up

:database:electronics:Audio:Amplifiers:HiFi:Class D:

Article Reference: 36531816A65F145B

001

World Wide Web Links:SGS-Thompson

Date: Fri, 18 Sep 1998 08:06:21 -0500

Original Subject: Re: Question about digital audio pwm

```
FYI: ST (SGS-Thompson) makes some Class D amplifier chips.  "http://www.st.com"

>Apart from the Harris web-site I have not found much on Class-D
>amplifiers.
>How do you find the sound quality of the Harris evalution board?
>Have you pushed out 100W or so through decent speakers with it?
>They use MOSFETs with a rating of 22A, which is a bit underated
>if you're driving a 4ohm speaker with 100W. I suspect they use
>them because they have quite fast on and off times, have you
>tried with higher current devices?
>What kind of quiescent (excuse spelling) current does the amp
>drawn when running off a 30-40v supply?
>I suspect that when the output stage blows it will take the
>driver IC with it as the gates tend to get shorted.
>
>-----== Posted via Deja News, The Leader in Internet Discussion ==-----
>"http://www.dejanews.com/rg_mkgrp.xp"   Create Your Own Free Member Forum
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Article Reference: DC2F5734715B46F9

bridge drive problems

Date: 24 Sep 1998 06:27:28 GMT

Original Subject: Re: Class D Audio Power Amps

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Jim Hellier wrote....

>There are instability ,overheating
>of driver chip problems preventing amplifier reaching its maximum power.
>--

I have used several of the Harris drivers based upon the same HIP process used
in the Harris class D amp chips.  We wasted significant time blowing output
stages because of the terrible noise immunity of the floating driver stages.
If you have to use a bridge/half bridge driver, look into IR's 21xx series.
They don't latch up as easily, and won't blow up with 15v supplies.  Check the
Harris max V spec on the bootstrap power.  16-17v and they will die...If you
have to use the Harris device, make sure you follow all of the app note data
(there are parasitic diodes coupled to the substrate) and it can be a nightmare
to keep them from conducting when the output from your bridge commutates into
the low side FET diodes.  If it swings more than 1-2 v below gnd, watchout!
Ground planes help too and mirror you power / gnd and keep loops to a min. good
luck,

Brian Faley
Shoreline Power Design
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Article Reference: B6A28F0711170BF1

class D FET amplifier

Date: 24 Aug 1996 00:40:05 -0400

Original Subject: Re: What's wrong with Class D amplifiers ?

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Nothing is wrong with Class D amplifiers !!!!. The Engineering group
where I work has just designed a Class D Subwoofer Amp that delivers
200 Watts RMS into 2 Ohms over a bandwidth of 20 - 200 Hz, THD is less
than .05 THD. The Amplifier is based on an all N channel totem pole
output stage, we chose not to use the Harris chip, instead rely upon
the classical approach of creating the pwm by modulating a linear
amplifier, (yes a linear amp, with a bandwidth 100 KHz) with a
triangle wave (35 khz) and using the the other input as the input from
an error amp. The entire design employs one TL072 as an error amp, one
cmos inverter chip as the triangle wave osc and 11 transistors
including the mosfet output devices as the "power comparator". The
output filter is a 2 pole network with a cutoff frequency of 1
KHz. S/N of the design is better than 100 dB A Weighted. Save yourself
the cost and silliness of Harris corps pre hatched "solution" and
design something simpler and superior.
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Article Reference: D237541546CB3625

class D headroom

Date: 2 Mar 1996 00:02:41 -0500

Original Subject: Re: PWM amplifier

```
I experimented with PWM amplifiers back in the 1960's (and wrote some
articles on them in AUDIO Magazine around '63 or '64). Back then, we
didn't really have fast enough power transistors to develop much power,
but it was interesting anyway.

My main problem was that I go in for classical music, for which you need
lots of headroom. PWM amps have no headroom; you must make absolutely
sure that you don't overdrive them or they sound TERRIBLE. The only way
around that is to either use an external limiter (which makes them OK for
some applications, such as a modulation amp for an AM transmitter, where
you limit anyway to avoid going over 100% modulation, but not very
desirable for classical music), or else run the amp sufficiently below
maximum output so that the peaks do not approach maximum output. But in
that case your efficiency is so low that it's questionable whether it's
worth the effort to use PWM.

Pete
pastark@cloud9.net
```

Article Reference: E963D9E4CD9DFF5A

class D link

Date: 26 Jan 1996 16:48:31 +0100

Original Subject: Re: class C or D audio amps?

```
In article <4e2p59$3c4@cloner3.netcom.com> jeffa@ix.netcom.com(Jeff Anderson) write

:>   From: jeffa@ix.netcom.com(Jeff Anderson)
:>   Newsgroups: sci.electronics.design
:>   Date: 23 Jan 1996 13:51:37 GMT
:>   Organization: Netcom
:>   Lines: 8
:>   NNTP-Posting-Host: pax-ca11-08.ix.netcom.com
:>   X-NETCOM-Date: Tue Jan 23  5:51:37 AM PST 1996
:>
:>
:>   I'd like to build an efficient switch-mode audio amp.  Audio output
:>   only needs to be 3 watts or so.  Does anyone know of any good
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:>   references that discuss these, or if chips are available?
:>
:>   Thanks!
:>
:>   - Jeff
Check out the HIP4080 from Harris. It might be a bit of an overkill
for your specification, though. The evaluation board is rated at 200W
on 4 ohm.

In case you didn't know, Harris has a www site at
"http://www.semi.harris.com"

Hope this helps

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```
Luis Palafox                           palafox@goofy.mpi-hd.mpg.de
Max-Planck-Institut fuer Kernphysik
Postfach 103980
D-69029 Heidelberg
```


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------------------------------------------------------------------------------
Maybe I'm wrong. It wouldn't be the first time, it won't be the last!!
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Article Reference: 40F45C39EEDBC02C

class D output filter

Date: Fri, 19 Apr 1996 12:29:40 GMT

Original Subject: Re: class D amp - crossover compatibility

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Richard Dudley <100633.112@CompuServe.COM> wrote:
>Class D amplifiers generate CONSIDERABLE amounts of ultrasonic
>hash, by definition, so you DEFINITELY need a low pass filter on
>the output. Not having one will result in an almost certainly
>fried tweeter ! The crossover circuit will separate out the audio
>frequencies correctly, but not do the ultrasonic removal.
>       You won't receive the efficiency benefits of Class D
>operation without this low pass, since the load on the PWM output
>stage must be negligible at high frequencies.

One problem with the filter is that it doesn't lend itself to changing load
impedances. The Harris app. note shows a design for 4R resistive load and
assumes that a Zobel network is used to make the load look resistive. The class
D amplifier is therefore good when you can tailor the amplifier exactly to the
speaker but not so good if you have to accommodate a range of speaker impedances
and types.

I simulated the effect of changing the load impedance in Pspice and a nominal 4R
looks nice, 2R looks really droopy and 8R has a big peak. Does anyone know a way
around this load matching problem? I have seen commercial switching power
amplifiers (eg. Peavey DECA) quote performance into a variety of loads, how do
they do it?

>       Check out Harris semiconductor's web page
>'www.harris.com' and search for HIP4080A which is their chip that
>does most of the modulation/drive work for you. They have a
>filter on the evaluation board for this chip which you could
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>usefully copy. They got the information to design it from
>'Electronic Filter Design Handbook' by A.B.Williams, McGraw-Hill
>ISBN 0-07-070434-1 ( not a cheap book ! ).

IMHO, the thing that is still missing from the equation is a switching power
supply for powering the amplifier from the mains. It is all very well making a
mega-powered amp with tiny heatsinks, but if you still need an enormous
iron-cored mains transformer and big capacitors you have thrown away most of the
potential advantage of class D.

Unfortunately, I think that any power supply designer would confirm that
designing an off-line switching supply of significant power (>100W) is _not_ an
amateur (or even non-expert professional) task!

Neil

--
Neil McGann
email: webster.house@dial.pipex.com
WWW:   "http://dspace.dial.pipex.com/town/square/ae331/"
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Article Reference: B8F9B7D0ADB269FD

design points

Date: Fri, 19 Mar 1999 16:44:16 +0100

Original Subject: info on Class-D Amplifiers

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Hi Fred

I have read all those discussion points on Class - D amplifiers and I'd like
to inform you that I  have already developed a class - D amp about seven
years ago (for my final thesis at the school of engineering).
Maybe my comments might be useful to somebody.
The design was of medium Quality i.e. THD around 0.4%, Frequency response DC
to 45 kHZ (+- 1dB with RESISTIVE load), P approx 200 Watts into 4 Ohms,
carrier suppression of > 80 dB.
It had a very "classic" design using a single summing integrator for the
forming of the triangle-wave and summing of input and feedback signals. The
output stage was a complimentary design which had the advantage of
simplicity and low cost amongst other designs at this low power (for higher
power however there's nothing that can compete with pure N-channel designs).
Gate drive was done with a circuit almost similar to the one used in the
Motorola AN1042 design but with some simple but effective changes, so that
the amp is capable to being driven to it's maximum output voltage and also
beeing severely overdriven with any signal down to DC without suffering from
latchup. Carrier frequency was 250 kHz.
Despite the THD of 0.4 % it sounded quite nice from a subjective point of
view. These 0.4 % could be made lower with a better layout (there was no
time left to improve this PCB layout).

The whole development process showed me that the most important things when
developing Class - D amplifiers were (in descending order):
1. PCB layout !
```

2. PCB layout !
3. PCB layout !
4. Circuit design
I.e. a great circuit design can be made performing worse than a crappy design by a bad PCB layout.

The output filter could also be a real source of trouble (as Neil McGann mentioned) because it can cause:
-Transient ringing.
-Frequency response depending upon load
-THD generated by nonlinear magnetic-core behaviour
-air cores on the other hand generate RFI and skin-effect related losses

There is a patent held by a British engineer named Brian Attwood, that shows a workaround for these points. The patent is about 20 years old already! To me it seems to be the only reasonable approach to build an analog PWM amplifier (apart from the high quality but much more sophisticated Crown BCA arrangement). The Peavey DECA amp is based mainly on Mr Attwood's patent.

The main points are:

The main output filter is consisting of a 2nd order lowpass filter with the advantage of having not too much phase shift.
Feedback is taken off AFTER the main output filter to include distortion caused by the filter.
Switching residuals are removed by notch filters after the main output filter.
At the output there is the need for the usual combination of a Zobel network and a series inductor as used with "ordinary" amplifiers to guarantee stability with no load or capacitive load or whatever.
The main problem is the feedback loop. It is not so easy to design this feedback loop in a way to achive a phase margin of >65 degrees (or even better 90 degrees) because of the output filter. Single pole compensation would of course be possible but would either give less phase margin or less feedback. The method of choice is a double feedback loop using lag filters. It gives a little more work for the design but the additional costs are marginal (or even less). A triple feedback-loop would give slightly better values but is much more complicated to design. With this double feedback loop the triangle wave has to be generated seperately from the summing network (which can be a source of problems in "classic" PWM designs anyway).

I did some P-SPICE simulations with a model of an amp with a carrier frequency of 250 kHz. The version with feedback taken after the filter showed much better transient performance and less THD (at an order of magnitude of around 10) than the "classic" version with the feedback taken before the output filter (even when using ideal inductors). Even though the Peavey amps use 500 kHz switching and also the papers by Mr Attwood recommend higher switching frequencies than my model uses, it is still possible to use a carrier frequency of around 250 kHz and achieve a quite good PWM - amp.

I think in terms of economy this approach is by far better than any other method that would be able to overcome the output filter problems (i.e. very high carrier frequency combined with high output filter cutoff frequency or any of those sophisticated oversampling/noiseshaping designs). From the quality point of view it could also be superior than the purely digital designs.

But if you are interested in the last ones, then have a look at www.tripath.com. They developed an amplifier principle using some kind of delta -sigma modulation. They claim it being superior than most analog amplifiers and than all PWM amplifiers. But from my point of view this is mainly sales talk. I think they compare their product to small IC amplifiers and multimedia crap when they state efficiencies of 30% for class

B and THD of > 1% for PWM.
I also assume that the really excellent THD figures they state will only be valid for full output power , but will increase with decreasing output power, because it is effectively a DIGITAL amplifier using discrete -time and discrete-value signal processing (the "classic" class - D , i.e. PWM amp is a discrete - time/continuous-value design), giving a finite resolution. They don't give any info about the modualtion scheme they are using but it is some form of a bit stream converter using sample rates of approx 2 Ms/s, as can be seen in their patent. There are also other patents around using such principles.
Their (i.e. TRIPATH's) biggest advantage is indeed the fact that they can offer reasonably priced turnkey products. I.e. ICs for low power amplifiers including the modulator, drivers and output stage (2 channels !). You will only have to connect the output filter (2nd order), power supply and some additional inexpensive circuitry. For larger power requirements they offer an IC containing the same as the smaller ICs except the output power devices. It is usable for amps with approx 150 Watts (x 2!) and uses a two N-channel design.
At the moment these ICs have analog inputs, but they are going to offer versions with digital inputs by the end of '99.

If you are interested in my PSPICE amp - models I can supply them. I've also made one for a bitstream amplifier but it's only advantage seems to be it's simplicity.

Regards

Charles Lehmann

P.S. Please excuse my errors but English isn't my mother-tongue.

Article Reference: 1441C5FFC0CF3589

output transistor drive suggestions

Date: Wed, 7 Oct 1998 23:23:05 +0100

Original Subject: Re: Class D Amplifiers

A few thoughts...

Drive each gate via a short (low inductance) connection via its own 10R resistor,
from a voltage source of +12 and -5V, such that the dv/dt is fast around the gate threshold voltage. This keeps the transistor out of its linear region as
much as possible.
(-5V also means noise is less likely to switch the mosfet on accidentally).

Make sure the gate drive can deliver at least 1.7A (17V/10R), and that it has
plenty of supply decoupling capacitors!
(eg at least several 100nF X7R 0805 multilayer ceramics in parallel with several 10uF 25V tantalums (for both +12 and -5V gate drive supplies)).

Fit 2x 15V protection zeners directly between each gate & source in anti-series, eg...

```
                     15V    15V
            G -----|>|-----|<|----- S
```

to prevent Vgs from exceeding maximum allowed.

Possibly also put a diode is series with each MOSFET output path to prevent the internal (slow recovery!) diode within the mosfet from conducting
eg...
D1 & D2 are fast recovery diodes. Power schottky types are good.

Also fit series R-C (22R, 2n2) snubbers across free-wheeling diodes (eg D2).

```
           +rail
               |
+ switch, similar to below.
(mirror image ish !)
               |
               *------------------------------- >>> to LC output filter
               |
               |
               *------------
               |              |
               |              |
               A             K
              D1            D2
               K             A
               |              |
               |              |
       |  |---   D          |
       |                      |
       |  |<--               |
       |       |              |
G __|  |---*  S          |
               |              |
   ----------*             |
               |              |
               |              |
               *------------
               |
               |
           -rail
```

To help protect the mosfet from voltage spikes, fit a few hundred pF directly between D and S of each MOSFET, and fit a ferrite core in the D1 line
to limit the dv/dt seen by the drain-source.

To clamp larger spikes... (described for lower MOSFET shown above)
Connect a fast diode, anode to D, cathode to a 100nF cap, with the other cap terminal connected to S.
Now connect a 10R 10W resistor between the cathode-cap junction and +rail.
This "spongy clamp" will absorb voltage spikes which are greater than the rail-rail voltage, with the resistor dissipating the energy.

Make sure both +rail and -rail are very well decoupled! (at least one 100nF Polypropylene from each rail to ground if not many of them!).
Copper busbars might help as well!

Check that the upper fet switches off maybe 200ns before the lower switches on and vice-versa, to make sure both don't conduct at the same time!!!

```
Keep all wiring as short as possible.

Make sure the output inductor doesn't saturate!!!


ps,

As the mosfets have an on-state Rds resistance, you can use this to protect
the device from over-current by monitoring if Vds is above a limit while the
mosfet
is enabled. (ignore Vds when the device is switched off as it will have a
very
large Vds!).

Some of this might help anyway  :)

Paul.



Tony B. wrote in message <361B9FEC.25EFD242@dynalec.com>...
>I am having difficulty understanding a specific phenomenon I have been
>observing with a new Class D audio amplifier design.  I am using an
>International Recitifer MOSFET driver to drive an H-bridge and, with a
>series gate resistance of around 33 ohms, the switching waveforms look
>great.  However, there is significant heating of the FETs.
>
>If I decrease the gate resistance or place a diode across the gate
>resistance (anode to gate so that the diode conducts during FET
>turn-off), the heat dissipation is greatly reduced.  The problem here is
>that I get a little notch around the zero-crossing of the switching
>waveform.  As a result, there is significant distortion in the output
>signal.
>
>Any thoughts?
>
>Thanks in advance,
>Tony B.
>
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page by Fred Stewart, fstewart@mediaone.net

Date: Mon Sep 17 08:23:38 2001