

Cabernet

A new kind of power amp – one of the current sourcing kinds

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*If you are not familiar with the current sourcing principle,
you better read my article about the SST first. This is about a really special design,
which makes it possible to drive a Full-Range speaker at its best.*

INTRO

Nearly 30 years ago I wrote an article about Single-Ended-Triode-Class A (SET-A) and Full-Range (FR) horn speakers. A copy can be found at my website in the

-> HiEnd-Audio Designs chapter, called

-> "A Word about Speakers and SET-A Amps".

In this article I describe, in which way a SET-A amp transforms input voltage to output current and why this performs better than standard amps. One more thing with this kind of HiEnd electronics is to show, why it is wise to make no use of any NFB loop.

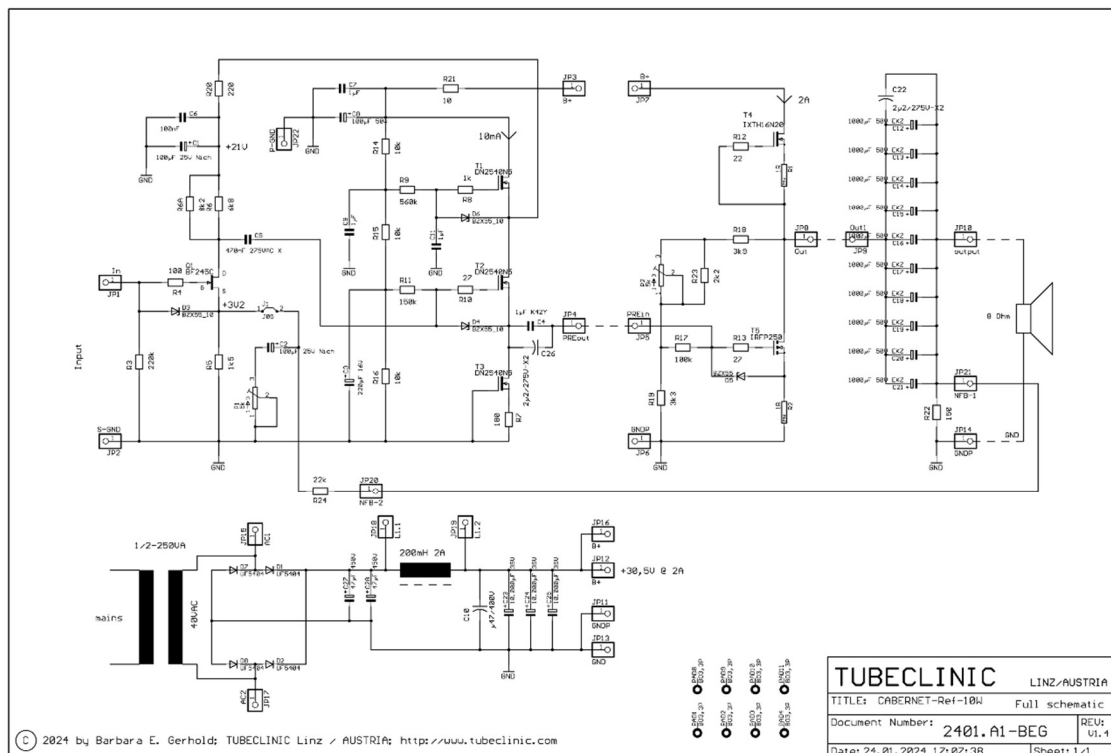
Then, in the early days of the 1990s I designed a Solid-State amp, which also makes use of this technology as well as it omits any NFB. It was powered by these days newly presented NMOS transistors from Hitachi. Today's NMOS are by orders of magnitude better than the early ones, plus we today have D-NMOS (depletion NMOS), which can be used very favorably for the needed constant current source (CCS) at the high side of the output stage. The D-NMOS are also by orders of magnitude better than any sophisticated CCS you could design 30 years ago. So, it was high in time to design a new current sourcing power amp by my SST-technology.

THE WAY IT WORKS

If you are a Pro in electronics, you should have learned about the "Lorentz' Rule" during your studies. It states that the resulting force [F_{res} in N] on a conductor in a magnetic field depends on the current [I_{rms} in A] across the conductor, the magnetic flux density [B in T] and the length of the conductor [L in m].

$$F_{res} = I_{rms} * B * l$$

Considering a loudspeaker and its voice-coil inside the magnetic gap as the above-mentioned conductor, two components of the equation (B and L), will not change during operation - so we can say, that the resulting force [F_{res}] is essentially only dependent on the current (I_{rms}) across the conductor. No dependency will take place vs. frequency, impedance et al.



THE FULL SCHEMATIC

The number of parts is really small. Maybe you remember my opinion, that a design is perfect, if you cannot take any more part off it. What you see here is an input gain stage, a driver stage and an output stage -nothing more.

I powered the amp very conservatively at 10W, because its Class A operation will then already sizzle 60W of power loss at a single channel. On the other hand, my horns give 102dB_SPL from 1 W, so it will be able to make more than 110dB of sound pressure... and this should be enough in the end. Former time SET -A amps were even less in power. And well yes, when calculated for an operation without a cooling fan, this power seems to be the top of the possible.

THE INPUT STAGE

It's the circuit around Q1, a BF245C but you could also use BF256C or J112 (which is what I used). BF245C is very low noise though rare, but since the amplification factor of this stage is low, that's no real issue and the other JFETs will show no real drawback. The $-U_{gs}$ of the JFET should show 3,2V @ 2mA -that's the most important value of this part. The design is the same as you would expect from a standard triode stage - and it performs exactly the same way. I made no use of capacitive bridging of the source resistor R_5 . Since this stage is the only one amplifying, its decoupling was made to its best and nearly 1/3 of the supply voltage was invested in a standard resistive decoupling. Amplification factor is app. 4x from an input impedance of 220k (range 20k to 220k is possible).

DRIVER STAGE

'Potentia copia omnis divisa est in partes tres' (the total power supply is divided into three parts). Pls. mind the 3# resistors R14, R15, R16 (each 10k): Since the main DC-supply is app. 30VDC, we get

+10V and +20V from GND. At the low side of the 3# D-NMOS transistors we find a DC-constant current source (CCS) built from T3. It gives app. 10mA. The mid transistor (T2) is voltage driven by the input stage and tied to the + 10V mark. Together they form a sophisticated driver of a voltage steered element "riding" on a CCS. It shows very high input impedance, low output impedance, really big frequency range (up to some MHz due to low parasitic capacitance) and unity gain. T1 on top of the high side acts like a "choke built from sand", called a gyrator. It also acts as a soft start feature.

OUTPUT POWER STAGE

It is built from a low R_{ds-on} NMOS at the low side and a D-NMOS at the high side. The D-NMOS (T4) acts as a CCS and makes for app. 2ADC constant current. The N-MOS at the low side is responsible for the voltage to current ($V \rightarrow I$) conversion. This $V \rightarrow I$ conversion is done close to what a power triode does. The source of the N-MOS (T5) is elevated via resistor R2 (1 Ω) by 2VDC due to 2ADC passing from the CCS (T4). This elevation must be added to the actual threshold voltage of T5 (app. 4,5V) to app. 6,5V. Because the threshold will be constant over a much wider band of operation than we need here, the voltage at the gate of T5 must only get adjusted to this total value of app. 6,5VDC to achieve the same dynamic internal impedance as the CCS on top. You must adjust it in a way, to measure half of the supply voltage at the drain of T5. This adjustment is kind of stable versus power and temperature, it will most cases not be necessary to readjust later on. But – anyway – to be absolutely sure I checked after the burn-in ...



THE V/I - CONVERSION

The signal from the driver stage is connected directly to the gate of T5. Therefore - if a level of say $1V_{peak}$ is applied $\rightarrow$

- during the negative half-wave of the gate voltage of T5, we have 0 to 1 V less at the gate of T5 which also leads to the same value less at the source (due to the highly constant threshold voltage of T5). Now +1V (instead of former +2V) also reduced the current to 1A (due to the current mirroring effect of R2). But the CCS will furthermore constantly drive $2A_{DC}$, so the difference of 1 A must go to the speaker via a charging current of the el.cap battery (C12 to C21).
- Assuming the same progress for the positive half-wave leads to an increase of current across T5 from 2A to 3A. Again, because the CCS only drives $2A_{DC}$, this additional current of 1A must come from the output as a discharge of the el.cap battery. Since all of the charging / discharging currents must pass across the speaker (connected in series to the el.caps) too, this resulting AC current forms the output signal.
- The output stage inverts the signal at the gate of T5. Together with the same inversion at the input (at Q1) the correct phase relation of 0° el. in between in- and output is granted.
- This conversion works so exactly, that we find less than 0,01% THD at the speaker terminals @ 1W (2,83V @ 8 Ω dummy; 1kHz). In my case that results in 102dB_SPL, a value you would not be able to withstand for even some minutes.

THE SUPPLY SECTION

Because we are mostly dealing with constant currents, it was best to build the DC-supply on a choke filter. Inductors act quite similar like a CCS, because they will be integrating the value of current over time. Therefore, a constant current will never make troubles with supply voltage. Additionally, hum and noise will get very, very low.

I took a single ring-core transformer with separated windings of 2x 40V_{AC} (loaded) @ 2,5A and built a separate supply section for each channel to achieve best crosstalk.

Most of the smoothing job is done by the choke of 200mH @ 2A. The 3# of 10.000µF el.cap behind it only act as a battery substitution. In the end I measured and plotted app. 98dB of SNR @ 1W. This value is absolutely necessary if you also drive very highly efficient speakers like me.

THE NFB

In my schematic you can see, that my design has an NFB-loop (R24, P1 and C2). I equipped it with a switch instead of Jumper J1. This switch is located at the backside of the enclosure and serves as means to show the difference in between current-sourcing (CS) and voltage-sourcing (VS) operation. Some colleagues state it's there to show the drawbacks of VS operation and I have to admit that it's close to that. It's important to adjust the output volume exactly to the one of CS mode, and that's the reason for P1. OTOH it's fine to have the opportunity to drive otherwise not suiting multiway speakers too.

But generally it's not needed and you can omit it anytime.

THE CHOICE OF COMPONENTS

- **Caps** are MKP-X2 by TENTA and/or WIMA as well as some additional NOS MIL RuPIOs (pls. refer to the schematic).
- **El.Caps** are only NICHICON Muse AUDIO GOLD, type UKZ. Only in the supply-section I used standard NICHICON, type VY.
- **Resistors** are standard metal film 1%, type SMA-0207 or -0414. The current sensors (R1, R2) at the output stage are metal-film TO-220 1Ω 1% 25W.
- **NMOS** are by IR and **D-NMOS** are by IXYS and/or by Supertex.
- **Diodes** and **Z-diodes** are from several brands. Pls. refer to the schematic.

OUTRO

Well - what's now the difference in between a standard amp and this new current sourcing amp? Like I already described about the "Lorentz' Rule", it will act at constant force due to constant current for the whole frequency and impedance range of the voice coil. And constant force will result in constant swing, which again will end in smooth overall response. And it will also end all resonance misbehavior, which occurs due to rapid and abruptly changes of impedance. Now the voice-coil will obey because it is reined by force. These are the requirements for good (let's say best) performance of any FullRange-driver (FR). As far as the driver is able to make sound by movement, the current drive will take care for the correct drive of the voice coil. Unfortunately, most multi-way speakers (better say most of all) with complicated crossovers inside will not work satisfactory, this must be said in advance to avoid complaints.

But OTOH, I am sure you did not hear a better amp/speaker combo before, than a FR-horn driven by a current sourcing amp. You cannot hear the speakers but a wall of sound and you will encounter the best imaging ever. Backroom is totally black and there is full, realistic empty space in between the instruments and voices.

Furthermore, there were a lot of guys in my home, who searched for the "hidden tweeter" but couldn't find any. One of my friends even called this behavior the "Non-Existing Billion Euro Golden

Ear Tweeter" ...



As you know - all information about the printable, full-size schematic and the PCB is again situated in a binder enclosed to this website.

If you have any further question, you can contact me via eMail.