Stan's Safari

STAN PONDERS THE RISE OF THE CLASS D AMPLIFIER, AND WONDERS WHETHER THE SNOBBERY OF SOME ABOUT THIS TECHNOLOGY IS DUE TO BE OVERTHROWN AS THE DESIGNS KEEP IMPROVING

There is an arguable case for saying that analogue audio amplifiers have reached something of a plateau if not a pinnacle in their development. And this is perfectly understandable because, after a number of breakthroughs in our understanding of amplifier design in the 1970s and 80s, the claimed improvements in performance have been quite incremental – and those increments have become progressively smaller.

In recent times we've seen designers investigating the thermal effects on components; the impact of vibration on components; and even the minuscule effects of induced magnetism on the signal path. Then, of course, there have been the ongoing "improvements" in component quality. the most relevant in my opinion being the development of custom power transistor packaging to house the naked silicon chip. We've also seen the use of high-tech communications-grade printed circuit boards, and even the selection of unusual solders on the basis that – with maybe a thousand soldered joints – even a minor effect on the signal path could add up to a degradation of the sound.

But then an equal argument could be offered that with so many joints any effect could be averaged out to zero – it all depends on the circuit design and the need for a very careful analysis before conclusions can be drawn.

By contrast. the development of Class D, Switching, or Digital amplifiers – choose your own description – is still at a relatively early stage, and they've seen only limited acceptance by the high-end audio community. On the surface such amplifiers have superb specifications – high-efficiency, low heat generation, virtually zero distortion, a wide frequency bandwidth and so on – so why don't they top every audiophile's shopping list? Well, that bit's easy.

Delve below the headline specifications and some things are no longer so grand. For example the efficiency drops significantly as power output drops and the distortion figures no longer look so impressive. So why have such amplifiers become almost ubiquitous in the rock music industry? Well, that's mainly because of their low weight and size, high efficiency, and because the potential efficiency and distortion failings aren't significant in systems where the musicians are driving them very loud, and thus hence mostly in the top end of the amplifier's power ratings.

By contrast most hi-fi amplifiers spend their lives pumping out a few watts, so their low-level performance is important. Nonetheless the pro audio industry has done hi-fi a great favour in promoting and financing the considerable development work on Class D amplifiers.

Better the devil?

Perhaps one reason why such amplifiers have seen limited acceptance in high end audio is inability of many designers to understand such new circuitry, so there tends to be a reliance on "bought-in" modules sourced from a small number of manufacturers who really understand what can be a tricky technology. I haven't designed a digital amplifier for many a year but I have designed some switched mode power supplies which share the same output stage configuration: namely a pair of high-power MOSFET transistors wired between the supply rails. These alternately switch on and off, and the faster they do so the higher the efficiency.

However each transistor has a small in-built switching delay so if the rate of switching is set too high both transistors can finish up switched on at the same time thus shorting one supply rail to the other. The result is a big bang and the smell of burning. I confess I went through several weeks of burning components before I mastered such output stages.

In fact Class D amplifiers are improving rapidly and one technique that has been trialled is the error correction system conceived by A.M.Sandman in the early 1970s. This is shown conceptually in Figure 1 opposite, where Amplifier A1 is the high power section, known to produce distortion, and A2 is a small 'near perfect' amp whose input is taken from the summing point of A1. At that summing point the signal is:-

Original – (Original + Distortion) = Distortion

This distortion signal is then inverted by amplifier A2 and fed to the loudspeaker: get the values right and this correction signal can cancel out