Class D audio power amplifiers: Adding punch to your sound design

Steve Taranovich - January 03, 2017

One of my first jobs as an engineer was at Empire Scientific Corp. in Garden City, NY in 1972. This was a high-end audio company that made speakers, turntables and phonograph cartridges. We had some direct-cut vinyl records with such incredible dynamic range that you could close your eyes and visualize the orchestra and the exact location of each instrument. When played through two Crown DC-300 amplifiers (Left and right channel) that, believe it or not, we used as Pre-amps into a Phase Linear Power Amplifier (First we had the 700B series and later the Series II).

Audio amplifiers have come a long way since then. Class D amplifiers were first conceived in 1958 and so much has been written about the different architectures that were cited just in 2016 which improve different aspects of their performance. I will share these recent techniques with you in this article.

Such recent design improvement techniques are Power Supply Rejection Ratio (PSRR) improvement, lower distortion, Electro-magnetic Interference (EMI) reduction, Intermodulation (IM) distortion improvement, quiescent current reduction, Total Harmonic Distortion (THD) reduction, and driving capacitive transducers in electrostatic loudspeakers.

The power transistor element

Audiophiles always said that the best sound came from vacuum tubes. There are schools of thought that say FETs are more closely related to the type of distortion and sound that vacuum tubes produce in audio than BJTs. An MIT report claims the opposite. The controversy continues as it has since the vacuum tube began to lose importance in most electronics designs.

If we look at the power element as a FET, such as Efficient Power Conversion (EPC) outlines in an app note, we see that good sound needs good THD, Damping Factor (DF), and IM Distortion. Efficient Power Conversion’s eGaN FET designs in class-D audio amplifiers provide excellent sound quality due to their excellent switching abilities because of lower open loop distortion that helps
lower THD and overall losses. This leads to a reduction in feedback which in turn reduces Total-IM Distortion and DF and improves sound quality. All of this over a wide range of output power which essentially enables better dynamic range performance in the audio realm.

**Class D power amplifiers lower, power dissipation, footprint, and cost**

Since all linear power output stages will dissipate power in class A, B, AB, and D amplifier configurations, we look for the one architecture that will enable the most efficient power so that battery operation can be employed, with the smallest board footprint to reduce size and weight in a portable or wearable design, and the lowest cost to enable expansion into a wider variety of market applications than ever before. The class D configuration is the one that stands out as meeting all these requirements (Figures 1 and 2).

![Figure 1: A linear CMOS output stage diagram (Courtesy of Analog Devices)](image1)

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![Figure 2: An open-loop Class D amplifier basic diagram including a nearly-lossless filter design architecture in which the only intentionally dissipative element is the speaker (Courtesy of Analog Devices)](image2)

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So how does a class D amplifier work?
International Rectifier (now part of Infineon) explained this topology very well in a 2005 application note.

The class D amplifier is simply a switching amplifier or Pulse-Width Modulation (PWM) amplifier as opposed to class A, B and AB which are all linear amplifier architectures. **Figure 3** shows a basic class D amplifier block diagram.

![Figure 3: A Class D Amplifier detailed block diagram](Image courtesy of Infineon)

The class D amplifier essentially operates similar to a PWM power supply; their topologies are essentially identical in operation (**Figure 5**). **Figure 4** shows the class D amplifier waveforms in action with its waveforms shown.

![Figure 4: Waveforms shown here in a class D amplifier operation](Image courtesy of)
Figure 5: The strong similarity of the Buck power converter and the class D amplifier
(Image courtesy of Infineon)

Eliminating the external low-pass filter while reducing EMI

Maxim Integrated has employed filterless modulation in its class D amplifier designs (Figure 6).
There is almost always a compromise to be made in circuitry architectures when improving a particular parameter. In this case, the added possibility of radiated EMI from the cables emanating from the amplifier to the speaker. Enter Spread-Spectrum modulation to the rescue. By using a dithering or randomizing method for the switching frequency in the class D amplifier, this technique spreads out the spectral energy of the output signal which in turn reduces any high-frequency energy peaks at the output\(^4\).

**Getting excellent PSRR with low THD+N performance**

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In the new millennium, more than ever, consumers are demanding new products with better audio quality, portable with a battery that lasts a long time before needing a charge, and a better resistance to interference in the presence of so many other RF protocols.

Class D amplifiers are better than Class AB audio architectures because they are able to provide efficiencies above 90% at maximum power levels with only a small increase in temperature of the system. This makes them a very good choice for mobile designs.
We present a design architecture here that employs a differential architecture using a class D amplifier with pulse-width modulation (PWM) common-mode control which will improve the power supply rejection ratio (PSRR) as well as the linearity of the audio power amplifier.

An input feed-forward method is employed as well using resistive summation in the loop filter that reduces internal voltage swing along with signal-dependent terms which will make a further improvement in linearity performance (Figure 7).

![Diagram](image)

**Figure 7**: The linear representations of the conventional Class D amplifier is shown in (a) and the Class D amplifier proposed in this new architectural technique from Reference 5 is shown in (b). $e_C$ and $e_D$ are shown lumped equivalently at the output of the amplifier. (Image courtesy of Reference 5)

The loop filter architecture uses a single amplifier biquad (SAB) method which enables a reduction of the number of op amps needed and leads to lower power in the system. Figure 7b shows the addition of a common-mode (CM) control loop to the conventional Class D design architecture. This addition suppresses $e_C$ which is the impairment related to even distortions, common-mode noise, and supply noise. $e_D$ is related to odd distortion or differential-mode noise. Here in Figure 7b, $e_D$ is suppressed by the differential loop gain $H_D(s)$ as it is also in Figure 7a, but now in Figure 7b the CM control loop also suppresses $e_C$ in the new architecture.

The loop filter in this architecture only uses two op amps due to the SAB technique (Figure 8).
This proposed technique of a PWM Class D amplifier improves not only PSRR but THD+N as well. The design has the low power, small size, improved resistance to external EMI and low distortion needed for today’s portable solutions.

Spread spectrum helps lower EMI in class D audio

The switching output stages of a class D amplifier can produce annoying EMI in an audio amplifier system. Adding a conventional filter would be an easy solution; however, this would add weight, size and cost to the architecture. Instead, we consider integrating a spread spectrum method that will reduce EMI by distributing the EMI over a wider frequency range, thus lowering noise level and peaks in the band of interest.

One problem that needs to be overcome is that spread spectrum techniques usually reduce good audio performance. By not using a filter, the amplifier is directly connected to the speaker. In most filterless designs, a three-level PWM technique is employed. A class BD modulation technique is usually used which will introduce a third level that reproduces the idle or low-signal conditions. Texas Instruments has a good app note describing class D LC Filter Design. Now a common-mode component is added to the output signal and will cause high frequency noise into the speaker through the wired connections. The solution was to develop a new modulation method.
A good, proper measurement setup must be created as in Reference 6. Two spread spectrum methods were evaluated:

1. A periodic modulation with an up-ramp profile and period of 12.5 kHz
2. A pseudo-random modulation which used a combination of linear feedback shift registers (LFSR). The repetition rate here is 6 Hz.

It was determined that simulation alone was going to be difficult since the frequency of the PWM signal is, at a minimum, an order of magnitude greater than the actual audio signal. To solve this problem, the developers chose to create a prototype of the class D amplifier that would match the implementation, performance and switching characteristics of the actual IC being considered for the design to be implemented in 180 nm IC technology (Figure 9).

Their design used a measurement setup using a PC with MATLAB which pre-calculated the carrier waveform that gets input to the carrier generator. An EMI receiver measures conducted emissions (CE) by using a 150 ohm Method and then an audio-analyzer monitored the audio performance. See Figure 10 for the test setup.
Figure 10: The test setup for the class D amplifier tests using two different spread spectrum techniques. (Image courtesy of Reference 6)

The final outcome was that the periodic spread spectrum method showed the best EMI reduction and the pseudo-random method had the least disturbance of the audio quality. There is always a compromise in every architecture depending upon what the final system performance parameters are most important.

**Pulse density modulation effects on IM distortion in a class D amplifier**

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We now look at two different modulation techniques used to drive a class D amplifier stage while examining how switching frequency affects total second and third order IM Distortion products.

It is stated that some audio engineers claim that IM Distortion is more important to sound quality than Harmonic Distortion. Be that as it may, let’s look at the differences in distortion for pulse width modulation (PWM) vs. pulse density modulation (PDM) techniques and how they affect the ultimate sound quality in this power amplifier architecture.

First let’s look at PDM. Any loss of information due to the modulation process will affect the quality of the final audio reaching the speaker. In PDM the number of pulses seen in a time window is directly proportional to the average value of the input audio signal. In this technique, the width of the pulses may not be arbitrary. The pulses must be quantized to multiples of the clock period of the modulator (Figure 11).
In order to reduce quantization noise in the modulator, the switching frequency will need to be much greater than the frequency of the input signal. Because of oversampling the switching frequency is usually pretty high which leads to higher power dissipation than the PWM method. On the plus side, the sigma delta technique offers better linearity than the PWM method.

IM Distortion can be filtered out; however, second order products are a problem in broadband cases where third order products are important in narrowband applications.

Using PSpice simulations we can examine the influence of switching frequency and voltage of a two-tone signal, used in an IM distortion test, on the distortion of the output signal of the class D amplifier.

In this example, we choose a PSpice representation of the class D amplifier composed of two Maxim Integrated MAX942 high speed comparators and a Texas Instruments TL061 input amplifier with a D-type flip-flop and two IRF9530 P-Channel Power MOSFETs as the output drivers. (Figure 12)
By analyzing the effect of switching frequency and voltage of the input signal of a class D amplifier with pulse density modulation, it was determined that second order distortion was pretty small at each frequency but the third order distortion could be kept under 1% at low frequency and low voltage input signals.

**Using feed-forward PWM-intermodulated distortion reduction (FFPIDR) in a class D amplifier**

In a closed loop class D amplifier configuration, FFPIDR enables a reduction of PWM-intermodulated distortion with an added benefit of high in-band distortion suppression without the need to increase input switching frequency. This technique resulted in a 2.4 mA quiescent current with a 1W power output capability that makes this architecture a good fit for mobile designs (Figure 13).
The final result of using such an architecture created a high fidelity sound while maintaining a 2.4 mA low quiescent current along with a THD+N performance of 0.0037% in a small IC footprint. The architecture has the ability to drive battery-powered mobile designs with a full 1 W of output power while maintaining long battery life.

A class D amplifier to drive electrostatic speakers

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Although thus far we have shown class D amplifiers for power amplifier applications, there is another area of need in the signal chain for good sound reproduction---the loudspeaker. Typical speakers are the weak link in the audio chain, in part due to poor efficiency of the electrodynamic transducer method used in most speakers. The capacitive transducer method for a speaker is used to create the electrostatic loudspeaker alternative. There is an alternative way to create a capacitive transducer is by the use of Dielectric Electro Active Polymers (DEAP). This method can make the speaker smaller, lower cost and most importantly more efficient if driven by a class D amplifier.

The push DEAP transducer may be used in loudspeaker drive designs but it is not able to act as a capacitor for switching frequencies above 100 kHz (Figure 14). To remedy this in a design in which the output filter is designed with only an inductor and the DEAP, film or ceramic capacitors can be placed in parallel with the DEAP transducer to extend the frequency response. Now a new problem emerges because the capacitive load now increases the reactive power of the amplifier output. The DEAP series resistance in the order of 1 to 50 ohms can also be a problem. To add to our difficulties, the DEAP transducer mechanical construction technique poses further mechanical stress problems.

Figure 14: A DEAP push transducer (Image courtesy of Reference 9)
Let’s start to solve this problem by first improving the ripple current in the DEAP transducer which is caused by the high series resistance. This can cause efficiency and reduced lifetime problems if the conduction losses get too high from high ripple current. Reference 9 determined that a 4th order output filter solution would give a loss of only 0.92 mW while a 3rd order one gives 8.43 W losses. The 4th order filter is the right choice, plus the film capacitor we added in parallel with the DEAP can be used in the first LC-filter stage. Now the high frequency will flow through a capacitor with an improved frequency response than that of the DEAP transducer. See Figure 15 for the class D amplifier design.

![Figure 15: A class D amplifier with a 4th order output filter and control circuitry. (Image courtesy of Reference 9)](image)

The experimental results in Reference 9 used the +/- 300V class D amplifier shown in Figure 15 composed as a half-bridge driving a 100 nF load in the frequencies between 0.1 to 3.5 kHz. The designers used a Si8235 isolated gate driver with SPA08N80C3 MOSFETs as shown in Figure 16.
The final test results showed that THD+N was below 0.1% while producing a maximum of 125 volt-amps reactive (VAR) power at an 87% peak efficiency.

These creative circuit architectures are only a handful of so many design improvements for class D power amplifier designs; the ones in this article were the latest and most interesting which I have found from 2016 developments and proposed design architectures. A really good source of these types of designs and many others is the IEEE XPlore Digital Library site which houses, what I consider, some of the best tech articles dating back over 100 years to the most recent ones published in 2017.

Let’s make the world a better place in 2017 with innovative electronics.

References

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5. A 118 dB PSRR, 0.00067% (−103.5 dB) THD+N and 3.1 W Fully Differential Class-D Audio Amplifier with PWM Common Mode Control, Wen-Chieh Wang, Member, IEEE, and Yu-Hsin Lin, IEEE JOURNAL OF SOLID-STATE CIRCUITS, VOL. 51, NO. 12, DECEMBER 2016


Also see:

- Add headphones to a Class D amplifier
- Class D audio-power amplifiers: Interactive simulations assess device and filter performance
- Digital-Input Class D amplifiers expand the benefits of traditional Class D and simplify system design