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Current Drive Can Overcome Pitfalls Of Class D Audio Amplifiers

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My background includes over a decade in the design of medical MRI digital gradient amplifiers. In many ways, MRI gradient amplifiers perform similarly to class D audio amplifiers except that the high power and absolute accuracy is well beyond what commercial audio amplifiers can achieve. In the MRI, the drive employs a controlled current to create an accurate magnetic field gradient across the patient without direct feedback of the magnetic field.

Amplifier bandwidth begins at dc and extends well beyond audibility with a constant delay across the spectrum. Data values can be arbitrarily controlled by 20- to 22-bit serial data with rates up to 20 MHz. Just as an example, one of my designs was for a low-power 0.35 Tesla open-MRI used in clinics. It provided ±180 A with ±350-V compliance.

A medical MRI commonly uses three amplifiers, one for each axis (X, Y, Z). In comparison with the low-power MRI design just described, most MRI systems are larger and can provide greater than 2x or 4x the current and voltage. The driven gradient coils are complex loads and are typically modeled as third- or fifth-order impedances. Driving these complex impedances with a controlled current rather than voltage, results in greater accuracy.

The same benefit can be obtained when driving a loudspeaker with a current and this approach also eliminates many of the errors associated with using class D amplifiers to drive speakers. It can also override the need to migrate from silicon to GaN power transistors in these amplifiers.

Most audio amplifiers have a fundamental error in application regardless of topology class, i.e. class A, B, AB, D, G, H, etc. They commonly drive speakers with a voltage, despite the fact that an electrodynamic loudspeaker functions most linearly by current. Nearly all audio amplifiers provide a voltage signal to the speakers, rather than current. Many scholarly articles and technical editorials have covered the subject over the years, yet the industry appears reluctant to change how speakers are driven.

What follows in this article is an explanation of simple concepts underlying the operation of class D amplifiers and is not intended as a comprehensive discussion of all the technologies used in the industry nor to challenge anyone's math abilities. The general idea behind class D amplifiers is to convert a signal's magnitude to a representative duty cycle or periodic pulse of fixed or variable duration. This concept needs to be understood before describing possible alternative solutions offering circuit design improvements such as the current-controlled amplifier that will be described here.

Commercial Class D Amplifiers

Many class D audio amplifier ICs are available in commercial distribution. So, for low- and medium-power levels (under 100 W), few audio electronic designers create amplifiers using discrete components due to the design convenience and low cost of these audio amplifier ICs. The manufacturers of such ICs are legion. They include Analog Devices, Diodes Inc., Infineon, Maxim, NJR, NXP, STMicroelectronics, and Texas Instruments.

Most class D IC manufacturers commonly create a time-modulated switched signal that drives a speaker. Simply put, the switched signal is a high-frequency carrier signal that is pulse-width modulated (PWM). That modulation follows the audio input signal such that pulse width per period represents signal magnitude. The pulses drive the speaker, and it responds to the averaged values.

Simplified Descriptions

To achieve this PWM signal, an analog audio signal is compared to a high-frequency ramp signal, often triangular in shape. The comparison result from a comparator component will be a pulse with a repetition rate matching the ramp signal and with a duty cycle that represents the trigger level of the original analog audio signal magnitude.

The audio is in effect “sampled” and the fixed switching frequency, F_{SW}, is the signal carrier and needs to be a least twice the frequency of the highest audio frequency of interest (bandwidth) to be sampled (the Nyquist limit) (Fig. 1.)

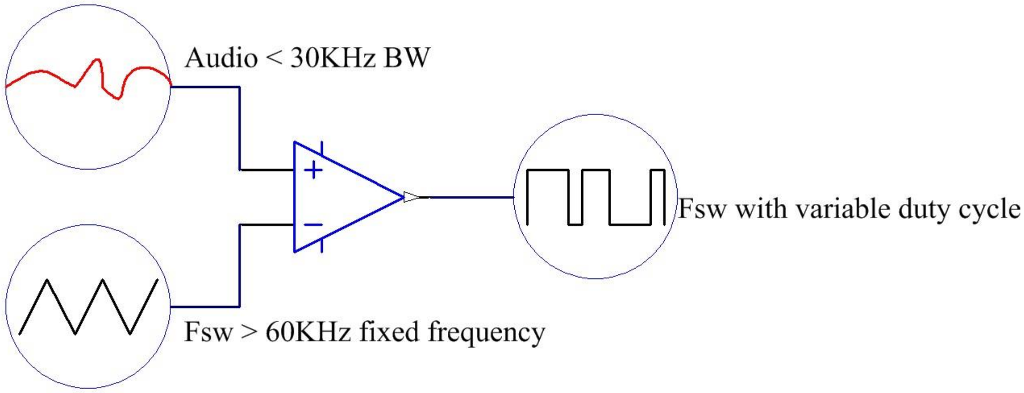

Fig. 1. A simplified open-loop example of how a class D amplifier generates its PWM signal.

The resulting PWM signal drives the speaker once it is buffered to provide the required voltage and current levels. The carrier frequency is inaudible because the mechanics of the speaker cannot respond to such a high frequency. However, a filter may be used before driving the speaker for the purpose of EMI reduction.

A closed-loop design improves accuracy with a feedback averaging network which removes the carrier noise and filters the output into a signal within the audio bandwidth with a short delay or low phase distortion. But a poorly designed feedback averaging network can create distortion and overtly colorize the audio result. Therefore, many designs opt for open loop and rely on the linearity of the circuitry for audio fidelity.

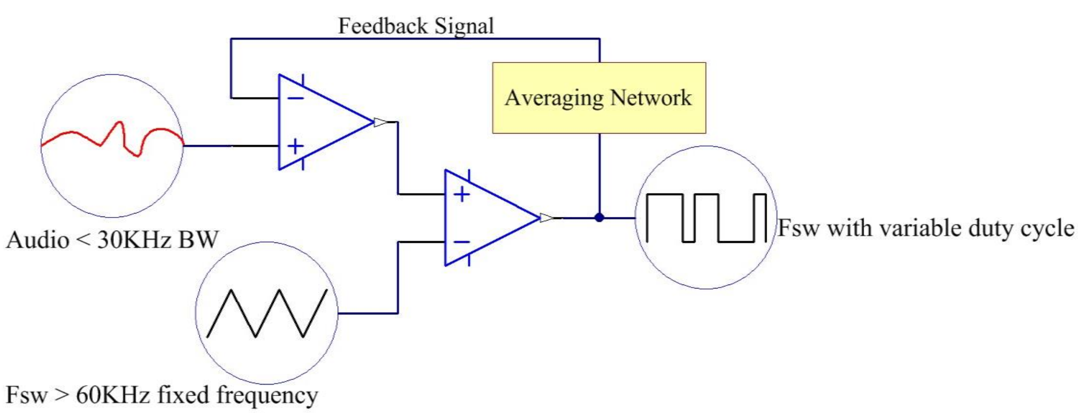

Fig. 2. A simplified closed-loop example of how a class D amplifier generates its PWM signal.

Problems With Common Class D Circuits

The Comparator

The comparator output switches states as it compares two voltage levels. But imperfect comparisons cause the absolute value of the audio to be transformed into a distortion that is either an expansion or compression from the true value. There are a few sources of comparator error:

- Nonsymmetrical offsets. An audio signal is sampled differently when rising versus falling and this results in different values of pulse width for the same audio magnitude.
- Delay variations or nonsymmetry. Any delay creates a phase shift in the resultant output. If non-symmetrical, the audio signal is sampled differently when rising versus falling which results in the representing pulse width to be of different values for the same audio magnitude. This may result in greater harmonic distortion as the audio frequency increases.
- Gain bandwidth. Since the comparator device will have limited gain, the smallest detectable signal determines its dynamic range. CD quality audio commonly uses 16-bit linear sampling and is described with a 96-dB dynamic range, or approximately a resolution of 1 part in 65,535.

Here's an illustration of how dynamic range affects the requirement for comparator gain. If a comparator circuit operates at 5 V and rail-to-rail such that the ramp signal magnitude is 5 V, then it needs to detect and switch output states on a signal difference of less than 76 µV with less than 16 µs of total delay. Therefore, the comparator's discrete voltage gain specification would be 62.5 V per mV. A value less than this would mean that the audio resolution would be unacceptable in the industry. So a very high-quality comparator device is required to detect such a small difference in input signals without drifting or varying with temperature.

For example, consider the Texas Instruments LM311, which is an ancient (over 40 years old) design, yet a respectable comparator IC. The specification states it may have a gain as much as 200,000 typically and switch delays of 115-ns T_{on} and 165-ns T_{off} for 100-mV "overdrive". However, when observing the delays in switching with the smallest input "overdrive" signal depicted from 20 mV to 2 mV, the delay stretches from 1.0 µs to 1.4 µs. Extrapolating by fitting 76 µV within an assumed inverse power curve, the resultant delay may be closer to 4.1 µs. If true, this is just tolerable to meet the minimum 16-µs delay requirement.

The Reference Ramp

This signal has two critical areas that effect error:

- Ramp linearity. A nonlinear ramp would create an error in the pulse width representing the audio magnitude. The absolute value of the audio would be transformed into a result that is either expanded or compressed where nonlinearity exists in the ramp.
- Clock jitter. The period of the ramp signal is varying and this creates a time variation in the comparison or the comparator's sample moment. The averaged result becomes an offset from the correct value, therefore representing a distortion.

An alternative to using the comparator with a reference ramp is to perform a digital comparison using an A-to-D converter (ADC). But eliminating the analog ramp error with an ADC introduces its own linearity issues, and has its own clock jitter and quantizing noise. The ADC is also susceptible to variations in the reference voltage that defines signal scale, therefore creating gain variations of compression and expansion.

With this approach, it is notable too that that audio quality ADCs are not inexpensive.

What Is A Speaker?

Herein, a speaker is an electrodynamic loudspeaker and electrostatic types are excluded from this discussion. A speaker is commonly a linear motor or solenoid where a current flows through a coil and creates a magnetic field that provides force opposing a permanent magnetic field. In the case of the speaker, the coil in question is called the voice coil.

This force results in a piston (the pole piece) pushing air via a diaphragm (the speaker cone) which has mass, and therefore inertia and momentum which consumes power in motion. The diaphragm is also held in a neutral position, which adds resistance to movement and consumes power.

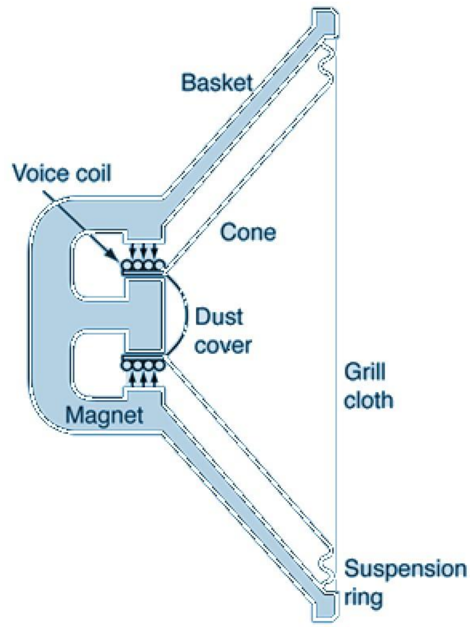

Fig. 3. Elements of an electrodynamic loudspeaker.

The three principles governing movement of speaker elements are:

- Inertia ≡ mass, Newton's first law
- Momentum = mass x velocity, Newton's second law
- The force that moves the diaphragm is related to the strength of the magnetic field inside the "voice coil" (a coil of wire or solenoid) and is proportional to N (coil turns per unit length) x I (current).

Fig. 4 represents the speaker electrically as a simple circuit and shows that a current provides the force that moves the diaphragm (speaker cone). Note that a more-accurate modeling would depict a very complex relationship that includes a coupling coefficient for the magnetic force and a more nonlinear reflected loading.

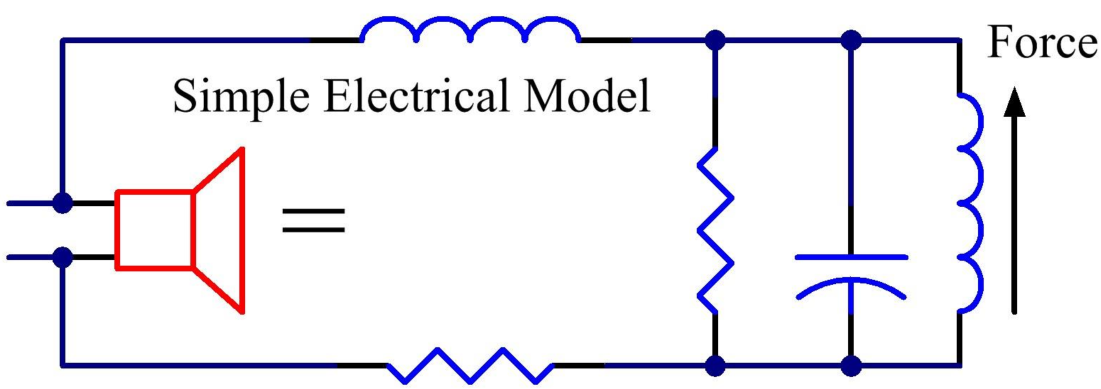

Fig. 4. A simplified electrical model of a speaker.

Importantly, this electrical model exposes a common error that speakers are commonly driven by a voltage rather than a current. Such resultant errors are somewhat small with regards to the movement of the diaphragm, but they exist regardless.

The amplifier-to-speaker EMI filter may introduce additional distortion and creates more complex loading. Fig. 5 expands the simple electrical model of the speaker to include the EMI filter.

This network depicts the difficult relationship between the average voltage driven versus the magnetic force used to pump air back and forth in a linear fashion. In a practical application, the diagram in Fig. 5 works satisfactorily only because the values of the filter and parasitic component impedances are relatively small in series and relatively high in parallel.

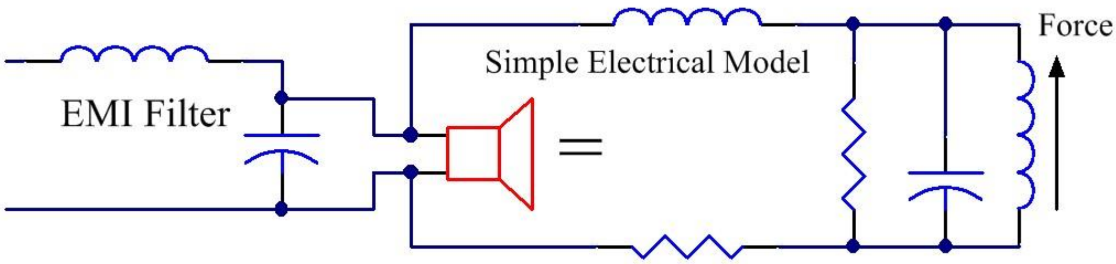

Fig. 5. A simplified electrical model of a speaker with an EMI filter added.

Nonetheless, signal delays through complex networks are distortion and if the delays at the various audio frequencies are different, the result may create unacceptable harmonic and spacial distortion. Therefore, common sense might suggest that the distortion effects become more marked as the audio frequencies increase.

Current-Sensing Feedback

Utilizing a current-sensing method as depicted in Fig. 6 improves accuracy between the drive signal and current supplied to the voice coil that provides the force to the speaker cone. We can observe that passing a voltage-averaged audio signal through a simple LC circuit introduces noticeable delay while a current-controlled drive signal can be passed through the same LC circuit and appear at the output with little or no delay. The series inductance (L) no longer limits the rate of current (or voltage) rise across the capacitance (C). Provided the C values are small, the delays are at a minimum.

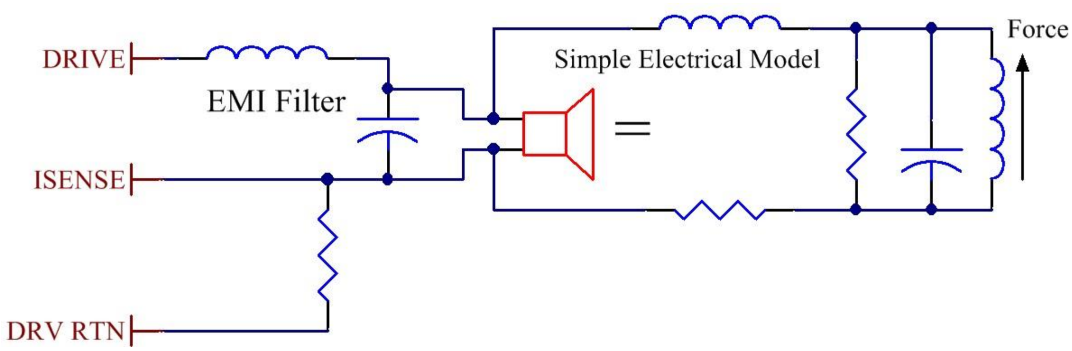

Fig. 6. Using current-sensing to drive the speaker with a current signal.

Adding both voltage and current feedback as depicted in Fig. 7 can improve most aspects of delay yet true feedback of voice coil position is not practical as it would require installing a power-consuming sensor on the coil. While use of both voltage and current feedback is practical, it’s not strictly necessary. If current feedback is used, voltage feedback (which corresponds to audio volume) has less significance.

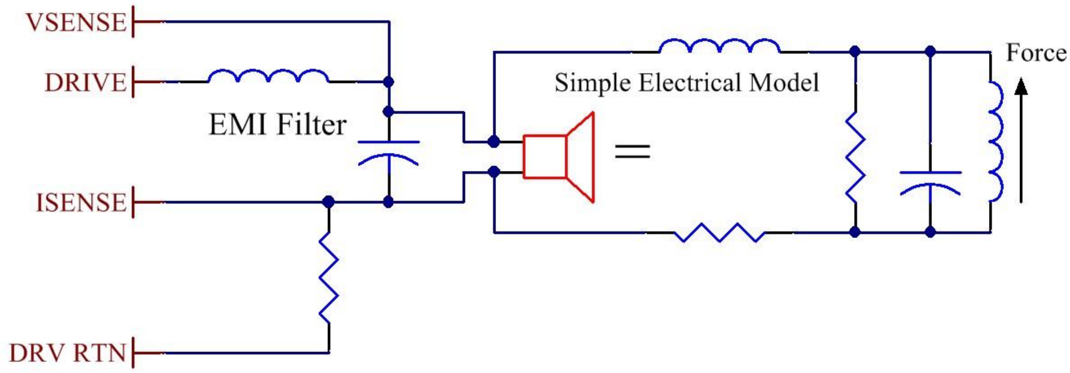

Fig. 7. Using both voltage and current sensing to drive the speaker.

A Current-Controlled Amplifier Approach

Assuming both voltage and current methods would perform nearly the same, the current mode is easier and more accurate to implement when the correct control circuitry is chosen. Fig. 8 shows how hysteretic control can be applied to achieve current feedback.

The output current varies at the carrier frequency just above and below the average value that represents the original audio signal. Note the current modulation ramping above and below the average output current in Fig. 9.

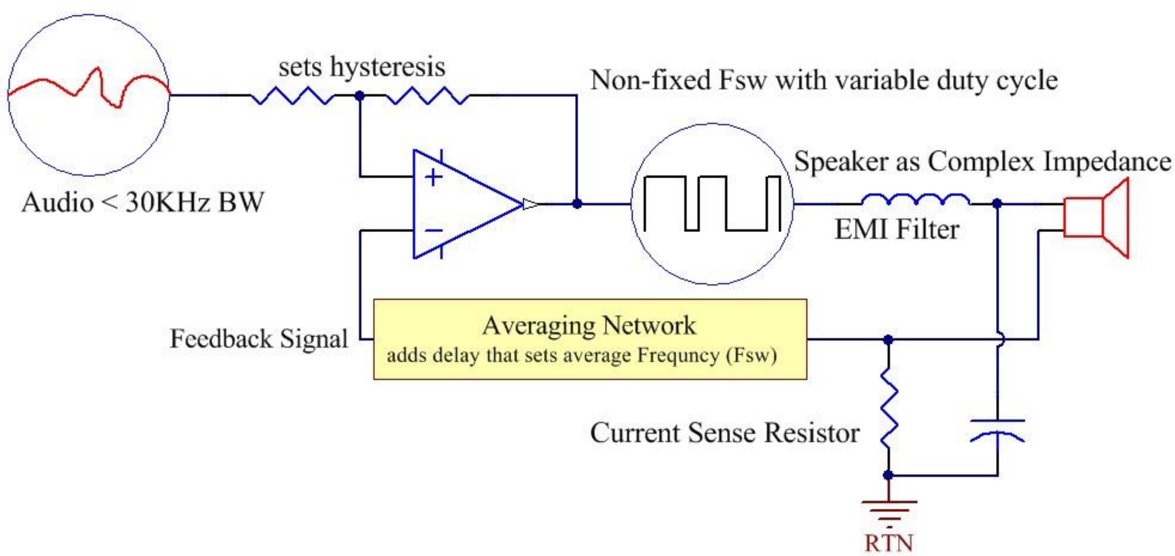

Fig. 8. A current-sourcing topology can be easily implemented using a "hysteretic" control system that self oscillates to create the carrier frequency (Fsw) as shown in this simplified circuit.

The resultant carrier frequency (shown in Fig. 9 as 80 kHz) is defined by the sum of the delays from the EMI filter, the speaker impedance, and the averaging network, and the current sensing gain versus the hysteresis levels set by the comparator resistors in the positive feedback to the audio input.

Fig. 9. A 2.5-kHz sine wave signal sampled at an average carrier frequency of 80 kHz.

Accuracy of hysteretic switching levels is not important and just needs to be small, perhaps 0.1% of the maximum current capability or overload limit (1 mA/A) as the feedback signal represents an average target value for the current flowing through the speaker.

Conclusion

Advantages of hysteretic control include the reduction of errors from comparator delay as it only becomes part of the averaging rather than a misrepresentation of absolute sampling. In addition the reference ramp is now removed such that its linearity errors are removed as well as concerns of clock jitter adding error to sampling. Hysteretic control features a bandwidth that includes dc, or extremely low frequencies with overload avoidance and little or no dc offset.

Fig. 10 shows measurements of a simulated speaker load (using the speaker-equivalent circuit in Fig. 7) that monitors the final current through the voice coil, which represents the magnetic force that moves the diaphragm. Measured results reveal that the fundamental and the second harmonic distortion track closely. However, the complex impedance of the speaker has an apparent effect on third harmonics and according to the experiment, this distortion is an order of magnitude lower when the speaker is current driven.

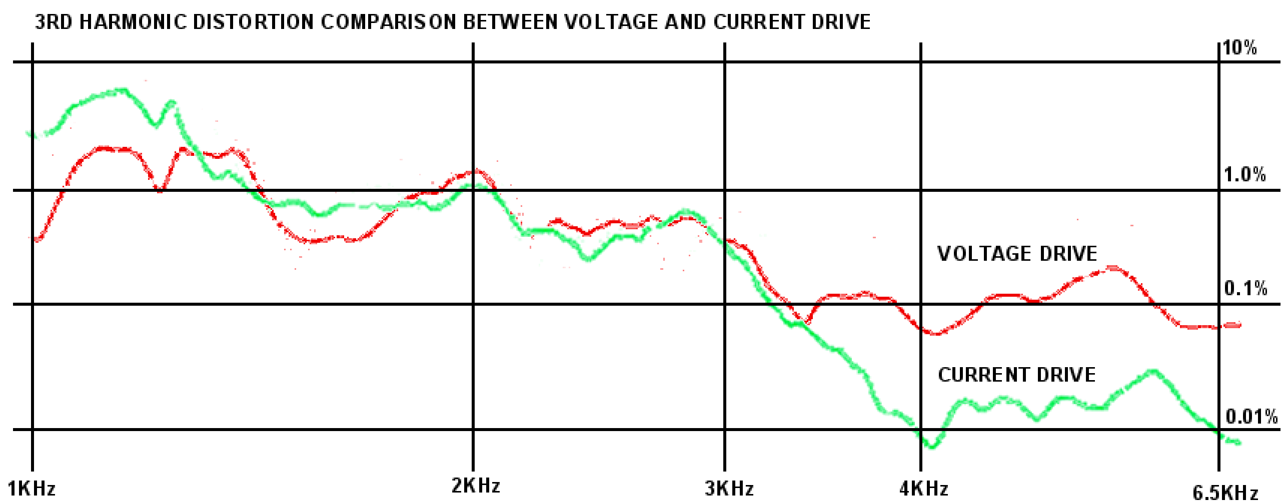

Fig. 10. Experimental data compares third-harmonic distortion values for voltage and current drive, indicating similar distortion performance until higher frequencies are reached.

From this measurement, it's apparent that the fixed and variable signal processing delays become significant for open- or closed-loop voltage control as audio frequencies increase.

The bottom line on current drive of speakers: it works, it's more accurate, it can provide good overload protection, it's easier to design, and it's less expensive to apply. For the reader, a good next step is to design your own circuit in Spice or build a breadboard and discover for yourself the benefits of this alternative circuit approach to class D amplifiers.

About The Author

CEO and owner of Avatar Engineering Corporation, Anthony Esposito, has over 35 years of experience in power conversion and precision instrumentation for all markets—consumer, industrial, telecommunications, automotive, green-power, medical, military, and space. Present activities include new semiconductor research, high-efficiency converters for military and medical applications, and miniature high-voltage transformers and circuits for implanted defibrillators as well as next-generation Taser weapons.

Prior work includes medical MRI gradient amplifier and systems design, telco cell site power systems design, as well environmental controls design for NASA's original Enterprise Shuttle. His New Product Development (NPD) software, NPDPRO.com has been used extensively by the SBA and Harvard Business and by prominent business colleges. Esposito resides in Arizona and has numerous patents in power controls and communications.

For more information on designing class D amplifiers, see How2Power's Design Guide, locate the "Application" category and select "Consumer and Entertainment".