JOHN ATKINSON

Weiss Engineering DAC502

D/A PROCESSOR

Hi-fi system resolution has long been the cause of heated arguments. But when it comes to converting digital data to an analog signal, there can be no argument. Data go in at one end of a DAC and an analog signal comes out of the other end, with a noise floor directly rated to the combination of the converter's digital and analog resolution. Ever since I started measuring digital products for *Stereophile*, I have been expressing a D/A processor's effective resolution in terms of the equivalent number of bits. With a typical FFT-derived analysis of 16-bit data, the levels of the individual FFT bins lie around 130dB below full scale. When the noise floor drops by 6dB, that's equivalent to another bit of resolution.

> **The immediate impression was of extraordinary clarity . . . as if the pixel count of the image had been increased.**

Two of the highest-resolution D/A processors I have measured have been from Swiss pro-audio company Weiss Engineering: the Medea, which Kalman Rubinson reviewed in February 2003,[1] and the Weiss DAC202, which Erick Lichte reviewed in January 2012.[2] Both offered superb resolution—almost 20-bit performance with the Medea and 21-bit with the DAC202—and both paired that resolution with sound quality to die for. "The Medea . . . remains in my mind as one of the only digital systems I've heard that could compete with the very best that vinyl has to offer while still doing

1 See stereophile.com/digitalprocessors/781/index.html.

2 See stereophile.com/content/weiss-dac202-firewire-da-converter.

SPECIFICATIONS

Description Roon-Ready, two-channel, digital-to-analog converter with volume control, color LCD touch screen, DSP, and remote control. Digital inputs: AES/EBU on XLR, S/PDIF on RCA jack and TosLink optical connector, USB Type A, USB Type B, Ethernet on RJ45 jack. PCM sample rates supported: 44.1kHz, 48kHz, 88.2kHz, 96kHz, 176.4kHz, 192kHz, 352.8kHz, and 384kHz, plus DSD64 and DSD128 (not all frequencies are supported with all inputs). Maximum input word length: 24 or 32 bits depending on input. Analog outputs: 1 pair balanced XLR, 1 pair single-ended RCA, ¼" stereo headphone jack, balanced headphone on 4-pin XLR jack, all short circuit–proof. Maximum RMS output voltage at 0dBFS: switchable, 6.8V (18.9dBu), 2.2V (+8.9dBu), 680mV (–1.1dBu), 220mV (–11.1dBu), balanced & 4-pin XLR headphone; 3.4V (+12.9dBu), 1.1V (+2.9dBu), 340mV (–7.1dBu), 110mV (–17.1dBu), unbalanced. Frequency response: 0Hz–20kHz ±0.25dB, Fs=44.1kHz; 0Hz–40kHz ±0.8dB, Fs=88.2 kHz; 0Hz–80kHz ±2.5dB, Fs=176.4 kHz. THD+N: –116dBr (0.00016%) at –3dBFS input level, –125dBr (0.000056%) at –40dBFS input level; –125dBr (0.000056%) at –70dBFS input level. Linearity: <±0.4dB deviation from ideal, 0dBFS to –120dBFS input level. Spurious components (including harmonics): at 0dBFS input level, maximum output level, 1kHz, all components <–120dB; at 0dBFS input level, maximum output level, 4kHz, all components <–115dB. Channel separation: >120dB, 20Hz–20kHz. Interchannel Phase Response: ±0.05°, 20Hz–20kHz, ±0.30° 20Hz–80kHz. Power consumption: 15VA max in use, 0.5VA max in standby.

Dimensions 17.7" (450mm) W × 11.8" (300mm) D × 2.9" (74mm) H (including feet). Weight: 11lb (5kg).

Finish Silver or black.

Serial number of unit reviewed 0318, "Made in Switzerland." Firmware version: v2.2.0 revision r2297.

Price $9850. Approximate number of dealers: 6 in US, 3 in Canada. Warranty: three years from original purchase date on defects in material and workmanship.

Manufacturer Weiss Engineering Ltd., Florastrasse 42, 8610 Uster, Switzerland. Tel: (41) 44 940 20 06. Fax: (41) 44 940 22 14. Weiss is distributed in North America by Bluebird Music Limited, www.bluebirdmusic.com.

what digital does best. In other words, there were warmth and musicality, staggering dynamics, and real silent backgrounds," wrote Kal. Erick concluded that the DAC202 was "easy to recommend for those who want a digital system that doesn't sound 'digital.' The Weiss . . . offers a sound that will be very pleasing to many audiophiles tired of fatiguing hi-fi sound."

Now comes the subject of this review, the DAC502, which costs $9850, and while it doesn't have the DAC202's FireWire input, it offers USB and Ethernet connectivity and adds a balanced headphone output.[3]

Design outside

The DAC502 is a utilitarian-looking component, with a black-finished steel chassis with a damped top panel—this connected to the main system ground with a wire—and a chamfered aluminum front panel that's 10mm thick. On the left of the front panel is a ¼" headphone jack. On the right is a rotary control, and next to it a fairly small four-color touchscreen. In use, this panel displays the source, whether the line or headphone outputs are selected or muted, the sample rate, the metadata text when the DAC is receiving network data, and DSP and Setup information. A short push on the rotary control activates the Menu function; the control then acts as a scroll wheel, allowing various options to be selected and adjusted with the touchscreen; more on this later.

On the back panel are the IEC AC power inlet; an array of digital inputs—AES/EBU, coaxial and optical S/PDIF, USB Type A for connecting external storage, USB Type B, and Ethernet (Roon Ready or UPnP)—balanced and single-ended output jacks; and a 4-pin XLR jack for driving headphones in balanced mode.

Pressing the power switch on the front-panel rotary control—short push for on, long push for off—or on the remote control operates a semiconductor relay that only switches at zero crossings of the mains voltage to ensure glitch-free switching. Mains-voltage selection is done automatically by measuring the mains voltage before power is applied to the rest of the electronics.

The DAC502's complexity resides within.

Design inside

Looking inside the DAC502 left an impression of a component constructed to a high standard. The DAC502's circuitry is split into two. A large printed circuit board running front to back behind the display carries the power supply with its two toroidal transformers. Separate voltage regulators provide power to the left and right channels. The input-receiver circuitry and the signal-processing module are also mounted on this board, and a small daughterboard carries a Texas Instruments Arm Cortex-A8 microprocessor chip.

A ribbon cable connects this board to a completely shielded module that houses the D/A conversion and output stages. Digital-to-analog conversion for each channel is handled by a pair of ESS Sabre 32-bit DAC chips, these clocked, Weiss says, with a high-precision/low-jitter generator. The clock and DACs operate at a fixed sample rate of "about 195kHz," which is the frequency that results in optimal performance from the DAC chips. While the DAC502 will accept all the standard PCM sampling frequencies up to 384kHz, as well as DSD64 and DSD128, the data are sample-rate–converted to PCM at 195kHz before being presented to the DACs.

Digital signal processing

The DAC502's core functionality is controlled with a fourth-generation Analog Devices SHARC DSP (digital signal processing) chip. The following DSP algorithms are implemented:

- Room Equalizer can apply high-shelf and peaking/notch filters to deal with low-frequency room modes.
- Creative Equalizer is a tone control with low boost/cut, high boost/cut and mid boost/cut.
- De-Essing automatically removes overly bright sibilance from human voices. Two modes are offered: "Surgical" and "Smooth."
- Constant Volume, aka Dynamic Adaptation, is a "party mode" that normalizes loudness for all the tracks played.
- Vinyl Emulation allows "that special sonic character of a record player based playback chain" to be applied.
- Crosstalk Cancellation (XTC) compensates for the fact that with loudspeakers, the left ear also hears the right channel's output and vice versa. This mode allows dummy-head, binaural recordings to be correctly played back on loudspeakers. The user has to enter their head width, the separation of the loudspeakers' centers, and their distance from the listening position.
- Loudness Control is promised for a future firmware upgrade. It will equalize the output to compensate for the ear-brain's differing frequency sensitivity at different listening volumes.
- Headphone Equalizer is another yet-to-be-implemented function; it will adjust the frequency response of the headphone output to suit the listener's ears.

Once you have chosen the parameters for each of these functions, the settings can be saved as a snapshot and recalled at the touch of a button. I report on the effect of some of these DSP settings below.

Setup

The DAC502 can be controlled in three ways: with the touchscreen and rotary control; with the supplied metal remote; or with a web browser, by entering the address http://dac502-serial number.local. Both web and front-panel interfaces provide access to volume, balance, mute, and polarity-inversion controls. Also selectable there is a choice of four full-scale output levels: "0dB," "–10dB," "–20dB," and "–30dB." I wanted to connect the DAC502's balanced outputs directly to power amplifiers from Lamm, Parasound, and Classé, controlling volume with the DAC502's high-precision volume control while remaining near the top of its range. The "–10dB" setting, equivalent to a maximum level of 2.2V, was the best choice for achieving those aims.

I connected the DAC502 to my network, opened the local webpage, and checked for firmware updates. ("FW is up to date," it told me.) The processor was recognized by the Roon app as "Weiss DAC502," and Roon allowed me to control its volume. (The Roon volume setting was immediately reflected in both the local webpage and on the front-panel display; the webpage duplicated Roon's transport controls and displayed the artwork of any album that had been selected with Roon.) I was ready to play music.

3 If you don't need the balanced headphone output, the smaller DAC501 ($8750) offers the same performance as the DAC502. The DAC501 was enthusiastically reviewed in our sister magazine *Hi-Fi News* in December 2018; see hifinews.com/content/weiss-dac501-usbnetwork-dac.

Listening

The immediate impression was of extraordinary clarity. This wasn't as if the edges of the objects within the soundstage had been enhanced, as can be done to images with PhotoShop, but as if the pixel count of the image had been increased. It didn't manage this by emphasizing treble detail, but to resort to an audio reviewer cliché, the DAC502 cleaned the window into the recorded soundstage to an impressive extent.

A decade or so ago, I was giving seminars on how the way recordings had been made influenced what audiophiles hear from their systems. As an example of how different stereo microphone techniques resulted in different soundstage presentations, I used "The Turning" by Maura Bosch from *While You Are Alive* (Cantus CTS-1208). I had recorded all the tracks on this album from Minnesotan male-voice choir Cantus with three pairs of microphones: a close, central ORTF pair of cardioids, a close, spaced pair of omnis, and a distant pair of omnis mounted on a Jecklin disc. (You can see a photo of the three pairs of mikes by scrolling down the page at tinyurl.com/y9hhkgsl.) For the CD, I mixed the outputs of the three pairs, but for the seminars, I played the three outputs separately. I wish I had used the Weiss DAC502 at those seminars, because the differences in perspective, ambience, and tonal character from each of the microphone pairs were superbly audible: clean and clear, with precise stereo imaging but insufficient reverberation and no low bass from the cardioids; rich, but a stereo image that was pulled to the sides with the close-in spaced omnis; and richly reverberant but insufficient stereo spread from the distant omnis.

This soundstage clarity was a consistent feature of my time with the Weiss processor. The first track on *Caverna Magica*, the second album from Swiss New Age harpist Andreas Vollenweider (16/44.1k ALAC, ripped from Columbia MK 37827), opens with a man and woman having a conversation; the recorded acoustic, excited by their footsteps and the sound of dripping water, opens up as they step into a cave. I've never heard this so clearly delineated as with DAC502. The layering of the soundstage on this album, with various different instruments accompanied by different amounts of reverb, was also very audible.

The DAC502's low frequencies combined clarity with an excellent sense of what the late Art Dudley used to call "force"—even with both ports open on each of the Vimberg Mino loudspeakers, there was an excellent sense of forward momentum with the bass guitar and kickdrum on such rock classics as the intricate arrangement of Paul Simon's "America" from the 2002 reissue of the Yes album *Fragile* (24/96 ALAC, ripped from DVD-A, Elektra/Rhino R9 78249). Similarly, the double basses in Peter Gabriel's reimagining of "Don't Give Up" from *New Blood (Special Edition)* (16/44.1k ALC file, ripped from CD, Real World 00038) were superbly well-defined but with good weight. On both these albums, I was again struck by the well-differentiated layering of the soundstage.

Special effects

While the D-Esser and Dynamic Adaptation functions held little appeal, I did experiment with the Vinyl Emulator.

MEASUREMENTS

The Weiss DAC202, which Erick Lichte reviewed in January 2012, was one of the highest-resolution digital processors I have measured, so I was intrigued to see if the DAC502 would match its predecessor's performance.[1] As I had done with the DAC202, I measured the Weiss DAC502 with my Audio Precision SYS2722 system (see the January 2008 "As We See It"[2]). Apple's USB Prober utility identified the DAC502 as "DAC501" from "Weiss_Engineering_Ltd." with the serial number string "0.0.1."[3] The USB port operated in the optimal isochronous asynchronous mode, and Apple's AudioMIDI utility revealed that the DAC502 accepted 32-bit integer data sampled at all rates from 44.1kHz to 384kHz. The AES/EBU and S/PDIF inputs accepted data sampled at rates up to 192kHz.

The DAC502's maximum output level at 1kHz with the balanced outputs or the headphone outputs feeding a high 100k ohm load was 6.85V with the level set to "0dB"; 2.17V with it set to "–10dB," which is exactly 10dB lower; 684mV set to "–20dB," 20dB lower; and 217mV set to "–30dB," 30dB lower. As expected, the maximum levels from the unbalanced outputs were half those from the balanced outputs. With its polarity button set to Normal, the DAC502 preserved absolute polar-

1 See stereophile.com/content/weiss-dac202-firewire-da-converter-measurements.

2 See stereophile.com/content/measurements-maps-precision.

3. Weiss's DAC501 and DAC502 differ only in their form factor—the 501 is narrower and slightly taller—and an extra, 4-pin headphone connector on the back of the DAC502.

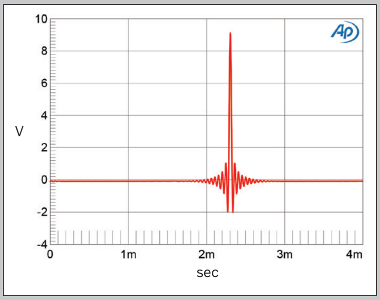

Fig.1 Weiss DAC502, impulse response (one sample at 0dBFS, 44.1kHz sampling, 4ms time window).

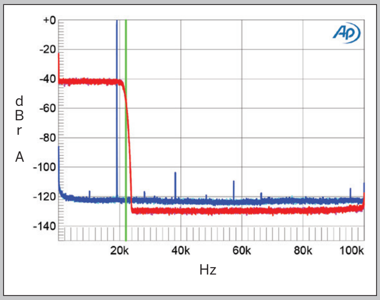

Fig.2 Weiss DAC502, wideband spectrum of white noise at –4dBFS (left channel red, right magenta) and 19.1kHz tone at 0dBFS (left blue, right cyan) into 100k ohms with data sampled at 44.1kHz (20dB/vertical div.).

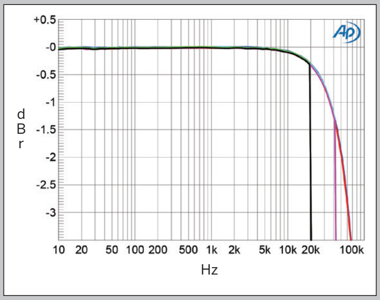

Fig.3 Weiss DAC502, frequency response at –12dBFS into 100k ohms with data sampled at: 44.1kHz (left channel green, right gray), 96kHz (left cyan, right magenta), and 192kHz (left blue, right red) (1dB/vertical div.).

Setting the Saturation control to "–3dB" gave a subjectively satisfactory effect. Even so, I ended up leaving the Emulator bypassed. A little of this effect goes a long way—an occasional splash of Sriracha sauce is nice, but the spicy effect is too much of a good thing when it is used with everything you eat.

Over the years, I have made several binaural/dummy-head recordings, but these need to be auditioned with headphones to project the soundstage outside my head. Played back on loudspeakers, binaural recordings tend to resemble "fat mono." Using the DAC502's local webpage for the XTC function, I entered the width of my head in centimeters, the distance of the speakers to my listening position in meters, and the distance between the loudspeakers. The maximum setting appears to be 2m; the manual says that the speakers need to be fairly close together, and if the geometry is suboptimal, the settings background turns yellow, which it did in my setup. I left the Mid Frequency and High Frequency attenuation at the factory settings of –5dB.

I resisted the temptation to play any tracks by Andy Partridge and cued up my binaural recording from the 1992 Formula One Grand Prix in Montreal, from *Stereophile*'s Test CD 3 (16/441.k ALAC, STPH006-2). I had made the recording sitting in the grandstand at the hairpin, and on headphones the cars rush toward me on the right outside my head, slow down to take the turn, then accelerate away into the distance on the left. With XTC bypassed and this track played back on speakers, everything sat in a wodge between the speakers, with some sounds smeared toward the left and right but with very little depth. Enabling XTC opened up the stereo image, the cars moved from the right speaker position to the left, and now there was excellent depth apparent as they zoomed away on the straight. But the degree of the effect was very dependent on how far away I sat and the separation between the speakers—easier to arrange with the standmounted KEF LS50s than the massive Vimberg Minos. The closer together the speakers, the better XTC appears to work; it would be a great function to use with desktop speakers.

The DAC502's Room Equalizer comprises five filters that can be set to peak/cut or high shelf for left and right speakers individually or together. To make use of this function, you download a FLAC or WAV file from the Weiss website that sweeps down from 200Hz to 20Hz. While playing this file, you note the time when the loudness is at a maximum. The Weiss processor's manual includes a table that correlates the time with the frequency of the sinewave. You then manually create up to five correction filters, a complex process that would be more user-friendly if integrated with an app like Room Equalization Wizard.

The DSP function I found most useful was the "Creative EQ." With the full-range Vimberg speakers, I left the frequency extremes alone but slightly suppressed the presence region. With the KEF LS50s, which have limited low-frequency extension, I added a 3dB-high low shelf with a Q of 1.4 and its corner frequency set to 92.5Hz. As long as I set the volume at a reasonable level, the EQ effectively fleshed out the speakers' midbass behavior, so that the organ pedals on my unreleased recording of Jonas Nordwall perform-

ity (ie, was noninverting) from all of its outputs. The balanced output impedance was 94 ohms at all audio frequencies; the unbalanced output impedance was 47 ohms. The front-panel headphone jack's output impedance was a very low 0.66 ohm. The DAC502 will have no problem driving low-impedance headphones.

Fig.1 shows the DAC502's impulse response with 44.1kHz data. It is typical of a conventional linear-phase filter with a symmetrical ringing before and after the single full-scale sample. This filter's ultrasonic rolloff (fig.2, magenta and red traces) reaches full stop-band attenuation at 24kHz with complete suppression of the aliased image at 25kHz of a full-scale tone at 19.1kHz (cyan, blue). The harmonics associated with the 19.1kHz tone all lie below –104dB. Fig.3 shows the DAC502's frequency response with data sampled at 44.1, 96, and 192kHz. The response with all three sample rates is down by just 0.3dB at the top of the audioband, with then a steep rolloff just before half of each sample rate. The response with 192kHz data continues the relatively gentle ultrasonic rolloff, reaching –3dB at 61kHz.

Fig.4 shows the effect of two of the DAC502's Creative EQ filters, high and low shelves at 110Hz and 10kHz, set to their maximum and minimum of ±10dB and measured with data sampled at 96kHz. The output reaches its specified boost or cut an octave below or above the selected turnover frequency; the boost or cut is 7dB at the turnover frequencies.

The effect of the Vinyl Emulator

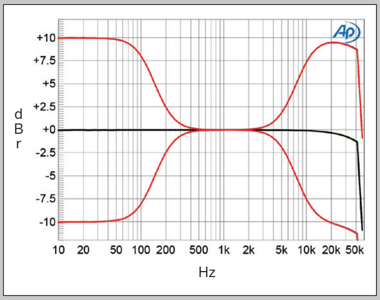

Fig.4 Weiss DAC502, typical EQ response at –12dBFS into 100k ohms with 96kHz data: high and low shelves at 110Hz and 10kHz set to "0dB" (left channel green, right gray) and set to ±10dB (left blue, right red) (2.5dB/vertical div.).

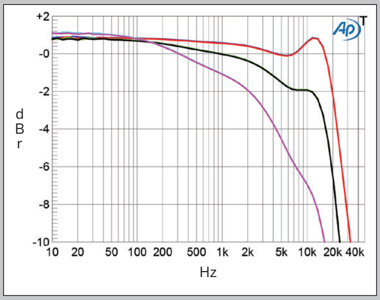

Fig.5 Weiss DAC502, effect of Vinyl Emulation at –12dBFS into 100k ohms with 96kHz data with Saturation set "–9dB" (left channel blue, right red), "0.0dB" (left green, right gray), and +9dB (left cyan, right magenta) (2dB/vertical div.).

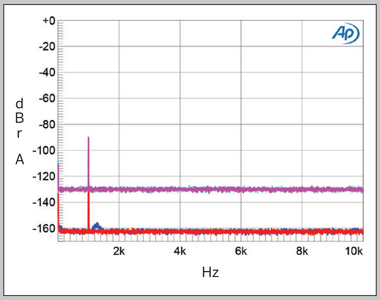

Fig.6 Weiss DAC502, spectrum with noise and spuriae of dithered 1kHz tone at –90dBFS with: 16-bit data (left channel cyan, right magenta), 24-bit data (left blue, right red) (20dB/vertical div.).

ing the Toccata from Widor's Organ Symphony No.5[4] had enough body.

With this small amount of boost in the bass, the KEFs' top octave now sounded a little depressed, so I added another filter, shelving up the frequencies above 10kHz by 1.5dB. This added a little top-octave air and better balanced the lows.

Comparisons

The PS Audio DirectStream ($5999)[5] has been my go-to D/A processor since I purchased our review sample in 2014. While not the last word in resolution, its smooth-sounding presentation allows the music to communicate and, since I added the original Network Bridge card ($899), allows me to use Roon to explore my music library. For the comparisons with the Weiss DAC502, the PS Audio was running the Snowmass firmware—I haven't yet installed the current Windom firmware, as I needed to keep the PS Audio's character, which I have become used to, unchanged.

With levels matched using the 1kHz warble tone from my *Editor's Choice* CD (STPH016-2) and the DAC502's EQ bypassed, the PS Audio's low frequencies sounded less well-defined than those of the Weiss DAC. With the big Vimberg speakers, I felt I needed to reinsert the plugs in their uppermost ports with the PS Audio—without the plugs, the Minos' lows were too loose-sounding—and with the little KEFs, I didn't feel the need to add midbass boost as I had with the Weiss. Nevertheless, listening to Martha Argerich's live performance with Nelson Goerner of Rachmaninoff's *Symphonic Dances*, arranged for two pianos (ALAC ripped from CD, Warner Classics 623594), the leading edges of the pianos' bass notes were still better defined with the Weiss.

The Rachmaninoff recording's midrange was warmer-sounding with the PS Audio but with less space around the pianos. Hugh Laurie's close-miked piano on "The Weed Smoker's Dream," from *Didn't It Rain* (16/44.1k ALAC ripped from CD, Warner Bros 535893), also sounded warmer on the PS Audio than it did on the Weiss, but Gaby Moreno's despairingly evocative vocal on this slow drag was equally engaging with both DACs, if a little more forward-

4 Readers are welcome to download this track from tinyurl.com/y4qdnjuz.

5 See stereophile.com/content/ps-audio-perfectwave-directstream-da-processor.

measurements, continued

function on the DAC502's frequency response is shown in fig.5. The response varies significantly with the setting of the "Saturation" control. The central green and gray traces in this graph show what happens with it set to "0.0dB": The output slopes down above the midrange, with a –3dB plateau between 7kHz and 12kHz referred to the low-frequency level, and then a steep rolloff. With the control set to "–9dB," the high-frequency balance is closer to neutral, but with it set to "+9.0dB," the DAC's output is down by 3dB at 2kHz and by 10dB at 10kHz. The Emulator also introduces noise with a spectrum that tilts up below 300Hz, adds mainly second-harmonic distortion, and reduces channel separation to around 20dB.

Channel separation with the Vinyl Emulator bypassed was simply superb, at >122dB in both directions below 3kHz, decreasing to a still-superb 113dB at 20kHz. An increase in bit depth from 16 to 24, with dithered data representing a 1kHz tone at –90dBFS, dropped the DAC502's noise floor by 30dB (fig.6). This implies a resolution of 21 bits, which is one of the highest I have encountered and equals that of the Weiss DAC202. When I played undithered data representing a tone at exactly –90.31dBFS, the waveform was symmetrical, with negligible DC offset, and the three DC voltage levels described by the data were free from noise (fig.7). With undithered 24-bit data (fig.8) the DAC502's very low analog noise floor means it can output a clean sinewave, even at this very low signal level.

Even set to its highest output level, the DAC502 produced very low levels of harmonic distortion with

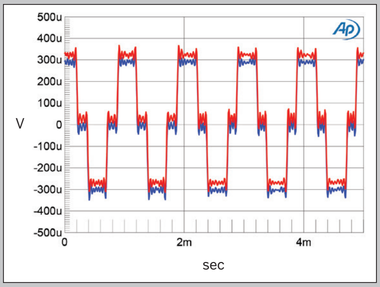

Fig.7 Weiss DAC502, waveform of undithered 1kHz sinewave at –90.31dBFS, 16-bit data (left channel blue, right red).

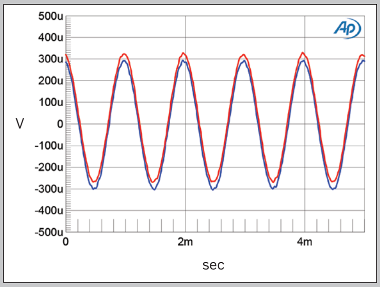

Fig.8 Weiss DAC502, waveform of undithered 1kHz sinewave at –90.31dBFS, 24-bit data (left channel blue, right red).

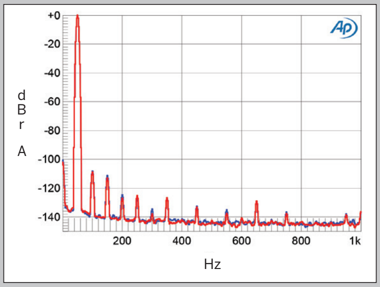

Fig.9 Weiss DAC502, 24-bit data, spectrum of 50Hz sinewave, DC–1kHz, at 0dBFS into 100k ohms (left channel blue, right red; linear frequency scale).

sounding with the DAC502.

The PS Audio gets the nod for its warm, easy-on-the-ear balance, the more-incisive Weiss for its transparent window into the recording's soundstage.

Headphone listening

I was editing one of my recordings with Adobe Audition while I had the DAC502 in-house. I took advantage of that processor's balanced headphone output to audition it with my Audeze LCD-Xes[6] using a balanced Nordost Heimdall 2 cable.

When you are creating a master file by splicing together musical selections from two different takes, it is critically important to be able to hear if there were differences in the reverberation and the noise floor between the outgoing and incoming takes. If there *are* differences, you get a "gear-change" effect at the splice point that gives the game away. Crossfades that I thought were okay with a pair of Sony MDR-7506 closed-back 'phones plugged into my MacBook Pro were revealed as needing more work when I listened to them with the Audezes driven by the DAC502.

Work done, I played a favorite Bill Frisell album, *East/West* (16/44.1k FLAC file ripped from CD, Nonesuch). I am enthralled by both Frisell's multifaceted electric guitar skills and his skillful arrangements. "Shenandoah," on *East/West*, starts off quietly with the guitar backed by a subtle loop then swells as the other three members of the quartet enter. Driving the Audeze headphones, the Weiss processor preserved the dynamics, the subtle ambience around the drums, the weight of the double bass, and the sheer force of Frisell's playing. Nice!

Conclusion

The Weiss DAC502 retrieves more information from the digits than any other DAC I have auditioned, with the possible exceptions of the Chord DAVE and dCS Vivaldi, both of which are long gone from my system and neither of which has either a headphone output or DSP functions. Both of those DACs are also more expensive than the DAC502. But the Weiss's resolution does come at a price: it is intolerant of problems with the rest of the system it is used with that would not be noticed with lesser DACs. During the six weeks I used the DAC502, I found I was continually fine-tuning my system before I could get the most enjoyment from my music.

But ultimately, musical enjoyment is what this product is all about. As I write this conclusion, I am listening to Stanford's hauntingly engaging song "The Blue Bird," performed by the Gabrieli Consort directed by Paul McCreesh (from *Silence & Music*, 16/44.1k FLAC, Signum Classics/Tidal). The interplay between the unaccompanied voices, the bell-like interjections of the high soprano, the setting of all the singers within a supportive chapel acoustic—the Weiss DAC502 made all these aspects clear, in service of the music. Which is what a great audio component should do. ■

6 See stereophile.com/content/audeze-lcd-x-headphones.

measurements, continued

full-scale data into the high 100k ohm load (fig.9). The subjectively benign second and third harmonics were the highest in level, but each lay close to a negligible –110dB (0.0003%). While the level of the second harmonic rose to –98dB (0.001%) when I reduced the load impedance to the punishing 600 ohms, the third harmonic remained below –110dB. The Weiss DAC has a bombproof output stage! Intermodulation distortion with an equal mix of 19 and 20kHz tones at –6dBFS was similarly very low (fig.10), with the difference tone at 1kHz lying at –115dB (0.0002%). Again the DAC502 wasn't fazed by the 600 ohm load. While the difference product increased in level, this was to a still-minuscule –106dB (0.0005%).

The DAC502 offered excellent rejection of word-clock jitter. Fig.11 shows the spectrum of the DAC502's output when it was fed high-level 16-bit J-Test data via USB. All the odd-order harmonics of the undithered low-frequency, LSB-level squarewave lie at the correct levels, there are no other sideband pairs visible, and the central spike that represents the high-level tone at one-quarter the sample rate (Fs/4) is narrow. The spectrum was similarly clean with 24-bit J-Test data via USB, though there was some spectral broadening of the Fs/4 spike with S/PDIF and AES/EBU data (fig.12).

I summed my measurements of the Weiss DAC202 by writing "The DAC202 is the best-measuring D/A processor I have measured in my quarter-century career at *Stereophile*. It just doesn't get any better than this." Weiss's DAC502 matched the DAC202 by also performing supremely well on the test bench.—**John Atkinson**

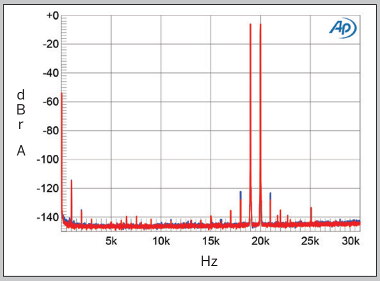

Fig.10 Weiss DAC502, 24-bit data, HF intermodulation spectrum, DC–30kHz, 19+20kHz at –6dBFS into 100k ohms, 44.1kHz data (left channel blue, right red; linear frequency scale).

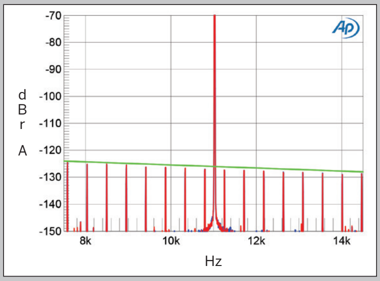

Fig.11 Weiss DAC502, high-resolution jitter spectrum of analog output signal, 11.025kHz at –6dBFS, sampled at 44.1kHz with LSB toggled at 229Hz: 16-bit USB data sourced from MacBook Pro (left channel blue, right red). Center frequency of trace, 11.025kHz; frequency range, ±3.5kHz.

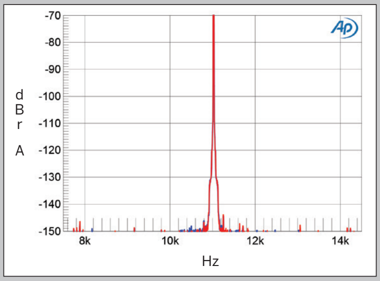

Fig.12 Weiss DAC502, high-resolution jitter spectrum of analog output signal, 11.025kHz at –6dBFS, sampled at 44.1kHz with LSB toggled at 229Hz: 24-bit AES/EBU data (left channel blue, right red). Center frequency of trace, 11.025kHz; frequency range, ±3.5kHz.