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The Hawk Audio Design Philosophy

by John van der Sluis

Audio equipment has the purpose of giving the user pleasure at leisure when listening to music. Therefore we're using electrical and electro-mechanical pieces of equipment and metal conductors transporting the signal in-between.
The development of modern electronics has brought us far ahead of ancient times. We all enjoy the various possibilities now available to make live easier. And the same electronics gives us the possibility to listen to music at a different time and place then where the music originally was played (live that is). This way of handling acoustic signals is called RE-production!

Considering what happens inside the various electronic components it's still a miracle that we can "almost" recognize what happened during a live concert. But let's face it, it really is "almost" (and sometimes even "hardly"). So, when talking about acoustic music such as classical and/or jazz, there's nothing better then going to a live concert. What happens there, and the emotional experience you'll undergo there, can not be recreated at home. At home there's the reproduction which differs from the original.

Having said all this, we, at Hawk Audio, are doing research and we design pieces of equipment with the idea of trying to get as near to the original as possible. And doing so within a reasonable budget so the potential user of our designs can enjoy his/her favourite music at reasonable price.
Starting off from "the original" we have to do with amplification. The small signal from the source, be it a record player, a CD/DVD-player or a radio tuner has to be amplified to be able to drive our electro-mechanical devices, loudspeakers that is.
Amplifying an audio signal is a simple task. Just take a tube or a transistor or transistors embedded in operational amplifiers (op amps) and at the output you'll find the bigger signal. Easy job ……. No! When using such devices they all have their own type of "character" or, better said, their own "non-linearities".
One might prefer using tubes because those are "more linear" then semiconductors. But it comes at a price. A tube need capacitors around it in order to couple the tube with the source and with the next stage. And in power amplifiers a non-linear transformer is needed to be able to couple the signal with a loudspeaker. And tubes are more dependant on the characteristics of the power supply.
Nowadays almost all manufacturers use op amps to "handle" the signal. One needs only two resistors to get it working, without capacitors. And, when fed from a symmetrical power supply an op amp will sufficiently suppress disturbances on the supply lines. That suppression is called "CMRR" or "Common Mode Rejection Ratio". Lately we found that the specified CMRR, which is only valid for very low frequencies, is NOT symmetrical. So symmetrical signals on the plus and minus supply lines are not eliminated automatically as it seems.

I have a colleague collecting record players. His large collection holds examples from the early Edison days (a roll telling us that "Mary has a little lamp") and so on. He once demonstrated me a big mechanical player. That player has a turntable, a hollow metal arm

with at the end a membrane and a needle, and a big wooden horn. It was made by Edison around 1920. The very special thing is that it holds a "diamond" needle, almost 50 years ahead of its time. The guy also had a "direct cut" record, with a thickness of around 10 mm, of a piano concerto. He played it and I got thrilled over my entire body. I never before had ever heard such quality and such dynamics!
The conclusion now might be that "electronics is the worst thing that ever happened to music". We can also say that when applying electronics in the audio signal path it should happen in as few places as possible. Sorry to say, but I had that idea already some 10 years before I heard that record.

Now let's consider applying electronics and start off with tube amplification. It's common knowledge that triodes have far nicer characteristics the penthodes and the like. So we start off with a triode. A triode has a high input impedance so it will not be a "load" for the source. The anode output has an impedance of several kilo-Ohms so we might get into trouble when coupling with a cable or a low-ohmic next stage. That's why we prefer using a second triode on top of the first one together creating a kind of "Series Regulating Push Pull" (SRPP) circuit. That circuit has a relatively low output impedance and it (to a certain level) withstands variations on the power supply. The circuit has an amplification factor of around 20 so an input voltage of 300 mV will give 6 Volts at the output. In practice we bring this down to 1 Volt using a resistor network thereby also creating an even lower output impedance. For the double triode tube in this example we prefer using a tube originally intended for use in a UHF receiver of (old) TV receivers. That gives a bandwidth of over 500 kHz and within the audio bandwidth there will be no phase shift.

Looking at the power amp there's a choice between "Single Ended" and "Push Pull". A single ended configuration (with just one power tube) has the advantage of being simple and when distorting only giving even harmonics. The Push Pull amplifier has the advantage of being more efficient and delivering more power. Whatever power tube is used they all have the same disadvantage: a high input capacitance. That capacitance is not a problem when driven from a low impedance. But tubes, by nature, offer a relative high impedance. So whatever the configuration, if you want full bandwidth and few phase shift within the circuit, be it SE or PP drive the power tubes from a low impedance source. In practice this can be an SRPP again or a (twin output) interstage transformer. We prefer the SRPP because of the bandwidth.

Power supplies for tube amplifiers can be more complicated then the amplifier itself. We have to do with disturbances from the mains supply, disturbances from the rectifier and the varying consumption of the amplifying circuit when amplifying a signal. In pre-amps and input stages of power amps, given the low current values, it is necessary to "stabilize" the supply voltage. Until now we mostly used a so-called "shunt regulator" but at present we're busy developing a series regulator, also because of the fewer power consumption and environmental considerations.

When looking at non-tube circuits we are facing the problem that hardly any new transistors are being developed. Happily some manufacturers, at last (!), understood the complaints from music lovers and they designed new integrated circuits in close cooperation with listen panels. So now we can choose out of a variety of op amps specially developed for high end audio purposes. As said before we have to look into the special requirements for the power supply. Preferably we stabilize each power line separately for every op amp. And recently we started developing a combination of a current source with a discrete (!) series regulator. We assume to be able this way to realize an amplifying circuit with a S/N figure of better then 100 – 120 dB.

Power amplifiers with transistors is an entirely different story. For over 15 years we've been delivering hybrid (tubes + transistors) and discrete (only transistors) class-A power amplifiers. Apart from that we experienced with the new class-D technology. Class-D has a very big advantage over other configuration in that it consumes only a fraction of the energy. At present we're busy developing some amplifiers with bigger power and class-D and probably we will also develop a new hybrid amplifier combining tubes and class-D.

In power supplies we always use an extra RC-filter in series with the supply lines. This is done in order to get rid of the bad habit of rectifiers to produce "sparks" interfering with the audio signal and at the same time suppressing noise from the mains supply. So we don't need a special "mains regulator" or the like outside the amplifier. A well designed amplifier should not suffer from variations and other "trouble" of the mains power delivery. There is an exception though in rural areas. In some areas the mains voltage drops over 5% of the given and specified voltage. In such cases a "mains stabilizer" is needed.
The consequence of that series resistor is that the continues output power is somewhat lower then otherwise achieved with the same voltage. But when playing music there's some more instantaneous power (also giving transistor amps a character comparable with that of tube amps).

All our printed circuit boards are made out of glass enhanced epoxy withstanding moisture and a thickness of 2 mm or more. The copper traces have a thickness of at least 70 microns, power supply boards and power amplifier boards mostly 105 microns.

The electrolytic capcitors in our power supplies are special (and expensive) types from Jackon. Those capacitors are specified to function at temperatures up to 120 degrees C and provide a double (or even fourfold) lifetime when compared with other types. Also these capacitors are specified having a low impedance up to 100 kHz. So there's no need to have small capacitors in parallel also avoiding the disadvantage of paralleling capacitors.
Coupling capacitors mostly are polypropylene types, some with low inductance. In correction networks (RIAA in our phono preamps) polystyrene capacitors are used.

Almost all resistors are of the metal film type. Sometimes we use two resistors in parallel in order to lower the inductance.

Loudspeakers from Hawk Audio are intended for real music lovers. And our main goal is to create loudspeakers giving a good stereo imaging anywhere in the listening room.
The loudspeakers are constructed in a circular tube made out of PVC material. PVC is a "thermoplastic" so it converts pressure into heat. In that way, and because it's circular, we offer a very stiff construction hardly adding any sound from that construction itself.
The circular form also has to do with the radiation pattern. Sharp edges near tweeters and midrange units create "lobs" in the radiated sound field and in that way the stereo imaging is far less when moving from the center (hot spot) of the radiated sound waves.
All filters in our loudspeakers are of the series type. A series filter, if used with the right drivers (!), has a big advantage over a parallel filter in that the filter frequency for both drivers is exactly the same. Lately we introduced our new full range loudspeakers. Those have no filter at all. And although it comes at the cost of some coloration it gives the listener an even closer approach to the original sound. We also could alter, thanks to new drivers, the filter in

our top model, the “Duke” loudspeaker. That one now only has some filter exclusively filtering the tweeter.
In most models the tweeter is protected from overload by a special type of lamp.
All filter coils, with the exception of the ones in our Count and Duke loudspeakers, are coreless or air wound. The capacitors in the filters are of the MKP type (polypropylene).