NAD Masters Series M2 Direct Digital

INTEGRATED AMPLIFIER

JOHN ATKINSON

A decade ago, many predicted that amplifiers with switching or class-D[1] output stages would come to dominate high-end audio. In a post–Peak Oil world in which the price of energy would always continue to rise, a class-D amplifier's very high efficiency in converting AC from the wall outlet into speaker-driving power would be a killer benefit. Although a conventional push-pull class-B amplifier has a theoretical efficiency of 78.5%, which would seem usefully high, this efficiency is obtained only at the onset of clipping; the need for the output devices to carry a standing bias current reduces that efficiency considerably, typically to around 50%. Class-A amplifiers are even less efficient, with a maximum of 25%; *ie*, three times as much power is dissipated by the amplifier as waste heat as is used to drive the loudspeaker (see "Sam's Space" in this issue).

By contrast, a class-D amplifier can, in theory, be 100% efficient, and practical circuits are at least 90% efficient. Watt for watt, all of the expensive parts of an amplifier design—the power transformer, output-stage heat-sinking, and chassis—can be smaller and thus less expensive: a high-power class-D amplifier can be small in size, light in weight, and cheap. Class-D amplifiers appear, therefore, to have had every competitive advantage. So why do audiophiles still mostly buy and use amplifiers with class-AB output stages?

The answer is the ultimate sound quality—so far, class-D amplifiers have found widespread domestic acceptance only as subwoofer amplifiers. There are exceptions: Bruno Putzeys's Hypex modules, as used in the Channel Islands amplifiers, have their followers; PS Audio's GCC-100 is a favorite of *Stereophile* reviewer Robert Deutsch; and the latest generation of Bel Canto's e.One Reference1000 monoblocks are to be found in "Music in the Round" columnist Kalman Rubinson's system.

Still, when I encountered NAD's M2 class-D integrated amplifier, which sells for a respectable $5999, I was torn between respect for the technology its design demon-

DESCRIPTION Solid-state, stereo integrated amplifier with remote control, four digital and two analog inputs, digital processor loop, RS-232 port, 12V trigger input and output, and soft clipping. Digital input sample rates accepted: 32–192kHz at up to 32-bit word length. Clipping power: >250Wpc into 8 ohms (>24dBW). Maximum continuous power: >200Wpc into 8 ohms (>23dBW), >250Wpc into 4 ohms (21dBW), >300W into 2 ohms (18.8dBW). IHF dynamic power: 300Wpc into 8 ohms (24.8dBW), 450Wpc into 4 ohms (23.5dBW), 600W into 2 ohms (21.8dBW). Maximum output current: >27A. Frequency response: 20Hz–20kHz, ±0.3dB (with –3dB at >85kHz). Channel separation: >80dB (analog inputs), >90dB (digital inputs), both at 10kHz ref. 1/3 rated power into 4 ohms. Damping factor: >2000 (20–200Hz). Rated distortion: <0.02%, 20Hz–20kHz at 100mW–rated power, with Audio Precision AES17 and passive 20kHz filters. Input sensitivity: 296mV for 100W output, 418mV for rated power. Input impedance: 36k ohms in parallel with 200pF. Maximum input level: 5.6V RMS at –9dB level trim setting. Signal/noise: >95dB A-weighted ref. 1W, >118dB A-weighted ref. 200W. Power consumption: 500W in normal use, 100W at idle, 1W in standby.

DIMENSIONS 17.1" (435mm) W by 5.8" (148mm) H by 19.76" (502mm) D including speaker terminals. Weight: 44.4 lbs (20.2kg) net, 56.8 lbs (25.8kg) shipping.

FINISHES Gray, silver.

SERIAL NUMBERS OF UNITS REVIEWED H99M200085, H99M200094.

PRICE $5999. Approximate number of dealers: 200 NAD dealers, of which 60 are NAD Masters Series dealers.

MANUFACTURER NAD Electronics International, 633 Granite Court, Pickering, Ontario L1W 3K1, Canada. Tel: (800) 263-4641, (905) 831-0799. Web: www.nadelectronics.com.

1 The *D* in *class-D* does not stand for *digital*, as some commentators have suggested. Rather, *D* was just the next available letter in the alphabet when amplifier circuit topologies were being classified. A class-D amplifier can be either digital and analog in operating principle.

NAD Masters Series M2 Direct Digital integrated amplifier

ERIC SWANSON

strated and skepticism about its ultimate sound quality. How good could a class-D amplifier sound? The only way to find out was to review it.

The M2

Although it's convenient to refer to the NAD M2 as an integrated amplifier, it's actually something rather different: the M2 is a multiple-input D/A converter with an output stage that can drive a loudspeaker. Although it has two pairs of analog inputs—one pair single-ended on RCAs, the other pair balanced on XLRs—these are immediately converted to 24-bit digital, with a user-selectable sample rate of 48, 96, or 192kHz. Sources are selected with the buttons below the front panel's blue fluorescent display or with the remote.

Both digital input signals and the converted analog input signals are fed to a digital signal-processing section, this specified as having an internal data path 35 bits wide. The DSP section includes the volume control; as the maximum bit depth handled by the control is 24, the 35-bit data path, in theory, allows there to be up to 11 bits' worth (*ie*, 66dB) of attenuation with no degradation of signal resolution. Level adjustments are made either with a front-panel rotary encoder or with the usual Up/Down buttons on the remote control, in 0.5dB steps. The DSP section also allows the user to choose from seven impedance-compensation filters, to allow the amplifier's top octave to be tuned to match the chosen speaker impedance in that region.

The rightmost front-panel button is labeled Menu; in conjunction with the rotary encoder, it allows the user to select the sample rate of the analog inputs' A/D converters, the speaker impedance-matching filter, a gain offset for each of the analog inputs (from 0dB to –9dB in 3dB steps), the amplifier's absolute polarity, and the alphanumeric name of each input. The volume control can be bypassed if the owner wishes to use the M2 with a separate preamplifier, while digital data can be routed via an external processing loop if desired.

The M2's rear panel offers two pairs of plastic-shrouded binding posts for each channel, to either side of the IEC AC inlet and main power switch. (A front-panel button switches the M2 in and out of Standby mode.) The balanced and unbalanced analog input jacks are on the far left of the rear panel, with the digital inputs and loop sockets vertically arrayed next to them.

The M2 is relatively hefty, but offers a superb level of fit'n'finish, as well as a high maximum continuous output power of more than 200Wpc into 8 ohms. In common with NAD's amplifier philosophy, more power—300Wpc into 8 ohms—is available for short-term transients. Also

MEASUREMENTS

The NAD M2 is not the first power D/A converter to have crossed my test bench: the Sharp SM-SX100, reviewed by Michael Fremer in July 2000 (see www.stereophile.com/integratedamps/253) had that honor. But as I wrote at the time in the measurements section that accompanied that review, "On the face of it, an amplifier accepting digital input data and operating entirely within the digital domain is a very attractive idea. But as Sharp's SM-SX100 reveals, it takes heroic engineering to make it work, and there are still some compromises involved, particularly in achieving sufficient dynamic range."

The question facing the NAD M2, therefore, was whether it had implemented the concept without compromise. But first, as class-D amplifiers have significant levels of ultrasonic switching noise present at their speaker terminals, even with the obligatory low-pass filter in series with their output, accurately assessing their performance is not trivial. This ultrasonic noise can drive the input stage of an analyzer into slew-rate limiting, leading to inaccurate distortion measurements. See Bruce Hofer's PowerPoint presentation on this subject, which can be downloaded from www.aes.org/sections/la/archive/2003/recaps/2003_docs/BHTesting ClassDAmplifiers.ppt. I therefore performed the distortion, noise, and channel-separation tests on the M2 using the Audio Precision AUX-0025 passive filter discussed by Hofer.

I measured the M2's performance with *Stereophile*'s loan sample of the top-of-the-line Audio Precision SYS2722 system (see the January 2008 "As We See It" and www.ap.com); for some tests, I also used my vintage Audio Precision System One Dual Domain. Before I test an amplifier, I run it for 60 minutes at one-third its specified power into 8 ohms. Thermally, this is the worst case for an amplifier with a class-B or -AB output stage, but for a class-D amplifier such as the M2 it represents nothing out of the ordinary. Even so, at the end of the hour, the M2's chassis was comfortably warm.

Looking first at the analog inputs, both the balanced and unbalanced inputs offered a maximum gain (with the volume control set to "10.0dB") of 39.55dB into 8 ohms. Both inverted signal polarity with the Polarity set to Positive. Without any Input Offset, the analog inputs overloaded at 2.025V RMS, which is a bit too close for comfort to the maximum output of many CD players. I recommend adding a little headroom by trimming the analog input's gain by –3dB with the M2's Menu button—which, I'm told, is how production samples are being shipped. The unbalanced analog input impedance was close to specification, at 33.2k ohms at low and middle frequencies, dropping lightly to 29.2k ohms at 20kHz. The balanced input impedance was twice these values.

The M2's output impedance didn't appear to vary with the loudspeaker-impedance compensation setting used. It was extremely low at low and middle frequencies, at 0.06–0.07 ohm, which is close to the residual impedance of the 6' speaker leads I use in my testing. At 20kHz,

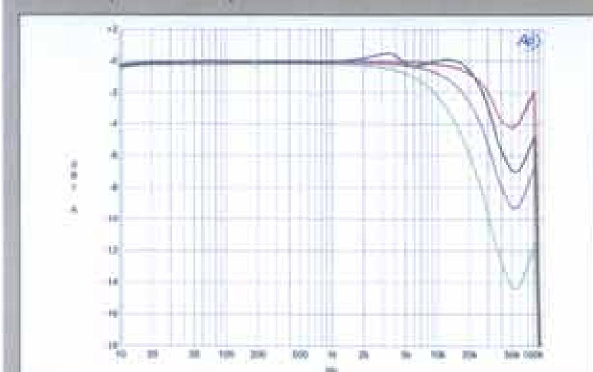

Fig.1 NAD M2, >8 ohms speaker compensation, frequency response at 2.83V into: simulated loudspeaker load (gray), 8 ohms (left channel blue, right red), 4 ohms (left cyan, right magenta), 2 ohms (green). (2dB/vertical div.)

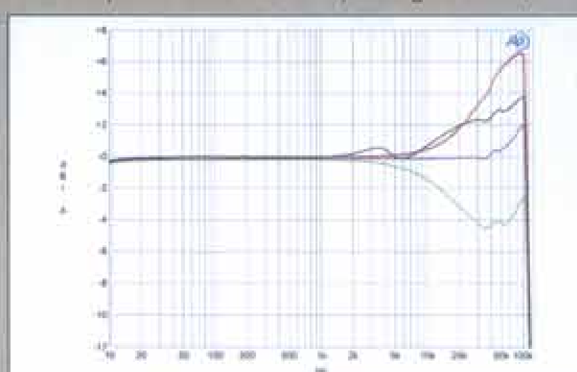

Fig.2 NAD M2, 4 ohms speaker compensation, frequency response at 2.83V into: simulated loudspeaker load (gray), 8 ohms (left channel blue, right red), 4 ohms (left cyan, right magenta), 2 ohms (green). (2dB/vertical div.)

reflecting NAD tradition, the M2 offers optional Soft Clipping, selectable with a rear-panel switch and realized in the digital domain. The M2 uses three power supplies, all switch-mode types: one for each channel's output stage, and a third for the input stage and control section.

Technical details

In its simplest form, the output stage of a class-D amplifier comprises two complementary switches (usually power MOSFETs), one connecting the output terminal to the positive voltage rail, the other to the negative rail. When no input signal is present, the switches alternately open and close at a very high frequency, sending a series of full-scale positive and negative pulses to the output–in the case of the M2, ±50V. The switches are never on at the same time, and as the average voltage at the output is zero, there is no output signal. With an input signal present, the oscillator controlling the switches adjusts its duty cycle so that the full-scale positive pulses last longer when the audio signal is in its positive phase, and the full-scale negative pulses last longer for the negative signal phase. The higher the signal level, the longer the switches stay closed for each pulse, and the higher the average voltage fed to the output for each signal phase. You have an amplifier! For obvious reasons, this operating principle is called Pulse Width Modulation (PWM). And because the switching devices are either fully on or fully off, no power is wasted and the efficiency approaches 100%. However, a hefty low-pass filter needs to be in series with the output in order to prevent the high level of high-frequency switching noise from contaminating the neighborhood, and to reconstruct the analog waveform.

In practice, of course, there are many

however, it varied between 1 and 1.5 ohms, depending on the load impedance I used to take the measurement.

The speaker-impedance compensation setting did have a major effect on the amplifier's frequency response, however. Fig.1 shows the response into 8, 4, and 2 ohms, and into our standard simulated loudspeaker, with the compensation set to ">8 ohms." (The digitizer sample rate was set to 192kHz for these measurements.) Despite the very low impedance, the variations into our standard loudspeaker load reach ±0.4dB in the treble, and the top octave can be seen to roll off into the lower impedances, with the 2 ohm response (green trace) down 6dB at 20kHz. Switching the compensation to "4 ohms" gives the family of traces in fig.2. The 2 ohm response is now –3dB at 20kHz, the 4 ohm output is flat to 40kHz (cyan and magenta traces), and the 8 ohm output rises to +6.5dB between 80 and 90kHz (blue and red). As you'd expect, the "2 ohm" compensation gives a flat audioband response into 2 ohms (fig.3), but with now a 12dB ultrasonic peak into 8 ohms. I recommend that you look carefully at the top-octave response of your loudspeakers, and take care not to set the M2's compensation lower than the average impedance in that region.

The A/D converter used to digitize the M2's analog inputs has a flat response within the audioband, with a sharp cutoff associated with the sample rate chosen (not shown). I recommend using a sample rate of 48kHz for CD-based analog sources, and 96kHz for phono preamps. The linear-phase digital filter associated with the A/D converter gives rise to the usual Gibbs Phenomenon symmetrical "ringing" with squarewaves: fig.4 shows a 1kHz squarewave digitized at 96kHz.

Channel separation at 1kHz via both analog and digital inputs was better than specified, at >95dB (analog) and >103dB (digital), both in both directions. The separation decreased at the top of the audioband, to the specified 80 and 90dB, respectively. Without any additional low-pass filtering on the output and with the input shorted but the

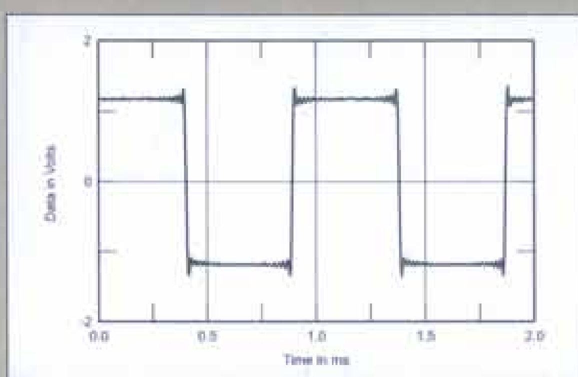

Fig.4 NAD M2, analog input, 96kHz sampling, small-signal 1kHz squarewave into 8 ohms.

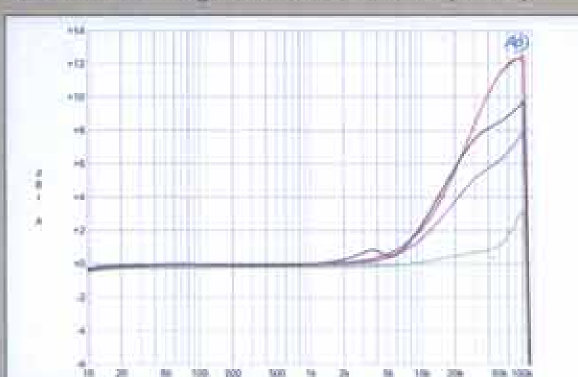

Fig.3 NAD M2, 2 ohms speaker compensation, frequency response at 2.83V into: simulated loudspeaker load (gray), 8 ohms (left channel blue, right red), 4 ohms (left cyan, right magenta), 2 ohms (green). (2dB/vertical div.)

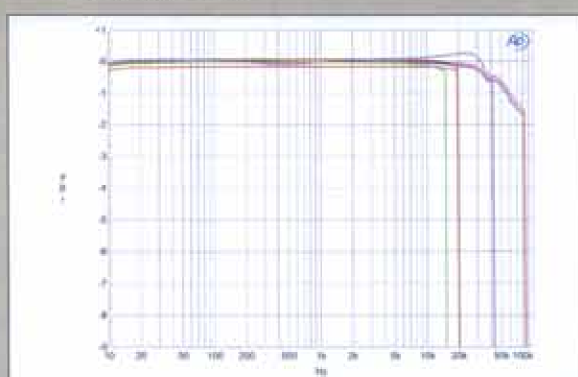

Fig.5 NAD M2, digital input, frequency response at 2.83V into 8 ohms with 8 ohms speaker compensation, with sample rates of: 32kHz (left channel gray, right green); 44.1kHz (left blue, right red), 96kHz (left cyan, right magenta), 192kHz (left blue, right red). (1dB/vertical div.)

engineering problems to be solved in the design of a PWM amplifier, and many proprietary solutions are offered. NAD has collaborated with a British semiconductor company, Diodes Zetex Ltd., which developed a novel feedback topology in which the output pulses are continuously compared with a reference to produce an error signal. This error signal is integrated, digitized (at 108MHz), and fed back, with noiseshaping, to the PWM modulator. The signal is also monitored at the output low-pass filter, to give a low output impedance. The Zetex team refers to their topology as a Direct Digital Feedback Amplifier, and the NAD M2 is the first commercial product to feature DDFA.

In the main, a PWM output stage follows a conventional small-signal analog amplification stage. However, if the PWM stage can be fed PCM data directly, there is no need for there to be any analog amplification at all. This is what NAD has done in the M2, which is why they call it a Direct Digital amplifier. From the block diagram in NAD's white paper on the design of the M2, it looks as if the PCM data are first converted to a noise-shaped bitstream that is then applied to the 108MHz PWM modulator, along with the feedback signal.

There have been similar products before, in which a digital input signal is directly fed to an amplifier's output stage. The original Wadia company bet the farm on what they called a PowerDAC, but couldn't bring it to market successfully. The TacT amplifiers did generate some marketplace traction, and the Toccata PCM-to-PWM interface used by TacT was licensed to Texas Instruments in 2001. The Sharp SM-SX100, from the start of the century, was functionally very similar to the M2 but was compromised in terms of dynamic range, and never sold

measurements, *continued*

volume control at its maximum, there was around 100mV of ultrasonic noise present at the speaker terminals, which was reduced to 20.6mV by the Audio Precision AUX0025 low-pass filter. The wideband, unweighted signal/noise ratio, ref 2.83V into 8 ohms, again with the analog input shorted, was thus limited to 42.7dB. This improved to a respectable 76.6dB when the measurement was restricted to the audioband, and to 84dB when A-weighted. The specified 95dBA ref. 1W into 8 ohms was probably taken with the volume control set to 0dB, therefore.

As shown in fig.5, the M2's frequency response via its digital input depended on the datastream's sample rate, of course. The input locked to data with sample rates ranging from 32 to 192kHz, and the display did show the correct sample rate when the status flag in the incoming datastream was correctly set. (If I left that flag blank, the M2 did operate at the incoming sample rate despite the display showing "44.1kHz.") A gentle rolloff above the audioband reached –1.5dB at 90kHz with 192kHz data (fig.5). The channel matching can be seen to be superb. With the volume control set to "0.0dB," a 1kHz tone with a digital level of –20dBFS resulted in a level at the speaker terminals of 5.935V into 8 ohms, suggesting that, unless the volume is backed off a little, the amplifier will be driven into a couple of dBs' worth of clipping with full-scale digital transients.

As a power amplifier capable of swinging at least 10 times the output voltage of a typical D/A converter, the M2 offered superb dynamic range. Fig.6 shows spectral analyses of the M2's output taken with a swept bandpass filter while the amplifier decoded 16-bit data representing a dithered 1kHz tone at –90dBFS, and 24-bit data representing dithered 1kHz tones at –90 and –120dBFS. (The external Audio Precision passive low-pass filter was in circuit for these measurements, along with an internal 20kHz brick-wall filter, as recommended by Audio Precision.) Not only are all the traces free from any harmonic or power-supply–related

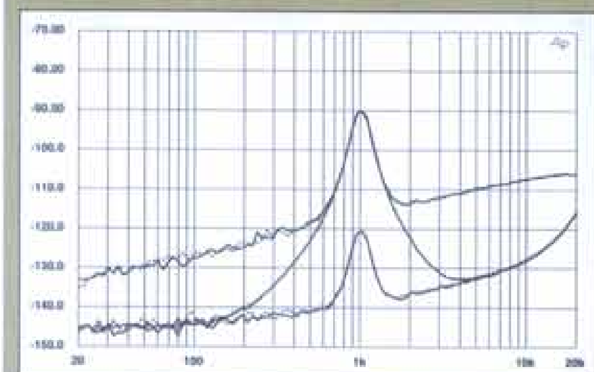

Fig.6 NAD M2, digital input, ⅓-octave spectrum with noise and spuriae of dithered 1kHz tone at –90dBFS with 16-bit data (top) and 24-bit data (middle at 2kHz), and dithered 1kHz tone at –120dBFS with 24-bit data (bottom at 1kHz). (Right channel dashed.)

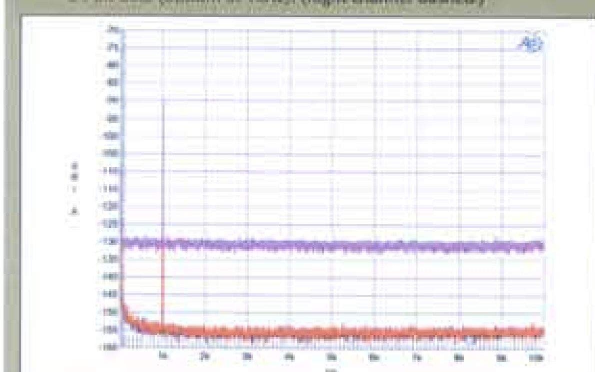

Fig.7 NAD M2, digital input, FFT-derived spectrum with noise and spuriae of dithered 1kHz tone at –90dBFS with 16-bit data (left channel cyan, right magenta) and 24-bit data (left blue, right red).

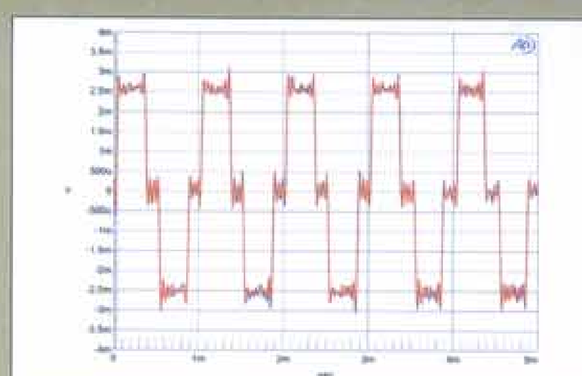

Fig.8 NAD M2, digital input, waveform of undithered 1kHz sinewave at –90.31dBFS, 16-bit data (left channel blue, right red).

in significant numbers.

The advantage of keeping everything in the digital domain is that, provided the math is done with sufficient precision, the only source of noise and distortion is the PWM output stage itself. The M2 thus has the potential for sounding better than a conventional analog PWM design.

Hiccups

A couple of operational niggles were apparent when I first set up the M2. The shrouded EuroNanny speaker terminals accept spade lugs from one angle only, which makes dressing speaker cables a hit-or-miss affair. The terminal's opening is also too narrow to take the thick lugs now found on some high-end cables, such as AudioQuest's K2. Fortunately, the terminals do accept 4mm plugs, which I fitted to the cables I used. I was also initially puzzled by the M2's sample-rate display, which obstinately told me that the incoming sample rate was 44.1kHz, even when I was feeding the amplifier data at a different sample rate. According to Greg Stidsen, NAD's director of product development, even though the M2 does adjust itself to operate at the incoming sample rate, the display shows the sample rate set by the appropriate status flag in the datastream. Unfortunately, if this flag is left blank, as can happen with some source components, the display will default to "44.1kHz."

I had a problem with the first sample of the M2 (serial no. H99M200085) I received for review. After a couple of weeks of operation, during which time I left it on continuously, I got home one evening to be greeted by the sound of clicking relays and the front-panel message "OVERHEAT." Unfortunately, I had no idea how long the M2 had been in this state. Following the instructions in the manual, I turned the

spuriae, the increase in bit depth lowers the noise floor by up to 24dB. FFT analysis (fig.7) shows a similar improvement, suggesting that the M2 has true 20-bit dynamic range, which is the state of the art of real-world digital decoding. There is thus plenty of resolution available to allow effective attenuation in the digital domain.

Linearity error with 24-bit data was negligible down to –120BFS, which, in conjunction with the superb resolution, meant that the M2's reproduction of an undithered 16-bit tone at exactly –90.31dBFS was essentially perfect (fig.8). The three DC voltage levels are clearly resolved, with superb waveform symmetry and the Gibbs Phenomenon "ringing" unobscured by noise. Increasing the word length to 24 bits produced an excellent sinewave.

Assessing the M2's maximum power was a bit tricky because, like all NAD amplifiers, it offers more power for brief transients compared with its steady-state power delivery. Fig.9 shows the result of using a level-stepped tone to plot the THD+noise percentage (with the AP passive filter and with Soft Clipping turned off) into 8 and 4 ohms with both channels driven, and into 2 ohms with one channel driven. At clipping (defined as 1% THD+N) with a 1kHz tone, the M2 delivered 303Wpc into 8 ohms (24.8dBW), 389Wpc into 4 ohms (22.9dBW), and 469W into 2 ohms (20.7dBW). (In each case, the speaker compensation was set to the appropriate value.) However, the amplifier is clearly happiest driving loads above 2 ohms, as shown by the plot of the THD+N percentage against frequency (fig.10, which shows useful data only up to 6.7kHz, due to the 20kHz brick-wall filter used for this test). The distortion is very low at low frequencies into both 8 and 4 ohms, but is considerably higher into 2 ohms. In addition, the higher the frequency and the lower the impedance, the higher the THD+N, though in this respect the M2 is still an order of magnitude better than some other class-D amplifiers I have measured.

Even with the measurement system's low-pass filtering, the residual distortion at low power lay below the noise floor. I therefore looked at the residual waveform accompa-

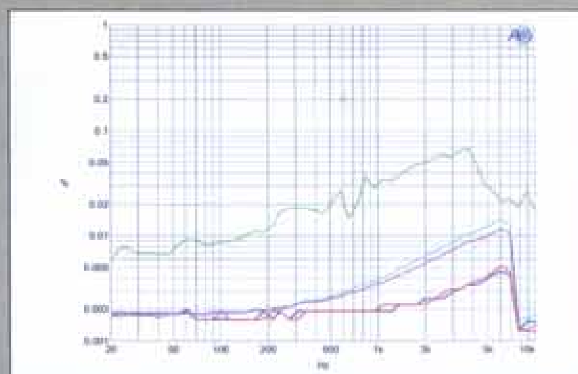

Fig.10 NAD M2, THD+N (%) vs frequency at 10V into: 8 ohms (left channel blue, right red), 4 ohms (left cyan, right magenta), 2 ohms (green).

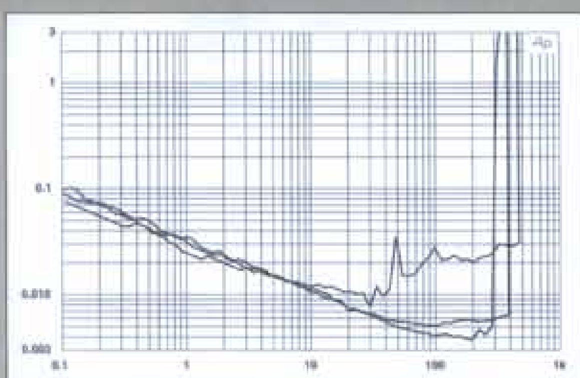

Fig.9 NAD M2, distortion (%) vs 1kHz continuous output power into (from bottom to top below 100W): 8, 4, 2 ohms.

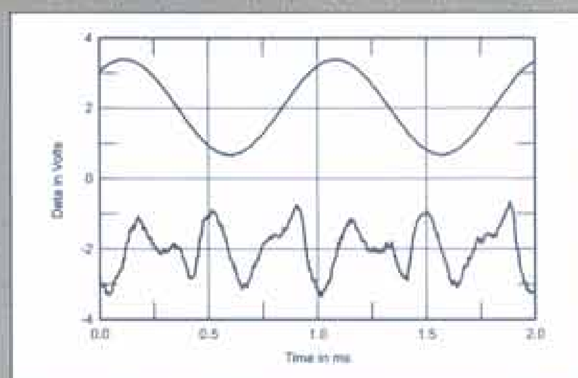

Fig.11 NAD M2, 1kHz waveform at 89W into 4 ohms (top); 0.0049% THD+N; distortion and noise waveform with fundamental notched out (bottom, not to scale).

amplifier off at the rear panel, unplugged the AC cord to make sure there would be a hard reboot, and left the M2 powered down for an hour. When I turned it back on, it passed signal for about 5 seconds, then displayed the "OVERHEAT" message again—the amplifier was now stone cold—and shut down. NAD shipped me a new sample (serial no. H99M200094), and I continued the review, including the measurements, with that sample, which performed perfectly.

Sound

I initially set the M2 up in its Fixed Gain mode, for use as a power amplifier, but after a New York minute's reflection, I realized that that was not going to take advantage of what the NAD could do. So I rethought the system's architecture, feeding to three of the M2's digital inputs the digital outputs of the dCS Puccini (for CDs) and Ayre Acoustics C-5xe[MP] (for DVD-Audio discs) players, as well as my Mac mini server via a Bel Canto USB-to-S/PDIF converter. For SACD playback, I fed the Puccini's analog output to the M2's balanced inputs, making sure its maximum output level was set to 2V so as not to overload the M2's own A/D converter (see "Measurements" sidebar). I left off the NAD's Soft Clipping feature. I also set the speaker compensation to ">8 ohms," which seemed appropriate for the Aerial Acoustics 20T V2 speakers with which I did most of my auditioning. The Aerial has a low impedance in the midrange, but much higher in the top two octaves; setting the M2 to "4 ohms," as I initially did, resulted in too much high-treble energy. For the PSB Synchrony Ones I used the "6 ohm" setting, to match that speaker's top-octave impedance.

I wasn't sure what to expect from the M2. My experience of class-D amplifiers has been somewhat limited, but from that experience I anticipated taut, dry lows, somewhat threadbare highs, and a flattened soundstage. I got none of those things. Instead, when I fed the M2 the

measurements, *continued*

rying a 1kHz tone at 89W into 4 ohms, and averaged 64 traces to drop the noise that would otherwise obscure it (fig.11). The primary harmonic can be seen to be the third, though some higher-order products are present. At low frequencies into 8 ohms, the distortion is predominantly second and third harmonic, though these spuriae are all below –112dB (0.00025%). Into lower impedances and at higher frequencies, the third and higher odd-order harmonics dominate (fig.12), though these are all still very low in level. With the decreasing linearity at high frequencies seen in fig.12, the M2's performance on the high-frequency intermodulation test was as expected (fig.13), with some high-order products present, though the second-order difference tone at 1kHz lay at a low –94dB (0.0015%). However, even the highest-level spuriae, at 18 and 21kHz, lay at –72dB (0.025%), which is fine.

Finally, I tested the M2's rejection of word-clock jitter by feeding a 16-bit version of the diagnostic J-Test tone from the soundcard of my test-lab PC via 15' of plastic TosLink. The resulting narrowband spectrum of the amplifier's output is shown in fig.14. The central spike representing the high-level Fs/4 tone shows very little spectral spreading at its base, and the harmonics of the Fs/192 LSB-level squarewave lie at the residual level. Other than the fact that the noise floor in the right channel (red trace) is a little higher than in the left (blue), this is state-of-the-art performance.

It is very satisfying to be able to discuss a component's measured performance without having to scratch my head over some or another idiosyncrasy. The NAD Masters Series M2 Direct Digital amplifier falls readily into that category.

—John Atkinson

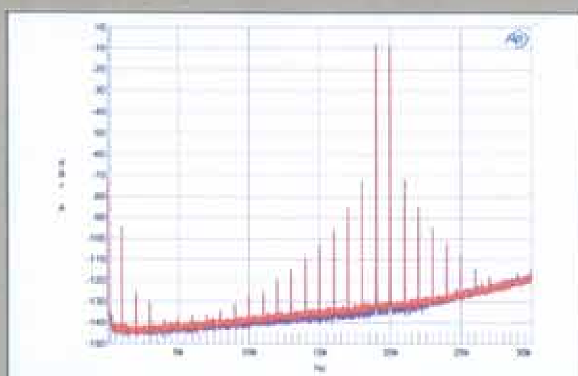

Fig.13 NAD M2, HF intermodulation spectrum, DC–24kHz, 19+20kHz at 192W peak into 4 ohms (linear frequency scale).

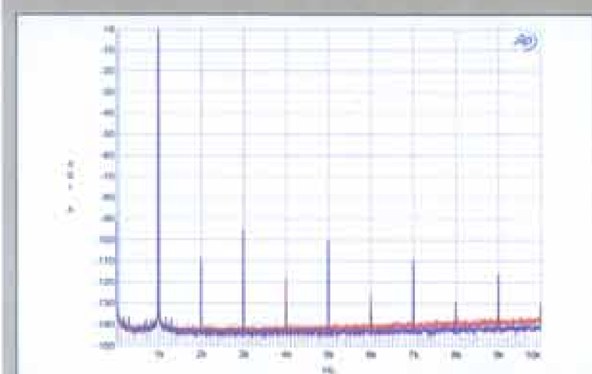

Fig.12 NAD M2, spectrum of 1kHz sinewave, DC–10kHz, at 97.5W into 8 ohms (left channel blue, right red; linear frequency scale).

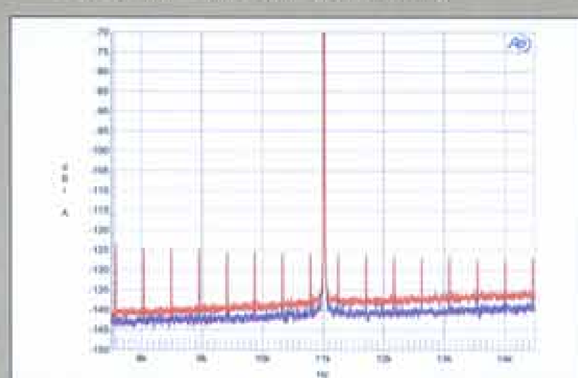

Fig.14 NAD M2, line output, high-resolution jitter spectrum of analog output signal, 11.025kHz at –6dBFS, sampled at 44.1kHz with LSB toggled at 229Hz, 16-bit data. Center frequency of trace, 11.025kHz; frequency range, ±3.5kHz (left channel blue, right red).

24-bit/88.2kHz master files of Cantus's *While You Are Alive* (CD, Cantus CTS-1208), the soundstage was wide and deep, the high frequencies silky smooth, the lows warm and rich.

With iTunes playing back CD data, well-balanced rock recordings such as Mary Chapin Carpenter's version of Jagger and Richards' "Party Doll," from *Party Doll and Other Favorites* (CD, Columbia CK 68751), had a delightful palpability in the midrange, and where reverberation had been used in the mix, the source moved back appropriately in the soundstage.

And even when the recording was not good, its poor qualities were handled by the M2 without exaggeration or tonal emphasis, allowing them to be mentally categorized and put to one side. One of my favorite Miles Davis albums, for example, *We Want Miles*, recorded in 1982 (CD, Columbia 469402 2), was not well served by the engineers, sounding bright, hard, and in-your-face, even on the original LP. Yet with the M2, the brashness seemed to be reproduced in a plane different from that of the music. "Kix" begins with a jaunty figure from Marcus Miller's bass, with first congas, then full drum kit accompanying, before Miles enters with a typically sparse melodic line. Miller uses a funky, hammered-on percussive style for this passage, followed by a mellower, thumbed walking-bass line in the solo trumpet, sax, and guitar sections. The M2 fully distinguished between these different tonal qualities and did a good job of retrieving the ambience around the bass and congas. And the M2's treble didn't exaggerate this recording's splashy-sounding cymbals.

Driving both the Aerial and PSB speakers, the M2 got right both the clarity and the weight of the piano's left-hand register. However, I did wonder if the bass region was a touch too ripe: while Marcus Miller's Fender on the thin-balanced *We Want Miles* sounded tonally right, Phil Lesh's bass on the Grateful Dead's *Live/Dead* (CD, Warner Bros. 1830-2) sounded fuller than I'm used to, though not so much as to interfere with the music. In fact, this fullness helped add a degree of majesty, not only to rock recordings and large-scale classical works, but even to smaller ensembles, such as the collection of Mendelssohn's complete String Symphonies, with Lev Markiz conducting the Amsterdam Sinfonietta (BIS 1738, more than four hours of music on one SACD), which I'm slowly working my way through.

The M2: display above, Input and Menu buttons below, Standby on the left, Volume Control on the right.

In fact, I kept returning to the M2's retrieval of subtle sonic cues in the mix, particularly of ambience. There was more *there* there, in the immortal phrase coined by Sam Tellig channeling Gertrude Stein, compared with the Simaudio Moon Evolution W-7, itself no slouch in this area. "Blizzard Limbs," from Attention Screen's *Live at Merkin Hall* (CD, Stereophile STPH018-2), for example, starts with Mark Flynn playing a repeated figure on kick drum, hi-hat cymbal, and a snare rimshot. In mixing this album, I used what I then thought was just enough of the distant omni mikes to give a sense of the hall's acoustic while preserving the immediacy of the instrumental sounds. With the M2 driving the Aerial speakers, and playing back the hi-rez master files, there was more of that hall's charcter evident than I remembered. Not that it didn't sound musically satisfying, but the soundstage was now both deeper and a little more reverberant than I had originally intended. On Attention Screen's purist-miked *Live at Otto's Shrunken Head* (CD, Stereophile STPH020-2), the image of the four musicians was more solidly resolved. And on Robert Silverman's set of the complete Beethoven sonatas, which I recorded in 2000 (CD, OrpheumMasters KSP-830, now sadly out of print), the acoustic of the relatively intimate performing space was sufficiently well resolved that the fact that the room was a little small for the Bösendorfer 9' grand piano could be more readily accommodated to, as it would be heard live.

All of the above comments describe the sound of the M2 as driven from its digital inputs. The analog inputs are certainly of high quality, but feeding the dCS Puccini's balanced analog out to the M2's analog input gave a sound that, with CDs, wasn't quite as well resolved as when I used the Puccini's digital output to drive the M2. (Setting the balanced Input Offset to –9dB on the M2 resulted in a 0.4dB difference in level that could be compensated for by adjusting the Puccini's own volume control.) During the review period, I was auditioning the gold CD reissue of Arturo Delmoni's recital of works for solo violin by Ysaÿe, Kreisler, and Bach (John Marks JMR14; the gold edition is available exclusively from www.stereophile.com). Yes, this recording is an ambiencefest, and that came though via the M2's analog inputs—but the violin was a little more forward in the soundstage than via the NAD's digital input. And a brief reference back to the Simaudio pre/power combination I used for last December's review of the Puccini revealed that for SACD playback–in which, of course, the player's digital output is disabled–the dCS system provided the ultimate sound quality with the Aerial speakers, edging ahead of the sound of the M2 fed by the dCS's analog outputs.

Overall, however, my time with the M2 was among the more enjoyable periods I have spent reviewing an audio component.

NAD'S MASTERS SERIES M2 IS **A WINNER** ALL THE WAY.

Summing up

This review proved a more difficult undertaking than I had expected. My system has been locked into the paradigm of Source Component(s) to Preamplifier to Power Amplifier(s) for the past three decades. When I decided to review the NAD M2, I had not appreciated just how radically it would shift that paradigm. The integration of a D/A processor and power amplifier into a single chassis eliminates the need for an actual preamplifier, instead substituting a digital-domain volume control and switching. Of course, the M2 does have two analog inputs, but as these are digitized ahead of the volume control, it doesn't affect the picture I have painted of the M2 as a paradigm-breaking component.

Once I had changed my system approach, the NAD M2 provided many nights of extended listening, with one album leading to just one more. And one more. While the M2 is relatively large and heavy for a class-D amplifier, runs warmer than you might expect, and is not inexpensive, when fed high-quality PCM data it offers sound quality that competes with that of the best conventional amplifiers. Given my long-term skepticism about the sonic benefits of PWM amplifiers, that was not what I was expecting. NAD's Masters Series M2 is a winner all the way.