Stan’s Safari Part 6

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I’m not one for participating in the many hi-fi forums now spread across the internet. First there is the matter of time; and forums are a great work displacement activity; and secondly there is the blow to self-esteem at finding that the more verbose regular contributors are labelled as “gurus” whilst those who occasionally dip in are, regardless of experience and knowledge, labelled as “novices”. None the less I took a look at the HiFi Critic forum where I came across a discussion entitled “CD Players. Much progress over the years?”. Now this is a question most of us would struggle to answer because life is about progress and of necessity the magazines focus upon what is new; exciting; interesting and each product is only compared to its contemporaries. Yet this topic did strike a chord with because I have found that a great many CD players are no better than those of 10 or 15 years ago; indeed a great many are far worse. Faced with the risk that I might be turning into a grumpy old man I have raised the topic with other well established designers, indeed with our esteemed editor, and found that I am not alone in my thinking.

Not long afterwards serendipity reared its head and my listening room was used to audition a number of “high-end” CD players for a group test. I provided the coffee and sat back in anticipation of a day’s musical pleasure only to be sorely disappointed. With one honourable exception I wouldn’t have given any of these expensive players house room. So it came to pass that when the formal listening was over I dug out a player of my own design from 1984 which was powered up whilst we retired to brew more coffee. Freshly fortified we auditioned the said player and discovered that it was musically in a class of its own. Yes, some of the fine detail was a tad masked but we are talking about capacitors and other components which are far from their best having laid dormant for 25 years without a flicker of electricity. This rather left the reviewer in something of a quandary; a quandary he quickly solved by pretending that the whole incident had never taken place. And so I decided I should explain some of my own thoughts about CD player design over the next issue or two. Inevitably in questioning the degree of progress I've needed to make comparison to one or two models from the past; and I've chosen models of my own design, not in a burst of egotism but because I know their designs intimately and, more importantly, why they were designed the way they were.

One of my worries about current player design is the sheer amount of processing that seems to take place with outstandingly complex over-sampling digital filters with their "pre-echo" artefacts. My first experience of the problems of "pre-echo" came not through CD players but when I helped developed the Digital Room Correction system for B&W Loudspeakers in the early 1990s. In simple terms this system took measurements over a portion of the room, then generated extra signals which would compensate for frequency response problems caused by the acoustics of the room. One of the dangers of such a technique is that it is possible in some room locations to actually hear the correction signal ahead of the original signal and when that happens you do sense that something is fundamentally wrong although you haven't got a clue what it is. We put a lot of effort into overcoming such problems but ultimately, for me at least, the system failed because of what I might term "over-processing".

The system certainly worked; you could put loudspeakers into the proverbial bathroom and it could deliver an acceptable acoustic environment for your listening pleasure. Yet something was lost. Even with 32 bit processing there was some rounding of the calculations needed and even with the instances where very little room correction was necessary it was apparent that in comparing "before" and "after" something was stripping away much of the vitality of the music. And that was happening with a superb processing system using software developed by great minds such as Peter Craven. The product never made it through to the shops but it did serve to make me wary of too much processing of the music signal in the digital domain. Yet since that time we have seen a veritable explosion in the way digital processing is used and recently the spin-offs from home cinema multi-channel sound processing have seen 24-bit 192kHz sampling becoming commonplace for high-end players.

Processing power is quite cheap so the temptation is there for designers to do more and more to the music signal in the belief that things can only get better. Yet I've always had doubts about exactly what happens to the music signal when passing through the complex digital filtering.

My concern is; if we are to have a perfect recording-playback format then the output signal must replicate the input signal in every way. In other words the well known straight wire without gain. Therefore whatever signal processing takes place at the Analogue-to-Digital Conversion stage must be exactly replicated by a "mirror image" at the Digital-to-Analogue Conversion stage. I had this thought in mind in the early 1980s when I designed the Cambridge CD1; the first two-box (or even three-box) CD player and one of the first true "high-end" players. At the time I measured several professional studio A-to-D converters and tried to replicate "mirror images" of their frequency and phase responses. For good measure I added a few filters that I felt were better sounding compromises of the real-world filters. The CD1 had a choice of 7 replay filters and you could certainly hear a difference with most owners soon discovering which filters worked best for particular CDs in their collections. Unfortunately this idea added a fair amount to the manufacturing cost of the player and the concept was not, to the best of my knowledge, repeated elsewhere.

At the time I thought I was onto something and perhaps I was because in the view of many the CD1 was the best sounding CD player of its era. But I soon learned that there were other consequences of the conversion process that could not be easily compensated for at the playback stage. Take the effect of jitter, those small variations of the clock frequency. In the A to D conversion process the jitter will cause a sample of the waveform to be taken either too early or too late relative to the last sample and so the value will be wrong. And this has more effect upon high-frequency signals because the waveforms are changing rapidly so with a fast rising waveform a slightly delayed clock will give a value that is too high. That value is then set numerically in stone and whilst processing can mitigate the error through averaging the fact remains that you can never ever re-create exactly the same waveform.

There is no doubt that the filtering used in a CD player, be it analogue or digital, does have a major influence on how the music sounds. Let's take a moment to remind ourselves why we have filters. In the course of reconstructing the analogue signal from the digital samples we also produce so-called Nyquist images in the band above half the sample rate frequency; above 22.05 kHz in the case of CDs. So to attenuate these so-called alias frequencies we need an analogue filter which can drop the output, say, 80dB between 20 KHz and 22kHz; a filter which could be made but which would ring to high heaven and screw up the phase response. Philips came up with an alternative technique which was to increase the sample rate by a factor of four using a technique called over-sampling. The Nyquist images are then shoved up to above 88 kHz where they are easy to filter out of the signal. Since that time we've had various iterations of over-sampling; up-sampling and re-sampling. Each is a different technique but all result in an output signal which has a faster sample rate and usually more bits of information. How can this be so? Well at its simplest we are gaining the benefits of averaging where the unwanted noise and least significant bit errors fall in level relative to the signal. For example if we average a group of 256 consecutive 20-bit samples we can add 4 bits to the resolution of the average, producing a single sample with 24-bit resolution.

Probably the high spot of my own player design work was the one that never was; an intended collaboration between Bowers & Wilkins and SME. At the time B&W was headed by Robert Trunz, a personable guy who loves his music and who is always open to new ideas. I was consulting to the company and one day he mused that what he wanted was the CD player equivalent of the SME Model 30 turntable. I sketched some ideas and started work on a prototype after which Robert had another inspirational thought – why not get SME to build it! Drawings were produced; costings were prepared; we even talked to Linn Products about building some electronics, but in the end it was too much for the two of us to get off the ground without taking valuable resources away from B&W's core business of making loudspeakers. None the less I did go ahead and finished a prototype which, whilst lacking the sheer elegance and the “fit & finish” of a SME made product, did actually work. I recently came across odd components and some of the drawings and in the one shown here it can be seen that the player was built like a record turntable with a proper suspension; a lightweight platter with an oversized motor which locked the actual linear velocity solidly to the servo voltage; and, get this, a vacuum suction system to hold the disc firmly in place.

Unusually the laser head was on top of the disc in the manner of a tone arm and at start up the arm moved across to the disc which was then scanned radially by the laser/optics assembly in a short stub arm. The whole assembly was very stable and measurements made on the servo system of the prototype showed that the focus corrections made were a fraction of those made on a conventional player. As a result of this and other design features this transport had extremely low reading errors so the processor’s error correction circuits could be almost entirely devoted to the errors caused in the process of moulding the disc.

Originally the plan was to sell the transport with a Digital-to-Analogue Converter then being built by SME but we designers always think we know best so I proceeded regardless with a new design based upon what would have been the conversion stages of the Cambridge CD1 mk.II. The concept of 16 times re-sampling was carried over from the CD2 but with considerably more complex processing and without any conventional digital filtering. Each channel was then fed to 4 banks each of 3 matched multi-bit DAC chips giving a total of 24 DACs in all. Finally given the now high sampling rate (705.6 kHz) there was no need for the feedback integrator found in just about every CD player. In addition it had finally dawned on me that the master clock seemed to be quite important and that disturbances of the crystal by, for example, vibrations, did degrade the performance. I wasn’t aware of the importance of minimising jitter as such but I did have an intuitive feeling that something had to be done to improve things. Not being an expert in the design of stable high-purity oscillators I took a pragmatic approach and lifted the design of a quite superb oscillator straight from a piece of Hewlett-Packard test equipment. All in all extremely high-end; particularly for 1991.

To my ears this system sounded totally natural and unlike any player I'd designed before. It wasn't perfect because the absence of any conventional digital filters did leave some potentially difficult artefacts about the place which, oddly enough seemed to help improve the D-to-A conversion process, but that's another story. The outcome was enough to convince me that it was better to pass the signal through the digital stages with a few rough edges than to process it to death no matter what the theory said. Of course there are some audio 'purists' out there who build CD players with absolutely no filtering and accept that there is a mirror image of the audio signal in the band above 22.05 kHz. The mathematics tell you that you risk horrendous inter-modulation distortion in the amplifier chain and tweeter as well as a mass of unwanted signal components scattered across the audio band courtesy of the real-life performance of the DACs. Well I have heard such players through extremely wide-band and fast amplifiers and there is no doubt that there is something to be said for the philosophy. The music seems to retain its vitality and flow, although for me there is something wrong about the sound during the louder passages. Although the simplicity of this approach has a certain emotional appeal it has to be rejected the problem of the Nyquist images frequencies is not solved; rather the listener goes into denial and pretends they aren't there. Still it wouldn't be the first time that hi-fi enthusiasts have denied reality !

Yet the industry has embraced even more processing with up-sampling and up-conversion being seen as essential parts of the specification. This is in part a consequence of some inexpensive chipsets being developed for DVD players which have a PCM format with a maximum resolution of 24 bits/96kHz. To ensure compatibility this system has to permit the easy upwards conversion of 16 bit/48kHz and 16 bit/44.1 kHz signals. This can be a more than trivial task. When changing the sampling rate, it is easier to work with a multiple of the original signal's sample rate. A two times over-sampling system will double the sampling rate, by adding an easy-to-calculate extra sample value in between each actual sample. However for an up-sampling filter to create a 96 kHz digital signal from a 44.1 kHz signal, some awkward mathematical calculations are necessary (the ratio being 2.1768707 or thereabouts) and you can find yourself wondering if one mathematician's filter algorithm becomes another man's "best guess". Certainly there is some approximation or "guessing" taking place because when I've fed the same digital audio signal to several processing chips and made the conversion to analogue using the same DACS; I can clearly hear the difference, indeed I can sometimes hear things that are just not right. And that just shouldn't be so.

I suspect I'm not alone in having my doubts about some of the digital filter technology which is regularly adopted by CD player manufacturers. I've mentioned already the designs which eschew all filtering but at the other extreme I've auditioned a player which tears up convention in the design of its filters (the Meridian 8082.i player) and again there is a difference which can clearly be heard. And you know what? There are still people out here in the Cambridgeshire fens who say digital filters are either "1s" or "0s" so they must be perfect!