Benchmark DAC1

It's not a brand that is that well known outside of the US but recent distribution developments will soon make it a name in Europe. ROB JAMES reports on that most innocuous yet most vital of things — a digital to analogue convertor.

EVERY SO OFTEN IT is a good idea to re-examine the things we take as read. One of the biggest threats to the maintenance of high standards, in audio as in other areas of life, is complacency. Sometimes, words we happily bandy about are defined as something rather different. When the DAC1 arrived on my doorstep from a US manufacturer I'd never heard of, the first thing I did was look up the meaning of 'Bench-mark' in my trusty Shorter Oxford English dictionary. Not for the first time, the original meaning came as a bit of a surprise. Turns out a bench-mark is a surveyor's mark, often chiselled into stone. It signifies the start or other point in a series of levels used to measure altitudes across a landmass. Now I've had the opportunity to evaluate the DAC1 this seems a singularly appropriate name for the manufacturer.

After objective measurement, the starting point for serious evaluation of recorded sound or sound equipment must be the monitoring system and environment. A lot of time and money is lavished on the room, loudspeakers and amplifiers. Assuming access to digital reference material of impeccable quality the only major variable is the analogue to digital convertor. The DAC1 is intended to be a reference quality, 2-channel, 96kHz 24-bit answer to the question, 'how can I be sure my A to D conversion is transparent?'

Physically, the unit is an imposing, black, 1U, half-width box with built in power supply. Two units may be joined side by side and bolted into a rack.

Ideally, the DAC1 will be the last device in the signal chain before the power amplifiers or powered monitors. To this end a front panel gain control is provided as an alternative to calibrated output. This gain control also controls the two front panel headphone jacks, driven by Benchmark's HPA2 headphone amplifier. XLR balanced, BNC coaxial and Toslink optical digital inputs are selected by a solid front-panel toggle switch. The XLR balanced outputs are factory calibrated to give +4dBu at -20dBFS with the unbalanced phonos at -10dBV. Ten turn trim pots allow these levels to be increased by up to 5dB or decreased by up to 15dB.

Internal jumper selected pads, affecting only the balanced outputs, decrease output level by a further 10, 20 or 30dB. A 3-position toggle switch selects between Calibrated, Off or Variable for the rear panel outputs. Further internal jumpers allow the front panel input select switch to be disabled and the input to be permanently set to any of the options, also the coaxial input can be terminated (default) or not. Despite the absence of external Word clock input, Benchmark claims multiple units will be phase accurate to within half a degree at 20kHz.

As you would expect with a device costing UK£799 (+ VAT), the manufacturer's performance figures are impressive. However, as most people are well aware, the figures do not tell the whole story, especially with digital audio. Immediately impressive is the graceful muting when the power is lost or a digital input is connected or disconnected. I know this shouldn't happen but it does and with high monitoring levels this is most welcome. The DAC1 is very fast. Lock is achieved in far less than a second from power up, or a change in sample rate.

The real test is, of course, the sound. I used a variety of challenging material I know well. This was ripped into Wavelab and played out of a domestic soundcard directly into the coaxial input of the DAC1. I used Marantz monoblocks and Leema Zen speakers since this is the most analytical combination I have. Imaging proved to be astonishingly solid and I heard some fast auto-panning in the background of one track that I'd never noticed before. But for me, the most impressive aspects of the DAC1's performance are its transparency and transient response. Again, I heard things I had not head before. Later, after reading the manual, I think I know at least one reason why. Unlike many other designs the DAC1 does not mute when the data is all zeros so nothing is lost when a fast transient follows silence.

From everything I heard during the review period, I believe the DAC1 redefines the entry-level price point for reference digital to analogue convertors. If anything, I found it more convincing than some convertors asking more than double the price of admission here. A bench-mark indeed.

Ultralock

The deleterious effect of excessive jitter in digital signals is well known and reasonably well understood. Jitter phase modulates the audio, which creates sidebands above and below every frequency present in the original. These sidebands are not musical because they are not harmonically related to the original sound. This makes jitter induced distortion more objectionable than mere figures would suggest. Another jitter-induced problem is degradation of the digital anti-alias filters used in oversampling convertors.

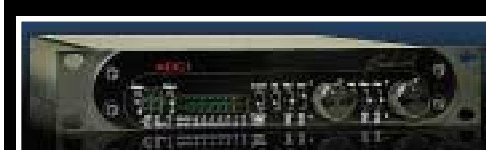
Benchmark has its own answer, dubbed 'Ultralock'. Interface jitter is a feature of all digital audio interfaces and Benchmark's solution uses a two-stage PLL (Phase-Locked-Loop) and isolates the conversion clock from the digital audio interface clock. Instead of phase-locking the conversion clock to a reference clock, Ultralock varies the convertor oversampling ratio to achieve the correct phase relationship to the reference clock.

Of course, this technology cannot remove artefacts already present in the digital signal, e.g. those induced by an A to D convertor with poor jitter performance.

PROS Super league conversion; convenient operation; versatile; highly tolerant inputs.

CONS Nothing I can think of except I can't afford it.

EXTRAS Initially known as the ADC1, the ADPre1 is a two-channel, 24-bit, 192-kHz A-DC with line, instrument and microphone input settings. The mic preamp claims THD+N as -110dB (0.0003%) measured with 40dB of gain. A 1dB noise figure is said to be maintained with gains as low as +30dB. Front panel level controls provide input gain adjustment between -3 and +65dB without an input attenuator.



The preamps have balanced 1/4-inch TRS patch capability and can be used independently from the convertor and vice-versa.

Main output sample rates are 44.1, 48, 88.2, 96, 176.4, and 192kHz, with a word length of 24-bits. The main output signal is available simultaneously on XLR, BNC, and 192kHz capable Toslink connectors. Output signal format choices include AES, SPDIF and ADAT.

A simultaneous secondary output provides 44.1 or 48kHz with switch selectable 16 or 24-bit word length for safety backup recordings.

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