'Measurements in Digital Audio'

Listening to music is easy. Appreciating music is easier still. Understanding the intricacies of how the music made its way from the concert hall to your front room is altogether more complex.

The measured performance and specs of any given component in the playback chain are often used to help people understand what to expect from it and sometimes what they actually sound like.

However, both reviewers and consumers can often find themselves focusing on one number that, in reality, doesn't actually tell you very much about the musicality of that particular hifi component. This is because you can get widely different products in terms of how they operate but with very similar scores. For example a THD+N of .003% on two boxes from different manufacturers might persuade you that they sound the same, but in fact their sound may be polar opposites.

Part of the problem lies in the fact that certain kit has eccentricities that defy single measures. This means you really need to consider performance in a number of areas before making a judgement on it. In our opinion the most important measurements in a **digital audio system** are as follows:

Linearity

Poor linearity results in a loss of fine detail and a system that is not musical. When looking at linearity we are talking about a change in the output being of the same level as a change in the input, and because of ***noise*** this is actually very difficult for standard measurement equipment to measure at very low levels, for example -120dB0 and below. Therefore a whole series of FFT's at varying frequencies (1kHz and 2.5kHz on different channels, 6kHz and more) will tell you the whole story about the linearity of any given unit.

At dCS in our production environment we drive our equipment with 24 bit data and put the output through an FFT analyser (for a DAC this would be via an ADC) and then compare the results against a limit file with a set of upper and lower limits. We are looking for a clean signal at the correct level, a clean noise floor and very narrow sidebands on the signal skirts, which is indicative of good jitter performance. In the dCS design environment we will again use FFT techniques, but in this case we will use a more iterative approach to measure the changes in the FFT caused by changes to the design against the required changes we are expecting.

Jitter Rejection

The word clock tells the DAC *when* to convert the audio sample to an analogue voltage. Any variations in its timing accuracy results in distortion of the output signal – i.e. the music. This is bad!

If jitter is periodic, sidebands appear either side of signal frequency – these have no harmonic relationship to the music and can be manifested in 'hardness' and 'glare'. If jitter is noisy or random, this can result in 'smearing' of signal energy and an increase in noise floor which masks fine detail in the music.

The measurements that help us to understand performance of a unit in terms of jitter rejection are quite varied. Firstly, you can measure the internal clock signals inside the DAC, and by looking at them with a spectrum analyser, you can verify that the clocks are clean.

You then need to analyse any possible clock contamination on the audio output of the DAC. We do this by stimulating the DAC with a high-frequency signal (jitter manifests itself proportionally with the signal frequency), and carefully examine via FFT if the noise floor is modulated with the signal, then we look for any sidebands around the signal that indicate correlated jitter. We have to do this at all sample rates, as the bandwidths involved will vary.

The key difficulty with this measurement is that the various results of the FFTs will give you various results in terms of picoseconds for sideband jitter, but how audible it will be depends a lot on where the sidebands are and how correlated they are – it is worth noting that most listening tests tend to favour low-frequency, uncorrelated jitter as being preferable to higher frequency correlated jitter.

There are some measurements systems in use that are based around the work of the late Julian Dunn, whereby the stimulus signal is a high level sinusoid at sample rate/4, with a low-level square wave laid on top. The idea behind this is that this signal causes the most problems with an SPDIF/AES style differential-manchester encoded signal, as bandwidth limitations in the cable cause data-related distortions to the waveform. As the signal is precisely aligned to the sample rate, quantization effects are effectively eliminated. The test then looks for sidebands around the signal and the level of the noise to generate a "jitter number", in picoseconds.

The problems with the measurement today is that the jitter it is attempting to excite is no longer present if using a different style of interface (e.g. USB, Firewire, Ethernet). Also, in these cases because the DAC clock and test equipment clock no longer have a defined relationship, the quantization effects will also be different than the test was defined for.

Finally, these types of tests have a problem in that if the device under test is sufficiently noisy to obscure the sidebands, the unit is given an unjustified perfect score for jitter.

Filter Performance

Digital audio is full of trade-offs. Which trade-off is best in terms of filter choice? Unfortunately, there is no "perfect" filter that will be optimum for all recordings. In essence the DAC filter you use affects amplitude response, phase response (transient performance) and image rejection.

For example, recordings made with poor filtering and lots of HF noise are likely to be improved with an asymmetrical filter, whereas those with good filtering will not cause the reconstruction filter to ring, and so the phase shift is unnecessary. At dCS we believe that it is therefore important to offer a choice of filters, so that the user can choose a solution to suit their music and tastes.

This variety of filter options means it is important to measure the flat signal bandwidth, the cut-off frequency and the image (or alias) rejection. It is also worth noting that filter characteristics become less of a factor at higher sample rates.

Analogue Output Performance

To measure analogue output performance effectively we are looking for a low noise floor (dCS DACs typically measure about -110dB0 in the audio band), low harmonics (particularly second and third) but also ensuring that higher order harmonics are not present.

The best measurements to determine the performance of an output stage are Output Impedance, Common Mode Rejection (for a balanced output stage) and Output Levels at full scale.

Common mode rejection is not often measured by reviewers but it is important because it is key to the operation of a balanced output stage.

USB Performance

USB is becoming the de facto standard interface for connecting a computer to a DAC or digital audio system. The benefits of using an asynchronous USB transfer protocol over the standard adaptive USB method are another discussion entirely but in the context of measurements firstly it's important to know whether the device you are using does actually support asynchronous transfer as spec sheets and the use of asynchronous rate converters can lead people to believe that a DAC is asynchronous when this is not the case.

Determining how a DAC operates in USB mode can be achieved with Apple's USB Prober and a Mac computer but unfortunately now to get access this to this requires you to be a member of the Apple developer forum. There are other similar tools out there so once the true nature of the device has been established repeating the same jitter performance tests on USB as with AES/EBU should give you identical results. What is important here is good jitter performance and bit perfect data.

Which measurements, if any, are frequently quoted but misleading?

Frequency response measurements are often reeled off by those in the industry, but crucially they often miss the most important aspect of a digital filter response – as it approaches the Nyquist frequency, and the image response past Nyquist. This is vitally important because it tells us an awful lot about how the digital filter was designed.

Similarly THD+N measurements quoted are often hard to read into – distortion products are more audible than noise, but you cannot tell the proportion of each harmonic, and the THD+N rarely has a bandwidth associated with it. Single figure jitter measurements are also largely hard to correspond to a sound quality, for reasons outlined earlier.

What are the main measurement instruments and what are their limitations?

Oscilloscopes are popular with those looking to measure audio equipment and they are fantastic assuming they are up to the job and you are mostly interested in time-domain measurements, but they are not very good at frequency domain measurements, so for instance you could measure noise with a scope, but not knowing the spectrum of the noise makes it a largely pointless measurement.

Spectrum analysers are an excellent tool for frequency domain measurements, but again they are not useful for time-domain measurements.

Popular Audio analysers (e.g. Audio Precision, Stanford SA-1, Miller) offer the ability to do lots of quite revealing measurements very quickly, but if you wish to do a measurement that falls outside the scope of the equipment this can prove to be very difficult – for example if you have a new interface (e.g. USB or Network Audio) or different signal format to PCM (e.g. DSD) that you wish to test you can run into real problems.

What Test and Measurement Systems do dCS Use In-House?

In house dCS has designed some bespoke test equipment (called AutoTest), which we use for both design and production testing. A typical test station consists of the following items:

- A PC for running test scripts and capturing and analysing data, running bespoke Windows™ software
- A Test ADC – a modified dCS904, with modifiable gain and input switching
- A Test Generator – a high-precision digital test signal generator, based around programmable logic to allow us to generate test signals on any kind of interface at any sample rate we desire
- A Test APX – a unit that provides an interface between a digital audio source and the PC. Again based around programmable logic to allow us to adapt it to new sample rates and formats as required.

By having full control over the software and hardware of the test equipment, we can very rapidly modify the kinds of test we do, and we can guarantee the quality of the results we get.

Additionally in-house we have a selection of Oscilloscopes, Spectrum analysers and a Stanford SA-1 which mean we can check our results against a well-known test reference. All in all this range of equipment and flexibility in measurements allows us to future proof a critical part of our R&D function.