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Class-D for DIY

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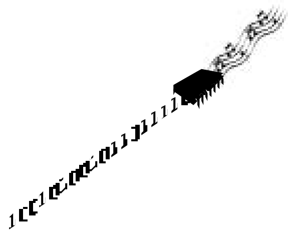

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Notes slide 1

Boring...

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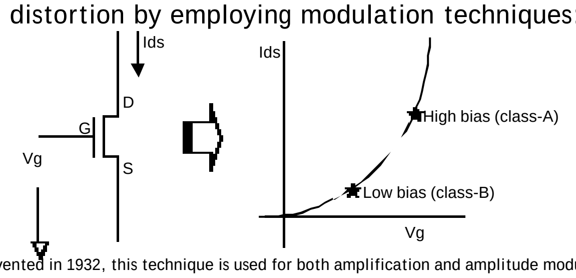
Introduction

- Biasing: Class A, B, C
- Class B and C always Push-Pull
- Extreme class-C: class D
- Reduce distortion by employing modulation techniques: class-S



Class S First invented in 1932, this technique is used for both amplification and amplitude modulation. Similar to Class D except the rectangular PWM voltage waveform is applied to a low-pass filter that allows only the slowly varying dc or average voltage component to appear across the load. Essentially this is what is termed "Class D" today. See References: Krauss for details.

Notes slide 2

Audio amplifiers are distinguished by their operating class. This class depends on the biasing. If the amplifying devices are biased so high that they conduct during 100% of a sinewave, the amplifier is said to operate in class A. If the conduction period varies between 50 and 100% of the sinewave, it is class B. In class-B, full sinewave conduction is still possible by adding a complementary device, the so-called push-pull mode.

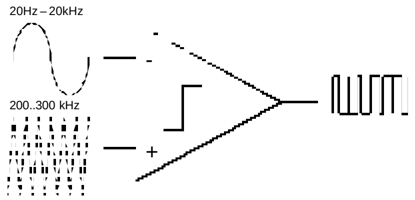
Class-C has devices on between 0 and 50% of the cycle. In this class, even in pushpull mode, the sinewave is reproduced less than 100% and output filters are necessary.

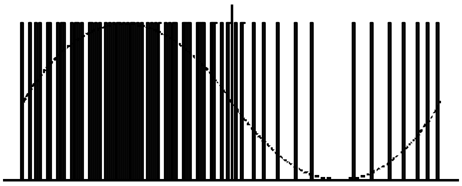
In class-D, the biasing is completely absent, and when the sinewave crosses a threshold the devices switch on completely. Thus, a sinewave is reproduced as a square wave. It is obvious that this causes a huge distortion and only in fixed-frequency amplifiers with large filters the result is acceptible. For audio, this class is unuseable.

However, by modulating the audio stream on a higher frequency and using a filter, it is possible to retrieve the original audio signal. This class, called class-S, can be used to achieve high fidelity amplification.

It is very confusing, but this class-S has become widely known as class-D.

Class-D classes





Analog:

Distinction by modulation techniques:

- PWM, PDM, PPM
- many other derivative forms (ICEpower etc)

Digital:

Distinction by binary format

Notes slide 3

Class-D amplifiers are, as said, amplifiers that involve modulation. The filtering retrieves the average signal. So if we want to reproduce the audio band (20-20000 Hz), we need to use such a high frequency, that the average signal can still reach 20 kHz.

Several modulation techniques exist. The simplest is Pulse Width Modulation, where at a fixed frequency, the audio amplitude is compared to a triangle-shaped wave and the duty-cycle of this comparison contains the original amplitude.

Pulse Density Modulation is essentially the same as sigma/delta modulation. The amount of fixed-width pulses per time unit is analogous to the audio amplitude.

Pulse Position Modulation is, as the name implies, a modulation mehod where the position or phase of the pulse in the duty cycle carries the information and hence largely compatible with PWM, but the energy content per cycle is always equal.

There are many derivative forms of modulation, which are all a combination or variation to the forms described above.

All the forms of modulation can also be synthesized digitally.

Typical properties

- rail-to-rail switching
- MOSFET design (Ron)
- 300 kHz < switching freq < 5 MHz
  - -> output filter
- power 1W - 1kW
- efficiency 78% - 97%
- applications: PC, portable, consoles, audio

Nodes slide 4

Typical properties of Class-D amplifiers:

- they incorporate only 2 voltage levels, typically both supply lines

- since the devices are switched fully on, only devices with a very low on-resistance are used: mosfets

- since the 'average' signal must still be at least 20 kHz, the amplifiers run on high frequencies. 300 kHz is a common trade-off between accuracy and device losses due to gate capacitances.

- Hence, an output filter can not be omitted.

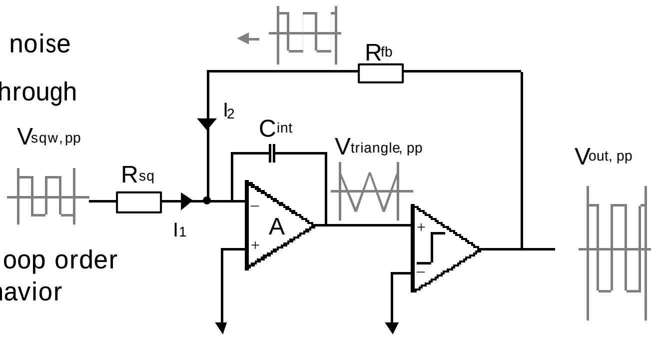
- Power outputs vary between 1W and 1kW

- Efficiency is at least 78% (otherwise you migth as well use a regular class-B amplifier) and 97%, depending on power output. Due to the switching, either the device is fully off (I=0) or fully on (V=0), in any case the dissipated power in the device P=V*I is nearly zero.

- class-D amplifiers are found mainly in low-power applications. Some high-end audio experiments however have shown to be very succesfull (i.e. the TacT millenium amplifier).

Feedback and higher orders

- Feedback not evil, lots of bandwidth
- aim: correct pulse area at output
  - PSRR
  - offset, noise
  - shoot-through



- Increasing loop order improves behavior
- Stability

Notes slide 5

Feedback is easily bashed in audio design, especially in the high-end community. However, class-D amplifiers need feedback to stabilize their modulators. This type of feedback has not a lot to do with the audio signal itself, since the amplifier is running at such high speeds.

It is comparable to the feedback used in the sigma/delta modulator: the errors caused by the output device can be regarded as quantisation noise, for instance by power supplies folding, or through the dead time that is needed between switching off on one device and switching on the other device. If the other device is switched while the other device is stilled partially active, the power supply sees a very low resistance (2 times the on-resistance per device) which can cause the amplifier to fail.

As is the case with the sigma/delta modulator, increasing the order of the modulation improves the noise shaping characteristics. Say your amplifier runs at 300 kHz, then your audio bandwidth is 100 kHz (rule of thumb). If you have a 2nd order modulator, the rejection decreases with 40 dB/decade, which means that at 1 kHz, you already have 80 dB of noise and power supply rejection.

Feedback helps improve the modulation, but can destroy it as well. If the feedback squarewave is exactly 180 degrees out of sync with the reference clock, the modulation wave is nullified. This form of digital instability is of course to be avoided.

$\sqrt{x^2}$ Digital class-D and PowerDAC

Pulse Density Modulation:

- two level signal !
- noise shaped
- amplify, filter, done?
    - speed <-> THD, power consumption
- Do SD conversion to multibit, then PWM (TacT Equibit)
- reduce switching (PowerDAC+)
- live with power consumption, parallel fast devices

$\sqrt{x^2}$ Notes slide 6

As said, class-D and sigma/delta audio are quite alike. They both have only 2 levels, a high-frequency modulation scheme and noise shaping characteristics.

It therefore makes sense to use a class-D amplifier with 1 bit audio: just apply a power stage after the bitstream, amplifying it directly, filter it and you're done. Unfortunately, it is not as simple as that since the bitstream runs at a very high frequency. If you want to switch a power stage at such a high frequency, you create large errors due to the dead-time needed to prevent shoot-through.

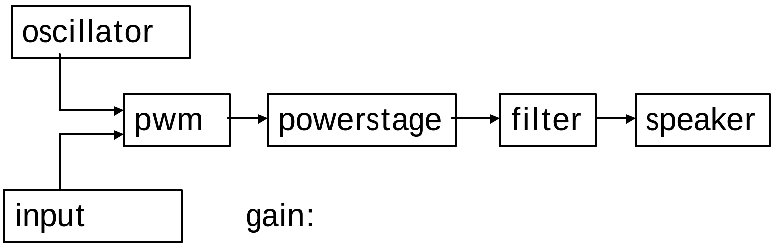
One of the solutions that has been applied, is recalculating the digital data to an 8 bit word (using noise shaping) and using the 8 bit info to create 256 possible duty cycles, and then running the power stage at 300 kHz with the discrete dutycycles. This is the Tact Equibit principle.

Another possibility is to still try and amplify the bitstream directly, but apply a PWM feedback over the PDM pulses. Using this feedback, errors have the effect that the pulse is widened to still have the same area (amount of energy). Hence, several pulses are 'smeared' toghether, and thus the switching frequency is lowered as well.

The high-end solution would be to accept the higher distortion and still run at a very high frequency, but that would imply a powerful driver as well to drive all those gates so fast.

DIY with class-D

Circuit 1



gain:

$$duty(oscillator) \bullet \frac{Vsupply}{Vswing}(comparator)$$

Notes slide 7

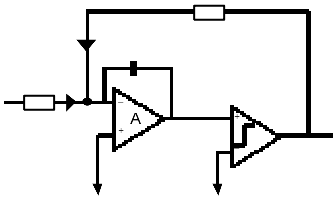
Now that we know a bit about the pitfalls of class-D design, we can start DIYing with it.

The simplest class-D amplifier has no feedback or noise shaping. Just use a simple 300 kHz oscillator (i.e. a 555, 4060 etc) and run that into a simple PWM modulator, which can be made with a comparator (NE519, LM311) or another 555. Feed their outputs directly to a complementary pair of HEXFETs, use two LC sections tuned at 50 kHz and you have your audio signal. The author has good experiences replacing those nasty STK modules with this type of amp. This is already much better than that.

The gain of such an amp can be easily calculated. The comparator feeds the power stage so the gain there is Vsupply(powerstage) / Vsupply(comparator). The comparator itself switches its supply voltage at a certain input swing so the gain is Vsupply(comparator) / V(inputswing).

Circuit 2

- 1st order loop



$$G = \frac{A_{comp}}{wR_{fb}C_{\text{int}}}$$

$$G' = \frac{2}{p}\frac{f_{sqw}}{f_{in}}\frac{V_{switch}R_{sqw}}{V_{sqw}R_{fb} - V_{switch}R_{sqw}}$$

Notes slide 8

The first circuit has zero power supply and dead time rejection. We can enhance the circuit to incorporate a feedback loop if we add an opamp configured as integrator.

This circuit also allows us to input a square-wave which, when mixed with the signal and integrated by the opamp, serves as triangle wave. Thus we can use a high-stability crystal.

The gain is as in the first example, but now also includes the gain in the opamp.

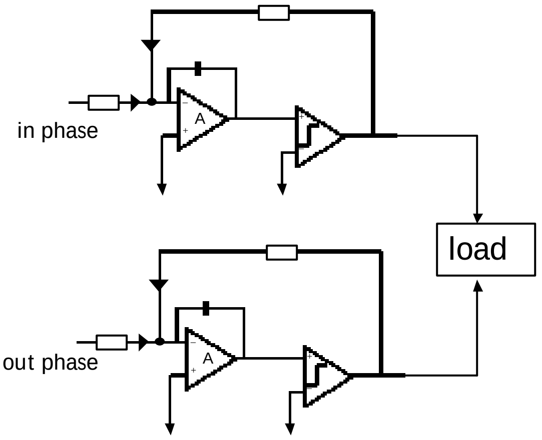
DIY with class-D

Circuit 3

- bridged outputs
- BTL: P'=4P

Possible problems:

- ground bouncing
- EMI
- reactive loads
  - PSU
  - 4 quadrant PP
- protection



Notes slide 9

The circuit that we now have can be expanded further by adding another opamp to create a 2nd order loop. This improves the modulation and PSRR. In theory, higher orders improve the behavior even further but in practise it creates more problems than it solves.

Another way of improving the behavior is by using two amplifier modules in a bridge configuration. The power output quadruples since the output voltage doubles ($P=V^2/R$).

The problems that arise when designing a class-D amp are:

- ground bouncing and supply pollution: the transient switching draws high currents, severely polluting the supply

- EMI is a definite problem: switching square waves at frequency x creates HF up to 100 times as high as frequency x

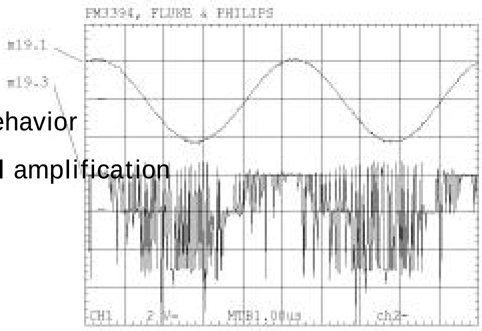
- reactive loads can fire back currents when there is no path to sink this current since the sinking device is off. This can be hazardous to the components and sound quality.

- short-circuit and other device-related protection schemes, easily implemented in linear amplifiers are hard to implement in a class-D amplifier.

Conclusions

Class D

- General: linear amps without linear components
- oversampling, minimum freq
- filter
- efficiency
- higher order improves behavior
- suitable for direct digital amplification



Notes slide 10

What have we seen about class-D amplifiers:

The modulation scheme used to linearize the switching nature of the amplifier allows an amplifier to be very linear, even if the devices used to create it are not.

This however necessitates that the amplifier must run at a frequency much higher than the highest audio frequency, allow the audio signal to average out and yield the right amplitude.

The averaging process must be performed by a filter located between the output stage and the load.

The switching nature allows for a very high efficiency, since at any time either the device voltage or current is zero.

Feedback is incorporated to correct the errors created in outputstage and powersupply. The higher the order, the better the rejection.

The 2-level nature of class-D lends itself to direct digital amplification of 1-bit signals. Smearing modulation is applied to reduce the average switching frequency.