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Benchmark Media ADC1 A/D Converter

FIELD TEST

by Barry Rudolph

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We've been waiting more than two years for Benchmark to release its half-rack companion to the DAC1 2-channel digital-to-analog reference converter, but the wait is over. The 2-channel ADC1 is a high-resolution A/D converter for recording, broadcast and either portable or studio DAW rigs. Both the ADC1 and DAC1 operate from 44.1 to 192 kHz, and offer up to 24-bit resolution.

The ADC1 starts with a built-in power supply that will operate worldwide from 95- to 285VAC mains. Overall construction is very good with a tight-fitting, heat-sinking cabinet; the ADC1 runs warm, so give it some breathing space. It has an anodized front panel and surface-mount components on a single circuit board. Both the digital and analog connections use gold-pinned XLR and BNC connectors, and each output is DC-isolated, ferrite toroidal transformer-coupled, diode-protected and current-limited. If a faulty cable shorts one output, the others will continue to operate.

LOW-NOISE ANALOG INPUTS

To maintain a full 0dBFS digital output from the ADC1, the unit's analog "front end" is optimized for the best signal-to-noise and THD+N performance from any audio input level source from -14 to +29 dBu. Analog Devices' AD797 audio op amps are used for the low-noise first gain stage, the subsequent differential amplifier and the variable, final gain stage. There is enough gain here to directly connect and record synths and other semi-pro sources.

Two variable input gain control paths with a 20dB range for the L and R channels are provided on the front panel: two 41-detent control knobs, and two recessed, 10-turn trim pots. Each path has a separate toggle switch to select between them, and both share a common three-position coarse gain range switch with 0, 10 and 20dB positions.

The trim pots are used for calibration to standard studio operating levels such as a console's stereo bus output. The knobs would be used to maximize the gain structure of any external analog signal chains, such as a mic preamp/EQ/compressor chain. This is a very useful and practical feature.

HOW IT CONVERTS

The ADC1 samples all audio at 218.75kHz/24-bit using AKM Semiconductor's latest AK5394A ADC chip. The ACD1's sampling circuit uses an oversampling ratio of 32:1--exactly 7 MHz (32 X 218.75 kHz). At 44.1 kHz, the ADC1 has an effective oversampling ratio of approximately 159:1 (7 MHz/44.1 kHz). Benchmark chose 218.75 kHz to prevent aliasing of signals around half the sample frequency. Sample rate conversion from 218.75 to 192 kHz and lower rates removes these aliasing components.

There are two Analog Devices AD1896 192kHz converter chips. The AD1896 is the first sample rate converter to exceed 130dB S/N ratio and it has jitter attenuation at all frequencies above 3 Hz, with more than 100 dB of attenuation above 1 kHz. Two SRC chips are used: one for the main output and another one for the aux output if it is set to a different sampling frequency; otherwise, both outputs are bit-for-bit the same and use identical drivers and isolation transformers.

To make a quick copy, you can record at 192kHz/24-bit using the main output to your DAW and simultaneously use the aux output to feed 44.1/48kHz with a TPDF-dithered, 16-bit word length to a CD burner or DAT machine. Triangular Probability Distribution Function, or TPDF, uses random numbers as a dither noise source instead of noise-shaped dither as used in mastering. This TPDF dither noise is added to the 24-bit audio before truncation to 16 bits, and it's a good choice for safety backups and demo CD-R recordings.

The ADC1 has four digital outputs--very useful for feeding several different digital recording systems in your studio at the same time without repatching. These are a single 24-bit, XLR balanced, AES/EBU output; two 75-ohm, coaxial unbalanced BNC connectors that spit either 24-bit AES/EBU or S/PDIF at all rates (BNC to female RCA adapters are provided); and one 5-mm Lightpipe Toslink Type-F05 optical that supports AES, S/PDIF, ADAT, ADAT S/Mux 2 and ADAT S/MUX 4.

THE WORD ON CLOCKING

Benchmark's UltraLock system technology uses a clock derived from a fixed-frequency crystal. The A/D conversion clock signal is completely isolated from the external signal received by the AES/EBU, S/PDIF, ADAT, world clock and super clock interfaces. The ADC1 produces digital audio at all times, and a poor-quality clock reference will not degrade the system's jitter performance--even during the presence of errors and interruptions or if the sample rate status bit is set incorrectly.

The clock input receiver always measures the sample rate of the incoming clock signal to synchronize the main output. This is called Auto mode, and the ADC1 will follow any vagaries of the clock signal such as sample rate change or type of reference signal. In Internal mode, the unit will act as clock master with all devices locking to it. There are BNC connectors provided for both word clock output from the ADC1 and external word clock input.

AT PLAY WITH THE ADC1

Setting up and using the ADC1 could not be easier. The well-written manual takes you through all choices of clock source, sample rates, bit depth and the main/aux output configurations using the toggle Mode switch. The ADC1's configuration is confirmed by a matrix of nine LED indicators. After optimizing the input gain controls to accommodate your analog signal's level, you're good to go. I liked the meter because it's digital and post-conversion and indicates all clips--even a clip of only one sample. The three-position meter toggle switch selects peak hold function on/off and resolution of either 6 dB or 1 dB per step.

For my evaluation, I fed identical audio from the studio's API console to the ADC1, an Apogee Rosetta 200 and Pro Tools|HD 192 converters. I recorded monaurally at 24-bit and at both 48kHz and 96kHz rates into Pro Tools. At that point, I wanted to see if I could hear any differences without the "distraction" of stereophonic sound.

I connected the ADC1's word clock using good, short 75-ohm cables and gold-pin BNC connectors to the Apogee and HD 192 (Digidesign's Sync I/O unit) external clock inputs. Likewise, I used proper 110-ohm AES/EBU digital cables to connect the digital audio outputs of the Apogee and Benchmark units to the AES/EBU inputs of the first two HD 192 units in the studio's Pro Tools system.

All of the converter's outputs instantly phase-locked from the ADC1's clock signal with no problem, and after a careful 1kHz level setup for all three systems using Pro Tools' metering, I was ready to listen to three simultaneously recorded tracks: one made with each converter and played back through the HD 192's D/A converters, patched to three equally set mono faders of the API console. I repeated my listening test after I switched the faders around on the API console and changed the HD 192 outputs to nullify any slight differences in the console's individual input modules or the HD 192 D/A converters.

I used a single B&K 4011 cardioid microphone placed three to five feet away to avoid any possibility of capsule overloading and an API 512C preamp. No other processing was used to record a variety of sources, including hand percussion (tambourines, blocks, bells, etc.), vocals, acoustic guitar and piano.

All three converters sounded great, and the sonic differences were tiny. But at the 96kHz rate, I was able to discern extra clarity with the ADC1 I would characterize it as a clearer, more transparent low midrange with smoother high frequencies. The HD 192 sounded good, but was slightly harsh in the highs most noticeable on tambourine something I would resolve with a different mic and/or tambourine choice. The Apogee was also wonderful-sounding: smooth high frequencies but slightly thick in the low midrange. This was noticeable within my acoustic guitar recording--especially when playing chords.

Apart from recordings that are tracked live, a lot of today's records are built one overdub at a time. When you consider that each of those overdubs might use the same A/D path, the converter's quality and clock--like the microphone and analog signal chain--become more important than ever.

I'VE BEEN CONVERTED

The Benchmark Media ADC1's superior down-conversion design made possible in part by the latest chip technology, the unique UltraLock™ low-jitter clock, optimized analog input circuitry and the four multiformat digital outputs makes it a worthy studio tool that will sound great for years to come. I highly recommend it as a way to instantly upgrade the recording quality of any studio.

Price: $1,775 MSRP.

Benchmark Media Systems, 800-262-4675, www.benchmarkmedia.com.

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