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The Magic Carpet Ride:

Class D Audio Circuit Design And Some Live Testing Results

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I once had a good friend that was really into classic audio. He had all sorts of wonderful equipment from JBL, AR, McIntosh, and many other vendors. We would listen to CDs, vinyl, and ¼-in. open reel tape. We'd watch wonderful TV shows like "Cheers", "Get a Life" and "Married with Children" in SAP stereo. He had a great setup. This is where I learned of presence, sound field, and depths of tonality that I'd never imagined could be reproduced.

And of course we used to play it LOUD. At least 20 dB over tinnitus levels. That was great fun. This is where I learned the differences between fine transducers like a JBL2404 and an OS80 Afterburner and the artists that bring these differences to life like Steppenwolf, Supertramp or the tireless pipes of Billy Corgan of the Smashing Pumpkins. This quest took me down some wonderful avenues of re-examining the spring and ball models used in a bass reflex enclosure design, and reworking and validating the works of A.N. Thiele that mapped some of the more common bass reflex tunings.

I also learned of the different effects of various circuit elements, for example, the distortion that can be caused by a saturated inductor core or the voltage coefficient of capacitance in a MLCC ceramic capacitor. Most importantly, it was fun. I later engineered all of this knowledge into some world-class large scale monitors built of classic, legendary stock like [JBL] D130s, LE15s, 2404s and 2426/2370A driver/horns for my own home theater 'world'. The monitors stood 6 ft. tall, weighed 500 lbs each and took over a decade of endless detail and effort to complete.

I've wanted to bring class D audio into my amplifier rack for a long time—but I was skeptical. This article will outline my skepticism, as well as the design, build, bench testing, and live testing of the IR class D audio solution. The results were nothing short of impressive. I'll guide you through the process step by step.

As a field applications engineer for IR, I've had the occasion to demonstrate the IRAUDAMP7S class D audio solution to some very high-end customers in some of the finest listening environments I can imagine. The results were always positive. The testing always ended in "wow, I can't believe that's a switching amplifier" after comparing favorably against known-good linear solutions. But to walk that amplifier into my home theater and plug into the opus of my loudspeaker creations requires much more than a favorable review. I needed to build the solution up proper, test it on the bench and address any concerns.

Reasons For My Skepticism

As a general topology, Class D audio has had some problems. The class D audio amplifier of yesteryear pounded the voicecoils beyond xmax during power up, inverted the modulator signals on clipping, and on A/B testing had a heavy sound that sounded like a bad linear amplifier with devices in the pre-amplifier stage that had an F_t (transition frequency, where small-signal current gain = 0 dB) in the kHz range played back with a phonebook taped over the transducers.

The efficiency claims and maximum power output claims in some cases could only be reproduced for a few seconds with a large heatsink or a flow of cold spray directed on the amplifier power stage. The amplifiers that claimed to require no output filtering needed LC filters for any appreciable power output. To mask this, the designers used small filter inductors that would create distortion products when pounded into and out of saturation. This has tainted the perception of class D audio in the marketplace. I'll attempt to go through these attributes one by one.

Clipping. Most amplifiers fell apart completely at clipping. They'd invert the outputs, and produce all sorts of abnormal, harsh distortion products well beyond that of a clipped sine or program. This was so bad that most that were building high-power class D audio amplifiers designed their own modulator stage out of discrete components and logic gates.

Power up popping or dc burst. Some of the finest class D power processing topologies available fail to address the need for common-mode rejection on the inputs of the first stage of the amplifier. When these amplifiers are powered up, a loud “thud” can be heard as the input stage comes into equilibrium and tries to throw the voicecoils across the room (a little beyond Xmax in most cases).

Distortion. This is a mixed-mode consideration that requires a lot of attention at the IC design level and in terms of power throughput. From the input amplifier, to the modulator stage, to the bootstrap capacitors and output MOSFETs to the output filter, every aspect of the amplifier impacts distortion. This is beyond simply not using carbon-comp resistors and avoiding micro-arcing in the bad old days of high-voltage tube circuits. Total harmonic distortion (THD) is one measure of fidelity, but it certainly doesn’t tell the whole story. THD simply measures a steady-state response. There is no dynamic information in a THD measurement. THD data is furnished as a plot at fixed frequency and variable power, and then at constant power and swept frequency.

Intermodulation products, noise floor and dynamic range. This is a little tricky. Class D audio uses a high-frequency ramp signal tied to the comparator in the modulator stage. The frequency of this ramp signal is the carrier, or switching frequency of the amplifier. This carrier frequency can vary from channel to channel, and from board to board. If the variance is in the audible frequency range, there may be a discernable problem or simply an irritating presence on high-frequency notes that just sounds odd. This can be difficult to measure, but the source of the problem is simple. Whenever there is a carrier frequency difference such that there is an audible component, there may be an audible intermodulation product. In multi-channel amplifiers, this requires a little frequency planning at the design stage. Most will space 30 kHz between adjacent amplifiers in a multi-channel system to avoid this problem.

In a class D amplifier, dynamic range and noise floor tell us how well the modulator and output filter are doing their jobs. We need a low minimum duty cycle for zero and low-level reproduction, a large maximum duty cycle for large-signal reproduction, and excellent linearity between those two asymptotes. Many class D solutions fall short in this specification. There are a couple of tests on the bench that measure this, but I’ve found that the best test is a live audition with a high dynamic-range program. The idle hiss should be minimal, and any sharp change (the cannon blast in Beethoven’s Fifth, for example) should be quick and effortless. You should be able to hear the pre-charge ignite before the big bang. Anything less will sound hollow and heavy.

Efficiency. The efficiency of the amplifier will be poor at very low power levels. We’d expect this. Any switch-mode power processing solution has overhead in the form of switching loss. At mid power levels, the efficiency pops up to a much higher value than linear amplifiers and stays there through high power levels. A lot of the integrated driver/power MOSFET amplifiers have an efficiency peak at mid power levels because they’re forced to use smaller MOSFETs. At higher power levels, the conduction loss begins to dominate these smaller ICs and they take an efficiency hit, but their low-power efficiencies look better. The ability to select the right MOSFET for the load impedance and general design affords the designer the ability to shape this curve.

Overload, short circuit and thermal protection. The crown DC300A or the Phase Linear model 400 would adamantly disagree (and I’ve overhauled enough of them to know this), but any audio amplifier should be able to handle a short circuit indefinitely, have thermal and electrical stability into low-impedance loads, and offer thermal protection. Most class D amplifiers have a shortcoming somewhere in these few parameters. Some can only sense current on the low-side MOSFET, leaving any short to ground during a positive excursion unprotected. Others offer no overcurrent protection at all, allowing the IC to simply fail catastrophically. And some were never designed to handle the higher currents of lower-impedance loads.

Output filtering. If you value your voicecoils, an output filter is needed regardless of the modulation scheme used in the amplifier. Without a filter, all of the high-frequency components around the PWM frequency cause eddy current and proximity effect heating of the innermost windings of the voicecoil. The goal of the filter is to reject high-frequency ripple from the modulator, integrate the signals from the power stage, and stay as far above the highest-frequency audio material as possible.

Clearly we don’t want a high Q in this application as that may excite some unwanted components. Most signal chain engineers view a filter as having fixed elements that never change. But what if they do? For example, what if large, low-frequency program excursions cause the output inductor to saturate slightly (or in bad designs, saturate completely) and drop in inductance? This change in inductance causes distortion products as the inductance swings. It’s the part of the equation that’s forgotten:

$V=-L di/dt + I dL/dt$.

Normally L doesn’t change. But in the audio world, when it does, we have distortion, not to mention excessive loss in the inductor core.

EMI. The carrier frequencies of these amplifiers run from 250 to 500 kHz in most cases. These are switching waveforms with fast transitions over large voltage ranges (high dv/dt) and similar current transitions (fast di/dt). Care needs to be taken to avoid coupling high-frequency noise to the internal circuits or anything external. I suspect my acrylic box could use a little shielding, but when I test my Class D audio amplifier I can't find a single birdie on my spectrum analyzer or on a swept receiver with microvolt sensitivity.

Amplifier Circuit Design

The Class D audio amplifier described in this article is based on the IRAUDAMP7S reference design. The schematic for this reference design is shown in Fig. 1. IR offers an evaluation board/kit based on this design, which I will be using here to build the amplifier.

Generally, when we design an amplifier we have a power output goal and ±Vcc established for our rails. The two go hand in hand. The power supply must be able to deliver enough voltage into the load to make rated output power without clipping. Once we have this goal, we can select the driver IC, which in this case is the IRS2092. From this point we select our MOSFETs for the output stage.

There are several optimal choices to suit your cooling and spatial constraints. For example, if you need the PCB to be very low profile with very low maximum height, you may want to use IR DirectFETs. On the other hand, if you have a preference for through-hole only devices, you will likely use the DIP version of the driver (IRS2092PbF) and through-hole MOSFETs, resistors, inductors and caps.

Please note that the power MOSFETs need to be carefully selected for optimum amplifier performance. The IRS2092 driver has excellent built-in delay matching between the high-side and low-side driver to the MOSFETs. If we maintain that matching through the Q_G of the MOSFETs, the duty cycles will have maximal symmetry throughout any modulator output. IR has a family of through-hole devices for this application. This is the IRFI4xxx family. These are available in 3- and 5-pin TO-220 footprints. They are also available in FullPak versions (totally plastic encapsulated, no sil-pad or heatsink insulator needed). These devices are ordinary trench-type MOSFETs with care given to Q_G matching as discussed above.

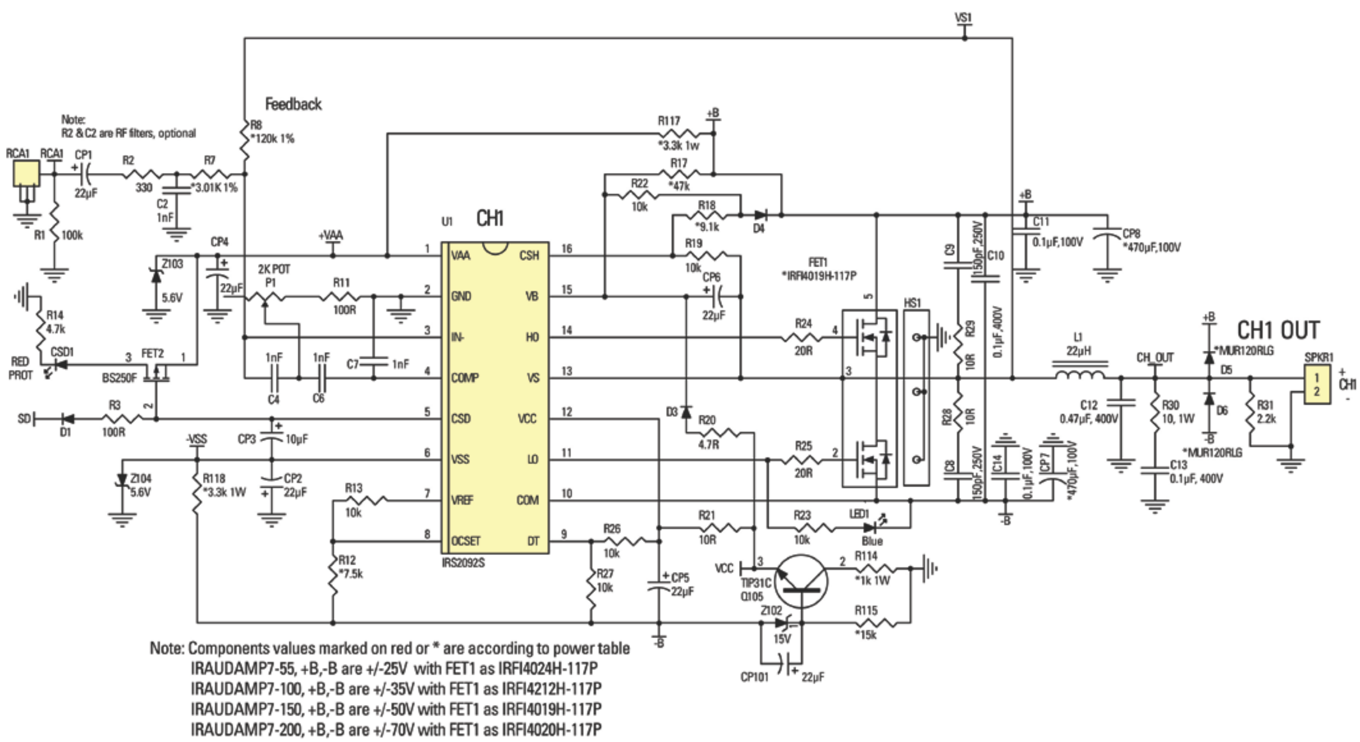

Fig. 1. Schematic of the IRAUDAMP7S class D audio amplifier reference design.

Once we've selected the MOSFETs and the driver, we can begin setting up the switching frequency of the driver IC, the output filter, and the overcurrent and deadtime control parts.

The switching frequency needs a little explanation. The oscillator in the IR class D audio amplifier is formed by a low-current feedback signal from the output of the MOSFETs (at the midpoint and NOT post output filter) to the PWM comparator. This signal feeds a resistive divider to ground (R8, P1, R11) and a capacitor network going to the comparator (C4, C6, C7). The capacitor network should stay with the recommended values of 1 nF in all three locations. The grounded leg of the resistive divider sets the free-running frequency of the class D amplifier—this is the switching frequency of the amplifier.

The IRAUDAMP7S evaluation kit is set up for approximately 400-kHz operation. It is important to point out that there is a compromise in setting up the switching frequency. If the frequency is too low, the output filter will impact the phase and possibly the amplitude of high-frequency components in the audio program. (This is usually only a problem at switching frequencies below 250 kHz.) If the switching frequency is too high, the maximum and minimum duty cycles will become more and more significant, impacting the dynamic range of the amplifier. The optimal range is something in the 250- to 450-kHz range for these amplifiers. Channel spacing should be 25 to 30 kHz.

The output filter is a conventional LC lowpass filter, terminated of course by the loudspeaker (in parallel with a Zobel network.) When designing this filter, bear in mind that any change in inductance with increasing output current causes distortion. For this reason, we want a well-designed inductor that will exhibit little change in inductance value at absolute maximum output-current levels as opposed to something that goes into hard saturation. The Q of the LC network needs to be designed for the highest load impedance. This is the condition that will present the worst peaking in the frequency response of the RLC filter.

The corner frequency of the LC network needs to be well above the maximum audio-output frequency of the amplifier, and below the switching frequency. The design presented shows an LC filter designed to have a Q of 0.5 into a 4-Ω resistive load, with a crossover frequency of around 50 kHz. If the load is changed to 8 Ω, there is slight peaking as the Q of the filter becomes 1.0. Please also note that the inductor was carefully chosen to exhibit minimum change in inductance over the full range of output current.

The deadtime is selected from four internally set values of 105 ns, 75 ns, 45 ns, and 25 ns. This is accomplished by setting up a low-current voltage divider (R26 and R27 in Fig. 1) from the driver supply voltage (Vcc) to the driver return (com pin on the IC or B- in circuit). The midpoint of the divider ties to the DT pin. For noise reasons, the current through the divider should be around 0.5 mA. The table lists the values of voltage divider output corresponding to the four selectable values for deadtime.

Table. Voltage divider values for programming deadtime.

Vdt (with respect to common)	Deadtime
0*Vcc	105 ns
0.29*Vcc	75 ns
0.46*Vcc	45 ns
1.0*Vcc	25 ns

The gain of the amplifier is determined by R8/R7.

PCB Layout For IRS2092

This is the most important part of the design task. Clearly, for a high-gain power amplifier, we want to keep the input connections as far away from the output connections as possible. This avoids parasitic coupling and erratic operation. Since this is a switching amplifier, we want to use good power layout practice on the output stage and sufficient bypassing as well as good signal layout practice on the input stage. The layout should minimize noise on the output as well as minimizing the susceptibility of the input stage to any radiated noise.

The low-level input-signal path should be laid out with fairly thin traces and kept fairly compact with good symmetry between + and – input for best common-mode rejection. Use a copper flood tied to input ground on the top and bottom side of the PCB to minimize any loop areas that may be formed between the input traces and other connections to the IC.

The power stage should also be kept tight. For best gate drive, the MOSFETs should be kept as close to the driver as possible. For best EMI performance, the LC filter components should be kept as close to the MOSFETs as possible along with all local bypassing capacitance for the B+ and B- rails.

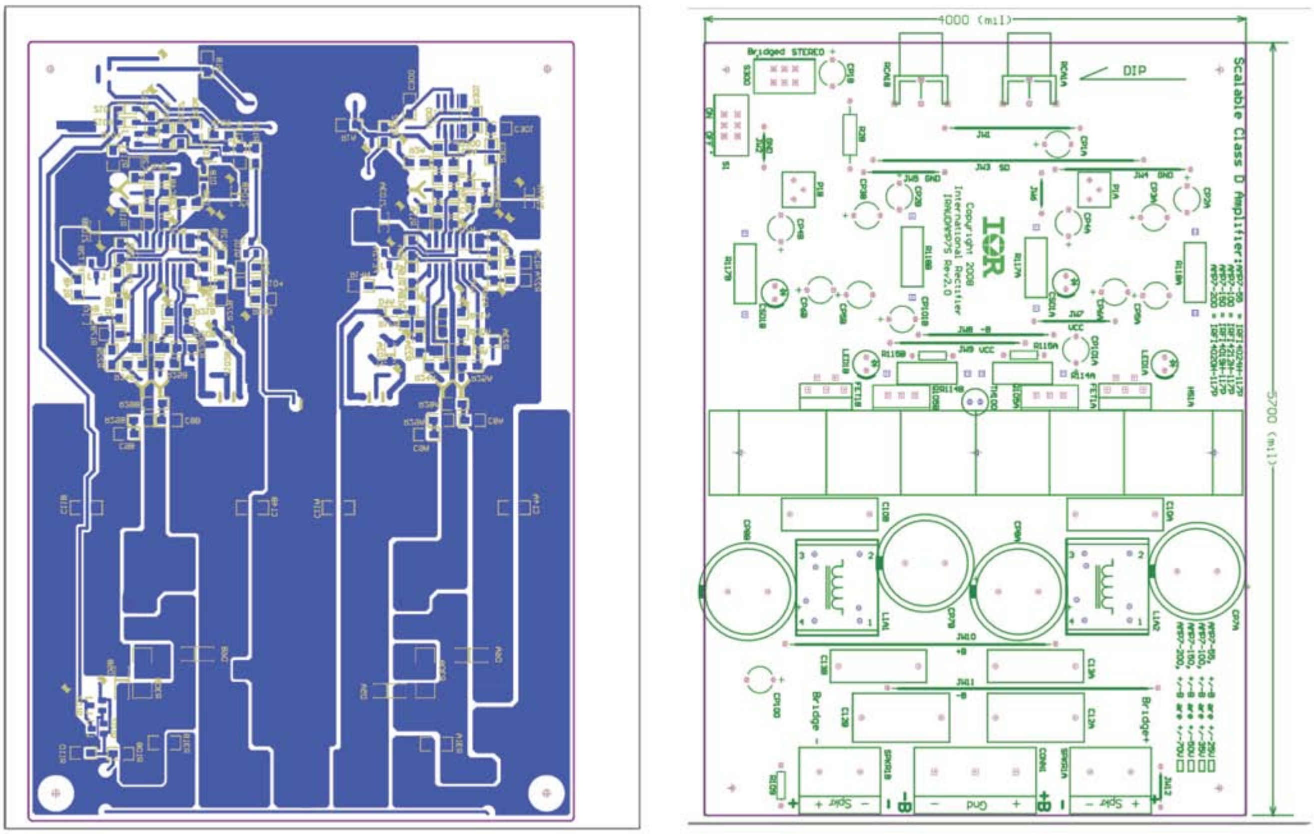

Fig. 2. Picture of well laid out PCB.

The Amplifier Build

I ordered two of the IRAUDAMP7S evaluation boards and a couple of spare IRS2092 driver ICs and brought them into my shop. I built up a reasonable chassis with clear 0.5-in. acrylic material and a simple toroidal transformer as the power supply. I screwed everything down and milled out some slots and holes for connections and assembly (Fig. 3.)

I spent a little time on shielding, but not much—the box is clear acrylic after all! After buying a surplus JBL2404 off eBay and running the class D solution into it for three months straight (post output filter), with a thermocouple glued to the diaphragm to make darn sure none of that high-frequency ripple was causing excessive eddy current loss in those perfect silky windings and causing that glorious adhesive and varnish to shift at all. (PS, I also checked the flux in the gap of that Alnico 5 with a gaussmeter before and after. There was no discernable change, but I can't imagine 2 V of ripple at 300 kHz could cause a whole lot of degauss or eddy current heating into a 2-lb magnetic structure.)

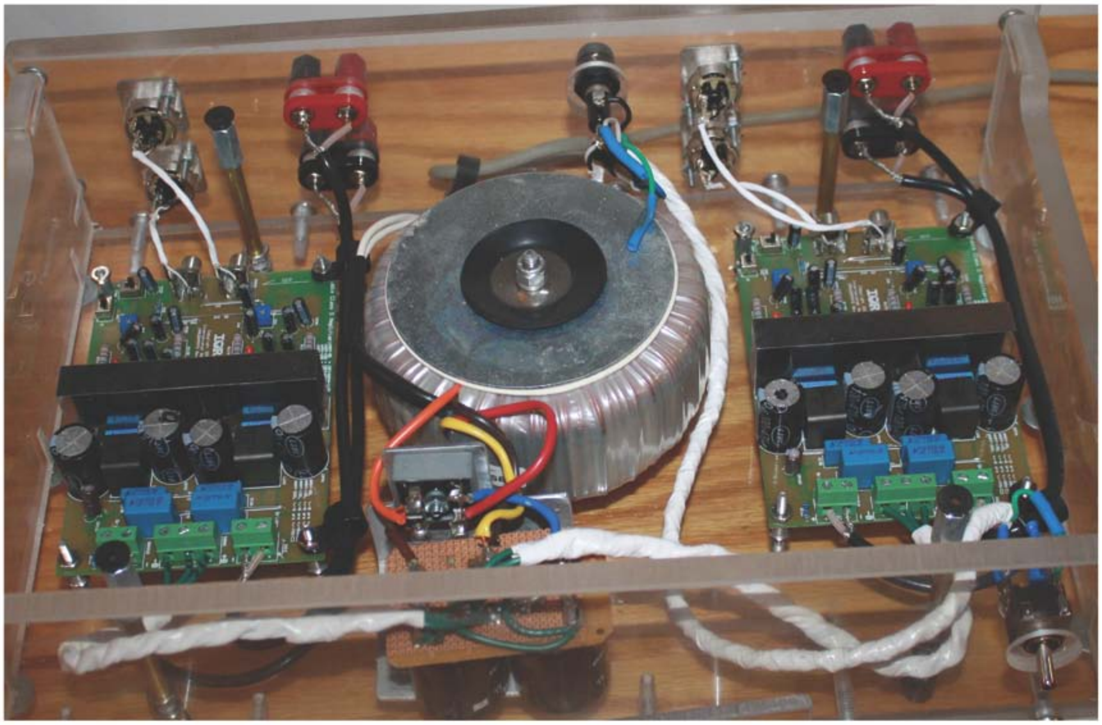

Fig. 3. Close-up view of finished amplifier with IRAUDAMP7S evaluation boards, power supply and chassis (cover edge is in bottom of photo).

Test Results

The THD performance of the IRAUDAMP7S is pretty good. It's certainly not a McIntosh MC2300, but outstanding when the efficiency gain is considered. While it may not impress the deaf skeptics that only see numbers on paper, in a live audition it performs very well. The THD of the amplifier was measured at 4- and 8-Ω loads, both channels driven.

Due to some budgetary restrictions, I didn't initially have the AP audio analyzer available for my measurements, therefore I used my trusty old HP334A and a 100-kHz, 4-pole filter with a Q of 0.7. The distortion and noise that I measured is shown in Fig. 4. Based purely on experience, I know my equipment typically measures THD slightly higher than what is actually present on switching amplifiers. So keep that in mind when viewing these results.

Later, I had the opportunity to measure distortion using the AP audio analyzer. Those results appear in Fig. 5.

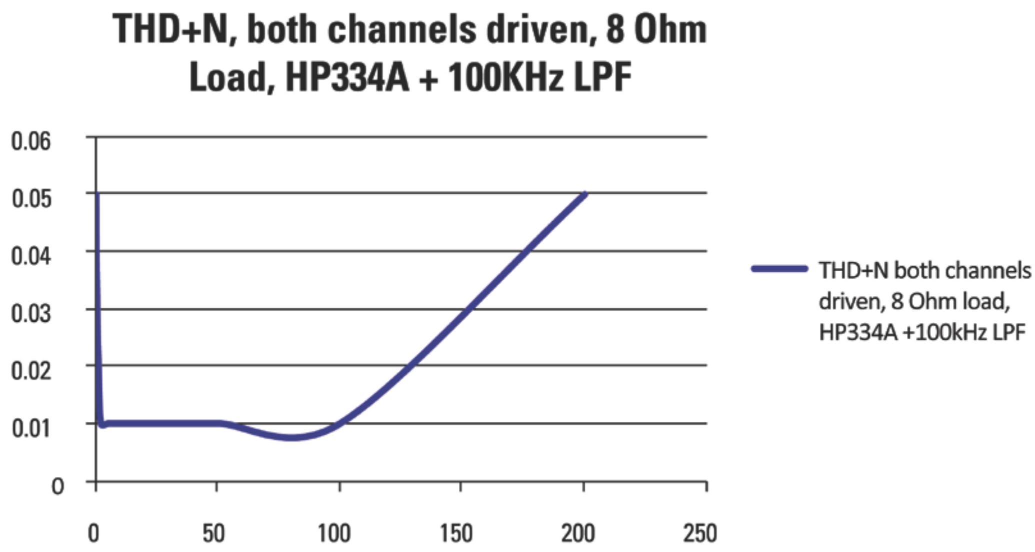

Fig. 4. THD at 8-Ω load, 200 W, both channels driven (measured with HP334A and 100-kHz LPF.)

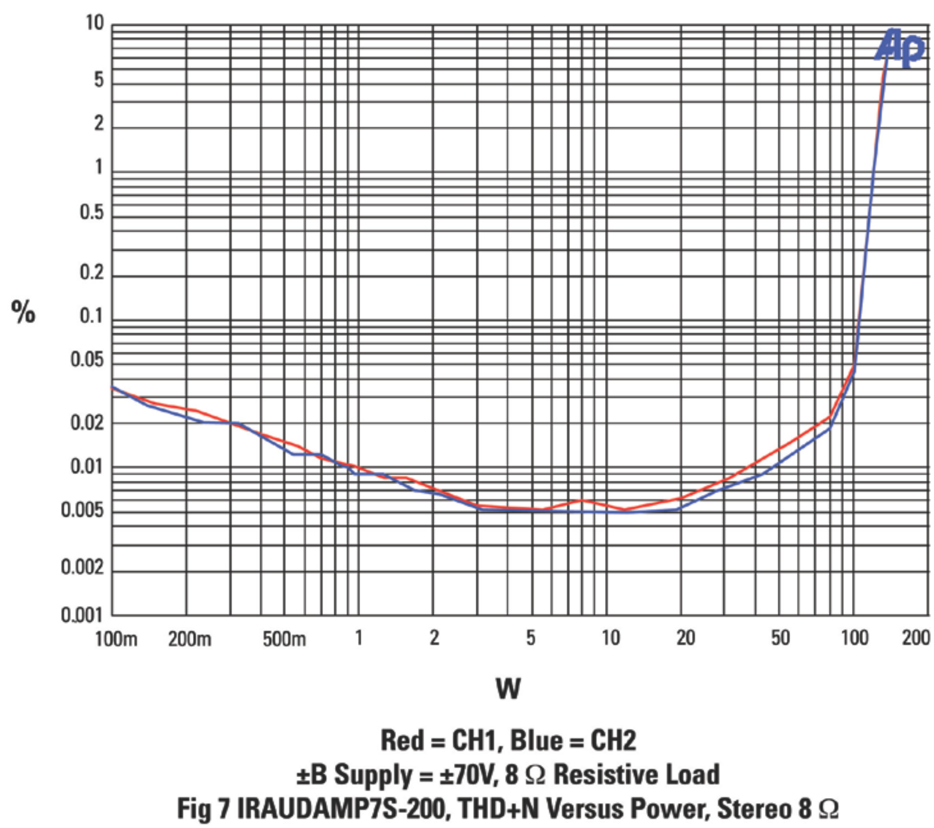

Fig.5. THD at 8-Ω load, 200 W, both channels driven (measured with AP audio analyzer.)

The square wave response of the amplifier was exceptional. The signal does show a little noise. Some of this is due to my nonideal setup, while some of this is simply from the output ripple of the class D audio amplifier. Here is a 1-kHz square-wave input measured at the output terminals of the power amp (Fig. 6.)

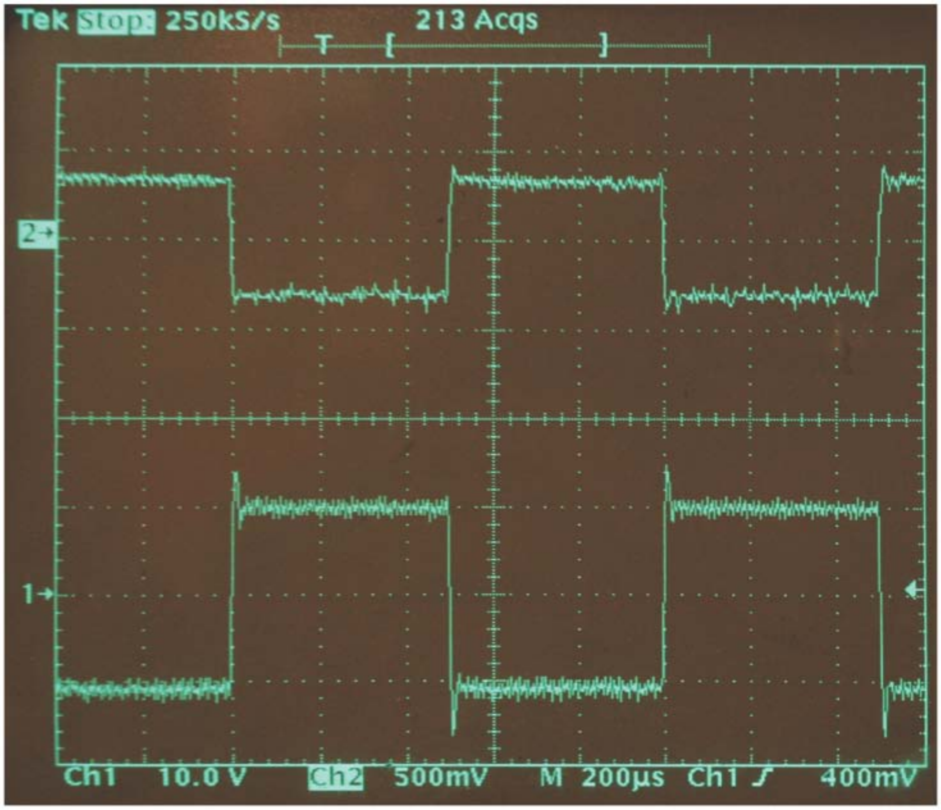

Fig. 6. Square-wave response.

The frequency response of the amplifier was as expected. You can see a little peaking at 8-Ω loads due to the Q of the LC filter (Fig. 7.)

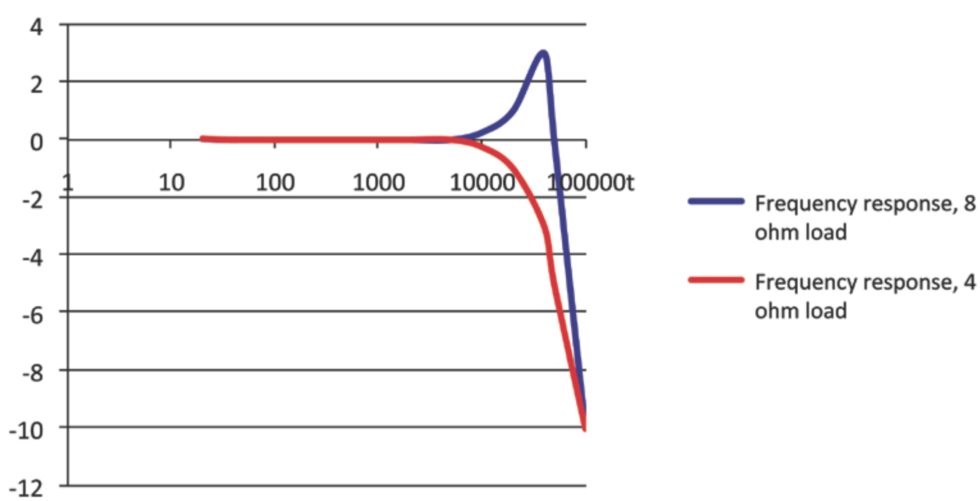

Fig. 7. Frequency response of the amplifier.

The efficiency of the amplifier was exceptional at high power (Fig. 8.)

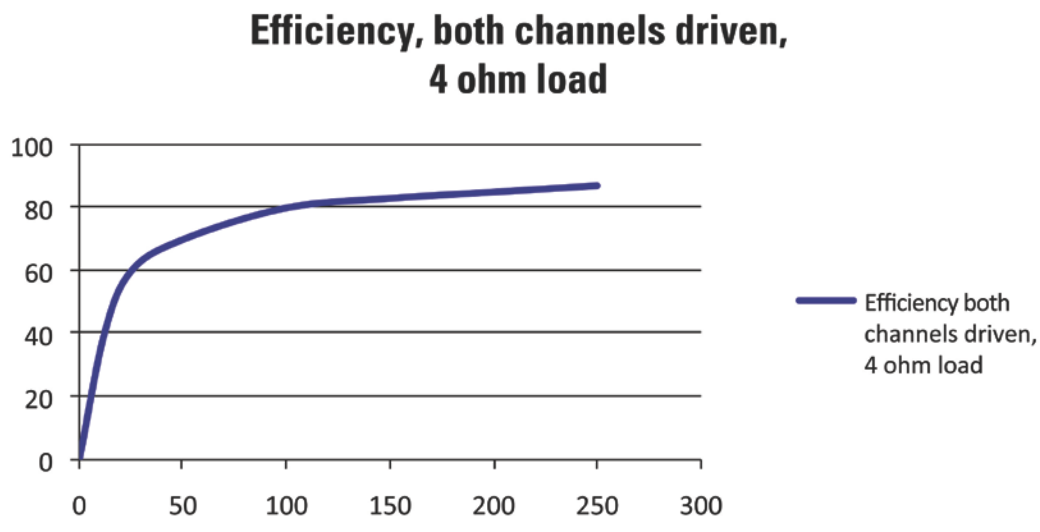

Fig. 8. Efficiency of the amplifier.

The IR class D solution exhibits no exacerbated distortion products at clipping and it clips as well as most any solid-state linear amplifier (Fig. 9.)

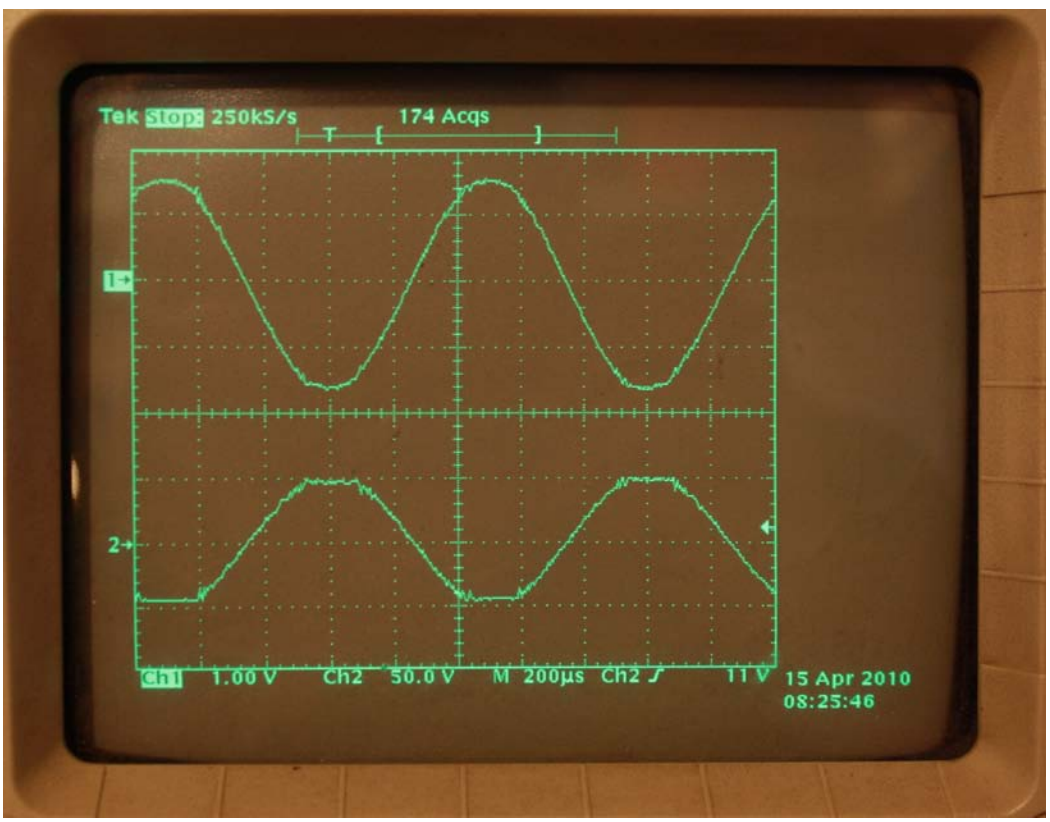

Fig 9. Input and output waveforms for Class D audio amplifier.

The IR Class D audio solution solves the power up popping problem with an innovative front-end amplifier, built in pre-amplifier regulators and some level shifting to the modulator reference point in the power stage, thereby allowing the input to be driven with signals referenced to common. The CMRR of this input stage is fabulous. Flipping the off/on switch several times, I can't discern or aggravate any popping condition on power up.

As for short-circuit protection, the IR solution uses external MOSFETs with internal sensing in the IRS2092 IC. The MOSFETs can be chosen with lowest $R_{DS(ON)}$ to run the most demanding low-impedance loads, all the while maintaining adequate short-circuit protection with the internal sensing circuitry on both the high- and low-side devices. There are also provisions for a thermistor to be placed near the MOSFETs on the small heatsink to provide additional thermal protection. I have no waveforms to illustrate, but in live testing, when the outputs are shorted, the amplifier is safe to itself and any load that may be across the terminals at the time.

Classic Stock Meets High Efficiency

Then I brought the IR class D audio solution into my home theater I plugged my mains, my beloved hand-made Mark Is into the class D audio amp, stereo mode, with my old McIntosh C26 as the preamp (Fig. 10.) Whoa! Just as I'd heard before, there's nothing missing, it's all there. All of the wonderful notes and the broad full sound field of Supertramp 'bloody well right'….it's all there. That 'soulful voice' that Clyde Orange sang of when he recorded 'Nightshift' was there in that warm soothing midrange, wonderful highs, and tight, seamless bass.

Fig. 10. Legendary sound meets efficiency in power processing.

Just for fun, in total distrust of my own ears and head, I brought in some golden ears. After some critical listening I told them that this was class D audio, to which they told me that couldn't be true. I let them touch the heatsink, which is a minimal aluminum slab that holds up the MOSFETs, no forced air, no fans, no cold spray. "It's not even hot! That heatsink is really small and we've been pounding on the amp for an hour! My Threshold (big heavy class A amplifier) would be on fire by now!"

We listened to all of the good stuff. Metallica's "One", Brian Johnson's (AC/DC) "Rock and Roll Ain't Noise Pollution," Stevie Ray Vaughn's "Cold Shot", Boston, the piercing pipes of Billy Corgan from the Smashing Pumpkins and some blues talent like Charlie Love, The Commodores, Jeff Healey and Marvin Gaye—only this time through the IRS2092 driver IC. Everyone agreed that it's an excellent amplifier worthy of mounting in the rack as a permanent fixture.

Further Reading:

1. The IRAUDAMP7S application note can be found at

http://www.irf.com/technical-info/refdesigns/iraudamp7s.pdf.

2. The IRS2092 Class D audio driver datasheet can be found at

http://www.irf.com/product-info/datasheets/data/irs2092.pdf.

About the Author

Paul Schimel spent his early years developing a wide ranging interest in all things electrical and mechanical, investigating and eventually repairing all types of equipment. He was working conventional machine tools and welding by 14. Through high school, he fixed most anything that was breakable including yard equipment, two-way radio gear, power tools, TVs, and stereos. He attended the School of Electrical Engineering at the University of Illinois at Urbana Champaign, specializing in power electronics where he earned a BSEE degree. After this, he spent eight years in successful design engineering roles in consumer equipment including power supply design for projection and direct view televisions and telecommunications equipment including ring generators, battery rectifiers/eliminators, dc-dc converters, and UPSs—both switch-mode and ferroresonant. He then moved on to field applications engineering where he has spent the last six years on power management support and design work for Unitrode/TI, Fairchild, and International Rectifier. He has assisted successful designs from mW to MVA and from prototype stages to finished end equipment. He moonlights in broadcasting, antique test equipment restoration, metal working, woodworking, TIG welding, loudspeaker building, and amateur radio (K5NJP). He holds a commercial radio telephone license, a refrigeration license, and PE license. Schimel holds several patents on magnetic structures for power electronics and novel circuitry.