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BRYSTON EXCELLENCE

DIGITAL AMPLIFIER TECHNOLOGY &ND LOUDSPEAKER DESIGN

THE FOLLOWING is a discussion I happened upon while surfing through 'Audio Asylum' [www.audioasylum.com] on the internet. I would like to thank Dan Agnanos for permission to reprint his words. I thought our readers might find it quite informative to hear about amplifier designs from a speaker designers perspective. On 'Audio Asylum' Dan Agnanos (then with Sony I believe) was answering questions regarding Digital Amplifier technology as it applies to a loudspeaker designer. His speaker design had just recently been rated Class A by Stereophile Magazine.

Dan With His Stereophile Magazine Class A Rated Speaker

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DAN:

"My feeling regarding digital amps - which I have evaluated and researched extensively for is that with the exception of Low Frequency reproduction…. they all suck.

Furthermore, there is just no reason for them except for use in low cost, low end systems. They seem to offer no performance advantages and only marginal efficiency advantages (which pretty much go away with high switching rates or if they are run at less than 80% full power). The more you understand about the intricacies of the design of such products, the more you realize how wrong-headed this direction is.

Why on earth would you design an amplifier which RELIES UPON and GENERATES SWITCHING TRANSIENTS and requires an ADD-ON PASSIVE LP FILTER just to operate! Add to this the fact that the S/N and PSRR are horrible because they are simply switching the DC power rails on and off directly and you get the picture.

Such designs require so many "band aid" repairs and circuit kluges just to fix all the inherent problems, that they just don't make sense. Now compare all of this digital mess to a simple, two stage Class A amp circuit like Pass Labs uses. No switching transients, no circuit kluges, no passive LP filters, etc. I could go on and on about all the problems inherent to digital amplification, but the point is they just don't sound good. And, there's just no reason for them in the high end or even mid-fi markets."

QUESTION:

Isn't the principal of the digital amplifications similar to the way DSD works? Would there be a problem during fast rising transients which also have a large amplitude level?

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DAN:

"There is a huge difference between DSD and digital amplifiers. First is the power level. DSD is dealing with low voltages and currents and can be implemented efficiently and accurately in monolithic ICs. High voltages and currents from an amplifier are quite difficult to deal with, and must be handled with (relatively low tolerance) discrete components for decent performance. At high power levels, you are correct that overshoot and ringing are very significant problems. Also, at these power levels, the amp acts like a fairly efficient high frequency generator/transmitter, depositing high frequency garbage on any nearby audio circuits, cables, speakers, etc. Also, power amps require high capacity power supplies which cannot be designed to have the S/N or freedom from noise that a DAC voltage supply does. And as you know, the cleaner the supply, the cleaner the output. In simple terms this is why amplifiers cannot have S/Ns as good as a DAC or even a preamp. Another difference is in loading. Amps have to deal with highly reactive speaker impedances, while DSD is usually loaded by high resistive impedances, with very controlled, low level current draw. Stability into such reactive loads is another problem - usually "fixed" by the LPF stuck on the output. DSD is fundamentally different from Class D amplification. Typical Class D uses an analog waveform to modulate a switching supply, so it's technically not really digital! I could go on and on, but the technological differences are very substantial. The point I am making is that the amplifier is one of the worst possible points to make the final digital to analog conversion. By far the best solution is a monolithic IC DAC. To add to this argument, analog amplification is probably the strongest link in the reproduction chain - something we can do very, very well NOW. So one must ask, why are we wasting time and effort focusing on the strongest link, while making zero progress at the two extremes (source and speaker), where nearly all the problems lie? Even more ironically, the best examples of "digital" amplifiers sound dreadfully bad compared to the finest analog examples. "Digital" amplification has been around for well over 20 years, yet it has not progressed much sonically. It is, in an engineering sense, a dead end solution to a nonexistent problem... A good engineer should always recognize when a design direction or technology path is wrong. Problems arise when heavy handed corporate directives bully the engineers into doing what they instinctively know is folly."

DAN :
"Yes, DSD does require noise shaping to work. But there are a million ways to do that noise shaping. It's the implementation that matters. Also, the quality or characteristic of the remaining noise is critical. For example, if the left over noise in the pass band is not white and is correlated to the input, you will have serious sonic problems - even if that noise is - 120 dB and below the noise floor of the amplifier. The same is true for PCM systems. An equally damaging problem lies in how the noise is distributed in the stop band. This is why DSD originally caused so many problems with so many amplifiers early on. Now sharp filtering must be used at 50 kHz or lower to alleviate the problem. I hope you get the point."

DAN:
"A better solution than high order noise shaping in DSD would be to utilize much higher sampling rates (e.g., 256 Fs) throughout the recording and mastering process and then down-sample to 64 Fs for the final disc coding. Unfortunately, the rush to commercialize SACD prevented that from happening. I even remember some people arguing this point (unsuccessfully) some 5 years ago. This is analogous to rushing a piece of software to market before all the bugs are out - forcing band aid solutions

down the road. One reason new CDs sound so much better than older ones is that they are recorded and mastered at 24/96 or better before being re-quantized to fit on a CD. Unfortunately, this is NOT the situation for SACD. Nearly all of the professional recording equipment is performing at the same level as the players! Hopefully this will change in the near future.

DAN:
"My feeling is that DSD does have a few benefits over PCM though. I think these relate to the ADC and DAC processes and to the simplification of the digital chain. Probably the greatest single advantage of DSD is its potential for future improvements. PCM is maxed out in this sense. DSD is very far from perfect and has lots of room for improvement. If you listen to a lot of live music, you would agree that it certainly needs it! Despite its shortcomings, SACD is the best format we have now (equal in my mind to the best analog tapes). Hopefully, though, it will continue to improve in quality, just as CD did during its lifetime. I remain optimistic."

QUESTION:
However worse off digital amps will sound compared to a properly designed analog amp, it does have its marketing appeal. Many ordinary customers, to my dismay, really don't care at all. I don't mean to say that it is justifiable for companies to make product for business sake only, but all that extra money from cash cows do sometimes end up where it does benefit in audio, now and then - in the case, developing of high-rez digital format to take-over the long-in-the-tooth 16-Bit/ 44.1kHz CD.

DAN:
"I think you're right - the single greatest attribute of digital amps is the marketability and "buzz" among those who are clueless but recognize and are enamored by all things "Digital"." Despite my slamming

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of digital amps, I do believe that - properly designed (99.9% are not) - they can be better for LF reproduction. First of all, they're using regulated supplies so are less prone to "softening up" with heavy bass output. It's also possible to do some clever digital feedback (which we are investigating now). Low frequency reproduction is where most of the power dissipation and efficiency benefits of these amps are useful and beneficial. Finally, HF garbage is more easily dealt with if the reproduction bandwidth is limited to < 500 Hz. Stability is still an issue but can be helped a lot if the speaker load does not include any passive crossover circuitry (which powered subs are free from). So, in my estimation, digital amps can be an effective solution for LF reproduction."

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