Graham Boswell and Ian Dennis

The two founders of Prism Sound still drive the technology of the manufacturer. They talk about the wordlength issue, true resolution and why not all convertors are created equal.

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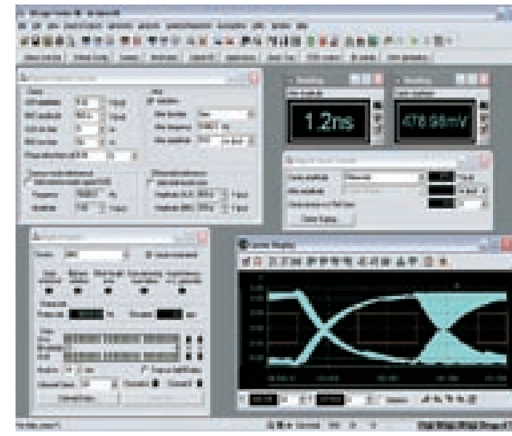

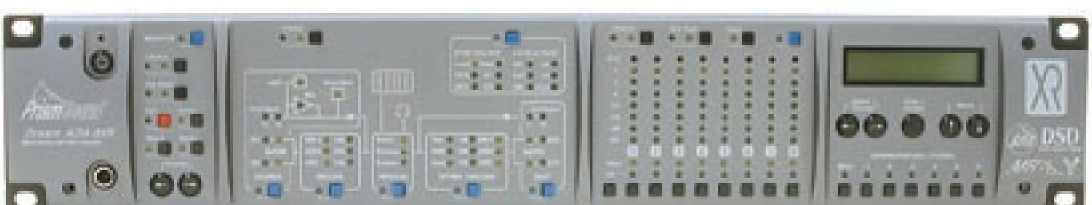

PRISM SOUND WAS originally founded in 1987, the brainchild of engineers Graham Boswell and Ian Dennis who first met when working at console manufacturer AMS-Neve in Cambridge, England. The concept was to develop an R&D consultancy specialising in digital audio applications.

By 1991 the company had grown from two engineers to a team of 12 and the Prism Sound brand had also grown from an initial business start up to a brand known for performance, quality and reliability, which it still commands today. One of the prestigious projects Prism Sound undertook at this time was the development of a large commentary and communications matrix, which was implemented by the BBC at the 1992 Barcelona Olympics. This product was a market leader for many years in the intercom market for television applications.

1992 saw the arrival of Prism Media Products, a company set up to develop, market and sell 'Prism Sound' branded professional audio products. The result was a range of products that put Prism Sound on the map and established it as a force in professional audio. The current range encompasses ultra-high resolution A-D and D-A convertors, including the ADA-8XR, AD-2 and DA-2, analogue signal processors, including the MEA-2 stereo equaliser, MLA-2 stereo compressor and the MMA-4 4-channel mic preamp, and test and measurement through the DSA-1 hand-held AES-EBU transmission analyser and dScope Series III analogue and digital audio test system. Prism Sound also produces multichannel networked digital audio and data recorders for monitoring and logging applications.

Today Prism Sound, continues to provide R&D consultancy to outside clients while still expanding its own portfolio of products.

What's special about Prism Sound products?

IAN DENNIS: A lot of different things, but in summary it's probably 'attention to detail'. We have tended to adopt a cost-no-object approach to each critical element of the box, whether it's clock generation and recovery, analogue signal path, isolation, PSU or the convertor element itself. We also like to provide all the features we can think of!

What were the significant technological steps historically that has brought conversion to where it is today?

ID: In the early days, when audio conversion technology was not well explored, digital audio in general got quite a bad name as a result. Dealing with basic things like adequate dynamic range and linearity made a huge difference in those days. Next of course were considerations of clocking and jitter susceptibility. In terms of the low-level convertor technology itself, advances in chip fabrication technology have brought us the possibility of monolithic sigma-delta solutions with good front- and back-end linearity and sufficient real-estate for a really good filter. But you can still do it better 'the hard way'!

What's the single most misunderstood aspect of conversion?

ID: It's become a highly technical subject in recent years. While most people could understand the principle of a base-rate ladder or successive-approximation convertor, today's sigma-delta technology, as well as dither in general, and sampling jitter effects require more mathematical capability than is common. DSP is integral to all convertor technology nowadays. On a practical level, a couple of things keep coming up: using the best box in the studio as clock master is usually misguided, since good boxes are equally at home as master or slave whereas bad boxes are usually OK as master but perform poorly when slaved. Also, the practice of setting digital output wordlength to, say, 24 bits when feeding a 16-bit input and thus creating undithered truncation distortion is still popular.

GRAHAM BOSWELL: While we're on the subject of misunderstandings, what about wordlength? It is still said that 16-bit resolution is not enough, but the 'resolution' is not limited by the wordlength — the noise is. With a good convertor topology (as Ian has described) and correct dithering, resolution is preserved to incredibly low levels. Of course, this has to be true, as most of the delta-sigma quantizers around today sample at nothing like even '16-bit' wordlength although they do run at blisteringly fast sampling rates up in the megahertz range. I think the trade-off between sample-rate and wordlength and the associated techniques of noise shaping are often misunderstood resulting in erroneous statements such as the example above. If it is not wordlength that matters — what is it? Well that is where Prism Sound comes in — we make convertors that have highly extended resolution.

All convertor manufacturers use largely the same sorts of chips so all convertors are largely the same — discuss.

ID: At the very top end there are still bespoke solutions out there that are unique, and bring some singular benefits. But I suppose in the high-end market as a whole, nearly everybody is using the same handful of very good chips. But there are still things that can make a huge difference outside the chip, perhaps clock generation and recovery being the most critical. But most of the other things that sort the sheep from the goats aren't really convertor-specific — they're

things like uncompromising analogue design, PSUs, interference elimination and, of course, feature set.

GB: These factors inevitably affect low-level accuracy and resolution — so just because there is a so-called 24-bit chip in the box you have no guarantee of truly high-resolution performance.

Sampling frequency or bits — which is most important?

ID: We see these as only two of many 'bottlenecks' in the ultimate conversion system (and these particular two are often dictated by the practices of the industry and so are often beyond our control). We often hear the observation from users that a really good convertor at 44.1kHz sounds better than a workmanlike one at 192kHz, for example. In general, for most applications, both might be argued to be adequate nowadays, with other factors dominating.

For example, current flagship chip performance is about -117dBFS unweighted noise and -110dBFS full-scale THD+N. In theory, we won't need to go beyond 24 bits until one of these drops below -142dBFS. Never say never, but there are significant boundaries of physics that will prevent this for the foreseeable future, particularly in the case of THD+N. As regards sampling frequency, there were a lot of limitations in 44.1k and 48k systems resulting from the proximity of the desired upper band-edge frequency and the Nyquist frequency; the 'transition band' for high-end systems was vanishingly small, resulting in the need for ambitious filtering that could rarely be adequately implemented. Once we were at 96kHz, these problems were largely resolved. I guess this may have become a marketing question, what do you think Graham?

GB: I think that this question also brings up the issue of the trade-off between sampling-rate and wordlength that we touched on earlier, which is part of the debate about what bandwidth ultimately the finished product should have. This is really a question for our users and we will make what they ask for. A really well-made recording sampled at 44.1kHz is hard to beat, but on the other hand as technology has provided us with 96kHz sampling, why not use it? It is unlikely (though not impossible!) that this could be detrimental. And then there is DSD/SACD. That is why the ADA-8XR is modular — we provide all the flavours.

What's the Prism Sound recommended 'best practise' procedure for 44.1/16 CD production?

ID: Unless you want to archive for the future in some hi-res format, you can't do better than a great A-D conversion at 44.1k, 24-bit for all analogue sources. Mixing in the digital domain, but with a system where all the math is looked after correctly, and delivery to 16-bit with a high quality WLR (noise shaping) system like SNS2 at the final stage. The practice of converting at higher sampling frequencies and downsampling later can't really inherently add anything, but it can really make things worse unless the downsampling process is beyond reproach, which it usually isn't since it requires a lot of processing power.

Why as an industry are we so obsessed with high sampling rates and more bits when most of the product is now listened to through data compression algorithms?

ID: I fear that those days may be coming to an end. There will always be a few of us striving for ultimate quality, but it's not as many as perhaps it once was. On the other hand, the recent universal acceptance of DAW technology has been very good for the convertor business, since it's easy to demonstrate the advantage of third-party boxes over most of the standard offerings.

GB: Actually I suspect that the big electronics companies are keen to see the cycle of re-invention continue so we keep buying new music players, phones and TVs and I suspect that the increasing popularity of high-bandwidth Internet connections (2M to 10M) will allow audio quality to be rediscovered to some extent.

What's the next great leap in convertor technology, where is it heading?

ID: Who knows? I suppose that this will probably end up being a format-marketing matter again. I'm a little sceptical about this, because there are still so many opportunities to improve what we currently do within existing formats. There have been quite a few 'quantum leap' technologies proposed recently, and the ones I know have all been either a quantum leap in the wrong direction or else largely irrelevant! But something exciting is sure to come along.

GB: I hope we can get back to something as good as Vinyl ......tk........tk........tk