

AH PRE-1 Highend-Preamplifier

Development Manual

Andreas Hünnebeck
ah@bruker.de

29th January 2000

Preface

Is It Really Worth to Make DIY Stereo Equipment?

Like most questions it is not possible to give a definite answer to this question. Regarding “lo-fi” stereo components like modern Mini systems or “mid-fi” components like small amplifiers, tuners and so on ranging in price up to 500 US\$ you must strictly deny it: any DIY project would be much more expensive because this type of stereo equipment is produced very cheap in very large quantities.

This changes however if you look at components which are made in small numbers where the price of the raw materials is much less than the price you have to pay to a decent dealer. This is especially true for the so called “highend” equipment-often the ratio between the dealer’s price and the raw materials is about tenfold or even higher. Just have a look into the innards of a preamplifier and make a short calculation of its sparse contents. Also this equipment is usually very beautifully made and built to last, with nicely decorated front sides and gold letters all over the place. This increases the costs of production even more and is something the DIY maker need not to do-a preamplifier sounds the same whether it sits in a regular 19” housing available at your local Radio Shack or in a platin housing with quarter inch golden front plates. Therefore it is possible to make very good highend equipment by yourself without paying much money as long as your time does not count.

Furthermore DIY of stereo equipment has several advantages: fine tuning of the electronic circuits allows the adaption to your personal taste, and you do not need to buy new equipment after some time due to new techniques-just replace old fashioned parts with modern ones to bring the component up to date.

DIY of Preamplifiers

The most preferable and thankful DIY projects are preamplifiers:

- it is not very expensive to built one
- you can get very high sound quality
- there is not much that can go wrong
- if something goes wrong there are no serious consequences (in contrast to power amplifiers)

Furthermore the preamplifier is the “heart” of the stereo system and you can put in as much (or as few) control elements as you like or need. Phono (MC or MM) or not, 10 line level inputs or just 2, no tone control or an equalizer-you name it, everything goes. The “highend” scene seems to be

in the age of minimalism¹ which means that even (or especially) the most expensive components have just 4 line level inputs, no phono of course, and leave out the tape monitor switch to improve the sound. The developer never heard of headphones and an external headphone amplifier cannot be connected due to the missing outputs.

Why Did I Make My Own Preamplifier?

There were several reasons that eventually lead me to make my own preamplifier:

- My old preamplifier had no headphone output.
- It had no separate source selectors for listen and record.
- While tuning the circuits of the old preamplifier I reached the limit of its design. No more improvements could be gained without developing new circuits.
- While tuning the circuits of the old preamplifier I realized that I could do it better.
- During “studies” of the innards of highend preamplifiers (e.g. Bryston 25BP) I realized two points:
 1. "Don't pay that much money for that small amount of content."
 2. "It should be no problem to make something equivalent for much less money."

Acknowledgements

My thanks go to Thomas Magin, who converted my manually drawn schematics into CAD format so it is now possible to make a pcb. Furthermore I have to thank my (former friend and) wife who missed my company for many evenings and weekends and instead had to survive uncountable listening tests.

¹This kind of minimalism reflects itself also in the circuits, see for example the products of Nelson Pass, Greenwall or Brain Audio (what happened to their super transistor?). They claim to use just one active amplifying part (usually a Mosfet) and very few passive parts. I'm just wondering then why you have to pay the same amount of money like for the “parts grave” of Mark Levinson et.al.

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Chapter 1

Requirements And Design

1.1 Requirements

The preamplifier should fulfill the requirements as listed below:

- Inputs
 - 5 asymmetrical line level inputs
 - 1 asymmetrical MC-phono input with about $150 \mu\text{V}$ sensitivity
- Outputs
 - 1 output for a tape recorder
 - 1 output to drive the power amplifier
 - 1 output to drive a headphone of normal sensitivity and medium impedance
- Control elements
 - separate source selectors for listen and record
 - no tone controls
 - no balance controls
 - a switch to route the output alternatively to the headphone or to the power amplifier
- requirements regarding the sound quality
 - external power supply
 - no coupling capacitors within the signal chain
 - no DC offset servo control loop to prevent “pumping”
 - frequency response should start at DC if technically possible
 - all switching done by relays located as short to the signal chain as possible
 - simple circuits with as few active parts within the signal chain as possible
 - separate power supply regulation for every circuit module

1.2 Design

The preamplifier contains two amplifier modules:

- a phono preamplifier on its own circuit board which amplifies the signal of a MC phono pickup to line level
- a buffer amplifier connected to the volume control potentiometer which buffers the line level signal and amplifies it by about 10 dB to drive both head phone and power amplifier.

There is no buffer before the volume pot simply because you don't need one and it would just add more distortion. The same is true for the buffer for the tape recorder output: there is none.

There is a 1 x 6 source selector switch connected to the volume pot which is used to select the source you want to listen to. The switch itself is a cheap one as the audio signals are switched by very good relays.

There is a second 1 x 6 source selector switch connected to the tape recorder output which is used to select the source you want to record. The switch itself is a cheap one as the audio signals are switched by very good relays.

Chapter 2

The Phono Preamplifier

2.1 Design

The phono preamplifier amplifies the audio signal of a MC¹ phono pickup with a sensitivity of $150\mu\text{V}$ to line level and also equalizes the frequency response according to the RIAA deemphasis curve (see [Hü97]). Additional equalization according to the IEC deemphasis curve is not done as my turntable does not produce significant drive noise and the electronics following in the signal chain is able to handle all signals produced by the MC pickup. Usually there is at least one highpass somewhere within the signal chain between pickup and speaker which has enough suppression of sub bass frequencies.

The preamplifier consists of several coupled stages (see the schematics in figure 2.1 on page 15):

1. an ultra low noise linear amplifier stage offering 40 dB amplification,
2. a passive lowpass with a corner frequency of 2122 Hz ($75\ \mu\text{s}$),
3. a second amplifier stage which also works as active filter for the poles $3180\ \mu\text{s}$ and $318\ \mu\text{s}$,
4. and finally an offset compensation circuit which eliminates the DC offset of the first amplifier stage.

You must use at least two amplifying stages because an amplification of 64 dB at 1kHz (84 dB at 20 Hz!) cannot be handled by one single stage in high quality. The THD ratio would be high because a single stage has not enough noise gain to compensate the inherent non linearities.

I choose two separate filters instead of one composite filter for the RIAA equalization because it is very difficult to construct a three pole filter where the poles are very close together and influences each other. If you look into the excellent paper in [Lip79] you'll understand what I mean.

2.1.1 The First Amplifier Stage

Nowadays you can get extremely ultra low noise operational amplifiers which compare with transistor pairs regarding price but are more easily available. Furthermore a discrete amplifier using a transistor pair needs very high precision resistors and a very good stabilized supply voltage. Operational amplifiers on the other hand have all these parts built in. Examples of ultra low noise operational amplifiers are LT1028 by Linear Technology and AD797 by Analog Devices.

¹Moving Coil

A none inverting amplifier is built with the feedback resistors $R_8 = 1k\Omega$ and $R_{12} = 10\Omega$ to get an amplification² of about 40 dB. Usually I do not use feedback resistors with such low resistance to prevent overloading the output of the operational amplifier (it gets warm and does not sound very good). This case is an exception because the input noise current of the operational amplifier flows through R_{12} which means that the output noise increases with the resistance of R_{12} . Therefore its resistance should be so low that its residual noise is small compared to the input voltage noise of the amplifier.

The input impedance is set by putting R_2 in parallel to the input. Its value should be selected according to the specifications of the MC pickup. Usually this is 100-150 Ω , for low output MC pickups, high output pickups might need up to 1000 Ω (check the owner's manual). I chose $R_2 = 100\Omega$.

This means that the none inverting input sees a source impedance of $100\Omega||6\Omega$ and the inverting input of $1000\Omega||10\Omega$. As the input offset current (which is relatively high) flows through both input impedances the imbalance leads to an additional input offset voltage which is then amplified by 40 dB. Usually this can be prevented by adding an additional resistor at the inverting input to recreate the balance. I did not do this because the input noise current also flows through this additional resistor, thereby adding unwanted voltage noise. The resulting offset voltage of the first stage is later eliminated by the offset compensation circuit.

At some later time I replaced R_8 in one channel by the serial connection of a 750 Ω resistor and a 500 Ω adjustable resistor. This allows to balance pickups with unequal output voltage.

2.1.2 The Passive Lowpass Filter

I chose a passive filter to realize the 75 μ s lowpass for the following reasons:

1. The 75 μ s lowpass must be the first filter to prevent clipping of the following amplifier stages by high frequency transient signals (e.g. due to scratches).
2. It is not advisable to realize the first amplifier stage as an inverting stage due to the high noise contribution of the input resistor. This input resistor of typical 100 Ω would be added to the output impedance of the pickup-approximately 6 Ω at my MC pickup-which would increase the input noise voltage by a factor of

$$\sqrt{\frac{100\Omega + 6\Omega}{6\Omega}} = 4.2 = 12.5dB$$

This cannot be accepted.

3. It is not possible to construct the 75 μ s lowpass as active filter using a noninverting amplifier, because the minimum amplification is always 1 (0 dB) which inserts a second pole into the frequency response. This second pole can be compensated by adding a second lowpass to the output of the active filter. Unfortunately the second pole depends on the amplification of the filter, which means that an adjustment of the amplification (which is necessary as output voltage and source impedance of moving coil pickups vary considerable, e.g. 50 μ V to 2 mV and 2 Ω to 100 Ω) requires an adjustment of the compensation filter. This is very impractical to implement.

²The amplification is calculated according to the equation

$$V = 1 + \frac{R_8}{R_{12}}$$

The passive lowpass consists of a simple RC filter. It is calculated as follows:

$$\begin{aligned} \text{From } \omega &= \frac{1}{2\pi f_g} \quad \text{and} \quad f_g = \frac{1}{2\pi R_{20}C_{35}} \quad \text{results :} \\ \omega &= R_{20}C_{35} \quad \text{therefore} \quad C_{35} = \frac{\omega}{R_{20}} \quad \text{or} \quad R_{20} = \frac{\omega}{C_{35}} \end{aligned} \quad (2.1)$$

The capacitor should be of the highest quality both in precision and sound. The only choice is a styroflex capacitor, which is available with 1% tolerance. The biggest value is just about 15nF. I chose $C_{35} = 10nF$ which leads to a resistance of $R_{20} = 7,5k\Omega$ according to equation 2.1, and this part is available with 1% tolerance. I got hold of a large bunch of 10nF capacitors and used a capacitance meter to select two capacitors of equal value (to improve channel balance) as close to the specification as possible.

2.1.3 The Second Amplifier and Filter Stage

The second amplifier stage also serves as filter for the poles $318\mu s$ and $3180\mu s$. It is possible to use a passive filter-however that makes no sense as then the signal must be amplified much more which leads to increased distortion, and this is not our target.

Therefore we need an active filter where the amplification is constant between DC and 50,05 Hz, decreases by 6 dB/octave up to 500,5 Hz and stays constant again for all higher frequencies. This is easily achieved by adding the serial connection of resistor R_{24} and capacitor C_{36} parallel to resistor R_{26} in the feedback loop. At very low frequencies the impedance of C_{36} is very high so that the amplification depends on R_{26} alone. At very high frequencies the impedance of C_{36} is close to a short cut so that the amplification depends on $R_{26}||R_{24}$.

To be able to calculate the filter exactly we must find the transfer equation of the circuit. Having done this we can compare the coefficients of the amplification equation with those of the filter equation. Let's do this assuming an ideal operational amplifier (having infinite difference amplification and input impedance and zero output impedance)³.

The transfer equation of the active filter is:

$$A_s = 1 + \frac{R_{26}||R_{24} + C_{36}}{R_3}$$

Replacing the filter components by the complex impedances results in:

$$A_s = 1 + \frac{1 + j\omega C_{36}R_{24}}{1 + j\omega C_{36}(R_{26} + R_{24})} * \frac{R_{26}}{R_{22}}$$

We reduce the equation to the lowest common denominator

$$A_s = \frac{1 + j\omega C_{36}(R_{26} + R_{24})}{1 + j\omega C_{36}(R_{26} + R_{24})} * \frac{R_{22}}{R_{22}} + \frac{1 + j\omega C_{36}R_{24}}{1 + j\omega C_{36}(R_{26} + R_{24})} * \frac{R_{26}}{R_{22}}$$

and work it out to:

$$A_s = \frac{R_{26} + R_{22}}{R_{22}} * \frac{1 + j\omega C_{36} \frac{(R_{26} + R_{24})R_{22} + R_{26}R_{24}}{R_{26} + R_{22}}}{1 + j\omega C_{36}(R_{26} + R_{24})} \quad (2.2)$$

³See a similar derivation in [Skr88] on page 141f.

The transfer equation of both poles $318\mu s$ and $3180\mu s$ results of the combination of the corresponding equations of the single poles (see [Hü97, equations (3) and (4)]) and a constant amplification k :

$$A_f = k * \frac{1 + j\omega 318\mu s}{1 + j\omega 3180\mu s} \quad (2.3)$$

If we compare the coefficients of both equations (2.2) and (2.3) we find the linear equation system:

$$k = \frac{R_{26} + R_{22}}{R_{22}} = 1 + \frac{R_{26}}{R_{22}} = DC - amplification \quad (2.4)$$

$$318\mu s = C_{36} \frac{(R_{26} + R_{24})R_{22} + R_{26}R_{24}}{R_{26} + R_{22}} \quad (2.5)$$

$$3180\mu s = C_{36}(R_{26} + R_{24}) \quad (2.6)$$

If we assume C_{36} and k as known or required, we can solve the equation system in a way that results in the equations for the resistors R_{26} , R_{24} and R_{22} ⁴:

$$R_{26} = \frac{3180\mu s}{\frac{10C_{36}}{9}(1 - \frac{1}{k})} \quad (2.10)$$

$$R_{24} = \frac{R_{26}}{9}(1 - \frac{10}{k}) \quad (2.11)$$

$$R_{22} = \frac{R_{26}}{k - 1} \quad (2.12)$$

Again I chose a $10nF$ styroflex capacitor for C_{36} , and $k = 171$ for the DC amplification. The resistors turn out to be:

$$R_{26} = 287883\Omega$$

$$R_{24} = 30116\Omega$$

$$R_{22} = 1693\Omega$$

Regarding the quality of both capacitor and resistors please refer to 2.1.2 on page 11.

Subjective listening tests comparing the new preamplifier with my old one⁵ by connecting the output of U_1 to the line level amplifier showed that the sound of the new phono preamplifier was more clean, had much more resolution and was much better at the frequency extremes, but was also more cool, sterile and metallic and not that musical. After countless tests I found a solution which added the missing characteristics: a class A output stage within the feedback loop using a power Mosfet (credits go to Nelson Pass).

Q_1 is a source follower using the source resistance $R_{28} + R_{29}$. The quiescent current running through Q_1 is calculated according to the equation

$$I_r = \frac{-(-U_B)}{R_{28} + R_{29}}$$

⁴If you choose $k = 10$ the equations turn out as

$$R_{26} = \frac{3180\mu s}{C_{36}} \quad (2.7)$$

$$R_{24} = 0 \quad (2.8)$$

$$R_{22} = \frac{R_{26}}{9} \quad (2.9)$$

which means that we can omit one precision resistor.

⁵Audioplan Homogen (alias B&K) made in 1986

as being 22,7 mA. Q_1 dissipates a power of

$$P_{tot} = (+U_B)I_r$$

equal 340 mW and R_{28} and R_{29} each dissipate 170 mW. Therefore Q_1 needs a small heatsink, otherwise it gets very hot⁶.

The signal is coupled to both the line level amplifier and the tape output via a serial resistor of 50Ω each, because I use 50Ω coax cables (RG58 / RG214U) both internal and external. This serial resistor prevents reflections⁷.

2.1.4 The DC Offset Compensation

Both stages of the phono preamplifier together have a DC amplification of approximately 85dB. Even when using the best precision operational amplifiers money can buy today it is not possible to get rid of the DC offset without reducing the DC amplification—which makes sense anyway in a phono preamplifier. There are several possibilities how to realize this:

1. The most simple way is to use an output capacitor. The disadvantage is that the output capacitor and the input impedance of the following stage (don't forget the tape recorder, if there is one as there is no buffer amplifier) form a highpass. To prevent that the corner frequency of this highpass gets too high you must use a big capacitor which usually leads to bad sonic performance.
2. It is possible to use capacitive coupling between the first and the second stage. In this case it is necessary to add a resistor parallel to C_{35} to ground the input offset current of U_1 . This resistor must be much bigger than R_{20} if it should not influence the filter characteristic, and this leads to an increased DC output voltage of U_1 .
3. You can reduce the DC amplification of the second stage to 0 dB by adding a capacitor in series to R_{22} . Then the DC output voltage originates of the first stage only.

Approaching a corner frequency below 20 Hz requires a capacitance of 4,7 μF. Subjective listening tests using a capacitance of 6,8 μF (leading to a corner frequency of 13 Hz) gave the impression of reduced bass dynamic (punch) compared to a version with full DC amplification. For a corner frequency of 1 Hz a capacitance of 82 μF is required, and a good one is very big in size and very expensive too. You cannot use a cheap electrolytic capacitor as there is no DC voltage applied which would lead to significant distortion.

4. It is possible to compensate the DC offset of the first stage so that just the DC offset of the second stage appears at the output of the phono preamplifier. The solution is to connect the DC voltage of the first stage to the inverting input of the operational amplifier of the second stage. In practice you connect the output signal of the first stage to a lowpass filter, amplify the filtered DC by a precision operational amplifier and connect the resulting voltage via a large resistor to the feedback resistor of the operational amplifier of the second stage. If the damping of the resulting voltage divider has the same value as the amplification of the precision operational amplifier the offset compensation is perfectly adjusted.

I chose solution 4. R_{14} and C_{33} form a lowpass with a corner frequency of approximately 1 Hz. The amplification of the precision operational amplifier U_5 is calculated according to the equation

$$A = 1 + \frac{RVAR_1 + R_{18}}{R_{16}}$$

⁶The voltage regulators of the phono preamplifier stage also need small heat sinks.

⁷It is very questionable whether reflections play any role in audio applications at all. On the other hand it does not hurt to prevent them.

and turns out to be 119..128 (41,5 .. 42,1 dB), depending on the adjustable resistor $RVAR_1$ ($1k\Omega$). R_{36} and R_{22} form a voltage divider the damping of which is calculated according to the equation

$$A = \frac{R_{22} || R_{26}}{(R_{22} || R_{26}) + R_{36}}$$

and turns out to be approximately 122,6 (41,8 dB). You can see that amplification and damping are almost identical, and the trim pot $RVAR_1$ is used to adjust the values to absolute identity.

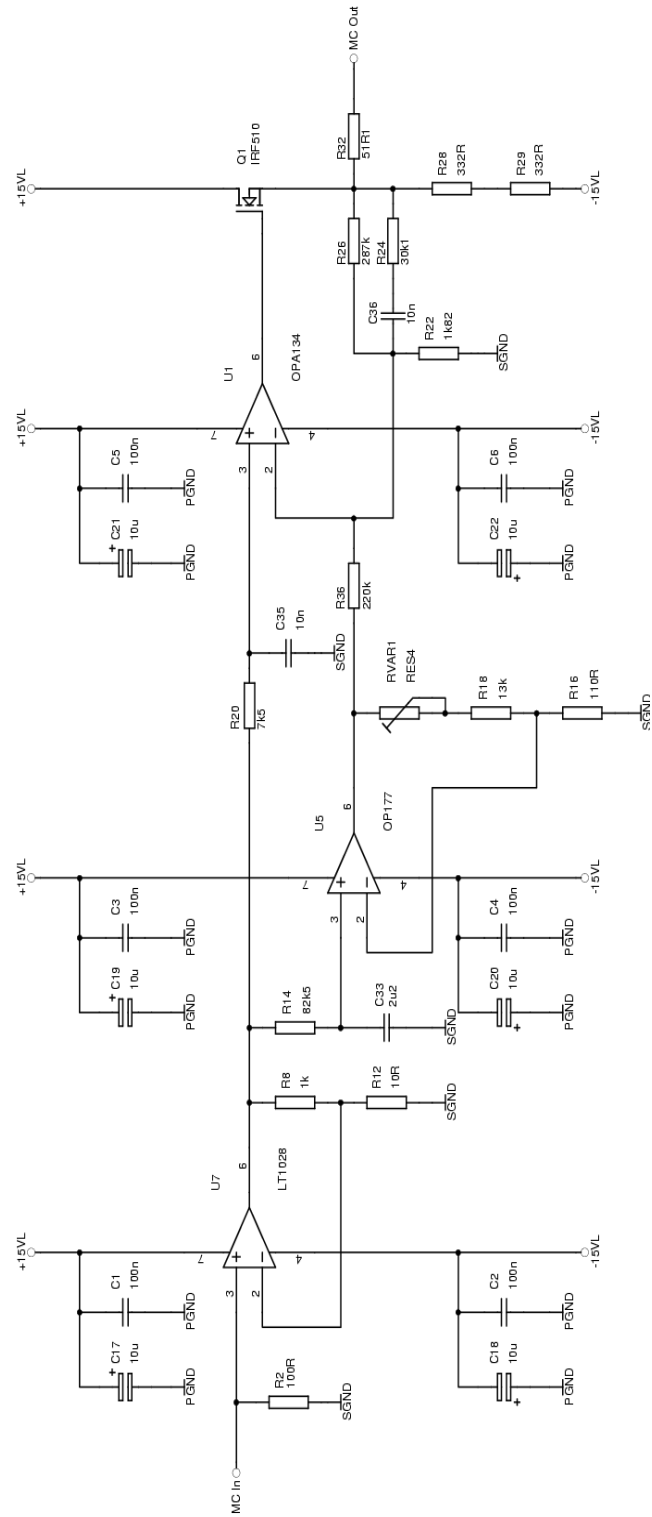


Figure 2.1: Schematics of the Phono Preamplifier Stage

Chapter 3

The Line Level Amplifier

The listening tests of the single ended class A output stage of the phono preamplifier were very promising, so I did not test different circuit variations of the line level amplifier. I just copied the circuit of the second stage of the phono preamplifier and did the following changes (see the schematics in figure 3.1 on page 17):

1. Instead of the active filter we set a linear amplification¹of approximately 10 dB using resistors R_{45} and R_{11} . The operational amplifier must be compensated for an amplification of 1 (0 dB), as the resulting input capacitance of the Mosfet and the open loop output impedance of the operational amplifier form a lowpass filter which turns the phase and decreases the loop stability. Counting on earlier experience I chose OPA134—its sound is very good and it has high impedance FET inputs with very low input quiescent current, which allows us to use a high impedance volume pot.
2. The quiescent current running through the Mosfet is increased to 45,5 mA to be able to drive the head phone. The dissipated power of Q_4 and R_{43} is 680 mW each.
3. The heat sinks for the Mosfet and the voltage regulators are bigger due to increased power dissipation.
4. The headphone is connected directly, while the output to the power amplifier is connected via a 50 Ω resistor.
5. A high quality volume pot is connected to the input of the line level stage.

¹The amplification is calculated according to the equation

$$V = 1 + \frac{R_{45}}{R_{11}}$$

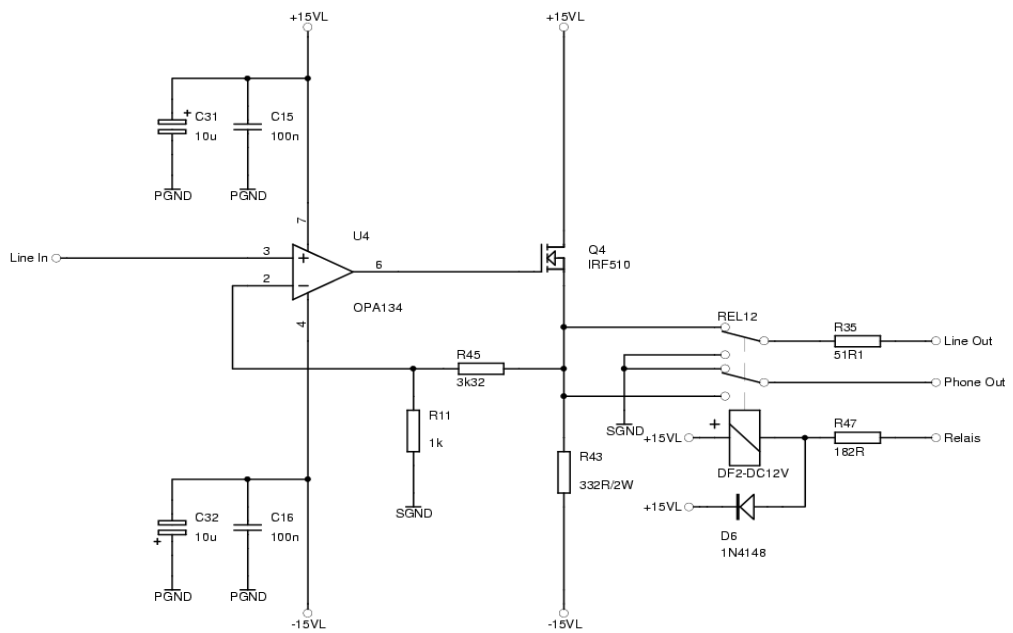


Figure 3.1: Schematic of the Line Level Amplifier Stage

Chapter 4

The Power Supply

Placing a humming and LF emitting lines transformer close to a phono preamplifier is definitely not a good idea. Therefore the power supply is located in its own housing which should be placed some distance away from the preamplifier housing. However, the power supply regulation should be placed as close as possible to its sink. Long distances¹ should be strictly avoided.

I decided to put the lines transformer and the big reservoir capacitors into an external housing. The power supply regulation is then separate both for line level stage and phono stage and left and right channel as well. This means that we must build the $\pm 15V$ supply voltage regulation four times.

4.1 The External Power Supply

A 120 VA toroidal transformer feeds two electrolytical capacitors of 15 mF each via a 25 A rectifier bridge. Each diode within the rectifier bridge is bridged by a 100 nF capacitor to get rid of fast transient spikes. For the same reason both 15 mF capacitors are bridged with 6.8 μ F foil capacitors.

4.2 The Local Supply Voltage Regulation

A high quality voltage regulation must be built using high quality parts. It is however no longer necessary to use discrete semiconductors as there are now very good integrated voltage regulator circuits available. I chose the adjustable regulators LT1085 (positive) and LT1033 (negative) which are compatible to LM317/LM337. The standard fixed voltage regulators of the 78xx and 79xx series produce much higher noise and have worse ripple suppression, although you can find those even within highend preamplifiers (Bryston 25 BP).

R_4 and R_{38} determine the positive output voltage, while R_5 and R_{39} determine the negative voltage. C_{39} (and C_{40}) works as an AC short regarding the adjustment input of the voltage regulator which improves the ripple suppression by about 20 dB. D_{13} (or D_{14}) leads the discharging current of C_{39} (or C_{40}) in case of an output shortcut away from the adjustment input, thereby preventing the destruction of the voltage regulator.

¹The opposite philosophy is held by Naim: their power supply regulation is either a simple one located in the power amplifier or an improved one within an external power supply (Flatcap, Hicap, Supercap). This allows to get higher sound quality when connecting the preamplifier to an external power supply. If the power supply regulation were located within the preamplifier then chances were high that a much better sound quality could be reached with the cheap power supply in the Naim power amplifier-but Naim would not sell that many external power supplies as well.

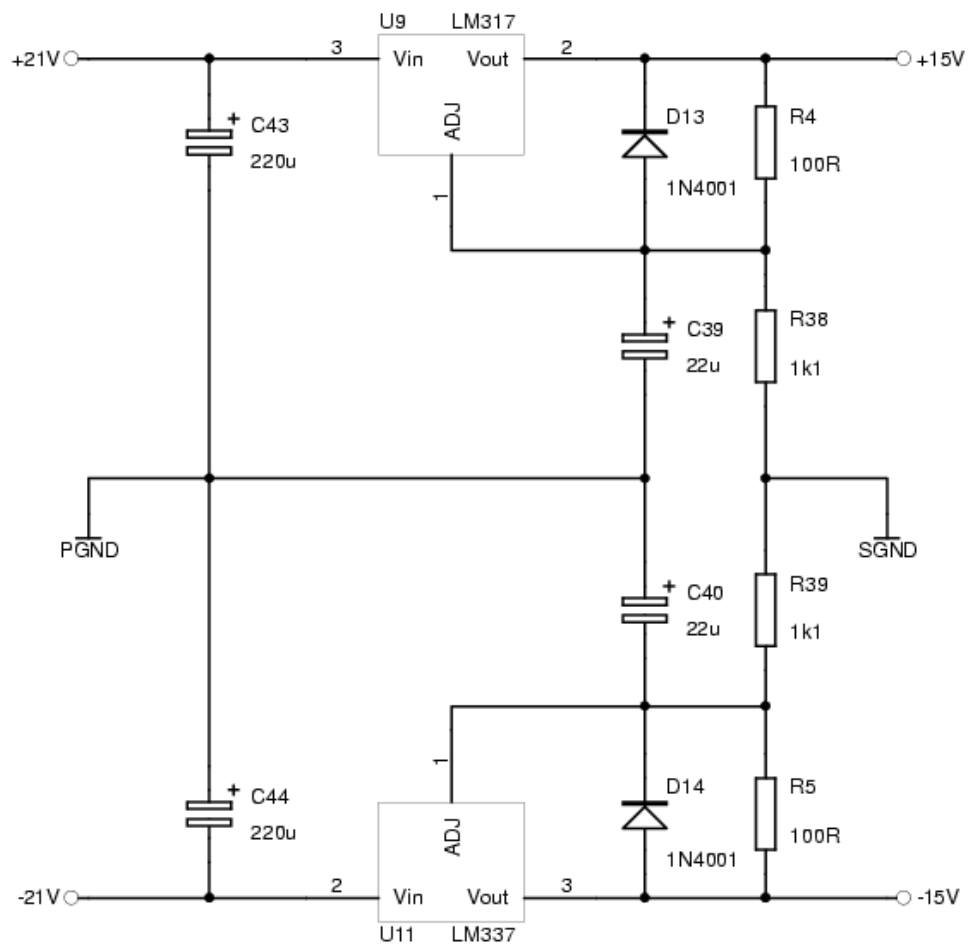


Figure 4.1: Schematic of the Voltage Regulation

Chapter 5

Mechanical Assembly

5.1 The Housing

The chosen housing is a 19" standard housing with one height unit (44 mm / 1.75") und 280 mm (11") depth, which I bought at Meyer-Elektronik (see appendix B.2 on page 26). The whole housing consists of sheet steel except the 3mm thick aluminium frontplate.

The housing is equipped with gold plated RCA jacks (good quality, DM 5.- each) which are electrically isolated. The jacks of one of the line level inputs were mounted on the front plate. This allows an easy connection to external sources without fumbling around at the backside of the equipment.

5.2 The Printed Circuit Boards

As I had no possibility to produce real pcb's I used doublesided experimental pcb's with solder points in european format (100 mm x 160 mm). The solder points on both sides have no electrical connection. Both channels of the line level stage including the voltage regulations were easily mounted on one such pcb. It was also possible to mount both channels of the phono preamplifier including the voltage regulations on one such pcb, but there was no space left. The electrical connections were done by soldering thin bare wire across the solder points. For the ground connections I used 1 mm silver coated wire.

All ground connections meet at the central point on a separate pcb. I used separate connections for signal ground (SGND) and power supply ground (PGND) to prevent crosstalk of spikes on the power supply ground into the signal ground. This is very important for the phono preamplifier.

Chapter 6

The Results

6.1 Sound Quality

Regarding the sound quality the following statements can be given:

- Subjective listening tests showed that the phono preamplifier is much better than the one in my old preamplifier regarding resolution, signal to noise ratio, dynamic, and bass and treble reproduction.
- Subjective listening tests showed that the line level preamplifier is also much better than the one in my old preamplifier regarding resolution, dynamic, and bass and treble reproduction. However, the sound quality limits of the source is easily shown, especially with digital sources (grainy)¹.

Subjective listening tests in comparison with a Linn Wakonda (commercial preamplifier, price DM 2000.-) at my local dealer showed the limits of the Linn, especially regarding dynamic, transparency, sound stage and at the frequency extremes. Both the dealer and I placed my high level preamplifier at a quality level somewhere between a Naim 72 and a Naim 82 (each with Hicap)—that is in a price range of DM 4000.- to 8000.- and all this at a financial expense of about DM 750.-.

6.2 Expenditure

I have to admit that I heavily underestimated the amount of time needed to design and build the preamplifier. First, because many minor circuit modifications were needed followed by extensive listening tests in comparison with my old preamplifier to get exactly the sound I wanted to have, and second, because the preparation and assembly of the housing and the pcb's (see 5.2 on page 20) took very much time to finish—I'm not very good in mechanical work but I wanted to build something I could lay my eyes on every day.

¹It is very interesting to note that the graininess of digital sources disappears for the bigger part if the preamplifier drives a better power amplifier!

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Further Reading

All those who want to modify my circuits (or want to design their own one) should have a look into the following books:

- [PH89] contains very good explanations without much formula, and many practical hints.
- [UT93] contains all those formula not mentioned in [PH89] and is therefore more concise regarding the theory of electronics, but you need a better understanding of basic electronics and mathematics.
- [Skr88] is a mecca of audio circuits, but expects some previous knowledge as well.

Appendix A

Recommendations for Modifications

A.1 Improvements

There are some recommendations how to improve the sound quality of the preamplifier, however without any warranty for success as I did not perform any tests:

- You may improve the dust, scratch and EMI sensitivity of the phono preamplifier by adding a lowpass of about 50 kHz to the first amplifier stage. This is done by bridging R_8 with a capacitor.
- The input noise of U_5 is fully added to the audio signal. As U_5 amplifies only DC it is possible to reduce the amplification within the audio band to 0 dB. This is done by bridging R_{VAR1} and R_{18} with a capacitor. Its capacitance is calculated according to the equation

$$C = \frac{1}{2\pi R_{16} f_u}$$

to a value of about $C = 68\mu F$ at $f_u = 21Hz^1$.

- Improvements of the power supply:
 - Additional filtering or regulation of the supply voltage of the first stage of the phono preamplifier.
 - Separate lines transformers and filtering for each channel.

However, as the PSRR² values of the operational amplifiers used are very good we cannot expect much improvement; this is much more important for discrete circuits.

A.2 Simplifications

You can also simplify the circuits to get a cheaper preamplifier but on the expense of sound quality loss:

¹This is not the corner frequency of the low pass but the frequency where the amplification reaches +3 dB.

²Power Supply Rejection Ratio

- Elimination of the offset compensation of the phono preamplifier:
Just chose an alternative solution to the DC offset problem as mentioned in 2.1.4 (see page 13). I would chose solution 3.
- Elimination of the class A output stages of phono- and/or line level preamplifier:
Just elimnate the FET and its source resistor (leading to the negative supply voltage). This will result in a sterile, metallic and somewhat boring sound; and in case of the line level stage you can no longer drive a headphone. It is advisory to use an operational amplifier with low output resistance and high short circuit current (e.g. OPA604), and to increase the output resistor from 50 Ω to 100..200 Ω .
- Use a cheaper potentiometer.
- Reductions in the power supply:
 - Common supply regulation for both phono preamplifier and line level stage.
 - Common supply regulation for both channels.
 - Use cheaper voltage regulators (7815/7915).
 - Use a smaller and cheaper lines transformer.
 - Use smaller and cheaper reservoir capacitors.

A.3 Other Modifications

You may also do the following modifications which may lead to a different sound—let’s call this the individual fine adjustment:

- Chose a different quiescent current of the class A output stage of phono and line level preamplifier by changing the resistance of the source resistors. Increasing quiescent current reduces the harmonic distortion, although this distortion is quite small anyway due to negative feedback.
- Replace the FETs by small bipolar power transistors (e.g. BD135).
- Replace the class A single ended output stage by a class A push pull output stage.
- Replace the class A single ended output stage by a class AB push pull output stage. Instead of a discrete circuit you may also use an integrated buffer (e.g. LT1010, BUF04, LH0033 o.ä.). This is especially interesting for the line level stage if you must drive a low impedance headphone but do not want the necessarily high quiescent current of a class A output stage.
- Using other operational amplifier:
 - Regarding line level stage and the second stage of the phono preamplifier any operational amplifier may be used, in theory. However, for the line level stage you should use an operational amplifier with FET inputs due to its very low input offset current. In the phono preamplifier I successfully used a OP37 in the second stage. In the line level stage you must use an operational amplifier which is unity gain stable or may be compensated for unity gain. An expensive alternative is the OPA627, cheaper is the OPA604.
 - Regarding the first stage of the phono preamplifier I could only think of the AD797 which has a similarly low input offset noise voltage. However, the input offset current of the AD797 is much higher which means that the MC pickup must be able to stand the higher input current. I succesfully used an OP37 and an AD846 (a current feedback amplifier!) but with increased noise, although the noise can be heard only if the pickup is lifted off as the noise of the records is yet higher than that of the preamplifier. An expensive alternative is the OPA637, but it still is more noisy.

Appendix B

Parts

B.1 List of Special Parts

Part	Value	Type	Dimens./mm	RM	Manufacturer	Distributer
$C_{35/36}$	10nF/63V	Styroflex	Ø5x15	20	Philips	RS
C	100nF/100V	MKS2	7,2x4,5x9,5	5	Wima	RS
C_{33}	2,2µF/63V	MKS4	8x15x18	15	Wima	RS
C	10µF/25V	Tantal E2	8x5x13	5	Philips	Mütron
$C_{39/40}$	22µF/40V	Elcap P2 EK	Ø9x13	5	ROE	
$C_{44/45}$	220µF/40V	Elcap B2 EK	Ø13x21	5	ROE	
D	B80C25000	Bridge	29x29x8			Meyer
U_{10}	LM317	V-Regulator	TO220		Nat. Semiconductor	überall
U_{12}	LM337	V-Regulator	TO220		Nat. Semiconductor	überall
U_{12}	LT1033	V-Regulator	TO220		Linear Technology	
U_7	LT1028A	OP-Amp	DIL8		Linear Technology	
U_{10}	LT1085	V-Regulator	TO220		Linear Technology	
$U_{1/3}$	OPA134PA	OP-Amp	DIL8		Burr-Brown	Rutronik Fenner
U_5	OP177GP	OP-Amp	DIL8		PMI	Reichelt
P	50kΩ log.	RK40312	40x33,4x34	10/7,5	Alps	Conrad
Rv	500Ω	3386P	10x10x5	2,5	Bourns	RS
Rv	1kΩ 25-Gang	3296W	10x10x5	2,5	Bourns	RS
$Q_{1/3}$	IRF510	Mosfet	TO220		Int. Rectifiers	Reichelt
Tr	2x15V/120VA	Toroidal	Ø110x37			Meyer

Table B.1: Special Parts

B.2 Hints for the Acquisition of Parts

Distributor	Address	Phone/Fax/Internet	
Conrad-Elektronik	Klaus-Conrad-Str. 1 92240 Hirschau	Phone:	0180/531-2111
		Fax:	0180/531-2110
		WWW:	www.conrad.de
Meyer-Elektronik	Postfach 110158 76487 Baden-Baden	Phone:	07223/52055
		Fax:	07223/52777
Mükra-Electronic	Postfach 847 73008 Göppingen	Phone:	07161/70031-33
		Fax:	07161/71929
Mütron	Postfach 103067 28030 Bremen	Phone:	0421/3056-0
		Fax:	0421/3056-146
Neumüller-Fenner	Raiffeisenallee 10 82041 Oberhaching	Phone:	089/613795-0
		Fax:	089/613795-80
		WWW:	http://www.fenner.ch
Reichelt-Elektronik	Elektronik-Ring 1 26452 Sande	Phone:	04422/955-0
		Fax:	04422/955-111
		email:	Reichelt@Post.de
		WWW:	www.Reichelt.de
RS-Elektronik	Hessenring 13b 64546 Mörfelden-Walldorf	Phone:	06105/401-234
		Fax:	06105/401-100
		email:	rs-gmbh@rs-components.com

Table B.2: List of Distributors

Appendix C

About this Document

This document has been made on a Linux computer (SuSE 6.1):

- It is created using *LYX-1.0.1* and exported as *TEX*. LYX is a graphical frontend to L^AT_EX. L^AT_EX is a professional book setting program. Compared to ordinary text processing programs L^AT_EX has no bugs—the last bug has been found and fixed some years ago (and the fix has been paid). The schematics have been imported using the *epsf*-package (`\usepackage{epsf}`).
- *TEX* is processed by *L^AT_EX-2e* into *dvi*, which is converted into *Postscript* by *dvips-5.82*, which is again converted into *PDF* using *ps2pdf* (Aladdin Ghostscript 5.50). *dvips* is the Postscript printer driver of L^AT_EX. Ghostscript is a printer driver for printers which cannot process plain Postscript. PDF (Portable Document Format) is a proprietary file format of Adobe (TM).
- *TEX* is converted into *HTML* using *latex2html-98.1p1*. All none automatic created links have been created using the *html*-packages (`\usepackage{html}`). *latex2html* is a *Perl* script which reads a T_EX file and creates HTML pages. Perl is a powerful system independent interpreter language and replaces the system dependent shell script languages.
- The schematics are converted from *Postscript* into *PDF* using *ps2pdf* (see above).
- The whole conversion and the installation onto my homepage is automatically done by *GNU make-3.76.1* using a *makefile*. *make* is a tool for the automatic build of projects of any kind which consist of several dependent files and which are build using command line tools.
- All source files are controlled by *CVS-1.10.5*. CVS (Concurrent Versions System) is a source code management program.
- All used programs are free software which fall either under the GPL (GNU Public License) or a similar license for none commercial purposes. The sources are open and may be compiled on most Unix systems.