My interest in electronics and audio stems from my love of music. When I was in my teenage years, I bought myself a tape recorder to record songs from the (pirate) radio stations, which broadcasted pop music all day long. But as I lived quite far away, the reception declined during the evening, so by recording the songs, I could have music in the evenings in my room as well. Not hindered by any knowledge of electronics, I put the microphone, which came with the tape recorder, in front of the loudspeaker of the radio and -fortunately for me- the automatic gain control made a recording possible. This went well for a while, but by the time I had also recorded the back door slammed tight by my brother twice, I thought it was time to consult an electronics expert in my school class. I knew that it was possible to record via a cable, without the use of a microphone, provided your radio had a connection for such a cable. Mine didn’t. So I asked my class mate whether he would be able to make such a connector on my radio. He said that he probably could, but it would be very helpful if I could provide a schematic diagram of the radio. When I bought it, the papers I got with it included also a large, folded sheet of paper with -to me at that moment- unintelligible scribbles on it and I suspected this to be the schematic he asked for. So I told him that I would bring this the next day. I dug it up that afternoon, had a look at it, found it completely incomprehensible and thought to myself: "He is never going to figure out what this all means...."

At the school yard, my class mate unfolded the sheet with the scribbles, had a quick look at it and said "Oh, its a simple device. No problem to make a cable connection to it for recording." That was my first frustration in electronics.... So he wrote down a shopping list for me to get at the local electronics store and we made an appointment that I would come with the components, the radio and the tape recorder to his place to get it done. When he opened up the radio, I was astounded as the schematic diagram had looked very complicated to me, but to see the interior of the radio was another order of magnitude more complex. But my class mate got a pair of test-pens and said: "When I am right, we can pick it up here." And it was right, so I had in record time my second frustration in electronics. He soldered some connections , made a hole for the chassis plug to be mounted and in no time, it worked. I thought.

I took the stuff back home and started to use the cable for recording. Indeed, I did have no more trouble with the slamming of the back door, but I encountered other problems. The signal was rather strong, which could no longer be handled by the automatic gain control and it also seemed to lack low frequencies. The first problem I could handle by changing to manual gain setting, but I was not able to find a solution for the lack of low frequencies. So I discussed this with my class mate and he said that he knew the cause: he had noticed a (too) strong signal for the input of the tape recorder and therefore he had put in a capacitor, which would also block any DC left at the point he had taken of the audio signal. But because of the low input impedance on the tape recorder, the result was a suppression of the lower frequencies. Thinking to be clever, I suggested to use a resistor instead as -I thought- a resistor would suppress all frequencies in the same way. He agreed and brought me the next day several resistors, explained the colour code, so I could determine which resistor had which value (of course, I had no meter at home!). Back home, I got myself a piece of electric wire and without soldering (I didn’t have a soldering iron either), I tried to put a resistor between the radio and the tape recorder. I tried several values and the larger ones seem to do a good job, the signal got weaker so that the automatic gain control did work again correctly and the low frequencies reappeared, except that there was a strong hum in the signal. How come?

"Did you use shielded cable?" my class mate asked. "Did use what?" "Shielded cable, of course". "Can you give me some subtitles?" So he explained to me what shielded cable is and why we should use it to avoid hum overwhelming our weak signals because of induction from the mains. So I tried to shield the resistor with aluminum foil and as long as you didn’t touch it, it worked, more or less. Taking all the stuff to my class mate to get everything soldered resulted in a much more reliable device. He also replaced the capacitor with a larger one and I could record directly from the radio without a microphone. The end result was a higher quality of the recordings and no more interference from noises from outside. I had taken my first steps in audio electronics under the guidance of my class mate.

P.S. - 1. We still have contact, my class mate is now professor renewable energy sources at the Eindhoven University of Technology and manager R&D renewable energy at Shell. I am proud that he was one of my paranimphs at the defense of my Ph.D. thesis.

P.S. - 2. A couple of years later, I had another look at the schematic diagram of my first radio and my class mate was right: it was a simple device.

P.S. - 3. Later on, I equipped several radios myself with a connector to record by cable...

The story begins in 1978. I used electrostatic loudspeakers for the mid-range and high frequencies in combination with a bass-system, electronic cross-over filters and bi-amping. The major source of high-quality sound came from my record player, equipped with a magneto-dynamic cartridge and a RIAA correction amplifier, which I had designed myself and had a response within ± 0.25 dB between 20 and 20 000 Hz. For those days, the design was revolutionary and it was published in a Dutch electronics magazine. A colleague of mine also used electrostatic loudspeakers and -as we all know- the step-up transformer is a PITA (= Pain In The Ass), so we got the idea to use feedback over the transformer to reduce its misery. After all, tube amplifiers also apply feedback over the output transformer, don't they?

So one night, we took a signal generator, an oscilloscope and measuring cables to my colleague's house and started some measurements across the transformer. Only to find out very quickly that the inductance of the secondary winding of the transformer created a nice (suction) resonance with the capacitance of the loudspeaker and thus a 180^o phase shift at something like 19 kHz, which would result in an oscillator when feedback would be applied. So that is the reason why nobody applies feedback over the step-up transformer.....

As we had lots of time left (the evening was still young), my colleague asked whether we could have a look at his pick-up cartridge, as he was pretty sure that something was wrong. He had a test-record with calibrated signals, so we started to have a look at the response of his cartridge, which, indeed, showed to be flawed and started to drop-off around 8 kHz. I asked whether he had the specifications of his cartridge, which, of course, he had and those told me that the cartridge was supposed to cover the range from 20 - 20 000 Hz within ± 2 dB. But it also told that the cartridge had to be loaded with 250 pF and 47 kΩ, which seemed rather odd, regarding the 947 mH inductance of the coil: a back-of-the-envelope calculation (portable computers were just arriving in those days!) learned that such a load would indeed give rise to an 8 kHz cut-off frequency, in line with what we saw. As we did not have the opportunity to contact the manufacturer, we decided to note down the results in detail, do the actual low-pass characteristic calculation the following day and I would look up the specifications of my cartridge.

The actual low-pass characteristic as calculated using the specified components were in line with the back-of-the-envelope estimate from the night before: above 8 kHz, the response declined with 12 dB/oct. So there was a big question: how come that the manufacturer specified the cartridge for up to 20 kHz whereas the electrical part already started to decline above 8 kHz? My own cartridge was a little less dramatic (its decline started above 12 kHz, using the coil inductance and the specified load), but still far below the also specified 20 kHz. But my cartridge was manufactured by Philips and as this is a Dutch company, I decided to give them a call.....

I got in contact with a commercial guy who didn't have a clue of what I was talking about even though I tried to explain it for almost half an hour. So, a bit desperate, I asked him whether he could put me through to somebody from the laboratory, who would have more detailed knowledge. At first he objected, arguing that there were new developments going on, but I replied that I was not asking about new developments, but about an existing product for which I had paid good money. He didn't have a good reply to this, so he promised that I would be called back either by him or by somebody from the laboratory. I thought to myself that if he would call back, I would not become any wiser, but to my surprise, an hour and a half later, I got a call from mr. Van Wijk from the Philips laboratory. I explained to him my findings and he said: "You are correct, but you are overlooking something". To which I replied: "I feared something like that, but I would like to know what." "Well", he said, " the magnet on the stylus resonates around 18 - 19 kHz, which increases the response of the cartridge and by introducing a roll-off in the electrical part, we compensate for that." "Yak, how dirty", I blurt out as a first reaction. "How do you mean?" he asked me. "That's dirty", I repeated. "I'm sorry, but still I cannot follow you", he said. "Well", I said, "that might be nice for continuous sine waves, but for impulses, this must be disastrous." From the splutter on the other side of the line I could make out that he was unpleasantly surprised that I was able to draw that conclusion so quickly. By then, I had my first inkling why move magnets measured better, but sounded a lot less well than moving coil cartridges. We continued our conversation for a while and I asked him whether the resonance could be described by a linear mechanical resonator and he told me that they had done some curve-fits to the resonance and those fitted a linear mechanical resonator well. I had learned a lot from this conversation.

In order to prove (or disprove) my suspicion about the response to impulses, I created a mathematical model for the response of the moving magnet cartridge in which the electrical part is provided by the specification of the manufacturer and the mechanical response is modelled as a linear mechanical resonator at roughly 19 kHz. The height of the resonance is derived from the electrical transfer function as the overall characteristic should be more or less flat. I was able to reproduce the response curves within $\pm$ 1 dB, so that gave me trust that my model was close enough to reality to predict the response to impulses. (**N.B.** The resonance is also reflected in the loss of channel separation at higher frequencies. If you can lay your hand on such graphs, you will see it). The outcome of the calculation of the temporal response to impulses was dramatic (see TYDGEDRG.pdf on this website) and this made it clear to me why moving magnet cartridges are inferior to moving coils when it comes to *listening* as the moving coils have a mechanical resonance at far higher frequencies because the electrical roll-off also starts at far higher frequencies. (**N.B.** In the end, it turned out that my colleague's cartridge was flawed, so if this would not have been the case, we never would have discovered this phenomenon, which would have severe consequences for future developments! See below.)

Of course, the obvious response is to throw the moving magnet cartridge over left or right hand shoulder (depending on the political preference) and to get yourself a moving coil cartridge. But to me it was a challenge to develop a compensator which would correct this resonance. The basic idea was to use an electronic simulator which would have an equivalent behaviour as a linear mechanical resonator. Use this as feedback in an amplifier and the response would be 1 / mechanical resonator, but -of course- such a system would be unstable because the resonance-simulator would have a 180^o degree phase shift. I solved this by approximating the resonance by a different curve, which would gradually push back the phase shift to zero. This showed to be doable and I was able to build a stable compensator, which indeed had a characteristic which was close (enough) to the inverse of the mechanical resonance. Correcting the properties of the electrical part was relatively simple and so I was ready to have my first listening test......

The design I used for the first prototype proved to be imperfect, but still I was amazed by the improvement of the sound quality. It was far beyond my expectations and I was speechless for a while. I solved the remaining problem of the design and the difference between the original sound and the response after the correction was very clear to me, especially with metallic sounds from e.g. percussion and the like. But was I just willing to hear differences or was it real? So I organised a listening panel of five people, two experienced listeners, two unexperienced listeners and a medium experienced listener. Another colleague of mine had a similar turntable and the same (type of) cartridge and we played the same record, so we could make an A-B comparison. Only I knew which one was playing over the loudspeakers. In 30 seconds (!), there was unanimity in the listening panel that i) there was a marked difference between the two sounds and ii) the sound from the processor was clearly superior. So we are not talking about subtle effects: it was a major difference. The outcome of this listening test was later confirmed by the application of the processor to other cartridges and by an independent judgement by an editor of the Dutch magazine in which I published the design. The most striking proof was the application to a moving magnet cartridge which had a mechanical resonance at 24 kHz (!), but still a clear improvement was audible.

A major conclusion which can be drawn from these tests is that the response above 20 kHz is still of major importance for what we hear. I used this argument during the discussion, which started shortly after these developments, about the CD. My statement was that because the phenomena, encountered with the moving magnet cartridges, the time-smear and sharp roll-off of the CD frequency characteristic would be audible. Philips was not very pleased with my input to the discussion, but years later, the effect of frequencies above the (continuous sine wave) hearing limit (usually set at 20 kHz) is generally accepted and can easily be demonstrated now with the SACD. The sharp anti-aliasing and reconstruction filtering has been the Achilles heel of the CD from day 1 and it has been criticized also from day 1. More details about my arguments can be found in "TYDGEDRG.pdf", which you can download from this website.

This concept is the basis of the MERC (= Mechanical, Electrical and RIAA Correction) processor which integrates the three required corrections for the record reproduction in an optimal way to improve the overall quality and signal-to-noise ratio.

The reproduction of low frequencies is (still) a major problem in audio, at least if you don't want to end up with loudspeaker cabinets which will certainly lead to a divorce. The problem is rather fundamental, physics tells us that it is impossible to combine a small (relative) enclosure with a good reproduction of low frequencies without effecting the temporal response (see "TYDGEDRG.pdf") and vice versa. So you can choose between a "woolly" low-frequency response (but going deep) or a better temporal response, at the expense of the lowest octaves. Numerous designs have been tried, of which the bass-reflex and the transmission line are the most well-known. But the bass-reflex employs the resonances of both the loudspeaker unit and the port to create the low-frequency response and as resonances need time to build up and time to decay, it is not surprising that the bass-reflex systems sound "slow". Not only that: they are slow! Transmission lines try to realise the quadrature of the circle: the line and port should be fully open at half the resonance frequency and acoustically "dead" at and above the resonance frequency to avoid that you build a pipe organ. The great Briggs summarized it as "*a clumsy way to achieve a limited goal*". I don't think that this can be formulated more to-the-point. So are we doomed to live without good low-frequency reproducing audio systems or live or lives in solitude? No! There is a way out, but that needs some "out-of-the-box" thinking.

Some elderly youngsters might recall the Motional Feedback loudspeaker systems (MFB) which Philips brought onto the market (around 1975) in which the motion of the loudspeaker was fed back to correct for the deviations, caused by the system (= loudspeaker unit properties in its enclosure) and thus were able to improve the response of the loudspeaker. A similar approach is used for record cutting tables where the movement of the head is fed back to improve the frequency response on the low frequency side and to reduce non-linearities. However, as usual the most fundamental law of physics, the Law of Conservation of Misery, immediately acts and this results in two major problems:

1. the system is basically a damped resonator, which means that a phase shift of 180^o is created and thus leads to an unstable system when feedback is applied and
2. the efficiency of the system remains as it is, thus requiring an *increase* of 12 dB per octave in power below the resonance frequency to correct the reduction of the efficiency at the low frequency side.

As 12 dB means a 16 fold increase in power, it is clear that it is -in practice- limited in its applicability as the limits of the available electrical power and the power handling capability of the loudspeaker unit are quickly reached. Unfortunately, the commercial design of the MFB systems was not really ideal, as the transducer, used by Philips to convert the loudspeaker movement back into an electrical signal was -diplomatically put- quite non-linear in its frequency response, only 6 of the 12 dB efficiency losses were compensated and the electronics took up half of the volume of the cabinet, which *increased* the resonance frequency by half an octave, compared to the situation where the same loudspeaker unit would have had the full cabinet at its disposal. So the improvement was -again diplomatically put- less than could have been achieved with a bit more optimum design. But it basically showed that low-frequency response in a relatively small cabinet is feasible if electronics are called to help.

The problem I faced was that I am not able to change the design and construction of loudspeaker units, so the application of motional feedback is basically out of the question. But a loudspeaker unit in a closed box (acoustic box), is basically -from a physics point of view- nothing but a damped mechanical resonator. The problem to correct its behaviour is -fundamentally speaking-identical to the problem to correct the mechanical resonance of a magnet on the stylus of a pick-up cartridge, albeit that ω is replaced by $1/\omega$. But given the properties of the loudspeaker unit and the properties of the acoustic box, it is possible to predict the response of the loudspeaker and thus it should be possible to correct it. Such an approach could be designated as "motional feed-forward" as the system corrects in advance what the loudspeaker will do wrong. It avoids the problem to change the design and construction of the loudspeaker unit, but does not correct the non-linear behaviour of loudspeaker unit.

I decided to give it a try and at that moment I was using a temporary bass-system as I was in a transition stage of my audio system. It was a simple acoustic box, but it sounded reasonable. So I calculated the properties of the system and made a design for the circuitry which I christened "ABC", which stands for "Acoustic Box Compensator". Basically, the first prototype worked! The only aspect which I overlooked was that the increased power also results in increased forces on the cabinet and unfortunately, this was not strong enough to withstand the forces which were now acting on it. And as a result, the sides were vibrating so strongly that it radiated audible power. As it would take a while to select a better loudspeaker unit and to build a better cabinet, an acquaintance of mine came up with the idea to stabilise the cabinet with iron bolts all through the cabinet, so I ended up with an enclosure through which the nuts and bolts stacked out. But everybody agreed that it was a road to pursue further as it did improve the low-frequency response significantly.

I selected a better loudspeaker unit, keeping in mind that it should be able to handle both the electrical power as well as the mechanical forces acting on it. And I had a far more solid cabinet constructed which had internal reinforcements to keep the sides from vibrating. And that gave, together with the adapted version of the ABC-2, a major improvement in the low-frequency sound reproduction, which was confirmed by several different people, some of who are also working professionally in the audio business. By sheer luck, I had the opportunity to have the system tested in the "dead-room" at the Delft University of Technology and after the first run, the man who did the test said that he had to repeat it with another octave range at the low end side added as he had never seen anything like this before. In the end, the response continued down to 15 Hz, - 3 dB.

This and similar concepts are also included in the "Diamond" design and although the woofer is housed in a cabinet of only 20 litres, its response goes down to 16 Hz (-3 dB). The advantage is on both sides of the coin, though: not only is the response down to very low frequencies, it is also "fast", meaning that the time delay is small, which avoids the "woolly" sounds of bass-reflex systems. This is very clear with percussion and obvious with instruments like Turkish drums, base drums and the like. But also organ sounds are reproduced better.

At the laboratory where I worked, a hot meal was served for young singles after working hours. One evening, a new colleague had arrived and by chance, the small talk at the table came to audio and amplifiers. By that time, I had build my first amplifiers of my own design and as the new colleague was -not surprisingly- rather sceptical about some youngster who claimed that his amplifiers sounded better than the (for us affordable) commercial types, the more so because he just had bought himself a new amplifier. So I invited him to my place to listen to my amplifier and one night he came for a visit. The end result was that he took the electronics out of his new amplifier to replace it by my design....

A colleague of mine had a neighbour who was not really pleased with his amplifier. So he asked whether I could have a look at it. "Bring me the specifications first", I suggested, so he brought me the manual from the amplifier and the specs looked quite good. "It will probably not meet its specs", was my first thought, so we decided to take it to the laboratory and have a check. Surprisingly, the amplifier did well on the test-bench and met -as far as I could test it- its specifications. So why did it not sound so well? Then it appeared to me that the amplifier might have trouble with load variations, so I gave it a tone-burst of a few cycles to eat. And immediately, the amplifier showed its weakness: the power supply was not able to cope with the changes in current, fed into the load. Actually, you could see the rectified tone-burst at its output! No wonder it didn’t sound well.

I am still amazed by the approach from the manufacturer, as the amplifier would have sounded ways better if he had provided a decent, preferably stabilised, power supply. The design of the amplifier was not bad at all, it just was equipped with a power supply which was underrated and therefore the supply voltage varied strongly in tune (so to speak!) with the input signal. I think this "penny wise, pound foolish" and I always recommend to apply stabilised power supplies to avoid problems like these.

An even more extreme example of what power supplies can do for you to eliminate quality was the amplifier which was published in a Dutch electronics magazine. It was very powerful for its days (50 Watt) and every trick in the book was used to get the best "state-of-the-art" amplifier. It was therefore also rather costly. And to verify the result, the amplifier was extensively tested and the results were also published. And those results had a nice surprise: the output power of the amplifier was indeed 50 W above 100 Hz, but below this frequency, it gradually dropped to some 15 W at 20 Hz. How come? You would expect that the power is *independent* of the input frequency until the frequency is so high that other problems start to surface. But in the range of say 10 - 1000 Hz, it should be constant. The cause, however, was a "power supply", which was hardly worth the name: it consisted of a bridge rectifier and two electrolytic capacitors of 4700 µF (one for the positive and one for the negative voltage) and that was it. At the low frequencies, however, the amplifier draws current from either the positive or the negative voltage for a period of time which is long, compared to the recharge cycle time of 10 ms. As a result, the voltage dropped too far, thus limiting the maximum output voltage. So if the designer had used a better power supply, he would have been able to keep the power at 50 W over the complete audio range. As the low frequencies usually demand most power, that would have been very wise. Now he ended up with an amplifier which was far less capable than what he expected, "thanks" to savings on the power supply, even though the amplifier itself was very costly. And when he had looked at his own test results, he could have known what the problem was. How stupid can you be!f