Measuring Clock Jitter Spectra

Paul Winser

After reading David Field's article (Issue 30) on his CD clock upgrading, I wanted to have another look at this topic. Of the half dozen or so people whom I know have tried a DIY upgrade of this kind, all have reported noticeable improvements. There's also been quite a lot of stuff written in Audio Electronics magazine, among others, on the subject of jitter over the last couple of years, and equipment reviews often now include measurements. I thought it would be interesting to see if the clock upgrade which I did a few years ago (Issues 9 & 11), and which has been progressed much further by David and others, could have any measurable effect. Not to validate the subjective results or otherwise, but to help confirm that our efforts have had the desirable objective effect, that is to lower the level of clock jitter at the DtoA chip as much as practical.

To make good measurements of the magnitude and spectrum of clock jitter, you need the right bit of kit, like a spectrum analyser as used in magazine reviews. Well I don't have access to one of those (please let me know if you do!) but I do have access to a fairly high-end scope which features a FFT mode which can be used as a spectrum analyser. The resolution is not very good, but it turns out that it is good enough to do useful measurements: as we'll see it is possible to distinguish a good DAC converter clock from a bad one.

The Tests

The best point in any system to measure clock jitter is right on the pins of the DtoA converter chip itself, because that is where it matters (you can argue that it is irrelevant everywhere else). My modified Philips CD player which some of you may have seen is a single box without SPDIF or AES interface. It contains both the original clock circuitry and my add-on clock design, and I can switch between the two merely by fiddling with DIP switches and swapping round a few cables. This makes it fairly easy to do 'before and after' comparisons. I clip the scope probe on the clock input (and associated ground) of the TDA1541A converter. The clock frequency here is approx. 5.64MHz. At the time of this test I also had access to a friend's converter which happens to be a professional rack-mount unit, but uses the same CS8412 decoder & CS4329 DtoA chips that lots of other HiFi units do. On another occasion David Field and I looked at his modified transport & DAX converter, although with a slightly different measurement technique. For all these units a spectrum of the clock signal applied to the converter device was measured with the scope.

Collecting The Data

Setting up the scope to extract the best data from the limited resolution available is not straightforward so I'll go through it for the record. If you want to cut to the chase, skip to the section below 'The Results'. I used a Tektronix TDS784A sampling scope that has a 1GHz bandwidth and records 4Gsamples per second, 8bits per sample. Bet I put that speed to good use, huh? Nope. For maximum frequency resolution of the FFT you need to collect as many clock cycles as possible to input into the FFT, and that means using the lowest sampling frequency that can reasonably reproduce the clock waveform. The frequency resolution is defined by the scope's record length, not its bandwidth. I used 25Msamples/sec and a record length of 50,000 points for the FFT input.

For best signal to noise ratio on the active FET probes I adjusted the vertical scale to just not clip the incoming waveform, and applied the 20MHz first-order low pass input filter. Next step was to set up the FFT, and the important thing here is choosing the right convolution window for the type of measurement. This is to do with trading off frequency resolution with amplitude resolution, and after a bit of experimenting in the session with David, we settled on using the Hanning window (the other choices being Box, Hamming, and Blackman-Harris).

The final step is to hit the 'average' button and select a large number of FFT results to average together – several hundred does it. This is necessary for two reasons: firstly the scope has only

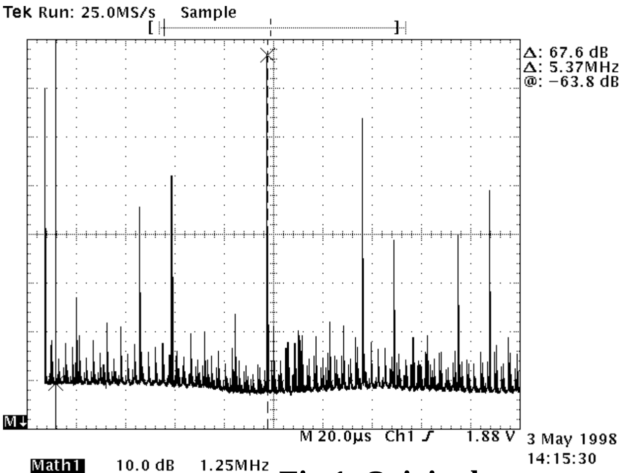

Fig 1. Original

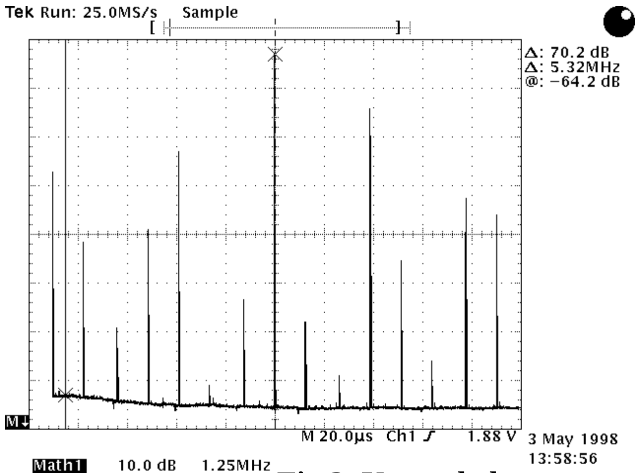

Fig 2. Upgraded

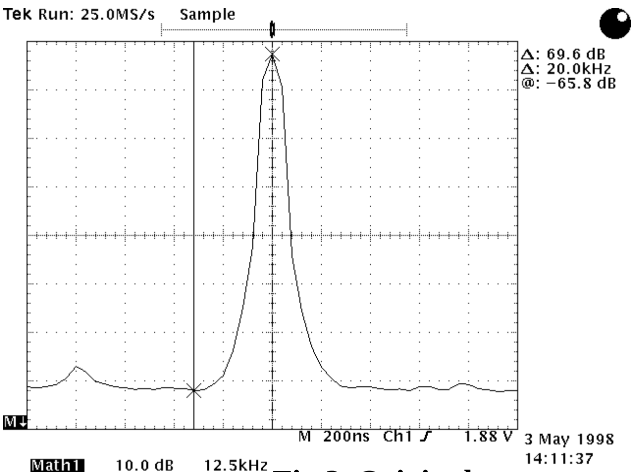

Fig 3. Original

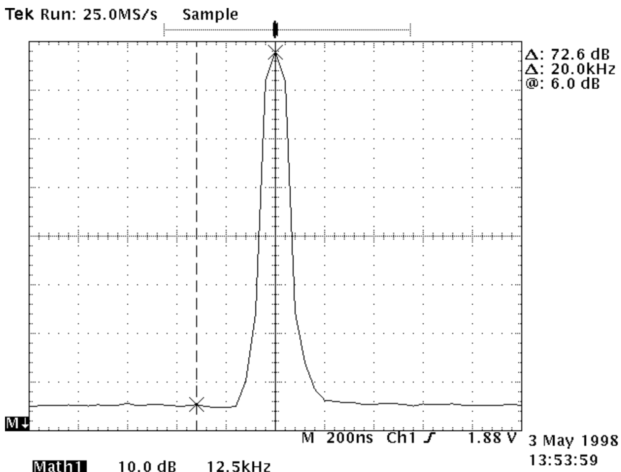

Fig 4. Upgraded

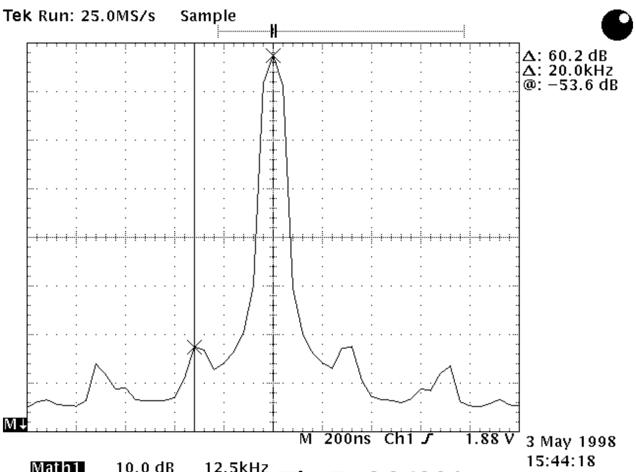

Fig 5. CS4329

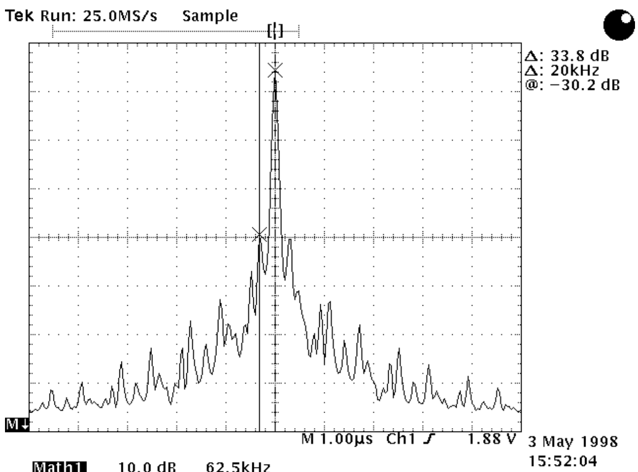

Fig 6. CS4329

an 8bit sample resolution, giving at best 48dB dynamic range, way too small for our needs. Secondly because we are sampling at a frequency (25MHz) which is only a few times that of the clock we are measuring, each individual FFT will have high noise in its frequency domain. By averaging a large number of these low resolution FFT's, you get a high resolution one, just as if you had sampled the input much more accurately.

One thing to note about the spectra produced by this measurement – the clock input contains lots of high-order odd harmonics, because it is a square wave. Some of these, especially 3rd but also some 5th and above, pass through the 20MHz input filter and are sampled at 25MHz. Those that are above the Nyquist frequency of 12.5MHz (that's all of them) will produce alias products in the resulting spectra. We would expect this to show as spurious frequency spikes above and below that of the clock itself, and that's exactly what we see. At first it's a bit alarming to see lots of big components around the clock fundamental, but they are just an artefact of the measurement process and are not really there, honest. It is fairly easy to distinguish these from actual rubbish in the spectrum which may be causing some jitter. It's also quite easy to predict where the alias components should show up, and then 'delete' them from the picture, although I have not done this in the traces here.

I did also attempt to look at the clock jitter in the time domain using a delayed trigger and infinite persistence display but the jitter levels were about the same as the triggering uncertainty of the digital triggering system of the scope, and I didn't get any meaningful results.

The Results

First up is my player set to use the original clock circuitry (the oscillator is built into the SAA7220 digital filter chip). Figure 1 shows the complete spectrum from DC to 12.5MHz, with the 5.64MHz clock fundamental just left of the centre graticule, and topped with a cursor cross. We also see several large spikes which are artefacts as a result of the aliasing in the measurement process as described above. In addition there is a 'grass' of low-level components throughout the spectrum. The noise floor of this grass relative to the clock fundamental (the difference in levels between the two vertical cursor lines with crosses) is 67.6dB. Now lets compare this with Figure 2 which shows the upgraded clock system (ECL oscillator, balanced transmission to DtoA chip). We still have the aliasing artefacts, but the grass has all but gone, and the noise floor is lower too at –70.2 dB. So far so good.

What really matters when we want to know about audio jitter is not the overall spectrum but only that part close in to the fundamental – that's where audible modulation might occur. Figures 3 & 4 are zoom-ins of Figures 1 & 2 respectively to look at the spectrum within +/- 60KHz of the fundamental. We see that the upgraded clock in Figure 4 has a narrower spike and also a noise floor 3dB lower than the standard clock at –20KHz from the fundamental. We can be pretty sure that this picture relates to lower jitter degradation of the DtoA process. Note that these traces are pretty close to the resolution limits of the measurement and a more detailed result would be found from better equipment. Still, it's good enough to show the difference.

Now lets look at the measurement of the Mytek Digital Workstation20 professional converter. This was driven via its SPDIF input from a PC sound card. The spectrum of Figure 5 (again close in to the fundamental) was measured on the SCLK pin of the CS4329 DtoA chip. Figure 6 shows a wider spectrum of the MCLK pin signal, which is the important input as far as jitter is concerned. Crystal Semiconductor's data sheet states that the single bit DtoA design of this device makes it insensitive to clock jitter, which is just as well as the clock spectra here are surprisingly poor, worse than the standard Philips consumer unit. This unit, plus others based on this chip sound pretty sweet, though, so the jitter insensitivity claim could well be true. Certainly in general it is more of a problem in multi-bit DtoA's than in the high oversampling single bit converters such as Crystal's.

So What?

It is worth considering that this last pair of spectra might be representative of many units that use the almost universal CS8412 decoder for SPDIF inputs. Those that use multi-bit DtoA chips may have an inherent jitter problem. One exception to this is David Field's Audio Synthesis DAX, which contains an additional phase locked loop after the CS8412 to attenuate jitter before it gets to the DtoA (I believe it cascades a PLL that has narrow loop filter bandwidth with a PLL that has wider bandwidth – the combination attenuates both incoming and VCO generated jitter). We measured David's clock after this second PLL, using his transport with his high quality clock source upgrade. The spectrum was very clean, similar to the upgraded unit without any PLL's, so it looks like Audio Synthesis has done a good job.

According to the datasheet, the CS8412 does not attenuate incoming jitter at all below 25KHz (depending on loop filter components). From there it's a first order attenuation until a second pole at 80KHz. The fact that the CS8412 seems to have quite poor jitter performance, either internally or in its ability to attenuate incoming SPDIF jitter (especially in the audio range), probably accounts for the success of the numerous add-on jitter reduction units designed for inserting between transport and DAC unit. Anything that reduces (or even simply changes the characteristics of) the signal input to the CS8412 will most likely change the jitter spectrum of the DAC, perhaps for the better. It should be remembered that theCS8412 spectra here were measured with a PC sound card used as the SPDIF source, and I would expect it's clock quality not to be good. Of course the same effect could also account for the differing sound of CD transports themselves…

It would be interesting to measure more combinations of units. If you want to eliminate the most common jitter source entirely, the solution is easy: do not use the SPDIF interface. Anyone curious about the clock quality of their DAC, and who fancies a day out to Bristol, give me a call.

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