At the European Triode Festival in Baarlo 2022 I had planned to do a presentation on the combination of positive feedback and negative feedback, and about a funny power amplifier based on SITs, but due to some latent health issues I could not attend. Peter van Willenswaard was so kind to give the talk. I thank him very much, and I follow his idea to write it out.
Here is part 1, as I split up the part on positive feedback as used in the preamplifier (line and phono) and a part 2 on the driver I made for the SIT 2SJ28-source follower amplifier (to follow)

Positive thinking about harmonics

For many audiophiles feedback is a menace. A grave danger. They like open loop amplifiers and single ended outputs. And distortion is often bad bad.
Negative feedback is generally seen as nasty – because in the loop there is a change in the phase of the signal as frequency increases to very high. When the change in frequency gives a phase difference near 180°, there is positive feedback and the amplifier becomes unstable. It might not be heard, nor even visible on a scope. Some oscillations are in the Gigaherz range. You need special probes to see it. But you can see the effect: the DC point of the output shifts. For instance, when you touch a component, even an isolated part. A Nyquist diagram can be plotted (I can't do it) to show at which frequency the phase becomes dangerous. I once made a push-pull amplifier where the para-phase had an oscillation at UHF frequencies. A shortwave radio! Just by moving a capacitor or a wire around it could be stopped. Local feedback turned to positive phase. It gets out of control quickly.

So – we avoid any design with positive feedback.

But.
Positive feedback also increases the dynamics of an amplifier. Used in moderation it can have a benign effect - just a slight bit can liven up the music.

And the question is, can a slight amount of local positive feedback be combined with a closed loop negative? Short answer: yes.

First, lets look at why harmonics matter, and what is a good harmonic structure. Because harmonics are not thát bad.

On harmonics in amplifiers

First of all. The structure of the harmonic degradation has always been important in audiophile circles. Musical instruments can be distinguished more by their harmonics that are generated than by the fundamental frequency - say 440 Hz. Slight differences can be heard – the personality of the maker, even up to the 9th harmonic. Amplifiers thus have to transmit that experience. Some amplifiers generate some harmonics themselves, while another amplifier is 'distortionless' and gives 'no' harmonics - both sound different, it is found.
Just like audio output with a low damping factor and some distortion is often more pleasing than one that has a high damping factor and virtually zero distortion, when using full-range

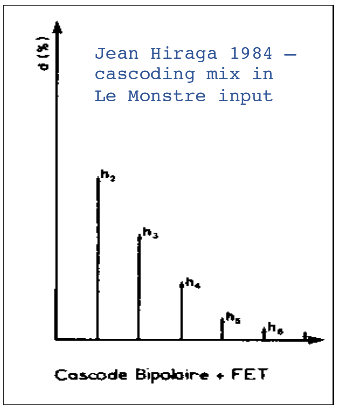
and horn speakers. Most audiophiles have speakers that expect some output resistance, high sensitivity speakers first of all, and many full-range speakers even improve with a damping lower than 1 because resonances are miraculously attenuated. Of course, speakers can be made for amplifiers virtually having no resistance. Audiophiles often like the amplifier with quite some resistance and harmonics more: their search is for audio with a realistic timbre and it is through listening they make a choice.

Audiophiles love triodes – they inherently give a second harmonic and a decay of the next harmonics. Though a good pentode has its place in the mix too with its propensity for third harmonics – it can help create a nice combination of second (from the output triode) and third (from the driver, a pentode). "Tubes just distort things in a very pleasant way." Note that loads of negative feedback reduces higher order harmonics more than lower – so undoes the nice mix. And it's the balance that is aimed for – *just like in making a violin*.

In the magazine L'Audiophile (Paris 1976-1992?) edited by Jean Hiraga, circuits were always discussed with a reference to which harmonics were produced. The basic idea was that, in order to reproduce harmonics to faithfully the sound of musical instruments, a slow decay of harmonics (2nd, 3rd, ..) is important. Having some handsome low order harmonics is better than no harmonics at all (that is, hidden in the noise floor) or an increased level of higher harmonics (4th, 5th, 6th, even 9th). Lots of feedback was seen as bad as it disturbs the gentle flow, even local negative feedback – cathode or emitter resistance - was thought to give rise to higher harmonics when the loop was applied. And it was common knowledge that a large amount of 2nd harmonics (even 1% to 3%) such as from a single-ended amplifier gives a warm and full sound, pleasing to the ear.

Negative feedback reduces the harmonics – but not all in a similar way. While open loop there may be a nice distortion, with predominant 2nd like with a triode, after closing it, the combination differs. Apply some local feedback, such as cathode decoupling? Enter 3rd or higher harmonics! Too much feedback kills the sound of an amplifier. It may measure well but not sound good. A good amplifier is art.

In L'Audiophile many a circuit has been discussed with attention to these harmonics. The positive effect of having a 6J7/6SJ7 pentode driver (that gives some 3rd harmonics) to mask or smoothen the larger amount of 2nd harmonics of the single ended 300B output in the WE-91 circuits may serve as an example. A regular decay is found more musical than a fast rise time and wide bandwidth. A second harmonic of -43dB (just under 1%) is not felt bad. Similarly, the precise positioning of a working point of e.g. a 300B by changing the load resistance was discussed in relation to the mix of 2nd and 3rd H.



Jean Hiraga also noted the effect of cascoding a FET and a transistor in the input of his Le Monstre amplifier of December 1983. This 'monster' gave 8 watts. [It conceptually was a simple 4-transistor circuit - but the input had a cascode and the output was a Sziklai pair, so the practical count was 8 transistors.] His important find was that the cascode at the input would give a very nice harmonic degradation. The 2SK170 jFET loaded with a cascode at about 5-6 volts Vd gave that wanted decay of 2nd, 3rd while a bare single jFET did not!

My build of the Le Monstre, which really was a monster with 25 Kg of power supply arranged in CLC and 12 large 56.000 µF to 68.000 µF cans, was heard in a show in NL in 1986 or so and highly acclaimed.

- By accident – I had different chokes – so my power rails differed somewhat.
- As well the pair of transistors had by intent slightly different mu (the selected pairs were bought in the Maison de L'Audiophile). So when trying to null the output automatically the amplification per half of the symmetrical amplifier will shift.

It gave rise to small differences in the amplification of the two halves of the symmetric amplifier and these probably were the cause of some 'extra' 2nd harmonics, negative phase. It was clearly seen on my analogue distortion meter. Anyway, my output was about 1% negative second at 2 watts. A great sounding relaxed amplifier.

Some ten years ago I also built the Nelson Pass F5 amplifier (structurally equivalent to the Le Monstre). I ended up with a configuration without source resistors (I had the best Mosfets 2SK175/ 2SJ55 I could get, matched even) but . . . it was less involving than with a 0,5Ω source resistor. Then Peter van Willenswaard suggested to totally decouple one half of the symmetrical amplifier, *so only one half does the job*, the other one is there for DC, in fact only being a CCS – as giving some sonic benefits. Also, Nelson Pass often designs output stages with a CCS, even at 1-2A. So that confirms the theorem of un-equal amplification to the extreme.
We see this configuring of input, choice of components, and power lines have an influence on the harmonic structure.

Another find to generate harmonics.
Mid-nineties I made a line stage built around a 6J6 input and E90CC output, with a Vb of + and – 150V. Both stages were Long Tail Pair. The coupling between the two was with a resistor string, no capacitor, and this gave a reduction that was almost equal to the gain of the first stage . . . And being so, it was DC coupled to the output. I had to bias the DC operating point of the output to come out at about 0,0 Vdc. Doing the balancing , I connected the output to my analogue distortion meter.

- Lo and behold, as I shifted the DC I saw that the second harmonic shifted in amount and phase. I choose the negative phase (by luck I must admit) and that distortion was about 1%. It sounded very good and warm.

That idea of a sliding edge of DC balance that influences the harmonic structure was

Fig. 12 : Effet bénéfique de la contre-réaction : le point de repos du di tiel passe par A et B qui est plus près du point d'inversion où la distorsi minimale.

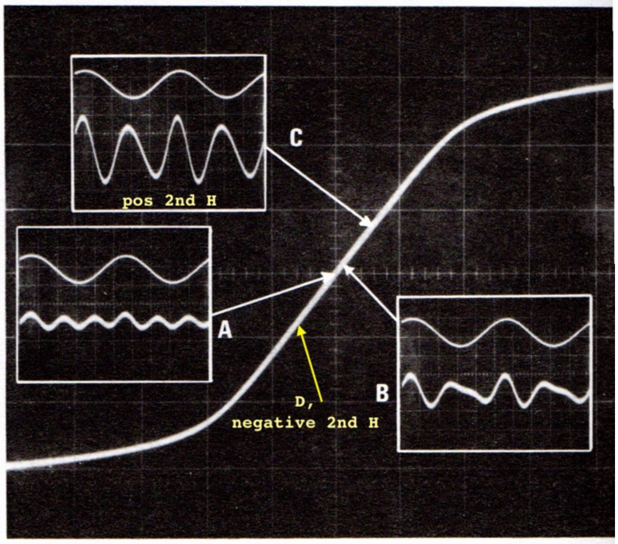


Fig. 13 : L'influence de l'offset sur la distorsion : elle est impressionnan A au point d'inversion, la distorsion est de 0,09 % et composée d'harm 3. Un peu plus loin en B, elle est de 0,18 % et composée d'harmoniques En C, elle atteint 0,44 % et est composée d'harmonique 2. On compre la polarisation est capitale pour les performances de distorsion et que, varie sous l'influence de la température, de la distorsion thermique, d mentation ou de la contre-réaction, l'impression subjective soit abom

exhibited too by Héphaïstos in L'Audiophile #2/NS dec '88 (see picture "fig 13") as part of a series of configuring feedback in differential (opamp) input stages. At the midpoint there is a little third harmonic, and he showed that on that slope of DC just a tad too much then second harmonics will arise. .
Héphaïstos thought that any operational shift would have a bad influence on the signal (sound): it is 'abominable' if polarisation would shift with changes in component temperature, in the power supply, or the DC induced by the feedback, leading to changes in the sound structure. He fought that by being concerned with an aleatory and haphazard shift in the DC due to for instance temperature or changing resistances in the gated lead (such as what arises when a Fet-input with its gate current is connected to a volume pot).

My own finding in my 6J6/E90CC preamp was that the second harmonic above and below the midpoint would have an opposite polarity - that the precise point the differential pair was set could flip around the phase of the second harmonic! As the opposite of point C imagine D on the other side of the midpoint A. So here we have another trick to create second harmonics and change their phase. Take a differential and DC-bias the two inputs differently. Or even: in a differential, load the two halves differently, the feedback one often is without resistance just a high V_{anode}, V_{drain} or $V_{collector}$, and as such they give a different amplification. *I added point 'D' to signify where the phase of the 2nd H would be negative.*

This concept that the im-balance in a differential pair will lead to the appearance of second harmonics is also mentioned by Jean Hiraga in L'Audiophile nr 2/NS, dec 1988 in the article about the hybrid Pacific amplifier. He states that it is obvious and common practice that a paired set in the differential is selected – but that a dissymmetry of a few volts in the output of the LTP of a few volts will give rise to the appearance of second harmonics; he suggests that the bias point of the cascode can be changed just a little bit to get a symmetry again with a given set of FETs/BJTs with their given Idss and Hfe – to reduce the harmonic level. But - as I interpret - this can also be leveraged by introducing dissymmetry to obtain a wanted mix. And it is Nelson Pass who did exactly that, if I understand correctly.

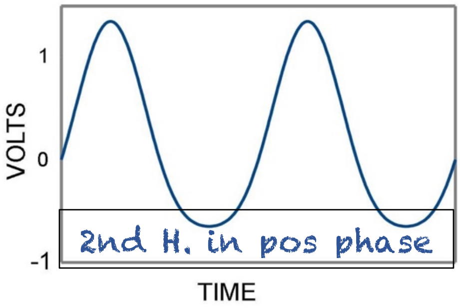
So we have another methods to influence the harmonic structure: a cascode of FET/BJT; a-symmetric power lines; + deliberate different amplification in two amplifying halves.

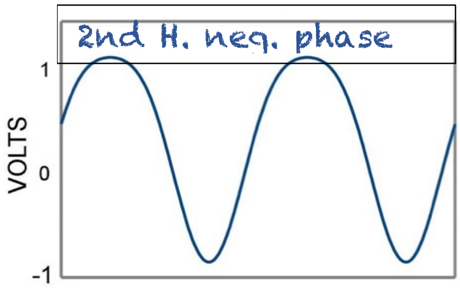
Recent advances
You'll understand I'm a Jean Hiraga / L'Audiophile fan. Fast forward from the eighties to 2010. Nelson Pass showed he could get very nice harmonic structures too from very simple circuits. He did with power mosfets what Hiraga did with the jFETS: load them such that the output contains a high amount of second harmonic. Nelson was the new Jean for me. Nelson did something else too: **he looked at the phase** of the harmonics. He asserted that a negative second harmonic was perceived as more pleasing than a positive second harmonic of the same amount. Just by flipping the polarity of the speakers this could also be heard. Or by using an output transformer to flip the phase.

In 2022 Nelson Pass came up with an adaption of his F5 amplifier (to me is a modernized *Le Monstre* circuit now with really 4 transistors) where he decoupled the source resistors of one of the output mosfets to get an effect of a disbalance of amplification in the two symmetric halves of a power amplifier, and that gave a smooth amount of positive 2nd harmonics.

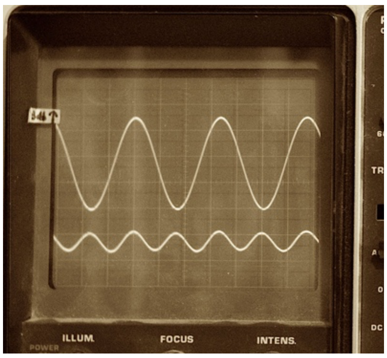
Now what is this phasy thing?

Why all the fuss about phase? Isn't just a percentage enough? Well, because the top of the second harmonic can be *in phase* – it adds to the top of the fundamental, elongating it and that is shown very *exaggerated* in this drawing – or *out of phase* – the top of the second subtracts from the top of the fundamental, blunting it.



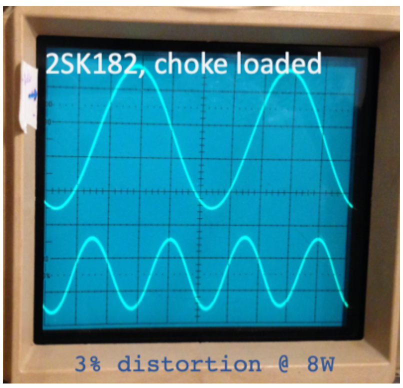


So a 2nd H in phase will rise faster and will give some brightness to the sound. It projects. This conceptual drawing by Nelson shows the effect is very important. But the oscilloscope photos of the 1613 and 2SK182 shows what the bare eye won't see. Only when it is so severe the amplifier is near clipping.



The relationship of fundamental and second is seen in this picture of a 1613 pentode, triode strapped, with a low 4 KΩ anode load intended as a preamp and > 200Vb. Ample 3% second, positive phase.

Imagine. Flipping it, such as with a 1:1 output transformer, it will get the wanted negative phase.



Here a 2SK182 V-FET is shown, running at 25V, 1,5A with a choke load on the drain. The 2nd harmonics phase is negative!

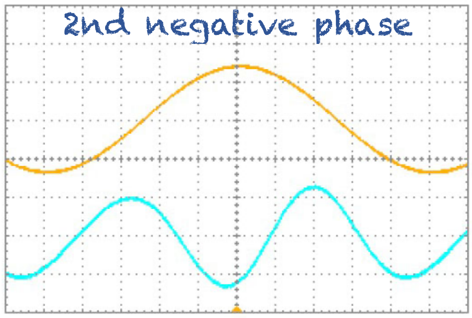
This is a pure circuit: one active component, the 2SK182 having a load consisting of a 120 mH choke and connected to the 8Ω of the loudspeaker on the drain with a capacitor. Result: warm sound. Like a big triode that is under-hung in voltage.

Again, just looking at the scope will not show it.

You need an analogue distortion meter.

How to explain that? I came up with the following. It is all about natural sound and speech, as Peter van Willenswaard tried to convince me in the eighties, when we were talking about the difference between the ESL57 and a conventional speaker.

Now I think so too: the positive second that adds to the crest will sound like shouting, it projects, like an ambulance siren coming to you, like a horn with a break-up; while a negative phase of the second harmonic is soothing, it slows the envelope of the sound, is muted and comforting, with its Doppler effect of a siren moving away.



The first I perceive with some anxiety and I call it audio nervosa. OK, just some words, but I hope the impression is clear.

Can we create a second harmonic out of nothing . . ?

Now the big question is: how to get that predominant second harmonic in a negative phase? Just good selection of components? Sure. Take a triode. . .

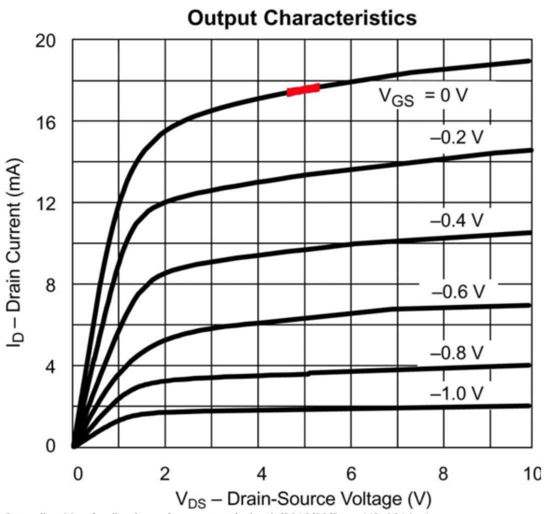
There is also something in the circuits.

The input cascode used by Hiraga in Le Monstre in 1983 and the low Vd loading of a jFET and mosfet by Pass are nice methods. A lower operating point of a jFET is similar to what Hiraga did with his cascode: ensure the load line is low and on the edge of the FET-curves (pinch it) to grab the 2nd. (Everyone in their right mind will do it differently but that is what these geniuses are for . . :-).

In his harmonic generator (2018) Pass loaded the jFET just where there was a curved slope of the Vg=0V loadline. Somewhere around there, he found there is a place that will give a 2nd H. So he loaded the stage right on that spot, red in the diagram above.

Just by chance, Hiraga had 30 years earlier placed the emitter of his cascode of the input of Le Monstre about there too at 5.3V (with a specified V+ of 12V).

I expect that a pentode can be loaded at a pinched-off point to give the same effect too.



(https://positive-feedback.com/wp-content/uploads/2018/09/Pass_H2_001.jpg)

Having a nice harmonic spectrum is my goal to attain. It is not simple, because a harmonic spectrum is not static. It changes with the volume, and it even influences the perceived loudness.

In simulating the output of a triode for instance, halving the output in one example reduces the distortion from -67dB/-114dB at 1Vpp to -72dB/-145dB at 500mVpp.

A rule of thumb I read in a 1930^{ies} triode documentation that: "Decreasing the output by N dB below the value given in the table improves the harmonic level by N dB." About what my simulation shows. And studying the 1930ies table of output, I see: doubling the anode load reduced the distortion with 5 dB. I love rules of thumb.

So maybe the best place to 'generate a nice harmonic" is before the volume pot, not after it. Then it will be more stable across listening levels. Specifically, when you have a tapped inductor potmeter, that is what you do. Maybe having a low impedance potmeter (5 kΩ?) after an amplifying stage would have the same effect of a more consistent soundstage.

<u>So far . . . history, sort of.</u>
Now let's look at some circuits and the thinking behind them. Like Jean Hiraga once wrote: 'It's not the circuit but the story behind it that is interesting'.
We have discussed the idea that a certain amount of harmonics can be perceived as nice, and that some designers have even taken a lot of effort to leverage a circuit to by intent give harmonics – as long as they have it in their hand, & are in control.

Looking in to Positive Feedback (PFB) in a (pre-)amplifier

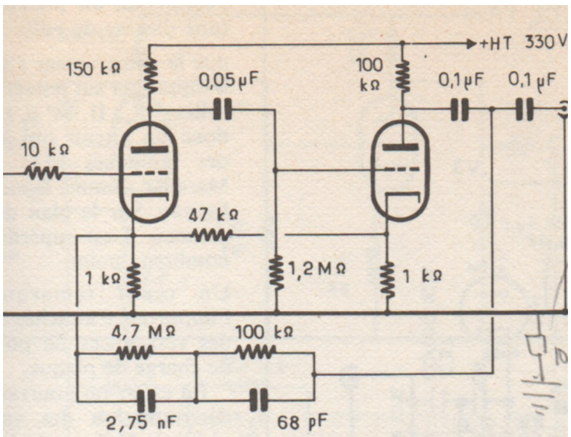
In 2022 I set off to make a new active phono stage (after having had several passive RIAA stages for twenty years) because I wanted to hear an active RIAA once again and my choice fell on the EAR834 from Tim de Paravicini, acclaimed in LencoHeaven. Main reason: his very short feedback loop of just one tube.

1: TdP used standard cathode decoupling in the stage where the RIAA is applied. As a standard practice, a capacitor is used to extinguish and nullify the signal there. But a capacitor is not very effective at low frequencies, so the amplifications starts to drop there – the headroom becomes lower. As well an electrolytic often gives the deadly smearing of sound. It has a memory effect; or the rising edge is influenced.

2: Jean Hiraga had a little trick that he incorporated in his 1981 ECC83 phono stage: positive feedback from the cathode follower to decouple and nullify the signal on the second cathode. Goal: To make it more silent than with an electrolytic smoothing capacitor & enhance dynamic. I wanted to use Jean Hiraga's design of his ECC83 phono stage that incorporated a PFB-resistor.

3: Looking at this PFB applied by Hiraga to the cathode I thought that it must have an influence on the harmonic structure. All at once there is a small residual signal and its phase that is also amplified.

Hiraga promised extra dynamics with his PFB set-up... What is the idea behind Hiraga's use of PFB?
He gave this example. The Dynaco PAS X phono stage has <u>positive</u> feedback to the first stage <u>to give a boost</u>: the open loop amplification is increased. We see a 47KΩ between the two cathodes both being 1kΩ. The second cathode has an opposite phase compared to the first one. The resistor that connects both of the cathodes of the two stages gives a PFB effect of 5 dB more open



In the Dynaco PAS X phono stage the 47kΩ gives positive feedback of 5 dB open loop. That helps the RIAA correction.

loop, enough to give the whole phono stage the required headroom for RIAA negative feedback.
Note it here the same node is used for PFB and the NFB with the RIAA. The PFB is claimed to be aiding in giving a lively impression. It is more than just an increase of the open loop amplification (necessary for the full RIAA gain range). Such a PFB between cathodes was also used in other renowned phono amplifiers.

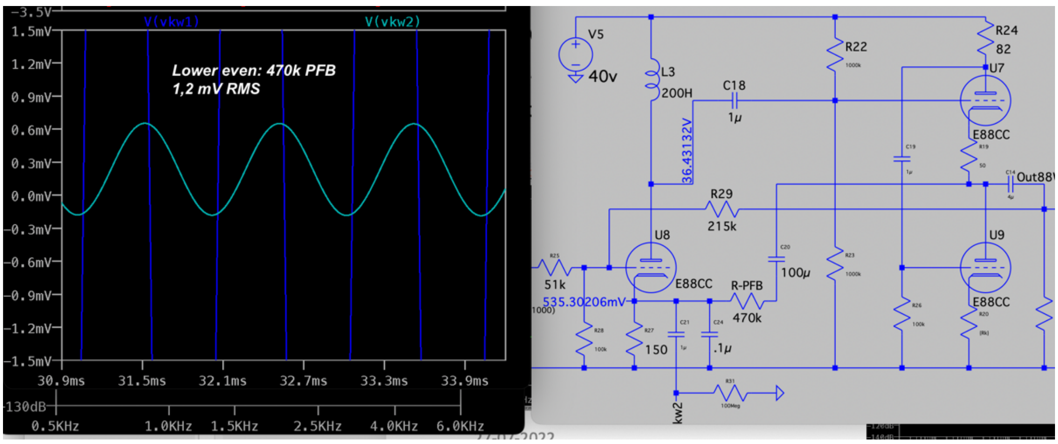
In the fifties and sixties such multiple positive and negative loops were common. Take a look at the Telefunken V72 microphone amplifier. It also has positive feedback between cathodes just like the Dynaco Pas X – and as well has some extra loops such as between the two anodes only for the HF - and a general NFB. It is acclaimed as one of the best.
I had a Krohn-Hite wide-band laboratory amplifier that had more than ten loops some PFB including cathode or anode loading enhancements, all to ensure the power phase would have a uniform shift up till 500 KHz at 100V out in 200 Ω. The result: a very pleasant sound.

Based on these known circuits and this thinking about PFB I started to simulate the concepts how to apply this PFB in a simple line stage, before going to a more complex phono stage.

Simulating PFB in a line stage

For showing the nulling that arrives from active decoupling.
In my initial simulation, I studied how a low Vb of 40 V could still be used to get a high output. I wanted to apply PFB and NFB. And I wanted to use tubes that still are expected to work at low Va like the ECC88. So that for my goal and constraints.

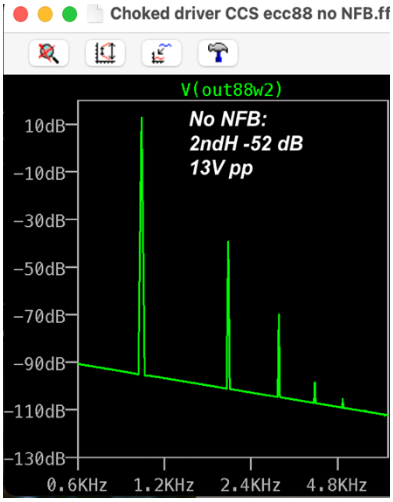
In this schema I studied positive feedback to nullify the cathode of the input tube, a ECC88. I used a choke load to effectively double the Vb – because I wanted to make a driver for a follower output, and use the same power rail. So, I needed an output that would be able to handle 25Vpp without clipping from Vb=40V. Therefore, I resorted in this design to the White cathode follower that stays linear over a wide range.



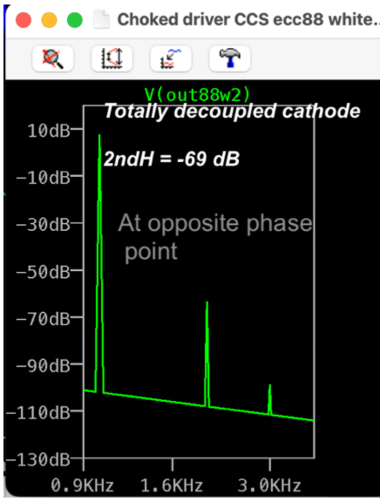
The blue line is the non-decoupled cathode = 30 mV pp; while the aqua line is the voltage on the cathode with PFB to decouple the cathode it gives 0,6 mV pp – it is almost nulled and <u>less</u> than can be reached with a reasonable decoupling cap. It shows: it works. Note you can see a slight phase shift of the signal.

Applying positive feedback has an effect on the distortion

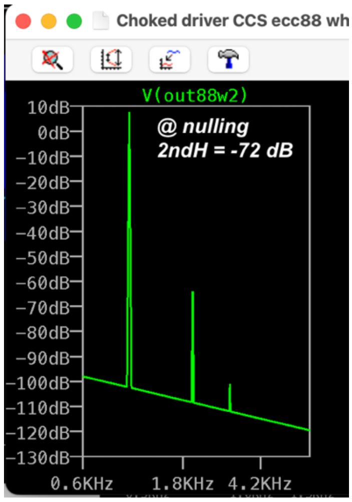
In this same circuit of an ECC88 input and a White cathode follower, I checked the effect of PFB on distortion by gradually lowering R-PFB from 470K.



1) non-decoupled cathode 150Ω



2) cathode decoupling with 100 µF



3) and as last exhibit, nulling the cathode with PFB, no capacitor

We see in the diagram 3) that exact nulling gives -3dB less generated distortion compared to the standard decoupled stage of 2) with a capacitor.

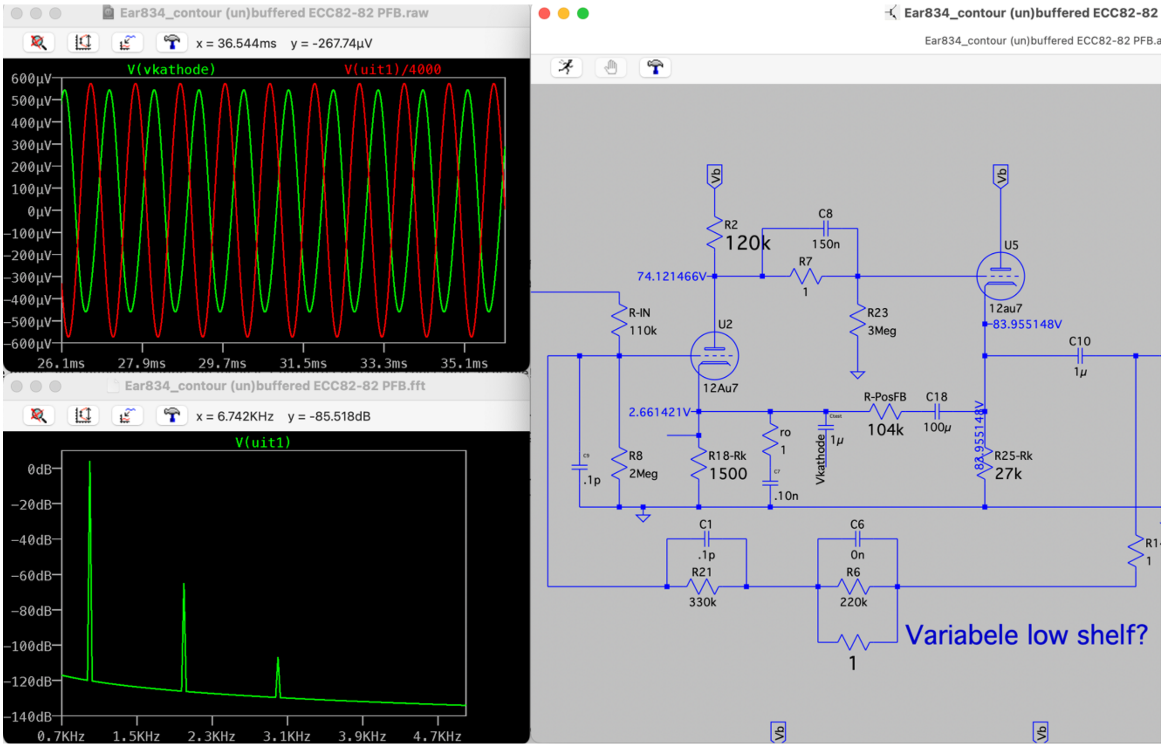
Going beyond nulling

I next simulated the concept of PFB in a simple ECC82 circuit with a cathode follower and a ‘normal’ Vb of 260V. So positive feedback across one tube only.

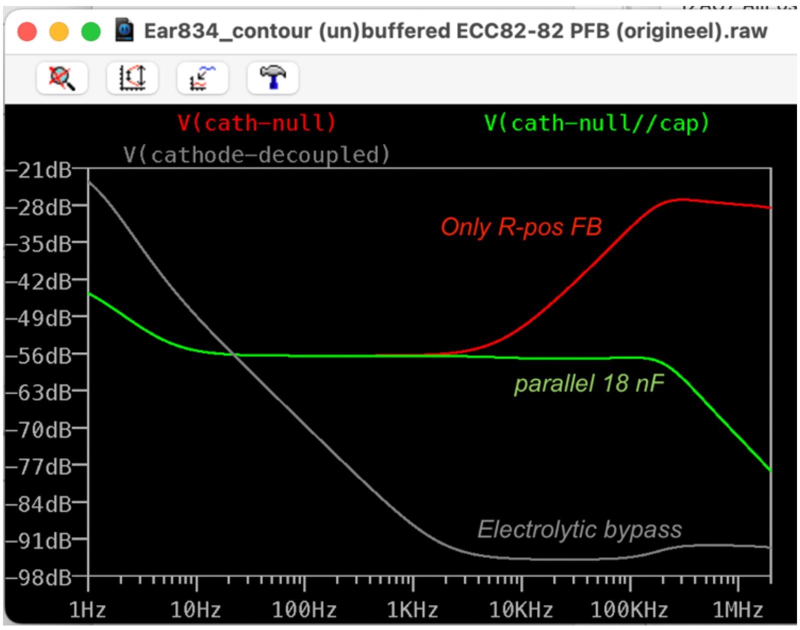
Applying the idea of PFB in a line stage is possible in a stage with a single ECC82 triode input stage and an ECC82 output cathode follower so we have a single stage negative feedback loop consisting of going from the inverting anode/cathode follower → input feedback, with feedback to the input of 100K/330K. Of course for the output we don’t really need the cathode follower, but we do for the positive feedback to the cathode of the driver.

I started with local PFB initially to null the cathode. See the first pictures below.
But one can go further. More positive feedback to give a boost, **real positive feedback**.

The ECC82 line stage with the positive feedback is shown in following picture. The circuit first with nulling (**Rpfb = 104K**) where the Rpfb is used to get the smallest possible signal on the cathode. This is the point of **nulling**. The small residue is still in phase with the input. The distortion spectrum is shown.



Nulling. Green in the top is the cathode, nulled to 1.100 µVpp. Red is the output divided by a factor to get the two in the picture (distance of Vk and Va is 66 dB or 4000x). We see the residual 1 mV is now in opposite phase - *cancelling*. In his article, Hiraga says that just changing Rk with a few ohms will make a difference in the exact nulling - a trimmer is needed!



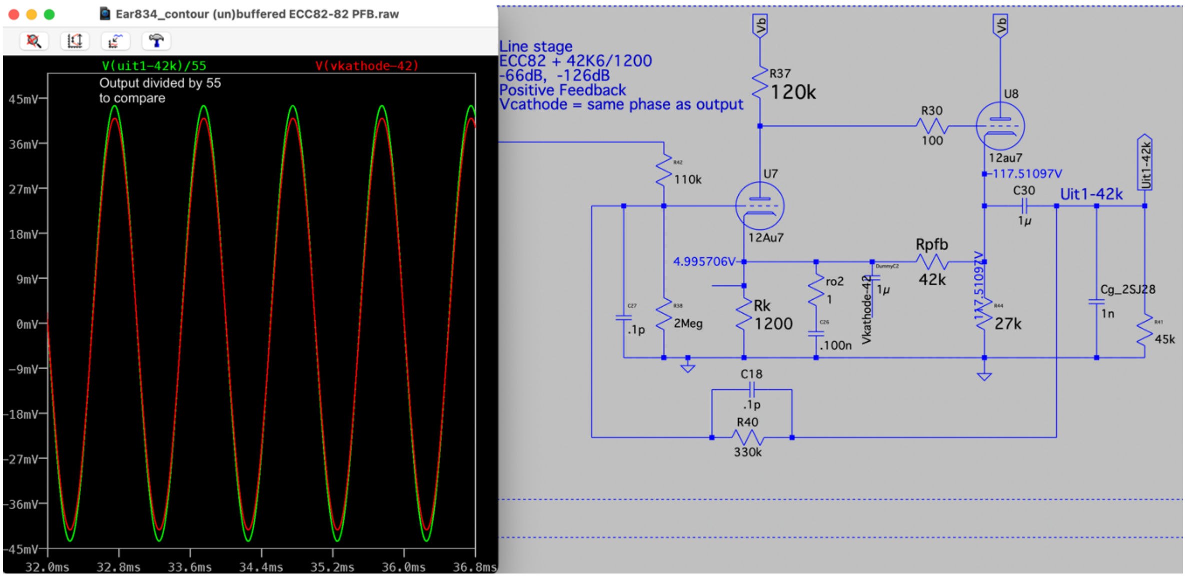
The red line shows the effect of the nulling using PFB in the ECC82line stage: 1 Hz at -45 dB! But at higher frequencies, phase shifts lead to less PFB. That can be compensated. Bottom line (green) is with extra a parallel 18nF.
Gray: Without PFB decoupling with a bypass electrolytic works but is frequency dependent – totally depending on the brand and structure (i.e. series inductance, ESR) of that cap. A decoupling cap of 100µF starts at 1 Hz with just -22dB and will drop rather low.

The voltage on the cathode of the amplifying triode, at the point of nulling, without and with // cap.

Well, if you like that gray line, why not use a negative bias? The signal on the cathode is then infinitely low (depends on the layout, that is). Someone at LencoHeaven did that with his EAR834, but here you would need an input capacitor.

Going beyond nulling.
We discussed the nulling. Now to get a positive FB, I came out at (**Rpfb = 42KΩ**) for a position where the feedback signal is in opposite phase to the input signal. This, open loop, gives a boost of 2 dB. Closed it is just 0,5 dB in this ECC82 circuit.
The result: added dynamic, specially so for an ECC82 which in my experience tends to be a bit 'sleepy', while it has a square wave with some overshoot - something the ECC82 is known for in my experience with only NFB. Just a tad of PFB will make the ECC82 more dynamic and livelier. And no tendency at all for instability.

The following simulation shows the extra amount of positive feedback with Rpfb=42K; this results in a signal on the cathode that is <u>totally out of phase</u> with the input, and in phase with the output. So, it **boosts** the output. That results in a more dynamic sound stage, extended differentiation of groups of sounds. There is no overshoot or instability visible on the scope, just a little benign effect on the sound. Added dynamics.



In the panel, I divided Vout by 55, to be able to compare and now it shows it is really opposite phase, really 180 degrees, with what should have been the AC of the input signal on the cathode just nulling. Just ponder that for some time. OK, it boggles me.

The now positive feedback effect results in a weak boost of 2 dB open loop (because it is an ECC82 with a µ of 20 – with a ECC83 it would be higher).
Well, the amount of 2 dB was there because it is at the point where the PFB gives the same charge on the cathode but in <u>opposite phase</u>. To explain: a 1 volt signal input will give 30mV pp sine in phase with the input on the cathode; the PFB can be 'dialled in' to nullify it (you can reach 100 µVpp easily) or you can go 'beyond nulling' and the signal on the cathode becomes in opposite phase of the input, in fact boosting it. Now slowly, as you increase the PFB, the phase changes, and I wanted a correct phase relation. So, I aimed for a 180° inverted signal, of same size, which also came out as 30mVpp.

Sound. The line stage sounds good. Better than my previous attempts to use the ECC82 or 6SN7 in a line stage. Great depth and fine details. ☺ Particularly recordings made by the students of sonology, who can trick you into believing that there is a sound somewhere, are impressive. That complex vocals sometimes become mish-mash is of course due to my full range. 😐

But then how about the second harmonics?
This is an important theme here. Well, I have no idea. Smart math minds are needed and a modern digital distortion meter. I have neither ☹ . But one thing is sort of clear to me: the PFB does reduce the distortion (probably: yes there is slightly more open loop, and thus more reduction overall, look at the Dynaco PAS) and it does sound dynamic but there is something else: while a larger amount of negative feedback tends to reduce the lower harmonics more than the higher ones, I observed in simulations that when some PFB is thrown in the mix *this is less so*.

- My intuition says that with a small phase shift, the harmonics must change. I now aimed for exact 180° shift.

I started out with awareness of the phase of the second harmonic. But I have no tangible evidence how it worked out. My choice for a positive feedback that gave a 180° shift with the input (pure boost); going beyond that the boost will be bigger, but the phase relationship will change. I could not measure it with my analogue distortion meter (going to 0,1%), and I should measure without NFB some time.

<u>Listening experience of PFB in the line stage:</u>
At the 2022 ETF this line stage was on demo, with a three-way knob; 1) cathode decoupled with 100µF, 2) a position with the PFB active, 3) with an active PFB and a decoupling with 100 nF (that gives a drop above 100 kHz). The panel agreed that the middle position sounded best. “Now this is music.”
Why is *dropping* above 100kHz not better? NO! Not what listeners heard. PFB without the 100nF sounds better. Could it be that positive feedback provides some sonic benefits and also results also in a loss of amplification of about 0,5 dB above 50kHz. That might be heard.

Later tweaks:

- as output I used 1µF tin foil capacitors at the ETF.
- I subsequently exchanged the output capacitor to a Bell Labs 1 µF capacitor. I got a more realistic stage. The lower mids improved. Once I had 1 µF Western Electric coupling caps – which I was so stupid to sell on ZeBay. Things like that happen. Oh well.

I also did some tube rolling. Inserting a 12BH7 as cathode follower darkened the sound. After some time, I resorted to the plain ECC82 I had there again because I liked the clarity it gives in this circuit. But I will look into it deeper.

So once we understand how this works in a line stage, we can go have a look at applying PFB in the more complex phono stage.

Applying PFB in the EAR834 phonostage

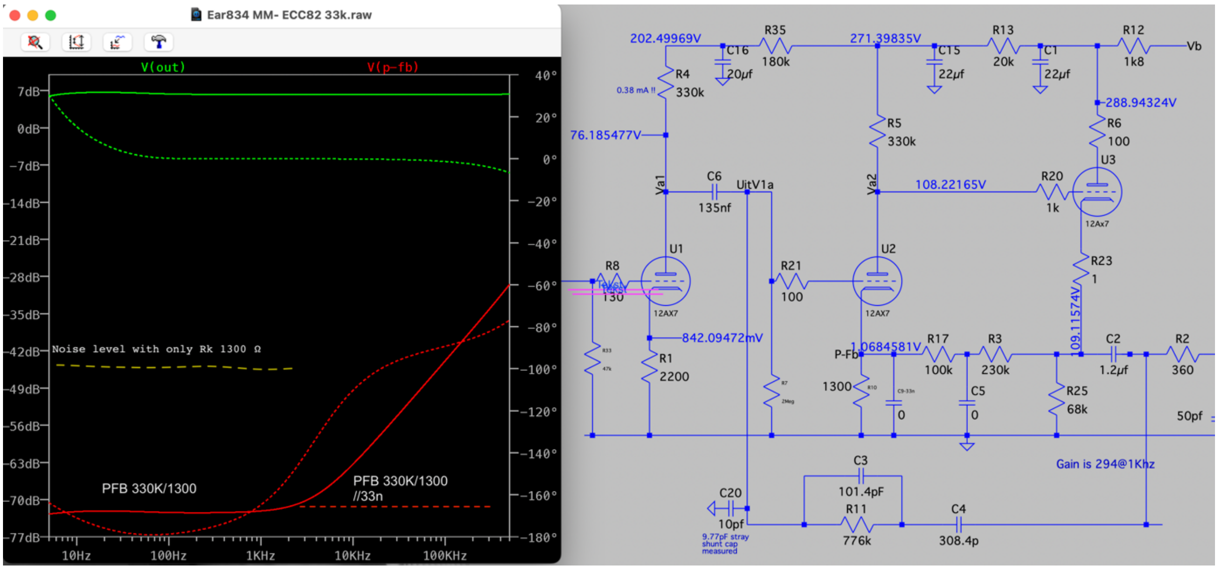
<u>Modding an EAR834 phono stage</u>
After having studied the effect of PFB in a single-triode line circuit, I went ahead to see how it would have to be configured in a single triode phono stage.

The EAR834 is a phono circuit that has an input ECC83 with a high anode load and local feedback due to a non-decoupled cathode (thus, the internal resistance is very high), and a second stage which carries the RIAA feedback loop – so we see single-stage feedback for RIAA. The first tube is not part of the RIAA feedback. That is very special.
What makes the EAR834 fall in a separate class?

> Normally the RIAA feedback is over two stages. Tim the Paravicini did it over one tube only. Thus, several negative effects of NFB (phase delay, overshoot, rise-time distortion) do not occur, the phase difference is too small.
> It works as follows. The output resistance of the first ECC83 is the passive resistance which is seen by the feedback loop. Effectively the feedback is from the anode to the grid. Because the loop is so small the feedback is not seeing much time-lag, no memory effects. No nasty phase differences and no overshoot from coupling capacitors. A very clear sound of the pick-up is the result. There have been several other designs with the active RIAA over one tube. But this is special.

In his standard circuit TdP decouples the second stage cathode with a 100µF electrolytic capacitor.

- That can be changed. Based on Jean Hiraga's use of PFB to evidently nullify the second stage cathode of his phono I decided to do that too. Experimentation is needed so the right values can be derived. Hiraga must have resorted to hours of laboratory change and test. I passed many hours of simulation ...
- That is, from the onset I guessed Hiraga went for nulling, not boost or enhancement.... while the Dynaco PAS-X aims for a boost.



The EAR834 phono stage. Red line: the cathode of the phono stage with PFB applied. The cathode is nulled >-80dB ref output signal. And where it goes up at about 5 kHz, a small cap of here 33nF could be added // Rk to keep that decoupling straight and -80 dB all along (red dotted line). See: ruler flat decoupling. Green is the output. The yellow dotted line is a not-decoupled cathode, only to show the reduction of some 30dB of the signal level (*erroneously the picture says* 'noise level'...)

In the end I came out at about similar values that Hiraga used for the PFB – I have the series of R3 and R17 with 330K/1300Ω, Hiraga used 300K/820Ω. And roughly that value gives nulling.

- I left the **EAR834 phono at the point of nulling**.

As you can see, the nulling of the cathode works until sub-sonic frequencies. A coupling cap does not decouple that good at all. [A very good alternative is omitting the cathode resistor altogether and using a negative bias (a Varta V370 lithium cell of 1.55V can be used and will stay in service >10 years). This is possible because there is no DC path across the FB loop.]

<u>The simulation bug</u>
Yes, it got me, the bug. Simulation is a nice way to see what-if.
Once you also start simulating you start believing what the outcome is.

- So, I measured the values of the core components and adapted the values of other components of the feedback until I got a ruler-flat output. Of course it is make-believe, because the output of simulation is only as good as the input: the models of the components.

Doing things by hand is even more interesting. By doing that in sometime around 1986 I made my first MC/phono with an input 2SK240 with a cascode of ECC88, topped by a CCS with a ECC88. Just experimenting. Following intuition. Can be so satisfying. Another example: I made a nulling filter for the 44 kHz clock of a DAC output. It worked like a charm. It got the steps out of the signal but did not influence the step-response of NOS. It took me days to experiment the right values of the filter.

<u>Second amendment</u>
Another simple tweak of the EAR834. A normal RIAA feedback circuit is R//C + R//C. So there is always a DC path from output to input. In the EAR circuit the feedback circuit is R//C + C. The second parallel resistor is not there. Its function is taken up somewhere in the physics of the first ECC83! In the EAR834 feedback is taken from after the output coupling cap so any low frequency deterioration is compensated for – but <u>it can also be taken before the output capacitor</u>. So no extra signal capacitor in the way. All other feedback-RIAA circuits have a DC path so any DC will fatally disturb the input tube - while the EAR does not! I found out that moving the feedback to before the output cap only has a slight impact on the lows. But there is one C less in the feedback to introduce strange storage effects disturbing the closed loop.

<u>And finally . .</u>
Of course the power supply can be made active to the first stage or at least, configured such that there are no electrolytics in the PS. On Bartola's site there is a blog on the impedance of the cathode and the minimum required decoupling capacitors (ETF 2015, capacitative decoupling of resistance-loaded triode gain stages). Of course, we over-do our implementations. Machos. We can get away with smaller better decoupling caps.
I can add: Using the supplied calculus we can use a 1 µF power supply decoupling in the first stage (not 400 µF like I used//0,5 µF) and 5 µF in the cathode decoupling. And if there is a small one this can even be used to compensate for low roll-off.
I tried a simulation with a 0,5 mA CCS (LND150) in the second stage, but the results were not good. But. I should instead just solder it. That often is a better school.

Feedback from listening

My phono amplifier and line stage as well as my SIT amplifier were heard at the ETF Baarlo 2022.

In my line stage I made put a three-way switch to check the effect of the positive feedback. In situ.
The switch applies

1. a 100nF that stops it above a certain frequency, in fact there it reduces the amplification with 0,5 dB. [Good readers remember I mentioned 18 nF – well that was in a redone simulation in May 2023, not the first one of September 2022]
2. Giving the PFB, with just beyond nulling, and some 0,5 dB over the whole bandwidth, no overshoot or instability seen
3. a total decoupling (a 100µF destroying the PFB loop)

This setting was tested. At the ETF 2022 these were the results:
1) with 100nF the effect was liked second best.
2) just plain PFB. Liked best.
3) 100 uF. This kills all PFB and it becomes just standard decoupling. This was liked least.
- It shows the concept works in our ears.

The sessions were made with a 2SJ25 SIT source-follower amplifier – alu case in the picture below.

My line stage/phono preamplifier in the black 4U chassis left in front of the big bass horn. The SIT amplifier is in the front.

Feedback from Torben, *positive feedback that is*, who made the big horns in the back of the picture: "a totally magic sound moment playing with your amplifiers + the abbas dac listening

[to] some Norwegian Christmas songs.....that was really really good. The phono-stage was also really good……"

Electravolt (Charles Azzolina-Michlin): "This tussle of gear sounding quite beautiful." ('**EFT Baarlo 2022 Friday**'); used on Torben's large horns. And Charles knows something of horns. On the front the V-FET with 2SK28 running as source follower in a Mu-Stage by Nelson Pass with a folded driver stage.