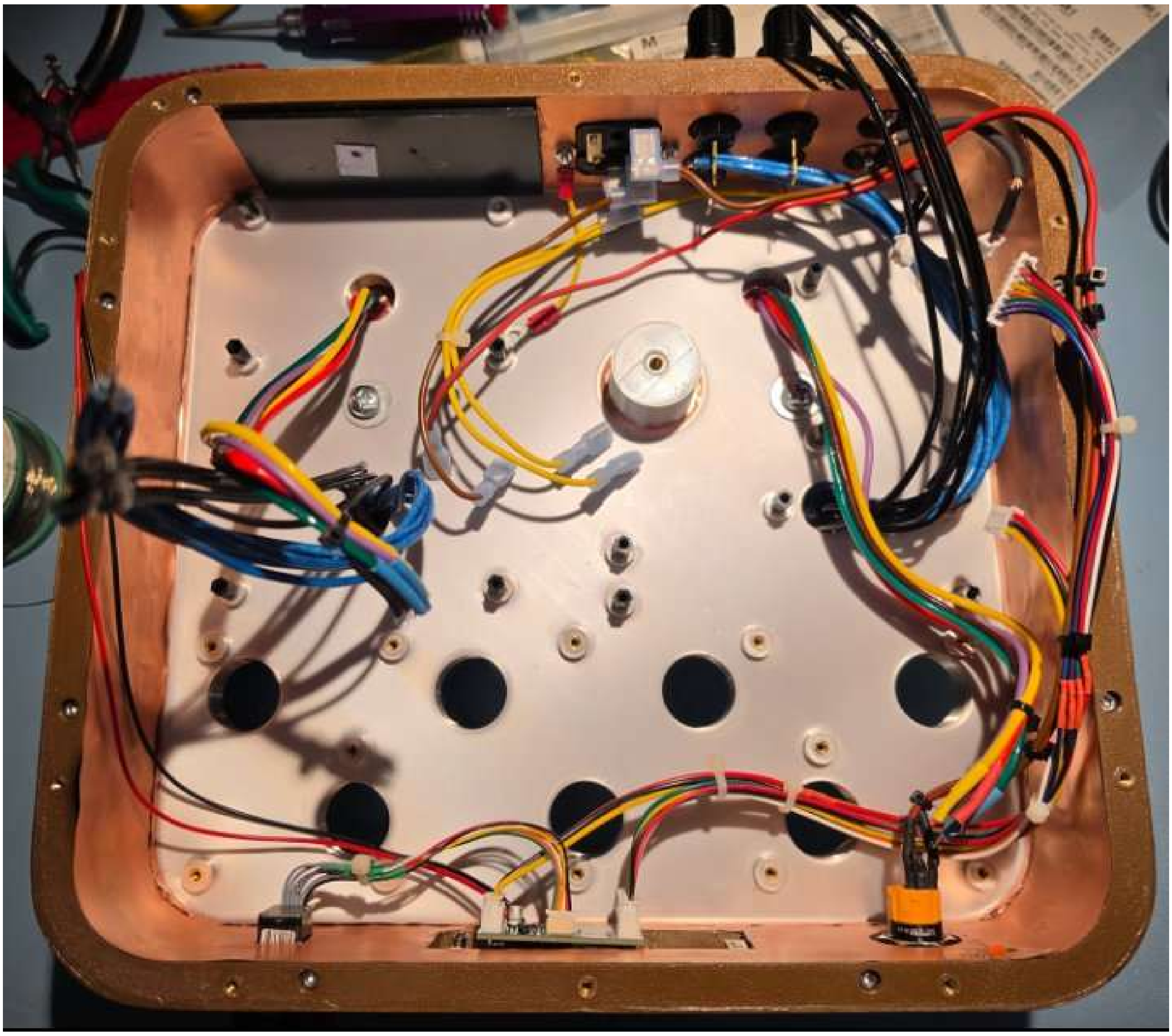


Figure 42 - The enclosure seen from the underside with copper foil in place, the copper foil is also beneath the grey cover. Standoffs are used to mount the pcbs.

Figure 43 - All glowing well. The pla does not get soft at all.

Figure 44 - Backside of the amplifier.

Software

All software for the microcontrollers was written in Visual Studio Code with PlatformIO. One project for the auto-bias and one other project for the display.

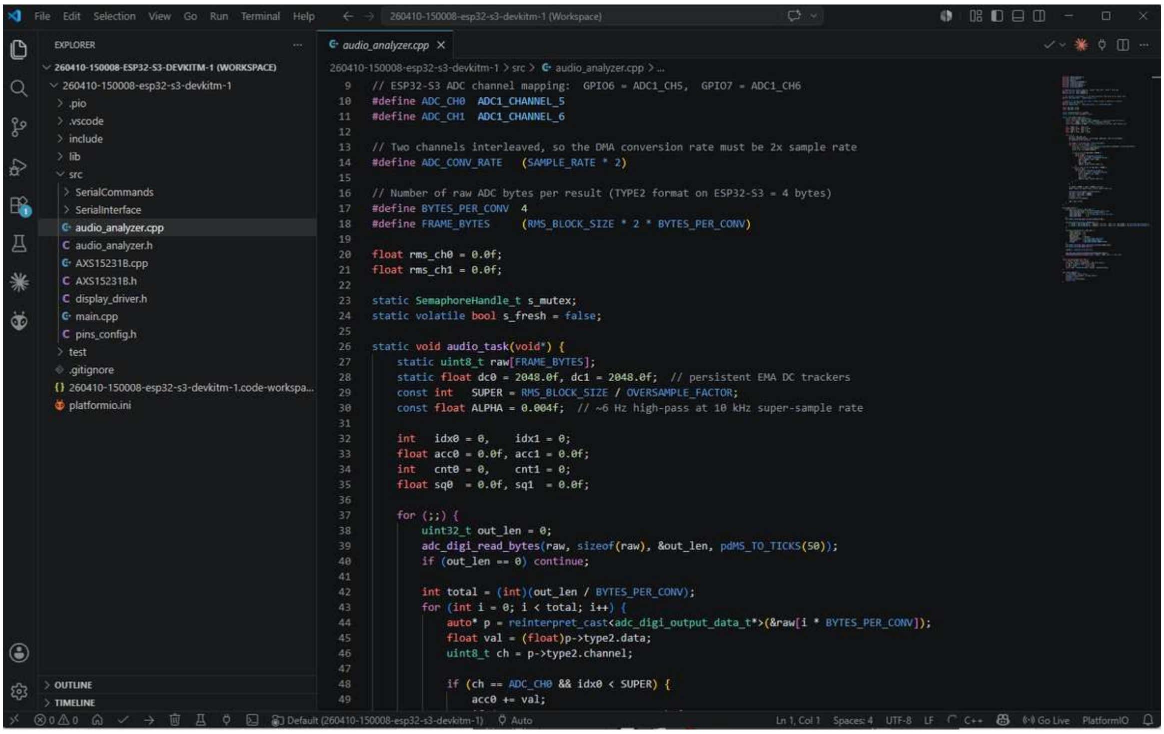


Figure 45 - Visual Studio Code

The screens were designed in Squareline Studio and then imported in Visual Studio.

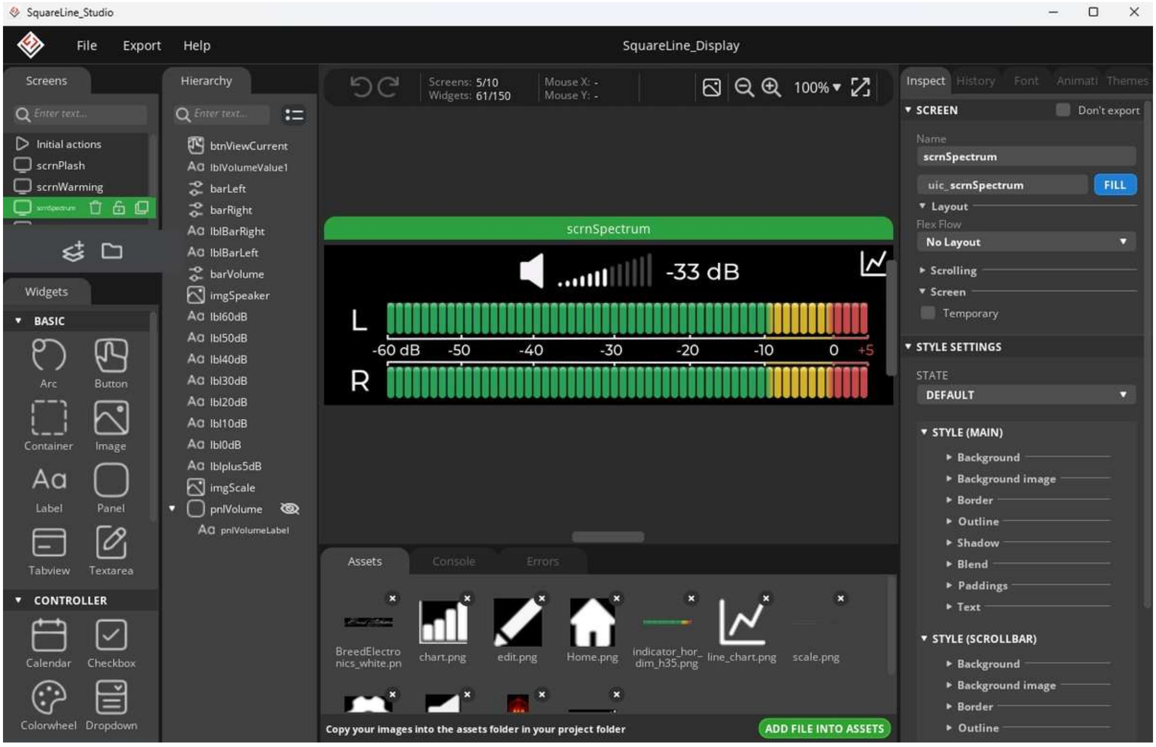


Figure 46 - Squareline Studio

Screen designs

Here are photos of the screens designed.

Figure 47- Splash screen

Figure 48 - Waiting for the current to stabilize during startup.

Figure 49 - The home screen which shows the volume and the signal level.

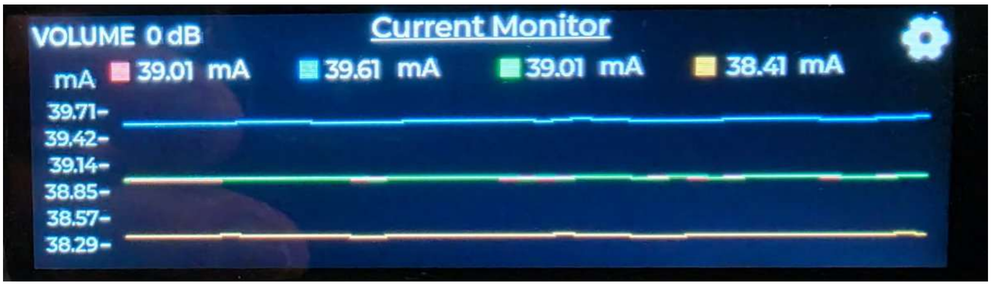


Figure 50 - The cathode current monitor to check the auto bias's performance

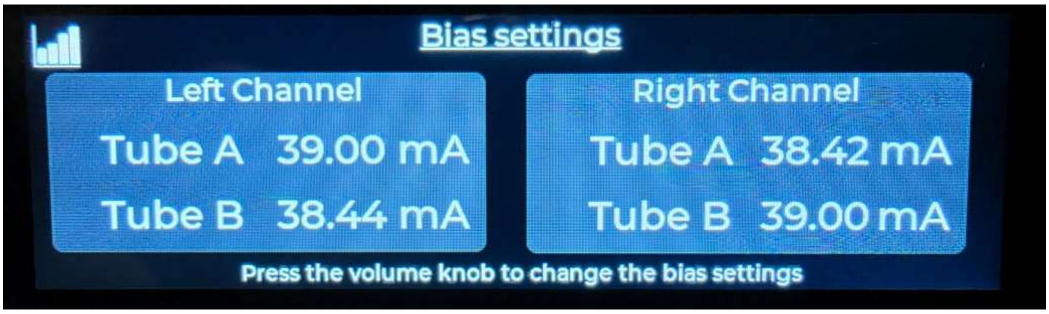


Figure 51 - With this screen the currents can be set