EQUIPMENT **PROFILE**

MARK LEVINSON REFERENCE DIGITAL PROCESSOR NO. 30

Both the transistor and CD, despite offering many technical advantages, often sounded worse at first than the technologies preceding them. In each case, subjective critics overreacted by condemning the new technology. Technical critics overreacted by claiming that the new technology's measured performance was superior and that the subjective critics had to be wrong. The resulting dialectic led to improvements that proved both sides right.

In the case of CD, this process has led to the discovery of digital distortions, the design of D/A converters which provide major advances in both sound and technology, and a steadily improving level of performance which has gradually allowed the Compact Disc to live up to its inherent promise.

The Mark Levinson Reference Digital Processor No. 30 is clearly intended to embody this new state of the art. The No. 30 combines a wide range of advances in D/A converter technology and sound quality. It is superbly built, beautifully finished, and has some very unusual features.

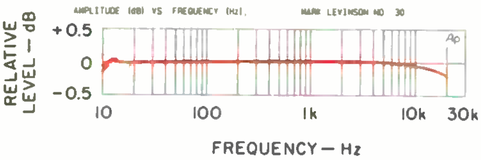
The No. 30 is in two pieces. The main unit has the digital circuitry and controls in the center, cradled between towers that each hold one channel's D/A and analog circuitry. The other unit contains separate power supplies for the main unit's digital center section and its two analog towers, each connected by a separate cable. My only complaint about the ergonomics is that the No. 30 is *big* and takes up a lot of space—though no more so than some preamps with separate power supplies.

The No. 30 also has as many inputs and outputs as many preamps. There are eight

SPECS

Frequency Response: 10 Hz to 20 kHz, +0, –0.2 dB.
THD: 0.003% at 1 kHz and 0 dB, A-weighted.
Dynamic Range: 98 dB.
S/N: 105 dB.
Channel Separation: Greater than 110 dB.
SMPTE-IM Distortion: Less than 0.005%.
Digital-to-Analog Conversion: Two custom 20-bit D/A converters.
Digital Filter: Eight-times oversampling.
Analog Filter: Bessel-tuned, linear phase to 40 kHz.
Low-Level Linearity: To –70 dB, deviation unmeasurable; below –90 dB, approximately +1.7 dB (undithered, referenced to 0 dB at 1 kHz).
Output Impedance: Less than 6 ohms.
Dimensions: Processor, $19\frac{3}{8}$ in. W × $7\frac{1}{4}$ in. H × $15\frac{7}{16}$ in. D (49.2 cm × 18.4 cm × 39.2 cm); power supply, $15\frac{3}{4}$ in. W × $5\frac{1}{4}$ in. H × $14\frac{7}{8}$ in. D (39.9 cm × 13.3 cm × 37.8 cm).
Weight: Processor, 34 lbs. (15.4 kg), power supply, 25 lbs. (11.4 kg).
Price: $13,950.
Company Address: Madrigal Audio Laboratories, P.O. Box 781, Middletown, Conn. 06457.
For literature, circle No. 90

digital inputs—five balanced female XLR connectors, one ST (AT&T) optical connector, and two EIAJ Toslink optical connectors—enough for most professionals. The Levinson Reference Digital Processor No. 30 also has three digital outputs, two male XLRs for listening and recording plus one EIAJ optical output. The analog outputs include two balanced XLR male connectors and one unbalanced RCA jack per channel. Two "Communication Ports" ("Master" and "Slave") hint at system extensions to come. All of the electrical connectors are gold-plated.



A

B

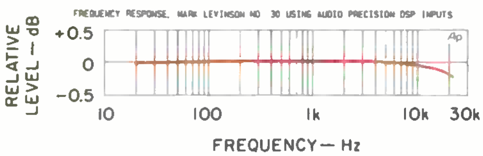


Fig. 1—Frequency response, both channels, using signals from CD player (A) and from test generator (B).

Besides input switching, the front panel has buttons for "Standby" (the unit is never turned completely off), "Digital Record Select" (which allows you to record any digital input while listening to another, effectively providing a digital tape monitor), polarity inversion, and display dimming. Indicator LEDs show status, blinking when the unit is in standby mode and glowing when the No. 30 detects copy-protection codes in the digital bitstream and when the digital input signal conforms to the AES/EBU standard. A large dot-matrix display offers a choice of four intensity levels; it also shows when signals are muted, locked, or absent and identifies the selected input and its sampling frequency. Internal switches allow you to rename any of the eight inputs as "CD," "LD," "DBS," "DAT," "VCR," or "AUX" to match your system's setup.

The input switching is highly sophisticated, using a crosspoint switch on the DSP board, the uppermost of the three boards in the main unit's center section. The digital input module, on the same board, has individual active termination, retriggering, and balanced driver circuitry for each input. This was done to avoid possible degradation of input signals, especially from unbalanced outputs. To prevent transient noise when switching between inputs, the first input is gently muted, and the signal only returns when the No. 30 has locked onto the new signal.

The No. 30 has clearly been designed for updating. The circuit boards are modular for effective upgrades through board-swapping, and key solid-state devices (including the EPROM that holds the No. 30's software routines) can be easily replaced.

Madrigal Audio Laboratories, which makes Levinson components, believes that purity and regulation of the power supply are critical to superior digital performance, and has put at least as much emphasis on the design of the power supply as on the digital and analog signal circuitry. A switching power supply is used for the digital section, providing the high power, freedom from variations in input power, isolation, and thermal stability that digital circuits require. The output of this supply is filtered and also has seven separate regulators to reduce noise on the d.c. line. The left and right analog power supplies begin with a.c. filtering and are said to be improvements of the designs in the latest Mark Levinson preamps. These power supplies also have an extremely low rejection ratio, which is rated at –80 dB in the power supply and another –40 dB in the main unit.

Madrigal contends that the data jitter caused at the digital interface receiver (DIR) that locks onto the incoming digital signal is more important in limiting sound quality than the problems dealt with by digital filtering. The company claims that most existing D/A converters have too much jitter to let the rest of the system resolve at 16 bits, much less the higher bit resolutions used by some D/A converters. Some DIRs, for example, are accurate to approximately 200 nS; the No. 30 has a rated accuracy of better than 100 pS.

The Levinson version of the DIR, called a Digital Audio Interface Receiver (DAIR), uses three separate narrow-aperture phase-locked loops (PLLs), each optimized for a single sample frequency (32, 44.1, or 48 kHz), instead of a single wide-aperture PLL that can cover all three sampling rates. The DAIR's jitter is said to be more than low enough for the No. 30's 20-bit resolution at eight-times oversampling.

Madrigal believes that the theoretical advantages of using general-purpose DSP chip sets with proprietary algorithms and high sampling rates are offset by the resulting requirement for much more critical jitter specifications. They also feel that existing transports and circuitry cannot use more than eight-times oversampling without a loss of sound quality because of jitter-related distortion and that, in any case, there is no practical advantage to oversampling rates higher than eight-times.

According to Madrigal, application-specific ICs (ASICs) are more desirable than general-purpose DSP computers because integrating the software into the silicon lets the ASICs run faster. Thus, from their point of view, only if the desired algorithms aren't available in an ASIC is a general-purpose chip set needed. An added benefit is that the best custom silicon devices have more accurate digital filters than have yet been implemented with general-purpose DSP circuits. After testing a number of devices and algorithms, the company found an application-specific DSP chip from Nippon Precision Circuits that met their requirements.

THE NO. 30 MEASURED BETTER, IN NEARLY ALL THE MAIN PARAMETERS, THAN ANY D/A UNIT SO FAR TESTED.

Each channel module of the No. 30 holds two linear 20-bit D/A converters, one for each polarity of the balanced digital signal. The positive and negative digital audio signals are processed separately, maintaining differential operation in the digital domain and providing differential analog signals to the analog output buffers. This eliminates the need for inversion circuits in the analog domain.

The final analog filter in the No. 30 uses a Bessel-tuned active filter in the output stage for maximum phase linearity. Instead of the voltage op-amps used in the feedback loops of most such circuits, Madrigal uses new and faster current op-amps that are said not to be affected by musical

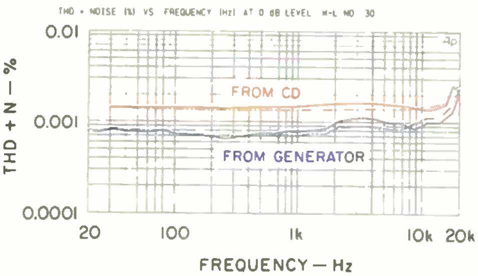


Fig. 2—THD + N vs. frequency, both channels, relative to maximum output level.

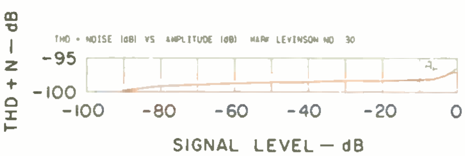


Fig. 3—THD + N vs. signal amplitude, both channels.

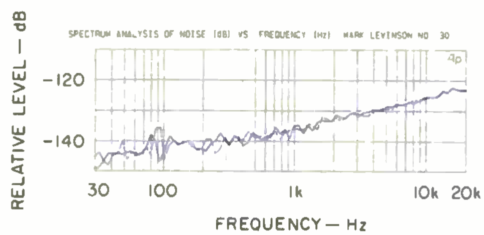


Fig. 4—Spectrum analysis of residual noise when playing "no-signal" track of test CD; both channels shown.

dynamics. Their output buffer has discrete components, with a d.c. servo—rather than coupling capacitors—to improve harmonic accuracy.

The No. 30 has two separate signal paths to enhance channel separation. Madrigal claims that the analog circuitry in each tower of the No. 30 represents an advance over the design in any of its preamps and amplifiers, partly due to the use of Teflon boards (whose dielectric constant is low and doesn't change with frequency) for the analog circuits.

The instruction manual is well written, fun to read, and easy to follow—a comment that rarely applies to high-end literature. But despite its sophistication, I was able to hook up the No. 30 before I read the manual; both connections and operation are straightforward and intuitive.

My subjective impressions of the No. 30's sound are in the usual place, at the end of this review. But to see how this technology measures on the test bench, as well as for a brief listening report, I turn you over to Len Feldman.

Anthony H. Cordesman

Measurements

While I will also have a few words to say about the superb sound quality of this incredible processor, my main task is to discuss its electrical performance. I did not see Mr. Cordesman's evaluation before writing this, but I must state, unequivocally, that the No. 30 performed better, in nearly all important parameters, than any D/A converter or digital processor I have yet evaluated—and that the measurements were consistent with what I heard in my abbreviated listening tests.

As I've done with other stand-alone D/A converters, I measured the No. 30 with digital signals from both a CD transport and my Audio Precision test equipment. The AP equipment generates 24-bit data words, then sets the dither levels and assumes that the device under test will simply truncate at the set level.

Figure 1A confirms that the frequency response of the processor, when fed from the digital output of a CD player, is flat within the –0.2 dB claimed by Madrigal, from 10 Hz to 20 kHz. Results are substantially the same for signals generated by the AP equipment (Fig. 1B).

The first difference to show up between CD player-fed signals and signals generated by the Audio Precision test gear occurred when I measured THD + N versus frequency (Fig. 2). While the results shown for signals fed from a CD player's digital output are the best I have ever obtained (approximately 0.0015% at mid-frequencies), results for the signals generated by the AP test equipment are even better, an incredibly low 0.0008% or so at mid-frequencies. In fact, my usual scale had to be expanded downward to "capture" this plot. Notice, too, that even at the treble extreme of either sweep, the increase usually seen in THD + N is present but minimal, never rising above 0.0025%.

Figure 3 shows how THD + N varies with signal amplitude. For practical reasons, this test was conducted using only the digital output from the reference CD player, with signals that ranged in amplitude from 0 dB (maximum recorded level) to –90 dB. Even at maximum recorded level, the reading is only about –97.5 dB, corresponding to an equivalent THD + N of 0.00133%. At levels below –20 dB, the

THIS IS THE FIRST D/A CONVERTER THAT HAD NO TRACE OF A.C. HUM OR HARMONICS.

readings (referred to maximum recorded level) are even lower, –98.5 dB or better.

To isolate the actual THD from the residual noise of the system, I used the FFT spectrum-analysis capabilities of the Audio Precision DSP circuitry while the CD player fed the No. 30 a 1-kHz steady-state tone. By "capturing" the results 16 times, the noise peaks were effectively averaged out, and only two significant actual harmonics of the 1-kHz tone were discernible, at 3 and 5 kHz. Their dB values, when calculated as a percentage of THD, worked out to only 0.00107%.

Channel separation exceeded the published specification by far and was greater than I have measured for any previous CD player or D/A converter. Separation was nearly 137 dB at 1 kHz and was still greater than 124 dB at 16 kHz. What's more, it was virtually identical for both channels.

Signal-to-noise ratio was the same whether I used the "no-signal" track from my CD-1 test disc or a "no-signal" digital output from the Audio Precision gear. In both cases, S/N was 117.8 dB for the left channel and 117.9 dB for the right channel. While these results are remarkable, the real surprise came when I ran a spectrum analysis of the residual noise (Fig. 4) and found absolutely no noise peaks at 60, 120, or 180 Hz from the 60-Hz line frequency. This is the only D/A converter I have ever tested that showed no measurable hum-related components. Of course, part of this "miracle" arises from the fact that the Mark Levinson No. 30's power supply is a separate component, connected to the processor via cables; no a.c. voltages ever reach the processor itself.

Figure 5 shows deviation from linearity, using undithered signals from 0 to –90 dB and dithered signals from –70 to –100 dB. The slight deviation of less than 1 dB for an undithered signal at –90 dB is better than claimed by the manufacturer and about as good as I have seen with any CD player or D/A converter. Bear in mind that the D/A converters used in this processor are not the increasingly popular one-bit variety that boast near-perfect linearity at the expense of some added noise (when improper noise-shaping is employed, as it sometimes is). Here we are dealing with 20-bit converters, which means that the converters used in the Mark Levinson No. 30 either have near-perfect low-level linearity or have been carefully calibrated externally during production.

In any case, when I used *dithered* signals in the range from –70 dB to –100 dB to

I ATTRIBUTE THE OPEN, AIRY SOUND TO THE NO. 30'S SUPERB LOW-LEVEL LINEARITY.

check out linearity at low levels, results were even better. As seen in Fig. 5, I could detect *no deviation from perfect linearity whatsoever!* My usual fade-to-noise test results, shown in Fig. 6, further confirm the No. 30's excellent low-level linearity as well as its low residual noise level and high EIA dynamic range (which I calculated as just over 100 dB). The EIAJ dynamic range exceeded the published claim, measuring 99.95 dB for the left channel and 99.91 dB for the right channel.

For further confirmation of the superb linearity of this processor, I plotted output versus input, using progressively lower digital input signals generated by the Audio Precision (Fig. 7). I could not easily translate this plot into *deviation* from perfect linearity, but close examination of the graph reveals that even at –100 dB, output matched the input perfectly. What little deviation there is occurs at levels below –110 dB, where the deviation from perfect linearity is still less than 1.0 dB!

At this point, eager to get the processor off the test bench and into my music system, I made but one additional measurement: A check of frequency (or pitch) accuracy determined by the crystal clock in the processor. That accuracy was within –0.0003%, which means that a middle A musical note recorded at the standard 440 Hz would be reproduced as a frequency of 439.99868 Hz. I suspect that even persons possessing the most perfect "perfect pitch" aren't likely to be upset by this minuscule "discrepancy."

Use and Listening Tests

Listening tests were conducted with a minimum amount of electronic equipment in the signal path. The digital output of my reference CD player was fed directly to one of the balanced digital inputs of the Mark Levinson No. 30 processor. The No. 30's analog outputs were fed directly to an amplifier equipped with its own input level control. The amplifier, in turn, fed my reference KEF 105.2 speakers. To be completely honest about it, I did not conduct any comparison tests against other state-of-the-art digital reproducing equipment. I did, however, listen to a fairly wide selection of my favorite CDs and can attest to the fact that they never sounded better. I attribute the open, airy sound quality of this setup to the No. 30's superb low-level linearity and to its ultra-low levels of noise and distortion.

Of course, I would not have expected a processor costing some $14,000 to deliver sound that was in any way flawed. Using the Levinson in my listening setup, I realized yet again that when the digital circuitry of an audio system is as perfect as this, the limiting factor becomes the source material. Comparing some of my earliest CDs with late releases proved highly revealing. Where software flaws were previously masked by hardware imperfections, suddenly they were easily identified. By the same token, well-produced recent CDs delivered the kind of superb sound that digital naysayers have maintained is inherently impossible in a 16-bit digital audio system with a 44.1-kHz sampling rate. If the claim for CD of "perfect sound, now and forever" was an overstatement back in 1983, it's closer to the truth when playing CDs and DATs using a processor such as the No. 30. I'm interested to see if Tony Cordesman agrees!

Leonard Feldman

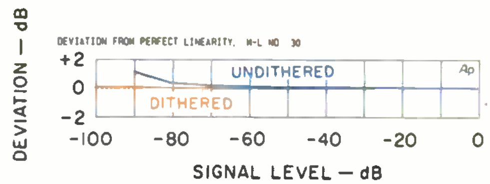


Fig. 5—Deviation from linearity.

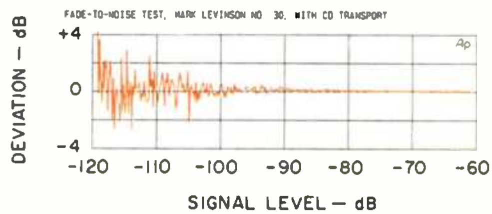


Fig. 6—Fade-to-noise test, using dithered signals from –60 to –120 dB.

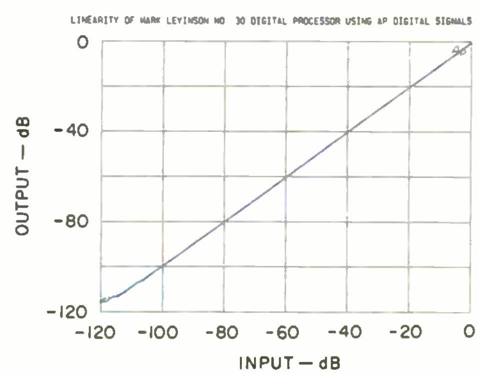


Fig. 7—Output vs. input linearity at 500 Hz, for test generator signals fed via AES/EBU digital input.