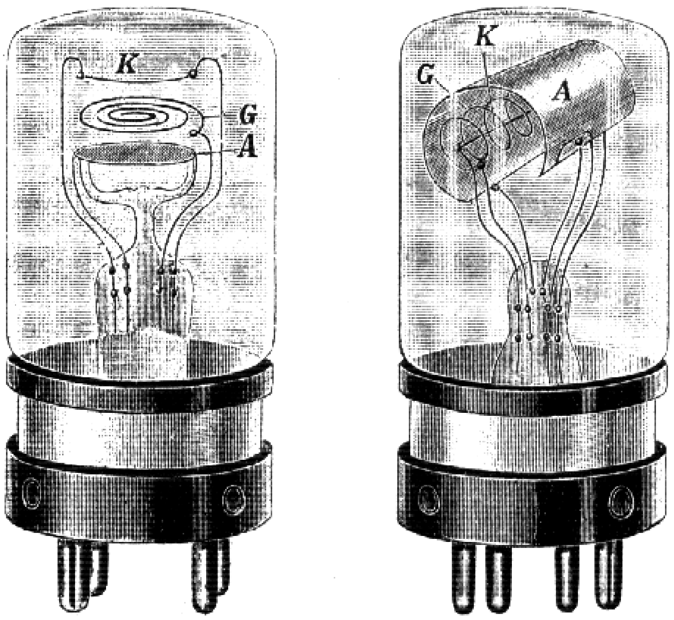
An Easily Constructed

High-Quality Single-Ended 8W Amplifier

Using Standard Valves and the

Lundahl LL 1664 Output Transformer



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Preface

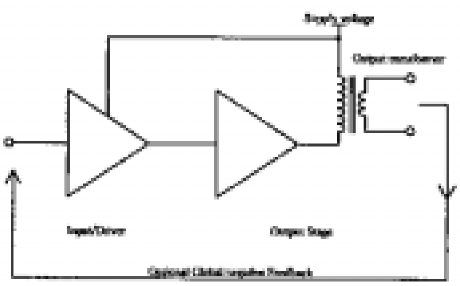
After the publication a year ago of my paper Power Amplifiers with Valves I have received a lot of comments and very much to my surprise I learned that a great deal of interest in single-ended (SE) amplifiers exists, and I have been asked for a similar paper concerning such amplifiers.

I admit that I have been very reluctant. I still remember when I was 17 and my financial situation finally permitted me to buy a decent transformer for 2 EL84^{s} in push-pull (PP) capable of handling 10 Watts, and I remember the joy that came from listening to my first PP amplifier. I never thought that I was ever again to show any interest in SE amplifiers.

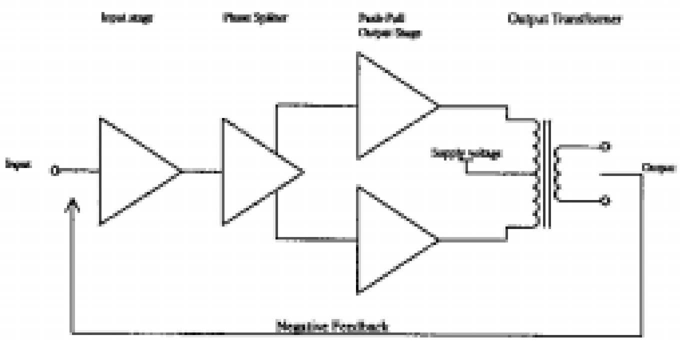
The output transformer is by far the most critical component in any valve amplifier. The problems to be solved by the manufacturer of output transformers are even more complex for an SE-transformer than for a PP-transformer. I shall return to this matter later, but I can without exaggerating say that the output transformer found in the SE-amplifiers of almost every radio receiver, tape recorder or record player in the late fifties and in the sixties was very poor as was the design of the amplifiers. The advent of stereo in these years did not improve this situation. On the contrary it worsened because the public would only accept a small rise in the investment costs when upgrading to stereo. Both amplifiers and speaker systems became simpler. Warning voices claimed that each of the two channels in stereo should be *at least* as good as your mono channel if full advantage of the new technique was to be taken. For many years these warnings were neglected, and the equipment I remember from the pioneering years of stereo made me loathe just the naked principle of the SE-amplifier.

Today I am perhaps more open-minded, and when Mr. Per Lundahl from Lundahl Transformers told me about his SE-transformers I became curious and decided to investigate the matter and in your hands you are holding the results of my efforts.

2. The Principle



SE Amplifier



PP Amplifier

As can be seen, the design of the SE-amplifier is simple. A combined input stage and driver drives the output valve, and a transformer matches the valve and the speaker system. A PP-amplifier must have a phase-splitter to provide signals of equal magnitude but opposite phase for the two output valves. The phase-splitter is a very critical part of the PP-amplifier, and its behaviour is of great importance for the performance of the amplifier. The design of this stage takes skill and care as explained in my earlier paper.

The SE supporters claim that simplicity and a very short signal path is the key to the survival of and even renewed interest in their favourite type of amplifier. The short signal path is however not a virtue *per se.* If you want to move from point A to point B the shortest path is of course a straight line, but if following this line makes you traverse a dunghill you will surely arrive at point B in a poor condition whereas a small detour around the obstacle would enable you to reach point B in a perfectly clean condition. Exactly the same applies to the signal travelling from input to output of an amplifier. If the short path should be the best, we have to take great care when paving it with the smoothest stones available and place them very carefully so that the surface of the path becomes impeccably flat.

3. The Transformer

As stated in the preface, the transformer is very important, and we shall in this chapter look into the properties and problems of the output transformer.

Basically a transformer consists of a core of a material that can be magnetized wound with two coils. The ratio between the numbers of turns in the two coils determines the transformation ratio of the transformer. The energy is transferred to an alternating magnetic field and back to electric power again.

If an output valve can deliver 10 Watts into a load of 3000 Ω, and the speaker system is 8 Ω, we use transformer to match the load from the formula

$$P = \frac{e^2}{r}$$

where P is the power in Watts

e is the voltage across the load

r is the resistance of the load

We can determine the signal voltage that is found on the primary of the transformer

$$10 = \frac{e^2}{3000} \quad \text{or} \quad e^2 = 30000 \quad \text{or} \quad e = \sqrt{30000} = 173.2 \text{ V}$$

We want the 10 Watts delivered in 8 Ω so the formula for the secondary is

$$10 = \frac{e^2}{8} \quad \text{or} \quad e^2 = 80 \quad \text{or} \quad e = \sqrt{80} = 8.95 \text{ V}$$

The turns ratio of the transformer must be $\frac{173.2\ V}{8.95\ V} = 19.35 : 1$

Which is the same as the square root of the primary impedance divided by the secondary impedance, $\sqrt{\frac{3000}{8}} = \sqrt{375} \approx 19.35$

In the extreme case where the secondary is unloaded or an open circuit, the primary should act as no load for the AC signal too, the impedance should be infinite. This will be the case if the induction of the primary coil is infinite. This will of course never be completely possible. The impedance of the coil is also frequency dependent, meaning that if induction is not infinite the impedance will decrease with decreasing frequency. The lower the frequency that a transformer must be capable of handling the higher the induction must be if losses and distortion are to be kept within reasonable limits.

Let us suppose another extreme: Short circuit of the secondary. If the transformer is ideal the primary should act as a short circuit too, meaning that DC resistance and AC impedance should be zero too. This will clearly never be the case. The DC resistance of the both primary and secondary remains, and because the turns of the primary and the secondary are not infinitely close coupled, a little inductance and therefore a little AC impedance of the primary will also remain. The remaining inductance is called the primary *leakage inductance,* and this inductance is harmful because it acts as an inductance in series with the load, in this case the speaker, compromising HF response.

Even a small leakage induction can be of serious consequences. 8mH leakage is a fine value for a single end output transformer 3KΩ/8Ω and it does not sound of much. But the impedance is

$Z = 2\pi \cdot f \cdot L_e$ where f is the frequency in Hz

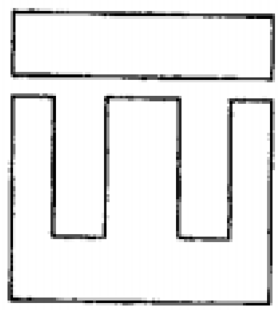
L_e is the leakage induction in Hy

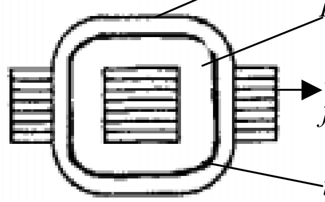
at e.g. 50KHz: $Z = 2 \cdot 3.14 \cdot 50.000 \cdot 0.008 = 1570\Omega$

which of course is not unimportant compared to the nominal 3000Ω primary impedance!

Another factor is the capacitance across the primary. This acts as an unuseful load in parallel with the primary, and since this load is increasing with frequency this capacitance is also deteriorating HF performance. It shows how important a low output resistance from the valve is because the lower the output resistance the less the additional frequency dependent load means.

The transformers of the fifties were wound on EI cores as shown:





The core is laminated to avoid eddy currents

The secondary was normally wound outside the primary, so coupling between the innermost turns of the primary and the secondary was not close, resulting in considerably leakage induction.

In the PP transformer the supply voltage for the two valves is applied to the centre of the primary. The anode currents are floating in opposite directions in the two halves of the primary. If the currents are equal their resulting static magnetization of the core will be zero. In the SE transformer the anode current of the single valve generates a static magnetic field that magnetizes the core permanently and causes serious problems. If the core saturates, the primary inductance drops almost to zero, and severe distortion occurs. If measures are not taken, saturation takes place when even a very moderate DC current flows. The core must cope not only with the static magnetic field but also at the same time be able to handle the alternating field generated by the signal current.

There are four ways to deal with this problem.

1. The number of turns in both primary and secondary can be lowered. The magnetic field is proportional to the number of turns multiplied by the current, so lowering the number of turns reduces the magnetic field.

   Unfortunately the inductance is lowered too, causing bass problems in the amplifier.
2. The core can be made larger. There will be more iron to magnetize, so the resulting magnetization for a given current will be lower.

Unfortunately this increases the diameter of each turn, which again increases DC resistance and capacitance between turns and between layers of turns, causing problems at both ends of the audio band. The price will of course go up too.

3. An air gap can be made in the core. Since the magnetic field travels thousands of times easier in the core than in the air, an air gap is a very efficient obstacle for the magnetic field.

   Unfortunately an air gap reduces primary inductance and decreases coupling between primary and secondary, so also the air gap is a hazard to the performance in both ends of the audio band.

   Note that the reduction of the inductance is very substantial, often by a factor 5 to 10, so when an air gap for say 80mA is introduced primary induction may very well go down by 75%.

4. The transformer could be equipped with an additional winding for a DC current generating a magnetic field of the same magnitude as the field generated by the primary but in the opposite direction. This field would cancel the magnetization caused by the quiescent current in the output valve.

   This is in theory an elegant solution, but unfortunately the extra winding requires space without being involved in the transformation process, and some regulating circuit to control that the counter-current is correct will be needed. The whole idea of simplicity will then be gone. I have to say that this solution has often crossed my mind. I have, however, never seen or heard about a transformer with a winding designed for this particular purpose.

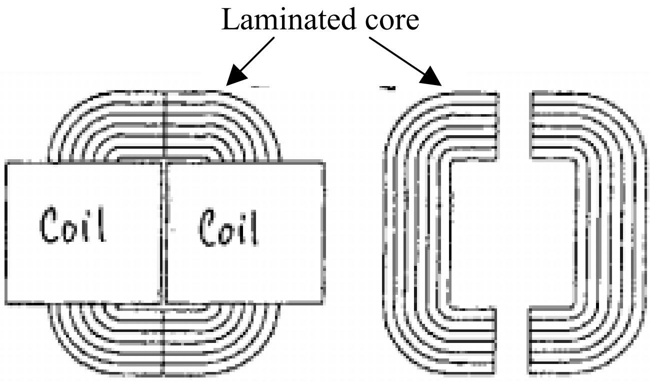
All factors are conflicting here and the manufacturer is put in a position like Ulysses on his way home from Troy: When he wants to pass Scylla in safe distance he gets so close to Charybdis that he is in great danger to be swallowed up. All factors are conflicting and only a very careful design and a superb core material can produce an acceptable transformer.

I was triggered by this information from Mr. Per Lundahl: He told me that he makes transformers with air gaps customised for the current you want to pass through the output valve. The air gaps are dimensioned so that the quiescent current generates a magnetic flux of 0.9 Tesla in the core. This is in the middle of the linear part of the magnetization curve for the core material, meaning that a variation of +/÷ 0.6 Tesla can be handled fairly linearily.

This also tells us that choosing a transformer for say 100mA when only 80 mA is needed “just to be on the safe side” is wrong. First you lose primary inductance and second, the transformer is not operating from the mid-point of the linear part of the curve.

The laminated cores are cut and the ends are precision machined and polished so that the air gaps are completely uniform over their total area. The benefit is a primary inductance which is very little dependent on signal current. This is clearly and positively reflected in performance in the LF end of the spectrum. In the SE transformers of my youth the air gap was formed by a piece of cardboard between the “E” and the “I”, and my impression is that the philosophy of the design was: rather too thick than too thin because the distortion caused by a saturating transformer is very nasty. The often generously dimensioned air gap meant a leaky and lossy transformer with low primary inductance and consequently poor performance at both ends of the audio spectrum.

In the light of this the initiative of Lundahl is very interesting, and I ordered and LL1664 3kΩ/8Ω SE output transformer with an air gap for 80mA. The transformer is rated for 8W at 30 Hz. It has the same size and shape as the 35W PP transformer I used in my PP amplifier last year. Lundahl uses method 2 and 3 in order to cope with the static field but not option 1, which I find is a praise-worthy method. Furthermore the Lundahl Transformers are wound on C-cores as shown:



Each coil has four sections
1. *Part of secondary*
2. *Part of primary*
3. *Part of secondary*
4. *Part of primary*

The two coils are identical so the transformer is in total divided into 8 sections giving a very close coupling between primary and secondary

This means that the grain orientation in the core can be perfect everywhere in the core and that a greater part of the core can be covered with coils. The secondary is wound in sections between layers of primary, and the turns are very neatly placed so that the space available is used entirely. No air due to a messy winding technique is found. These precautions reduce leakage induction considerably, and optimising the air gap for the actual current means that the decrease of primary inductance is at least not bigger than absolutely necessary. The actual primary inductance is 22H and the leakage is only 8 mH, an excellent achievement.

This short introduction will not make you an expert on transformers and there are of course many other aspects of great importance like e.g. phase shift. I just wanted to familiarize you with a few of the main obstacles the manufacturer must deal with when he wants to make a good output transformer.

4. Design Goals

Before embarking on a project like this it is wise to state exactly what to achieve and by which means.

I want to build an SE amplifier capable of producing 8W, because this is what my transformer is intended for.

I shall try to use widely available standard valves in order to keep costs reasonable, but I am prepared to make a series of investigations to make it possible for me to choose the most suitable solution to the very important question: the configuration of the output stage. Valve tables are not very helpful here. Their application suggestions are always limited to the standard pentode coupling with emphasis on maximum power.

I shall try to make the circuit simple and understandable, and I want to make it possible to use the amount of negative feedback that suits the user best, ranging from 0 to 10dB, and I shall limit myself to two valves.

Since no mains ripple cancellation takes place in the transformer as it does in PP amplifiers, I am prepared to use a filter choke in the power supply.

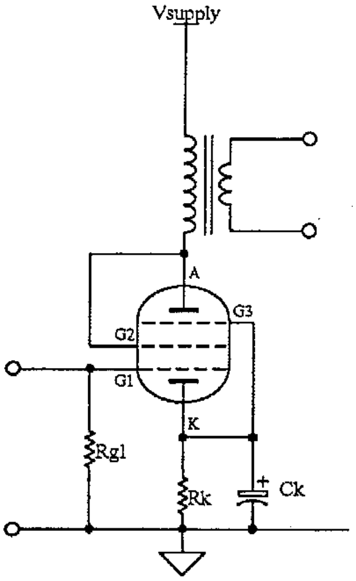
These considerations lead me to choose EL34 as my output valve. I want it to behave as closely as possible to a triode without sacrificing more than 25% of the power that the valve can provide in a normal coupling.

I tried the five circuits shown on the next page with two different supply voltages, 275 and 375 Volts. The circuits with a tap on the primary for the screen grid were tested with taps at different points.

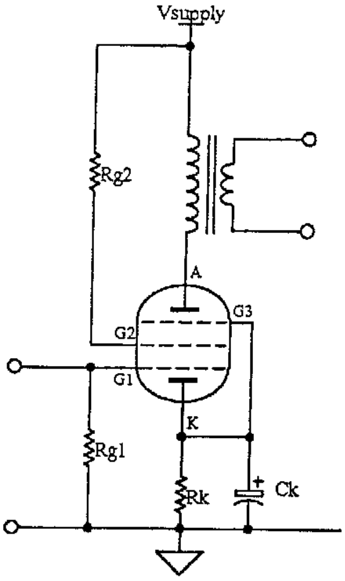
The first thing that I found was that maximum power provided by the triode configuration was so low that this circuit was left out of any further consideration for me.

The next thing was that with 275V supply, tapping at a lower point than 16% would also limit power unacceptably, whereas even tapping at 50% did not affect power significantly when supply voltage was 375.

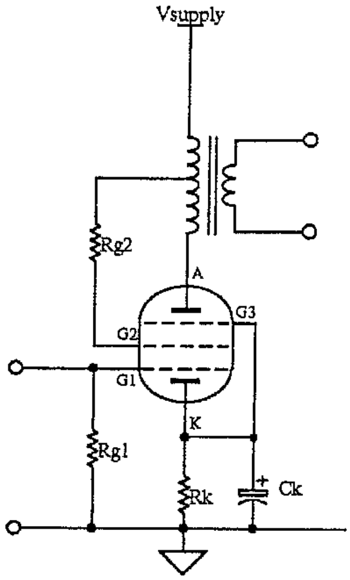
I measured input voltage requirements, total harmonic distortion at 1000Hz and output resistance for the circuits 2 to 5, and the results are given in two tables on the page after the circuits.



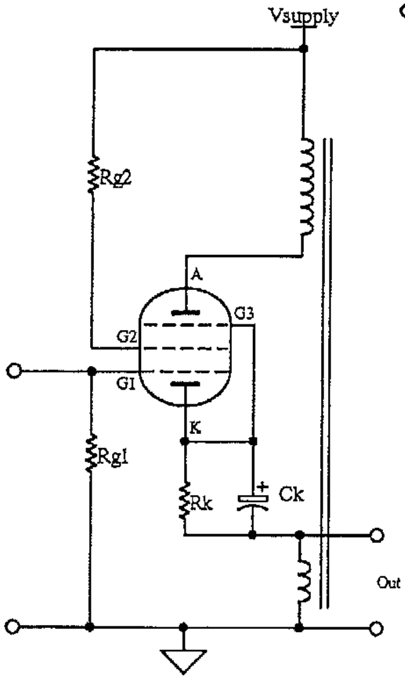
Circuit no. 1
Pentode coupled as
Triode



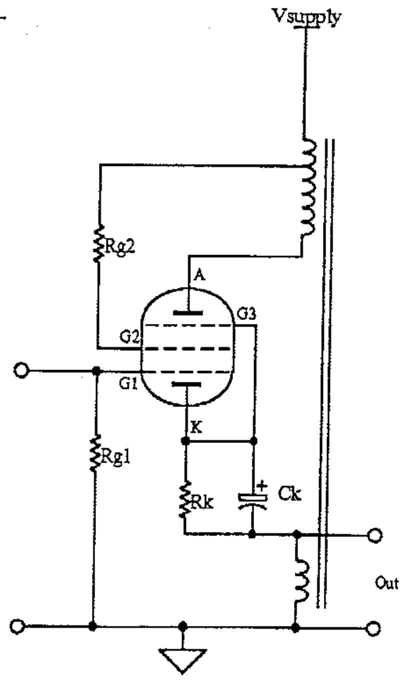
Circuit no. 2
Normal Pentode



Circuit no. 3
Pentode with screen-grid connected to
tap
at 16% - 50% og the primary.



Circuit no. 4
Pentode with current
feedback
through transformer
secondary



Circuit no. 5
As No. 4, but with the screengrid
connected to tap on primary

A very interesting point is the output resistance[1], not only because a low output resistance provides electromagnetic damping of resonances in the speaker but also because speakers are designed to have the flattest frequency response when fed with a frequency independent voltage. Since impedance of speakers varies considerably with frequency, an amplifier with a high output resistance will produce higher drive voltages at points where the impedance curve peaks, and this is where the speaker has resonances. A high output resistance tends to aggravate resonances and should consequently be avoided. I prefer an output resistance less than 1/10 of the nominal speaker impedance giving a damping factor of 10, but I learned that for many people the damping factor seems to be less important, and what is called "a more rounded, warmer bass" is achieved. It seems that a damping factor of 2 to 5 is enough. Even though I do not *a priori* agree, I am prepared to give my experiments the benefit of the doubt.

As expected the output resistance drops when the screen grid is connected to a tap on the primary. We know that from PP amplifiers where this way of connecting the screen grids is referred to as "ultra linear" or "distributed load".

It was also expected that output resistance would go down when the secondary of the transformer was used as a cathode feedback coil too. Normally when this method is used, a separate coil with about 10% of primary turns is used. The famous QUADII uses such a transformer. I have however never found a commercially available SE transformer with a cathode coil, but since I wanted to study the effect of transformer coupled cathode feedback, I tried to use the secondary both as such and at the same time as a cathode primary. The ratio $\sqrt{\frac{3000}{8}} \approx 19$ is only a little more than half the normal but nevertheless the effect is striking. With a 375V supply and tap at 50% for the screen grid the inclusion of cathode feedback halves output resistance.

The output resistance tells us about the anode resistance in the valve, which in turn tells us about the steepness of the curves representing anode voltage versus anode current. The lower the anode resistance the steeper the curves, the more triode-like they are and the lower the output resistance will be.

From the tables it can also be seen that when cathode feedback and distributed load create what I call an "artificial triode", the necessary drive voltage is increased and distortion is reduced precisely as if the EL34 was a real triode. The distortion is mainly second harmonic, again very similar to the distortion generated by a real triode.

The generation of second harmonic distortion in a triode is due to the fact that the relationship between grid voltage and anode current becomes less linear at low anode currents, meaning

[1] The term output resistance is used throughout in this paper, since it is the normal term used in audio circles. The term "internal impedance of the stage" would be more correct.

that amplification of the positive part of the signal is greater than amplification of the negative part at high signal amplitudes. Pentodes are more linear than triodes at low currents. We normally say that pentodes “can go closer to zero” than triodes.

It is now very easy without guesswork to choose the best configuration, and it is without any doubt tap at 50%, cathode feedback and 375V supply.

Only experiment could reveal this. Almost everything about the behaviour of a valve working with a resistance as a load can be predicted as long as the load is purely resistive. A load line can be drawn into a set of curves of anode current plotted against anode voltage for different grid voltages. But when the load becomes partly reactive as in this case when we use the primary of a transformer as the load, and the secondary of this transformer is loaded with the complex load of a speaker, the load line will no longer be a line. It will open up and become something like an ellipse. Not necessarily a correct ellipse but maybe an ellipse-like figure thicker at one end than at the other. So instead of trying to predict the unpredictable we can stick to the crude application suggestions of the valve table or we can use our other option: a series of experiments with the real-world combination, the valve in question, the transformer in question and the load in question.

Try to compare the results of the investigations made here with the suggestions from Siemens (pocketbook 1964) or Philips (pocketbook 1958). See appendix. You will understand what I mean.

Measurements showing correlation between input voltage (Vin), total harmonic distortion + noise (THD+N) and output resistance (Rout) for the circuits 2-5.

Circuit	**2**	**3** **tap at 16%**	**3** **tap at 50%**	**4**	**5** **tap at 16%**	**5** **tap at 50%**
Vin for 5W	6.5V	8V	9V^{+}	13V	15.5V	15V^{+}
THD+N for 5W	5.6%	4.6%	3.0%$^{+}$	3.6%	3.3%	2.2%$^{+}$
Rout	16.8Ω	9.6Ω	6.4Ω	4.15Ω	4.0Ω	3.3Ω

Test conditions: Vsupply = 275V, R_{g2} = 1kΩ, R_k = 180Ω, C_k = 470μF

Anode current = 80mA, R_{g1} = 470kΩ, Transformer 3kΩ/8Ω

With tap at 50% the maximum power was restricted to 3.5W. The measurements marked with $^{+}$ are made at 3W. All other configurations were able to provide at least 7W.

Circuit	2	3 tap at 33%	3 tap at 50%	4	5 tap at 33%	5 tap at 50%
Vin for 5W	7V	11V	13V	13.8V	18V	20V
THD+N for 5W	5.5%	4.5%	4%	2.4%	2.0%	1.7%
Rout	30Ω	6.4Ω	6.4Ω	4.2Ω	4.0Ω	3.3Ω

Test conditions: Vsupply = 375V, R_{g2} = 1kΩ, R_k = 330Ω, C_k = 470μF

Anode current = 74mA, R_{g1} = 470kΩ, Transformer 3kΩ/8Ω

At a supply voltage of 375V, power is not restricted by a 50% tap

All configurations could provide at least 8.5W

The output resistance r_{out} is calculated as shown here.

Ohm's law for the load says that

$e = r_{load} \cdot i$ where e is the voltage across the load

r_{load} is the load resistance

i is the current through the load

Ohm's law for the whole output circuit says that

$E = (r_{out} + r_{load}) \cdot i$ where E is the no load voltage at the output terminals

These two equations can be solved for r_{out}, and since the current is awkward to measure it is substituted by $\frac{e}{r_{load}}$ and we get:

$$r_{out} = \frac{(E - e) \cdot r_{load}}{e}$$

We simply measure the output voltage under load and without load and calculate the output resistance.

Passing the cathode current through the secondary produces of course a DC voltage drop that will be presented to the speaker. In this case about 35mV. This is not more than the DC offset of many high quality solid state amplifiers and will only mean a dissipation of 0.15 milliWatt in the speaker. This is allowable and can be considered insignificant.

The reader may be confused learning that a series injected current feedback *lowers* output resistance when the opposite is usually the case. The explanation is simple: When output voltage drops because of the load, feedback voltage drops too, and the grid sees a higher drive voltage. So the stage tries to restore output voltage and consequently output resistance decreases.

5. The Practical Circuit

It is now easy to draw the complete diagram to the amplifier and only a few things need an explanation.

Normally feedback is applied to the cathode of the input valve. This cannot be done here since the phase is 180^0 to what is needed, and it can't be reversed, so I mix feedback into the input signal (parallel injected voltage NFB). Let us assume that the feedback potentiometer has its wiper at the grid of the valve. The 150kΩ resistor is the upper limb of a voltage divider where the lower limb consists of the 100kΩ resistance of the potentiometer in parallel with the 22kΩ resistor in series with the output resistance of the signal source. This output resistance is now playing a major role in the feedback circuit, and since I don't want that, I use the second half of the ECC83 as a cathode-follower forming a buffer stage between the signal source and the input of the amplifier. The output resistance of the cathode-follower is low, less than 1kΩ, and it can perfectly handle a load of minimum 22kΩ with a signal never exceeding 2.5Volts.

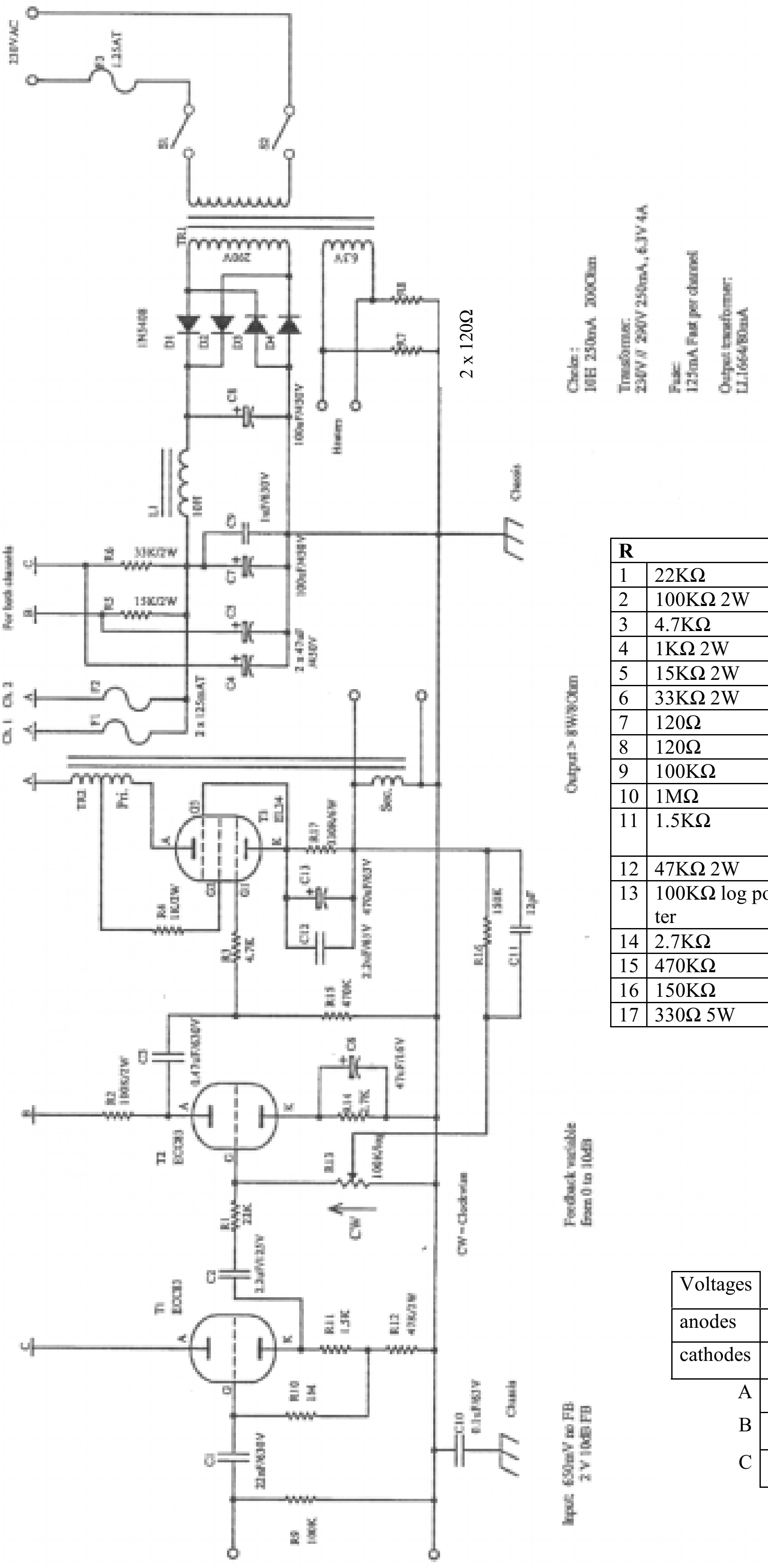
The feedback potentiometer should be a logarithmic potentiometer and it must be connected so that feedback increases and sensitivity drops when the potentiometer is turned clockwise as indicated in the diagram. This gives the most sensible regulation curve.

If the amplifier is to be fed from a signal source with a low output resistance (preferably less than 1/10 of the 22kΩ resistor) the cathode-follower can be omitted and the signal applied directly to the 22kΩ resistor.

If a potentiometer with a total resistance of 10kΩ is used in front of the amplifier, and this potentiometer is fed from a signal source of maximum 2kΩ, the output resistance from the wiper will be min 0Ω and max $\frac{10+2}{4}$ kΩ = 3kΩ. This variation is still acceptable.

Input sensitivity varies from about 0.6V to 2.0V (max NFB) for full output.

Since this method for applying feedback is different from the normal, the consequences for the HF Cut-offs in the amplifier should be investigated.



R		C
1	22KΩ	22nF 630V
2	100KΩ 2W	2.2µF 125V
3	4.7KΩ	0.47µF 630V
4	1KΩ 2W	47µF 450V
5	15KΩ 2W	47µF 450V
6	33KΩ 2W	47µF 16V
7	120Ω	100µF 450V
8	120Ω	100µF 450V
9	100KΩ	1µF 630V
10	1MΩ	0.1µF 63V
11	1.5KΩ	12pF or 22 pF trimmer
12	47KΩ 2W	2.2µF 63V
13	100KΩ log potmeter	470µF 63V
14	2.7KΩ	
15	470KΩ	
16	150KΩ	
17	330Ω 5W	

Voltages	1	2	3
anodes	305	260	360
cathodes	65	2.1	23.5
A	370		
B	345		
C	305		

The resistive adding network used to mix the feedback signal into the signal presented to the grid means that the amplifying stage is fed from a higher resistance than would normally been the case. In the worst case (no NFB) the source resistance is 22kΩ (the series resistor) in parallel with 100kΩ (the grid resistor). This combination makes 18kΩ which in conjunction with the input capacitance forms a low-pass filter. The question is: how bad is this?

Before we can answer that we must know the actual value of the input capacitance. For ECC83 the capacitance between grid and anode is 1.6pF. Suppose the grid voltage *drops* by 1Volt. Amplification is 52 times, and the stage inverts so the output voltage rises by 52 Volts. The capacitance grid to anode will be charged to 52 + 1 Volts = 53 Volts. This capacitance will act as if it was not 1.6 pF but 1.6 pF x 53 = 85 pF + inevitable stray capacitances. This is called the Miller effect. We may conclude that input capacitance of the stage is 100pF. The cut-off frequency of an RC-filter is

$$f_o = \frac{1}{2\pi \times R \times C} = \text{(R in K}\Omega\text{ and C in nF gives f in MHz)}$$

in this case $$\frac{1}{6.28 \times 18 \times 0.1} = \frac{1}{11.3} = 0.088\ \text{MHz} = 88\ \text{kHz}$$

88kHz is the frequency where the signal has decreased by 3dB.

The output resistance of the driver in conjunction with the input capacitance of the EL34 forms a HF cut-off too. The output resistance of the driver is the anode load resistor of 100kΩ in parallel with the anode resistance, i.e. the resistance through the valve seen from the anode. This can be taken from a valve table and is around 65kΩ. The valve tables often use the term internal resistance. So the output resistance is $\frac{65 \times 100}{65 + 100} \approx 40$ kΩ[2]. The input capacitance of the output stage is about 40pF so the cut-off point will be

$$f_o = \frac{1}{2\pi \times R \times C} = \frac{1}{6.28 \times 40 \times 0.04} = \frac{1}{10.05} \approx 0.1\ \text{MHz} = 100\ \text{kHz}$$

When feedback is injected to the cathode, it is applied as *current* feedback, which raises input and output resistances. Here feedback is injected to the grid as *voltage* feedback, which reduces input and output resistances. 10 dB NFB reduces amplification and resistances by a factor 3 so

[2] The resistance, R of r_1 and r_2 in parallel is $\frac{r_1 \cdot r_2}{r_1 + r_2}$

the new internal resistance is $\frac{65}{3} \approx 22$ kΩ and the new output resistance will be 100 kΩ in parallel with 22kΩ,

$$\frac{22 \times 100}{22 + 100} \approx 18 \text{ k}\Omega.$$

The new cut-off frequency is:

$$f_o = \frac{1}{2\pi \times R \times C} = \frac{1}{6.28 \times 18 \times 0.04} = \frac{1}{4.52} \approx 0.221 \text{ MHz} = 221 \text{ kHz}$$

The input resistance of the driver is also reduced so the first HF cut-off is raised too.

Even in the worst case and even given they are cumulative, these HF cut-offs are no matter for serious concern because the combination of the output valve and the output transformer will still be the most significant cut-off. It is below 50 kHz.

It must be considered an advantage that instead of lowering the critical cut-off between driver and output stage as it is normally the case, this way of applying NFB does the opposite. Had NFB been cathode injected current feedback, the output resistance of the driver would have been close to 100 kΩ and the cut-off frequency would have been in the 40 kHz region and this would be serious.

As the promise you make when you get married is not to be taken lightly, neither is stability of an amplifier under influence of global NFB. In this case we can relax. Apart from the output transformer there is only one HF cut-off within the feedback loop, and when full NFB is applied this cut-off is 2 octaves above cut-off of the output stage. The amplifier is rock-stable for any combination of resistive and reactive loads.

The power supply is straightforward. The choke is, as explained earlier, necessary and should preferably be generously dimensioned.

The last reservoir capacitor of 100μF is bypassed with a 1μF/630V foil capacitor to reduce HF resistance of the supply. Also the cathode bypass capacitor of the output stage is shunted with a foil condenser. Nothing seems to be gained by bypassing other electrolytic capacitors but I admit that in theory they ought to be bypassed. It is a matter of conviction and taste, and you can of course do it if you want to.

The power dissipation in the EL34 is 25Watts (anode + screen grid), which is still safe. Operating in class A always means maximum permissible power dissipation.

6. Components and Layout

Since the amplifier is designed around a high-quality transformer, only top grade components should be used throughout. Luckily there are only few components at all so costs will still be reasonable. I use ceramic sockets for the valves and metal film resistors except for the 330Ω cathode resistor for the EL34 where a 5W wire wound type clamped to the chassis with heat-sink compound for cooling is employed. 125mA fast fuses are used in the supply-lines for the output valves. Fuses are not very linearly behaving components and may impair the final result. It is a matter of principle whether to use them or not. I prefer them, you may not, and they certainly can be omitted.

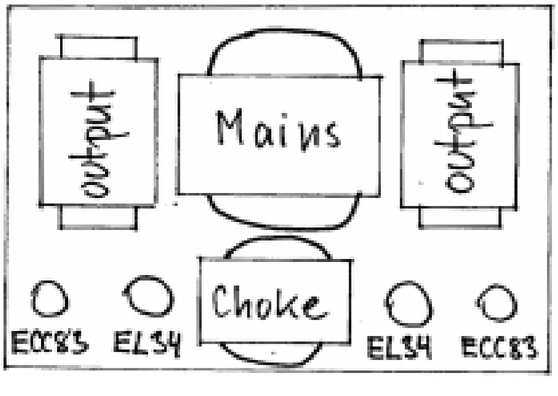
Normally I prefer monoblocks to stereo stages, but with this relatively small amplifier I made an exception so the negative supply is connected to chassis at the common ground point of the two 47μF capacitors, the 100 μF reservoir capacitor and the 1μF HF bypass capacitor. Two bus bars are taken from here, one to each amplifier. They go to the ground points of the output valve and further on to the input valve and end at the ground points of the *isolated* input sockets. These two ground points are then decoupled to chassis with 0,1μF/63V foil capacitors. This is clearly shown in the diagram.

My mains transformer is a leftover from a batch that I had made for an earlier project and it is a normal EI type. Minimum requirements are 290V 250mA and 6.3V 4A.

I made the amplifier on a TEKO box 300x160x70 mm. It is cheap and made from aluminium and not very ugly and it is easy to work in aluminium.

I have not made a printed circuit board for the amplifier. It is easily built components soldered directly to the valve sockets and a tag strip a few centimetres apart from the sockets, and layout is in no way critical as long as you keep distance between the leads to the primary of the output transformer and the input valve and associated components. Heater leads should be twisted and pressed into the corners of the chassis.

The output transformers are oriented so that their coils are at right angles to the coil of the mains-transformer and a symmetry line is shared by all three transformers.



Suggested layout

Connections to the output transformer are shown on the next page. The primary is divided into two sections each with a tapping point. You have two options:

1. Supply voltage connected to 1, 5 strapped to 11 and anode connected to 7. You have tapping points at 16%, 50% and 83/%.
2. Supply voltage connected to 5, 6 strapped to 1 and anode connected to 11. Tapping points are now at 33%, 50% and 67%.

I have chosen my tap at 50% so both configurations will do. It is essential that polarity is right when the secondary is included in the cathode circuit of the output valve. Make sure that output signal *drops* when you make these connections. Otherwise feedback becomes positive and output resistance will increase.

LUNDAHL
– TRANSFORMERS –

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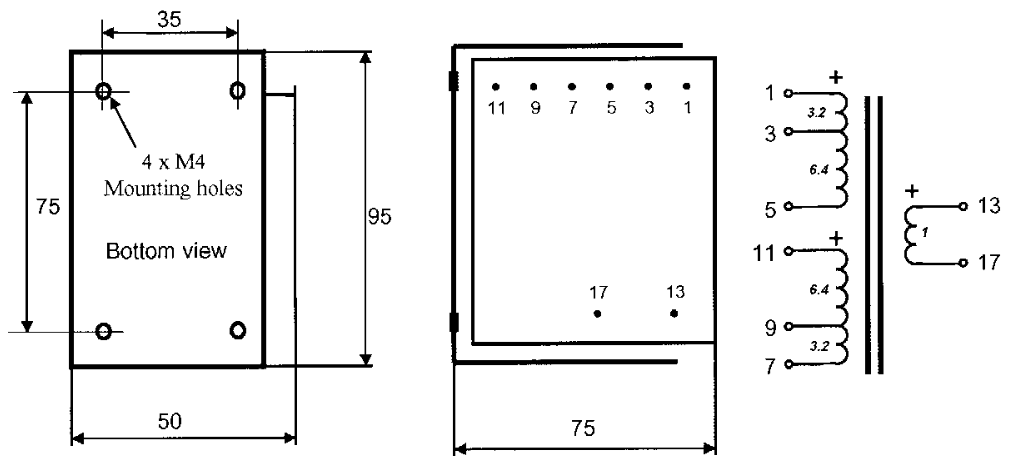
Tube amplifier output transformer LL1664
3k : 8 ohms

The LL1664 is a four-sectioned dual coil C-core tube amplifier output transformer for 3 k: 8 ohms impedance ratio available in PP and SE versions.
The coil is wound using our standard high internal isolation technique with isolation foil between each copper layer. The core is an audio C-core of our own production.

Turns ratio **9.6 + 9.6 : 1 or (3.2+6.4)+(3.2+6.4) : 1**

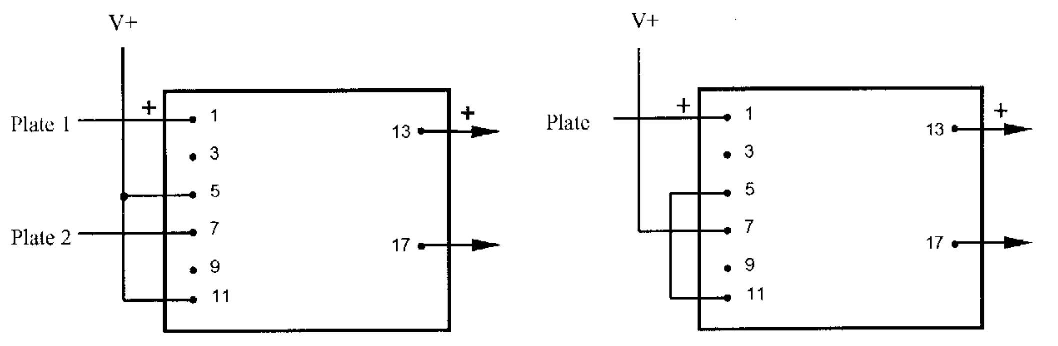
Winding schematics, physical dimensions, pin and mounting hole layout (all dimensions in mm)



Weight:	1.35 kg	
Static resistance of each primary:	74 Ω	*Primary inductance for LL1664/80 mA is 22H*
Static resistance of secondary:	0.5 Ω	
Isolation between windings / between windings and core:	4 kV / 2 kV	*Leakage inductance of primary is 8mH*
Max DC current through any primary winding:	200mA	
Primary leakage inductance, primaries in series:	8mH	

	LL1664/PP	**LL1664/50mA**	**LL1664/100mA**
Primary inductance		35H	17H
Max primary signal	410V R.M.S. @ 30 Hz	180V R.M.S. @ 30 Hz	180V R.M.S. @ 30 Hz
Max output power @ 30 Hz	55W (8Ω spkr)	10W (8Ω spkr)	10W (8Ω spkr)

Suggested use:



LL1664 used in Push Pull
Plate-to-plate impedance
3k at 8 ohms load

LL1664 used in Single End
Primary impedance 3k at 8
ohms load

R010921

Global Negative Feedback

Facts and Myth

The matter of overall or global negative feedback (NFB) seems to provide material for endless discussions *pro et contra*. The debate is often heated by emotions and since the participants very often know only little about the facts the debate tends to be futile.

Let me arm you for the debate with some facts.

Under the right circumstances NFB has several benefits to offer. The most important are:

1. NFB stabilizes gain
2. NFB flattens frequency response
3. NFB reduces output resistance
4. NFB reduces distortion
5. NFB reduces phase shift
6. NFB minimises the influence of ageing of valves.
7. NFB reduces hum and noise

But these very desirable benefits are only without negative side effects *if and only if:*

1. The amplifier has an infinite frequency response *prior to NFB*
2. The amplifier shows no linear or non-linear distortion *prior to NFB*
3. The amplifier has no phase shift *prior to NFB*

An amplifier complying with these demands would not need NFB at all except for reduction of output resistance and for minimising ageing symptoms, which is almost the same as stabilizing gain.

NFB can be compared to a medical drug prescribed for a disease. It has benefits and it has negative side effects. The art is to adjust the amount of the drug so that life-quality is improved optimally for the patient, and the less he is attacked by the disease the less of the drug he will need and the better he can withstand the side effects.

From this it should be clear that NFB is not a tool to make up for a bad design. On the contrary from this we can learn about the importance of good engineering, and the most critical factor is phase shift, which has to be kept as small as possible at least in the audio band and one octave above, or higher depending on the amplification and the frequency response. Only minimal phase shift can ensure that NFB stays *negative* all the time.

Amplifiers with too high a degree of NFB with respect to phase shift and frequency roll-off may become unstable and prone to oscillations at certain signal levels and difficult reactive loads. This cannot always be revealed under static conditions (a sine wave into a dummy resistor).

Manufacturers have always competed over distortion figures and have for that reason often used more NFB than I feel necessary and healthy.

Let us look into the influence of NFB on distortion. When say 3rd harmonic distortion is fed back to the input, this distortion is of course reduced by the feedback factor but since the amplifier apparently generates 3rd harmonic distortion the feedback itself will be distorted and 9th harmonic distortion is the result, and 9th harmonic distortion is a lot more unpleasant than 3rd. Obviously we must try our very best to design the amplifier to be as perfect as we possibly can prior to NFB.

For me the three most important benefits of NFB are stabilization of gain, lowering of output resistance and minimising of ageing symptoms in the valves.

In valve amplifiers gain tend to fluctuate a little due to fluctuations in cathode emission and the heat-inflicted changes in distance between the electrodes. Gain fluctuations are of minor importance in a mono installation, but in a stereo set-up stability of the image is very much dependent on equal and stable gain in both channels. Equal and minimal phase shift in the two channels is of course also of major importance.

The effects of the output resistance are explained earlier.

The importance of the distortion figures is in my opinion somewhat overrated. The reason for this is to a great extent historical. Apart from output power, distortion at 1000Hz is the easiest parameter to measure, and it is easy to verify a manufacturer's claim, and as stated earlier THD figures became an issue for competition between manufacturers from the early days of Hi-Fi, which in turn draw the attention of the buying public to these – often remarkably low - figures.

I feel that when it comes to harmonic distortion by harmonics up to 6th, figures up to 1 or maybe 2 percent at high power levels are of minor interest since music always contains these harmonics, and a lot of people find that especially a small amount of even harmonic distortion is even agreeable.

To me linearity is a matter for concern, because lack of linearity produces intermodulation which is a very annoying type of distortion. Our ears produce intermodulation, but this seems to come from the centre of our heads, and we have learned to ignore it if it is masked by signals of interest. But intermodulation created in an amplifying system has its origin in the speaker. Consequently it has a *direction* and we will no longer ignore it.

After this digression, which I hope will prove valuable to the reader, we are back to the actual amplifier. Provision is made for adjustable global negative feedback. This enables you to judge about benefits and possible drawbacks, and you will soon become a more qualified participant in the debate. You should however not forget that when you compare an amplifier with NFB to the same amplifier without NFB you are also comparing how your speakers react when fed from sources with different output resistances, and this could very well prove to be the most important reason for the change in sound. If a fair comparison is to be made the amplifier with the lower output resistance should be fitted with a series resistor in the speaker output to compensate for the difference.

The only thing that remains to be explained in the diagram is the capacitor across the 150kΩ feedback resistor. It is adjusted for optimal square-wave reproduction.

If you have access to an oscilloscope and a square-wave generator you can inject a 6-10kHz square-wave and adjust level for about 5 Volt out over 8Ω with full NFB. The capacitor is adjusted so that no overshoot is present and rounding is minimal. In this case the capacitor is replaced by a 22pF trimmer.

Nothing else needs any adjustment. You should however check voltages at the points where they are shown in the diagram. Note that deviations of 10% are not uncommon.

Further Improvements and Alternatives

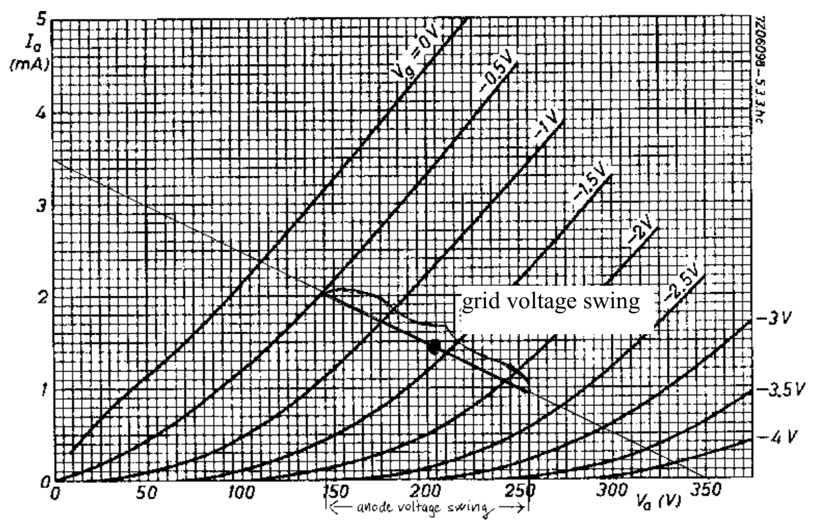
Equipped with the EL34 the output stage is as perfect as it can possibly be. I have however been told that another version of the EL34 called EL34S exists. This valve should not be a normal pentode but a *beam tetrode*. If this is true I would expect it to behave even better than the normal version in this amplifier. I have never seen this valve, so I don't know, but it is easy to verify whether the EL34S is a beam tetrode or not. Measure the cathode current and the screen grid current. All other factors being equal a beam tetrode draws less current through the screen grid, due to the alignment of the meshes in the grids.

Another question would be how the amplifier would perform if the EL34 is substituted by the famous KT88. I do not know, but surely an excellent amplifier could be built based on this valve of reputation. It just wasn't my project. You may remember that I wanted to use widely available and affordable valves. If you wish to try, I recommend a series of experiments like those described to ascertain the optimal working configuration for the valve.

It is of course also possible to avoid any compromises and use a real (and very expensive) triode like the 300B. This valve has a reputation for being one of most distortion-free valves ever constructed. Sadly the drive requirements are severe: almost $70V_{eff}$[3] or $200V_{peak\ to\ peak}$ is needed, so a completely redesigned driver with a very high supply voltage must be made. Again a completely different project.

The driver stage seems to leave room for improvement. The normal cathode coupled stage is not very linear when high output swing is required.

The first of the two main reasons for distortion is that –Vg curves are not equidistant along the working line seen from the working point:



[3] Even though the term V_{rms} would be correct V_{eff} is used throughout because it seems to be the most commonly used among audiophiles. For a sine-wave voltage $V_{eff} = V_{rms}$

Working line is drawn for V_{supply} = 350V and R_{Load} = 100kΩ
The working point is marked with a dot and the maximum usable part of the working line is bold. The working line is sometimes referred to as the load line.

It can be seen that the V_g curves are squeezed closer together the more we move to the right side of the working point, which is the normal point recommended by most manufacturers, and this is the point where the lowest distortion can be expected for a reasonable output swing.

The line can be used to about V_g = -0.5V (onset of grid current). Since the working point is at V_g = -1.4V, the stage can handle an input swing of 0.9V in both directions, resulting in an output swing from 145V to 255V or 110$V_{peak\ to\ peak}$ = 39V_{eff}. In the table below the recommended working conditions for one section of an ECC83 are given for a load resistance of 100kΩ and supply voltages ranging from 200 to 400V. The 350V Column is interesting here because this is the actual value in this amplifier. I used this working point as I started my experiments.

Supply voltage	V_b	200	250	300	350	400	V
Anode resistor	R_a	100	100	100	100	100	kΩ
Grid resistor next stage	$R_{g'}$	330	330	330	330	330	kΩ
Cathode resistor	R_k	1800	1500	1200	1000	820	Ω
Anode current	I_a	0.65	0.86	1.11	1.40	1.72	mA
Voltage gain	V_o/V_i	50	54.5	57	61	63	-
Output voltage (I_g = 0.3μA)	V_o	20	26	30	36	38	V_{eff}
Total distortion	D_{tot}	4.8	3.9	2.7	2.2	1.7	%

The information we gain from the curves is, as we might expect, consistent with the informations given in the table. It is, however, a good exercise to verify that because by doing so we learn a lot about why the stage behaves as it does.

We saw that an input swing of 1.8V produced an output swing of 110V. The amplification is $\frac{110V}{1.8V}$ = 61 times, precisely as given in the table. The anode current at the working point is 1.4mA. The table provides exactly the same information.

From the table we see that distortion is 2.2% for 36V_{eff} out. From the curves we can only see that maximum output is just below 40V and that some distortion must be expected because the distance between the -2V and the -2.5V curves is smaller than between the -1V and the -1.5V curves.

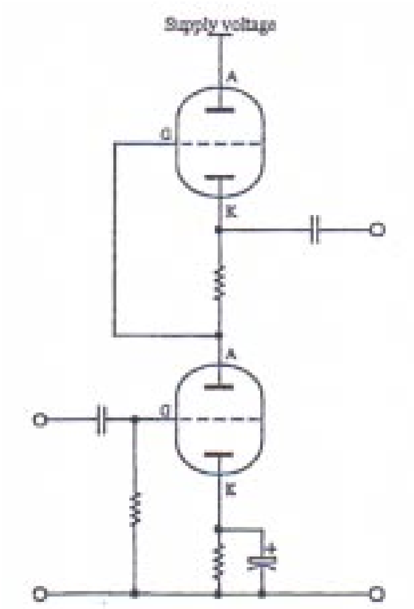
We can also ascertain that the $25V_{eff}$ needed for the output stage can be safely provided, although with some distortion. The table tells us that the stage can handle a load down to 330kΩ (grid resistor of next stage). The overload margin, $\frac{36V}{25V} = 1.44 \approx 3dB$ is not very impressing.

The second reason for distortion is the loading of the output. The output is not only loaded by the grid resistor of the following stage. There is also a capacitive loading, which becomes more and more heavy as frequency increases. The output stage exhibits about 40pF input capacitance, and this presents a load of about 220kΩ at 20 kHz. No matter how big the grid resistor we use, the loading of the driver will produce increased distortion in the upper end of the audio band, and the frequency response will be affected too.

But these two components are not the only loads to the signal. Since the positive supply rail has negligible resistance to ground for AC signals, the anode load resistor itself loads the signal, and the *effective* load is the parallel resistance of all components. At low and medium frequencies the capacitive loading may be left out.

It seems that what we are looking for is a device that eliminates the loading of the signal by the anode load resistor and forms a buffer between the output of the amplifying stage and the input of the next stage.

A compound stage, often referred to as an SRPP stage (shunt regulated push-pull) or a μ-follower, which term I prefer, meets all the requirements.



The lower valve is a normal cathode coupled amplifier as we already know it, but the anode load resistor is replaced with a valve coupled as a cathode-follower. The term cathode-follower has been used since Adam was a boy, but is in fact misleading because what really happens is that the cathode voltage is forced to follow the grid voltage so the term grid follower would be more consistent with what happens - but the term cathode follower sticks and will probably never change.

The active load works as follows: When the anode of the lower valve goes more positive, the grid of the upper valve goes more positive, and since the cathode follows the grid it goes more positive too. The amplification of the upper valve is very close to unity. Consequently the upper cathode follows the lower anode closely and very little signal activity will take place over the upper cathode resistor, indicating that the loading of the signal has disappeared. The grid of the upper valve presents no resistive load, and since gain is unity the capacitance is very low too (no Miller effect).

The output is now taken from a cathode-follower with very low output resistance and thus not affected by the loading of the next stage. Even the influence of the frequency dependent component is minimised.

Since the cathode follower has an AC resistance close to infinity the working line will be almost horizontal and amplification will be close to the theoretical limit, the amplification factor, μ of the valve, hence the name μ-follower.

I tried to replace the simple driver with such a stage. With a supply voltage of 350V a sensible working point is achieved when the upper and the lower valve share the voltage evenly.

An output swing of $50V_{eff}$ with low distortion can be expected when the anode current is about 1.2mA.

The upper cathode will now be at a potential of 175V and allowed to swing 70Volts up and down. This presents a problem since the maximum permissible voltage between heater and cathode for the ECC83 is 180V. Instead of referring the heater potential to ground it must now be referred to +90V so that the no-signal potential at the upper and lower cathodes is now + and - 90V respectively with respect to heater, and for the output stage the voltage between heater and cathode is now 60V (the cathode is already at +30V). This is permissible since EL34 allows 100V between heater and cathode.

I have now used both section of the ECC83 for driver, and unless I can feed the amplifier from a source with low resistance I shall need an extra triode for the input stage. This was not part of the plan as you may remember. The question is: will improved performance justify *oleum et operam?*

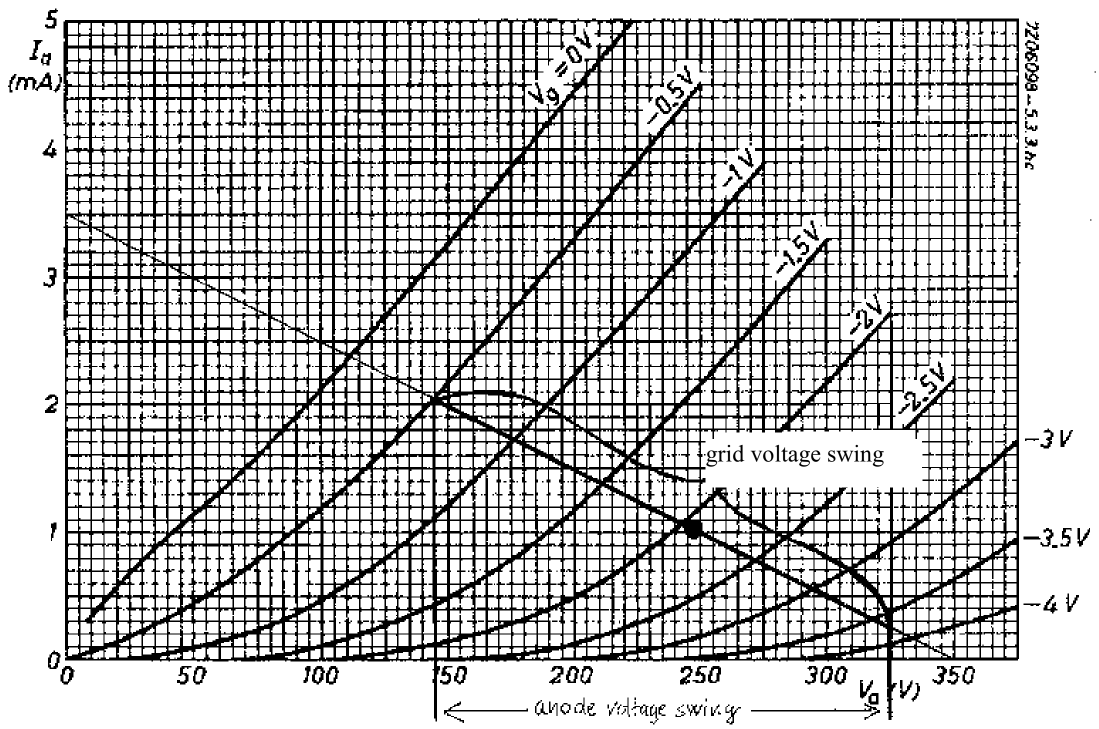
Surprisingly the answer is NO!

The distortion of the amplifier did not go down - on the contrary it went up a little. The sonic performance was not better on the contrary it was not as good as before. Differences were not big but they were certainly there. The only parameter that improved slightly was frequency response. The 3dB point moved upwards, but since leakage induction and capacitance in the output transformer worsen the conditions for the output valve as frequency rises, an extension of the frequency response much beyond the audio band is not desirable.

So the result of my effort was disappointing. Why does a simple distorting stage perform better in this context than an excellent stage much favoured by audio enthusiasts, possessing all the qualities one can ever ask for to do this job?

The answer is lack of synergy. Earlier I explained that the distortion generated in triodes and triode-like stages originates from the fact that the positive and the negative parts of the signal are not equally amplified - look again at the curves for the ECC83 below.... Since both stages invert, the part that is least amplified in the first stage will be the part most amplified in the second. It is therefore possible - at least partly - that the shortcomings of one stage can be compensated by shortcomings of the opposite nature of the next stage.

This seemed a field worthy of further investigation, and since the output stage cannot be altered without loss of available power, the parameter to change is the working point of the first stage. Since a high output swing is needed we shall need as high a supply voltage as possible. This is still 350V and since the stage has to withstand the load of the output-stage we cannot use a load resistor higher that 100kΩ. This means that the working line will remain as it was so the sole changeable parameter is the position of the working point, i.e. this bias voltage. Both measurements and listening test revealed that the best performance I achieved was when bias is -2.2V, see below:



Again the working line is drawn for V_{supply} = 350V
and R_{load} = 100kΩ. The Working point is marked with a dot
and the maximum usable part of the line is bold.

The anode current is 0.8mA and the corresponding cathode resistor for -2.2V bias must be $\frac{2.2V}{0.8mA}$ = 2.75kΩ. 2.7kΩ bing the nearest standard value. The new working conditions depicted here show that the stage can handle an input swing of $\pm$1.7V before we reach the critical -0.5V where onset of grid current can be expected. This swing produces an output swing from 145V to 325V. So an input swing of 3.4V yields an output swing of 180V. Amplification has fallen slightly and is now $\frac{180V}{3.4V}$ = 53 times, which is still sufficient.

The maximum swing of $180V_{peak\ to\ peak}$ equals 66_{eff}, but distortion has risen considerably as can be seen from the difference in distance between the V_g curves to the right and the left of the working point. This distortion is however counteracted by a similar but opposite distortion in the output stage, so as mentioned above synergy is what makes this amplifier surpass so many competitors. It should be remembered that still only 25 V_{eff} is needed to drive the output valve.

I have taken you through this long and maybe for the practically minded Do it Yourself audio enthusiast a slightly boring, theoretical chapter to show that an amplifier cannot always be looked upon as a series of isolated stages that can be optimised one after another. Sometimes they must be regarded as a whole and the stages must be optimised *together*.

I shall end my explanations here, but since I found the outcome of these experiments extremely interesting I made further investigations. I tried to use a cascode as a driver and I even tried to use a little power triode as a driver. None of these solutions were able to compete with the simple cathode coupled amplifier stage optimised for this particular use.

The amplifier is as demonstrated is very close to what can be achieved when the design goals are kept in mind. It is simple with a short signal path, it is fairly cheap, it is very easy to build and in no way critical, and the sound is indeed very satisfying.

Everything in this world can be improved, but this amplifier makes the optimal use of all invested components, and a real improvement means a totally different approach and far more serious financial implications.

Measurements and Results

By means of Audio Precision equipment a series of measurements was carried out on the completed amplifier with and without feedback. The most important results are found in the table below, and some of the original plots can be seen in the appendix.

Frequency response	**With 10 dB NFB**	**Without NFB**
at 1 W out +0/-1 dB	10 Hz - 35 KHz	12.5 Hz - 19.5 KHz
at 8 W out +0/-1dB	18 Hz - 35 KHz	18 Hz - 19.5 KHz
Total Harmonic Dist. + Noise*		
at 1000 Hz 1 W out	0.20%	0.60%
at 1000 Hz 8 W out	1.68%	3.59%
at 100 Hz 1 W out	0.27%	0.82%
at 100 Hz 8 W out	2.64%	4.46%
at 6.3 KHz 1 W out	0.21%	0.63%
at 6.3 KHz 8 W out	1.61%	3.25%
Second Harmonic Dist. at 500 Hz 5 W out	0.50%	1.12%
Third Harmonic Dist. at 500 Hz 5 W out	0.02%	0.10%
Output resistance 20 Hz - 20 KHz	0.9 Ω	3.3 Ω
**Phase-shift 20 Hz - 20 KHz	-32^0 - $+33^0$	-32% - + 32^0
Noise on output Input terminated with 20 k Ω:		
20 Hz - 20 KHz (“Fremdspannung”)	90 µV	330 µV
CCIR 468	45 µV	140 µV
A-Weighted	20 µV	60 µV

* Distortion products measured up to 30 KHz

** Phase Shift is measured from grid of driver to output i.e. within the feedback loop.
The phase shift reaches maximum, 90^0, at 60 KHz. Then it starts to decrease and at 120 KHz it is 27^0.

The figures for distortion and frequency response are not impressive and undoubtedly the observant reader will notice the absence of figures for intermodulation.

As pointed out earlier we would expect distortion to be mainly 2nd harmonic, and the measurements confirm this. Remember that this is bought at the expense of linearity (less amplification at high levels of the negative half period in both stages), and intermodulation is always the inevitable result of lack of linearity. It was however found that the amount of intermodulation is very dependent on frequencies and amplitude relations, so giving exact figures makes no sense.

In this respect this amplifier behaves almost like an analogue tape-recorder where non-linearity in the relationship between signal amplitude and resulting magnetization of the tape causes intermodulation of exactly the same nature as the one found in this amplifier. Note that figures for intermodulation were never given in the specs for even the most respected (and expensive) studio tape recorders - and for the same reasons as here.

Only when it comes to the noise figures this amplifier can compete with modern designs. The noise is extremely low and even with the ears in the cone of the woofer or at the dome of the tweeter it is hard to tell whether the amplifier is switched on or not.

The open-loop amplification i.e. the amplification without feedback is 22dB, and given the plots for phase-shift and frequency response it can be seen that the amplifier, as predicted, is unconditionally stable at any level of negative feedback.

Note that the phase-shift is virtually unaffected by feedback - yet another proof of an excellent output transformer.

The valves used were SOVTEK EL 34^{s} and TELEFUNKEN ECC 83^{s} from my old stock.

As mentioned earlier the specs cannot compete with even a modest modern amplifier. They are, however, very good for a single-ended valve amplifier. But even though measurements are interesting the main thing is: how does it sound?

Evaluation and Conclusions

I have now lived with this amplifier for the last six month. I have listened to it every day and must confess that I have enjoyed it. So the old question pops up again: The relationship between what we measure and what we hear. How are the figures for distortion and all the other parameters linked to our perception? Not as closely as manufacturers of Hi-Fi equipment want us to believe, and I am convinced that at least some important qualities in a sound reproducing chain are still not tangible by measuring equipment. This does of course not mean that measurements are worthless. We can read a lot from them, but this amplifier raises the great question of how much? And how much is relevant?

It is almost impossible to describe in words the qualities of sound. Many - and in my opinion too many - words are used by reviewers in magazines, so I shall try to limit myself to a minimum. Good SE-amplifiers have a reputation for an excellent midrange-reproduction, and this one exhibits extraordinary resolution in this critical domain. The high end of the frequency spectrum is reproduced without any trace of aggressiveness, and these are to me the two main features because bad midrange and harsh high frequencies are the main reasons for what is nowadays referred to as "listening fattigue".

The bass is far better that one would expect from the measured performance. The relationship between 2nd and 3rd harmonic distortion is responsible for this. The 2nd harmonic tends to enhance the fundamental (due to intermodulation in our ears) while 3rd harmonic tends to draw our attention away from the fundamental. The subjective result is that bass reproduction is richer than expected and still very clean even given a substantial increase of distortion below 50Hz. This increase must be attributed to the not infinite inductance of the primary of the output transformer. The impedance drops with decreasing frequency and the loading of the output valve becomes increasingly difficult.

The amount of NFB that I prefer is about 8 dB. At this level of feedback output resistance is low enough to damp resonances in my speakers (DYNAUDIO C2 monitors) and stereo-imaging seems optimal to me.

A very interesting question is: how does intermodulation affect the perceived performance?

The answer to that is difficult because the amount of intermodulation is level- and frequency dependent, and audibility of intermodulation is programme dependent.

At high levels intermolulation is clearly audible when a wind-quintet or a part-singing choir with many female voices is reproduced, and unfortunately this types of programmes are among the more power-demanding. A full symphony orchestra can be reproduced up to full power without audible intermodulation.

So the answer depends on your musical taste, your listening-level preferences and the sensitivity of your speakers.

Single-end enthusiasts maintain that the "Quality of the First Watt" is the most significant, and they are to some extent right because the bulk of all listening takes place at power levels below one Watt, and it is exactly in this power domain that the typical run-of-the-mill amplifier, at least in the past, showed problems like cross-over distortion or transient distortion, but an amplifier must of course be able to reproduce peaks faithfully too.

Recently the symphony orchestra of the Royal Academy of Music played the 8th Symphony by Dvorak, a full-bodied late romantic symphony in the concert hall of Aarhus. It is a fullscale professional orchestra. It has this winter recorded an internationally available and highly acclaimed CD with music by Malcolm Arnold. At the dress-rehearsal I measured the Sound Pressure Level (SPL) at different positions in the hall which has 1400 seats. In 10th row (normally considered the best seats) the SPL never exceeded 90 dB unweighted and just in front of the conductor never 96 dB unweighted.

I recorded the concert (it is my job), and in my editing room the amplifier was fully able to reproduce the music at the level in the conductor's position in my normal listening position about 1.5 meters from my DYNAUDIO Speakers, which are of average to low sensitivity, and I found the sound very satisfying.

So given speakers of reasonable sensitivity and a taste for clear and non-aggressive, not too loud reproduction this amplifier presents an attractive alternative to e.g. a 10W Push-pull amplifier with 2 x EL84 in the output stage, and I have found a use for it where it seems unbeatable. I use it in the room where I prepare my historical reissues. I have never heard anything reproduce 78 rpm records so beautifully and for this use the limited power is not felt as a problem.

My conclusion is that this is an amplifier with a personality, a cultivated, friendly and forgiving personality that I would never stress nor challenge it beyond its power.

I believe that it was the 17th century scientist Gallilei who encouraged his students to "measure everything measurable and to make measurable what yet is not."

I shall conclude this chapter by passing his words on to the makers of instruments for audio measurements.

There is still a lot that we don't know.

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He has listened with patience to my considerations and his advice based on decades of experience has proven extremely valuable to me. Last but not least he has made Audio Precision equipment for the measurements available to me and he also helped me through the measurements.

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The Author

Claus Byrith, b.1941, is a sound recording engineer. Since he was a boy he has designed and built all sorts of audio equipment. In 1968 he joined the team of the Department of Electronic Music and Musical Acoustics, a joined venture of the Royal Academy of Music in Aarhus and the University of Aarhus. In 1973 he became head of the studio at the Royal Academy. He has done innumerable recordings and received the Swedish “Fonogrampriset” in 1983 for recordings of Quartets by Stenhammar. He has also received an American Critics Award. In later years he has taken great interest in restoring of old recordings, and he has prepared many reissues of Danish recordings of the thirties and fourties.

His re-awakened interest in valve technology is based, as he puts it, on the fact that valve equipment is simple and easy to understand, serviceable and when well designed, extremely well sounding.