the absolute sound

Buyer's Guide to Digital Source Components 2016

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the absolute sound
Buyer's Guide to **Digital Source Components 2016**

From the Editor

Welcome to the new Buyer's Guide to Digital Source Components!

What would we do without digital?

For this year's all-new 2016 Guide to Digital Sources not only have we, the TAS editors, hand-selected **27 full-length product reviews** for you to explore, but you'll also find:

On the Horizon – A sneak peek at the hottest new digital gear about to be introduced—12 items across all price-points and categories.

MQA Feature Package – Through a series of informative articles, Robert Harley explains what you need to know about MQA, from tech details to FAQs.

Oppo Review Premiere – Neil Gader puts the new Oppo BDP-105D Blu-ray/SACD/CD disc and multimedia player through its paces.

Top Picks – Check out our expert reviewers' favorites across all digital source categories.

Whether you're into digital music on the go or while ensconced in your favorite listening lair, here's to enjoying the magic of those ones and zeros!

Happy listening!

Julie Mullins, Editor

On the Horizon

Hot New Products Coming Your Way

Neil Gader

Oppo Digital Sonica DAC

The Sonica DAC is a stand-alone high-performance DAC and a network hi-res audio player. In pursuit of outstanding digital-to-analog audio playback, Sonica DAC is equipped with an ES9038PRO 32-bit HyperStream DAC chip, the flagship of the ESS Sabre Pro series. The Sonica DAC's power circuitry includes a linear power regulator drawing from a massive toroidal power transformer, which provides a clean and robust power supply to the audio components. The asynchronous USB DAC input supports PCM up to 768kHz/32-bit and DSD up to 24MHz (DSD512). Optical and coaxial inputs are also available. The Sonica plays hi-res audio files from USB drives attached to it, from computers and servers connected via Ethernet or Wi-Fi, or from a smartphone or tablet running the Sonica app. The outputs are balanced XLR and regular RCA, with a volume control that can be bypassed.

Price: $799. oppodigital.com

Wyred 4 Sound DAC-1v2 and DAC-2v2

Wyred 4 Sound has recently announced two new DAC models, the DAC-1v2and the DAC-2v2. These DACs feature all-new, just-released ESS Sabre DAC chips, fully-balanced circuitry, hi-res PCM and 4x DSD capabilities, plus upgraded architecture and refreshed casework. The DAC-1v2 sports the new ES9026PRO chip while the DAC-2v2 will tout the new ES9028PRO chip. The updated DAC designs, coupled with these groundbreaking new chips, enable both models to deliver cutting-edge design at real-world cost. Retail pricing is $1299 for the DAC-1v2 and $1999 for the DAC-2v2. However, during Wyred 4 Sound's Black Friday Sale, Nov. 18–Dec. 5, these components will be on sale for $999 and $1499, respectively.

Prices: DAC-1v2, $1299; DAC-2v2, $1999. wyred4sound.com

Brinkmann Audio Nyquist DAC/Streamer

Although Brinkmann is justly renowned as a designer and manufacturer of state-of-the-art analog components, the company's first DAC, the Zenith, debuted in 1986 and still enjoys a devoted cult following. Everything Brinkmann learned with Zenith—combined with three decades of subsequent engineering experience—has resulted in the Nyquist full-featured DAC and streamer. An entirely new design includes proprietary high-voltage power-supply technology, a hybrid circuit topology with tubes, and completely separate PCM and DSD conversion paths. Nyquist has been optimized to accommodate the latest digital formats including MQA, and files up to DSD256 and 384kHz/32-bit PCM. Hardware and software of Nyquist's digital module are user-replaceable for unprecedented longevity. Nyquist offers flexible connectivity options, including Ethernet, USB, AES/EBU, and SPDIF. Balanced, single-ended, and headphone outputs are all standard. Nyquist streams music directly from a NAS drive, supports Roon, and includes access to Tidal and other online music services.

Price: $18,000. brinkmann-usa.com

On the Horizon

NuPrime Omnia P1 Portable Player/Server/DAC/Headphone Amp

The P1 is a portable hi-res music server, player, streamer, DAC, and headphone amp that is designed to replace several components within a music system. A revolutionary design feature sets it apart from other computing platforms dedicated to playing music: To isolate the audio processing from performance degradation due to high CPU utilization or signal drop-off due to wireless congestion, a proprietary Audio Processor Unit (OAP Unit) operates independently from the CPU and provides caching of digital music and completely eliminates jitter.The NuPrime Omnia P1 is capable of replacing a high-end desktop DAC and headphone amp, owing to separate professional, true-balanced, three-pin XLR left and right outputs, and through a NuPrime-designed OPA chip that produces 400mWpc of power. A customer-replaceable DAC module offers unique customization and upgrade paths. The USB-C port allows the Omnia to connect to a high-speed external hard drive as a music source.

Price: $1295, available January, 2017. nuprimeaudio.com

Aurender A10 Integrated Music Server

The Aurender A10 is an integrated music server/streamer with high-performance dual-mono MQA-certified DAC with balanced and unbalanced analog outputs. The A10 features a 4TB internal HDD and 120GB SSD cache for playback, variable outputs for direct connection to amplifiers, and a linear power supply. It can be a great solution for those replacing their aging CD players. A USB output and optical input provide additional connectivity options. As with all Aurender models, the A10 makes use of the integrated Aurender Conductor app, known for its performance and intuitive operation. This iPad app was developed in-house with managing large music databases in mind, so it provides exceptionally fast browsing and searching of your favorite music. Aurender Conductor also supports TIDAL lossless streaming with full app integration, support for NAS and USB drives, internet radio streaming via AirPlay, and access to remote technical support directly from the app itself. Available in silver or black.

Price: $5500. aurender.com

Esoteric N-05 Network Audio Player

Esoteric's newest digital source offering, the N-05 network audio player, is also one of the most entertaining components you can find. The N-05 can play virtually any music file over your network or directly from its USB port. You can also browse and play music from Tidal, all with the easy yet powerful Esoteric Sound Stream iPad app (free from the Apple App Store). In addition to being easy to use, the N-05 is built for sound. Multiple AKM D/A converters per channel allow 34-bit processing. All balanced circuitry, a high-current output stage and toroidal power supply give the N-05 the ability to get the most from your digital library. Have other digital components? The N-05 also features USB, coaxial, and TosLink inputs. (See review in this Guide.)

Price: $6500. esoteric-usa.com

On the Horizon

Astell&Kern AK70 Portable Player

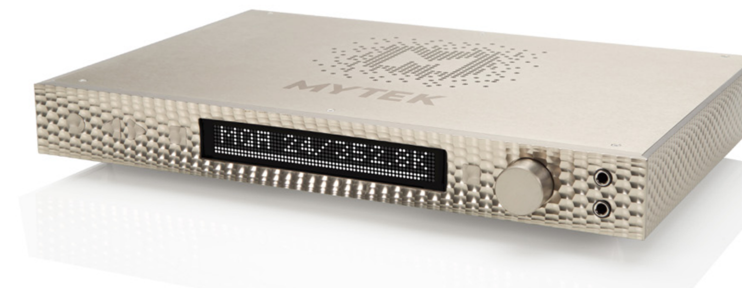
The AK70 is the latest entry-level, Wi-Fi-enabled, portable high-resolution audio player from Astell&Kern that brings audiophile quality music playback and value-added features to customers at a reasonable price. Features include Tidal streaming music service, OTA firmware updates, and DLNA 1.0 support for streaming music from other devices on the same network. With 64GB of internal storage and a single microSD card slot for up to 200GB of additional storage memory, the AK70 supports high-resolution audio up to 32-bit/384kHz PCM (downconverted to 24-bit/192khz PCM) including double-rate, 5.6MHz DSD (converted to PCM). Outputs include USB digital out, 3.5mm unbalanced and 2.5mm balanced headphone jacks, and support for 24-bit/48kHz audio playback over Bluetooth with apt X HD. The AK Connect app, available for iOS and Android devices allows full remote control of your AK70 player.

Price: $599. astellnkern.com.

NAD M32 Direct Digital Amplifier

The M32 is an elegant, top-of-the-line, BluOS-ready integrated amplifier with a host of features that offer maximum flexibility and excellent efficiency with reduced noise and distortion. With the M32, NAD is reimagining the integrated amplifier as we know it. Its DirectDigital amplification combines all pre-amplification and power amp functions into a single stage. With 150Wpc on tap, the M32 is a true digital amp (not just Class D) that is computer-controlled and amplifies entirely in the digital domain, giving it the shortest signal path possible. There are four MDC (Modular Design Construction) slots, and all are 24/192 capable. The optional BluOS module that can occupy one of the MDC slots is an advanced operating system for music management and control that includes support for local NAS drives and premium internet streaming audio services such as Spotify, Tidal, and many others. The M32 also includes a phono input and a dedicated headphone amplifier to accommodate music lovers of all kinds.

Price: $3999. nadelectronics.com



Mytek Digital Manhattan II Reference DAC

The new Manhattan II DAC is a reference network DAC/preamplifier with MQA decoder, 384k PCM and DSD256, phono and line analog preamplifier plus reference headphone amplifier. When used with the optional phono card it integrates digital streaming and vinyl analog playback for reference listening on headphones and speakers—and an optional Roon Ready network card turns the DAC into a streamer. Featuring the newest 130dB Sabre ES9038 32-bit DAC chipset, it handles PCM up to 384k, 32-bit, MQA, native DSD up to DSD256, DXD, with a 130dB dynamic range. A new Mytek Femto Clock in the company's C777 Clocking architecture pushes jitter and noise floor below measurable levels. Wordclock input and output allows stacking of multiple units for multichannel, multichannel DSD, and SACD.ISO operation. Two oversized, isolated power supplies—one each for analog and digital stages—lower noise and impedance. There's a greatly improved, low-distortion, 1dB analog attenuator circuit for main out and headphones, with a choice of 1dB-step digital 32-bit attenuator, or purist relay bypass. Shipping is in late November. Original Manhattan is upgradeable by manufacturer.

Price: $5995; phono preamplifier card, $1495; Roon Ready network card, $995. mytekdigital.com

On the Horizon

MSB Technology Select DAC

The Select DAC is MSB Technology's advanced reference DAC produced for the high-end market. Intended for longevity, it offers a "future-proof" modular design and is backed by a ten-year warranty. Following the release of MQA, MSB supports this format with a simple input module change. Inside the solid uni-body chassis there are eight Hybrid DAC modules with MSB's proprietary ladder DAC technology. The Select DAC II is powered by our manufactured-in-house femto clock technology for a high level of jitter control resulting in analog-like sound. All MSB components are manufactured in Santa Cruz, California, with its very own CNC machine shop and SMT line for careful quality control—all in the pursuit of greater "believability" in audio reproduction.

Price: $89,950. msbtech.com

NAD M50.2 Digital Music Player

The M50.2 is the top-of-the-line extension to NAD's 40th-anniversary M50/M51/M52 digital music suite—it combines the functions of both the M50 and M52 in one sleek package, and boasts worthy upgrades to boot. It offers high-resolution music listening, multi-room wireless streaming to other BluOS-enabled speakers, 24/192 storage, and CD ripping all in one elegant component. Outputs are digital only and include AES/EBU and HDMI; however, customers can add a complementary DAC or digital preamp (such as the company's M12 and M17 models). With the M50.2, music lovers can digitize and centrally store their music library, and then make it available to other BluOS-enabled components to network music throughout their home effortlessly; all your music can be available with merely a tap of the finger on a smartphone or tablet, or a click of a universal remote.

Price: $3999. nadelectronics.com

Berkeley Audio Design Alpha DAC Reference Series 2

The award-winning Berkeley Audio Design Alpha DAC Reference Series 2 together with the Alpha USB interface elevate audio reproduction to a level that is closer to reality than ever before. The ground-breaking resolution of the first Alpha DAC Reference Series enabled Berkeley Audio Design to design circuits and algorithms that achieve new levels of realism in the Alpha DAC Reference Series 2. Listening to the Alpha DAC Reference Series 2 is the only way to truly appreciate its excellence. The Reference Series 2 will also soon support MQA with a field-installable upgrade. (Please see the review in this guide.)

Prices: Alpha DAC Reference Series 2, $19,500; Alpha USB, $1895. berkeleyaudiodesign.com

Feature

Digital Emerges from the Dark Ages

Master Quality Authenticated (MQA) Is Here

The Long-Awaited Digital System Debuts

Robert Harley

It's not often that an audio technology comes along that has the potential to revolutionize the music industry as well as greatly improve sound quality for all listeners. But I believe that Master Quality Authenticated (MQA) may do just that.

MQA is a digital encoding and decoding system that delivers better-than-hi-res sound quality with a bit rate low enough for streaming. It's also backward-compatible with all existing music-distribution infrastructures and consumer hardware. MQA files are formatted as standard LPCM files (FLAC, AIFF, WAV, etc.); if you don't have an MQA DAC, the file will play on any DAC with slightly-better-than-CD sound quality. If you do have an MQA-capable DAC, the file will "unfold" the high-resolution information and deliver the resolution of the original studio master.

A number of high-end and mass-market audio companies have announced that they will offer DACs with MQA decoding, and the streaming service Tidal is very close to offering MQA files. Although the major music labels have not announced their intentions regarding MQA, indications are that they are engaging with the technology. (Visit theabsolutesound.com for the latest updates and announcements regarding the release of music in MQA.)

MQA was developed by Bob Stuart of Meridian Audio in collaboration with British mathematician Peter Craven, but the technology has since been spun-off into MQA Ltd, a separate company. I first heard MQA at CES two years ago in a before-hours, closed-door session, when the technology was an unnamed work-in-progress. Knowing nothing about the development effort or its goals, I sat down and was startled by what I heard. There was a complete absence of the artifacts that have always been part and parcel of digital audio—synthetic timbres, glare, lack of dimensionality, and a treble that is simultaneously bright and airless. I thought that I must be listening to some breakthrough super-high-resolution format with an enormous bit rate. After about 20 minutes of listening, Bob said with a sly smile, "We're listening to 1.3 megabits per second." I was astounded; that's less than the 1.4Mbs rate of 44.1kHz/16-bit PCM, and about one-seventh the bit rate of 192kHz/24-bit. Yet the sound was better than I'd heard from even the highest bit-rate PCM or DSD, never mind from CD.

Since then I've heard MQA in a number of demonstrations, but never at my leisure in my own reference system—until now. Meridian supplied me with its new 808v6 Signature Reference CD player/DAC with MQA decoding (see accompanying review), and MQA sent me a NAS drive loaded with full albums and many single-track samples from other titles. I was also able to stream MQA files from Tidal (through a special Sooloos account), and to play MQA-encoded music that had been stored on CD.

Listening to MQA

I had previously heard MQA only through Meridian's Digital Active loudspeakers, save for a short demo through Vandersteen Model 7s at the last Rocky Mountain Audio Fest, and through Wilson speakers and Dan D'Agostino electronics at the most recent CES. I was eager to hear MQA at length through the microscope of my reference system, which includes Magico Q7 Mk.II loudspeakers driven by Constellation Hercules II monoblock power amplifiers wired with top-of-the-line MIT Oracle cable. The Q7s were augmented with the EnigmAcoustics Sopranino electrostatic supertweeters, which extend the high-frequency response to 40kHz. AC power to the system is supplied by four dedicated 20A lines, and is conditioned extensively with Shunyata's Triton 2 and Typhon conditioners, and Shunyata Sigma power cords. All sources and amplification are supported by Critical Mass Systems Maxxum racks and amplifier stands. Acoustic treatment is provided by Stillpoints Aperture panels and ASC tube traps. I connected the Constellation amplifiers directly to the Meridian 808v6's variable-level balanced outputs, removing the preamplifier from the signal path. The MQA source was a Meridian Sooloos system running on an HP touchscreen PC that was network-connected to a Qnap NAS drive storing the MQA files as well as to the Meridian 808v6. I also evaluated MQA by listening to it on my work PC via Foobar music-playback software and Meridian's $299 Explorer^2 DAC driving my desktop speakers (Audience ClairAudient 1+1 V2+) as well as Audeze LCD-4 and

PSB M4U 2 headphones. The headphones were driven directly by the Explorer[2], or by the Moon by Simaudio 430HA headphone amplifier.

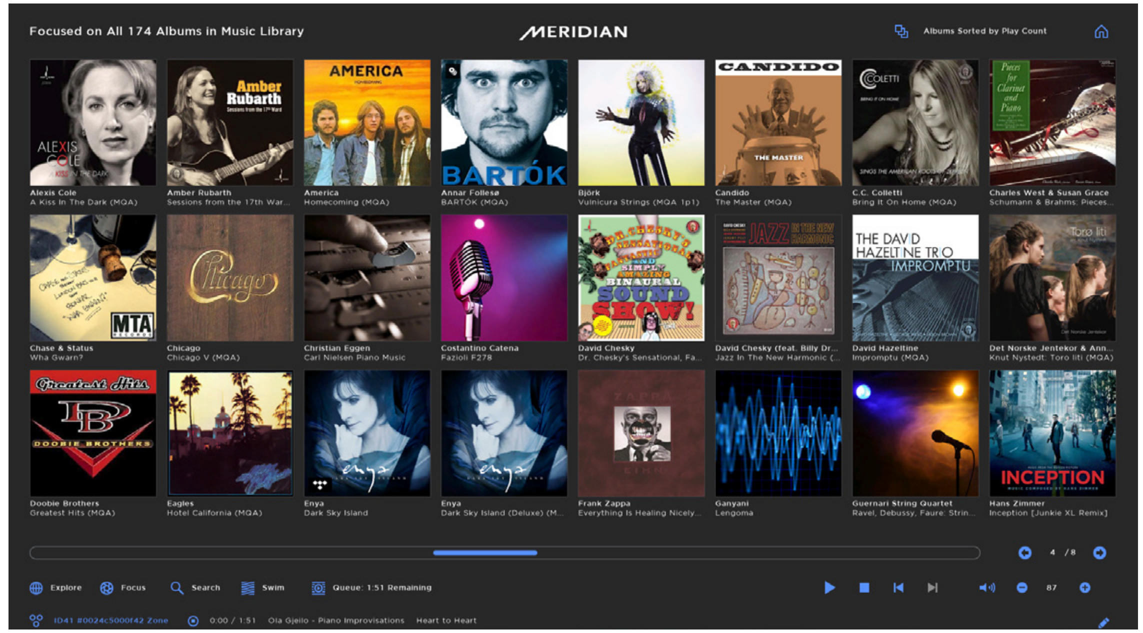
It's difficult to describe the sound of MQA, not because what it does is subtle, but rather because the improvement is so profound. No matter what I write, it won't fully convey the experience of listening to MQA.

I've been describing the sound of audio components as a full-time occupation for nearly 27 years, writing that this amplifier, or DAC, or other product delivered greater soundstage depth, or more accurate timbre, or increased transient fidelity. MQA also does all that, but on a scale that dwarfs all other differences I've heard between components within a product category or between digital formats. Moreover, the gestalt of listening to MQA is fundamentally different than, for example, listening to different examples of DACs, or even comparing CD-quality audio to so-called "high-resolution" digital audio with high sample rates and long word lengths. MQA is so much better sounding than any other digital that it's like hitting the "reset" button on an entire branch of audio technology.

On a macro level, the most significant sonic improvements rendered by MQA are in the spatial aspects of music (soundstaging, depth, bloom, air around images), timbre and tone color, resolution of fine detail, and transient fidelity. These specific sonic improvements translate directly into a sense of ease, naturalness, and musical communication that eludes even 192kHz/24-bit "high-resolution" files.

Starting with soundstaging, MQA completely eliminates the flatness, congestion, and homogenization of conventional digital. We've all lived with this distortion on standard-resolution (CD quality) digital, and heard it to a lesser degree on high-bit-rate recordings. But MQA is simply a game-changer in this regard, even compared with 192kHz/24-bit. By comparison with MQA, conventional digital miniaturizes the acoustic space, shortens reverberation time, truncates instrumental decays, fuses instrumental images together, and destroys the dimensionality that fosters the impression of hearing individual objects in space. On every single MQA recording I auditioned (hundreds of tracks from dozens of albums), I was struck by the completely natural and organic spatial presentation. Even close-miked studio recordings benefited, with greater separation of instruments and less of a homogenized, closed-down feeling.

But the more space and depth on the recording, the more that MQA reveals that space. Fortunately, one of the most spacious orchestral recordings I know happened to be on the Sooloos server in MQA—the spectacular Keith Johnson recording of *The Firebird* on Reference Recordings (with *The Rite of Spring* and *Song of the Nightingale*). I have the CD version, along with an excerpt from the 176.4kHz/24-bit file, which is a bit-for-bit copy of the bitstream created by the A/D converter when the recording was made. Note that the original 176.4kHz file has a bit-rate of 8.5Mbs, and the MQA version a bit-rate of 1.4Mbs. The MQA rendering of this recording was simply astounding, even in comparison with the high-bit-rate version. The back of the hall behind the musicians seems to expand rearward, and with it, the spaces between the rows of musicians, like pleats in a cloth unfolding. Instruments in the back of the hall



were not just portrayed as several feet behind the speakers, but as dozens of yards behind the loudspeaker plane. The reviewer's cliché of the wall behind the speakers vanishing took on a entirely new meaning with MQA. Moreover, the sense of transparent air between the front and back of the soundstage was palpable, charged, and vibrant, just as you hear when listening in a concert hall. The listening room walls disappeared, replaced by the hall's expansive acoustic.

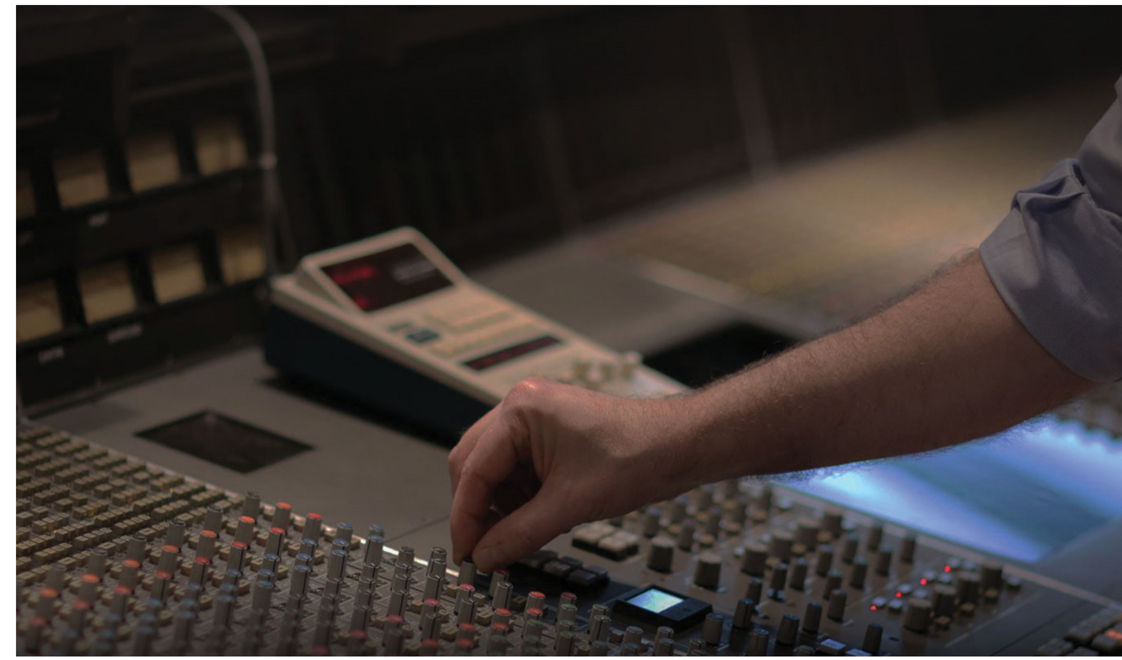
Within this vast space, individual instruments, and sections of them were clearly delineated from each other. MQA had greater clarity and vividness, largely because of the greater timbral fidelity (which we'll get to in a minute) but also from the palpable bloom around tightly focused images. The sound was the opposite of thick, congested, and confused. Moreover, the bloom seemed to extend in three dimensions, the sounds of instruments radiating in all directions, further increasing the sense of 3-D body and tangibility to images. By contrast, conventional digital sounds flat and two-dimensional. Moreover, MQA had a tremendous sense of what Jonathan Valin calls "action"; the sense of sound expanding outward from an instrument in response to the dynamic envelope created by that instrument. This was particularly apparent on the *Rite*, with its sometimes violent attacks by the horn section.

It wasn't just the outward bloom of instrumental attacks that was three-dimensional, but also their decays. Reverberation tails were more closely associated, in space and in timbre, with the instrumental images that created them. These qualities imbued both instrumental and vocal images with a palpability and presence that was almost spooky. I could cite many examples, but listen to how Eric Clapton's guitar amplifier on "Three O'Clock Blues" from *Riding with the King* isn't just another sound within the sonic tapestry, but a distinct presence, with a compact image surrounded by the studio's space and air. The sound conjured up a vivid impression of the amplifier in the studio, with Clapton standing next to it. I compared the sound of this track streamed from Tidal in CD quality against the MQA file; that *frisson* of realism was simply missing from the CD-quality version. This track is notable for another phenomenon of MQA that I'll describe later but mention here: the startling immediacy of instrumental or vocal entrances. Clapton's vocal just pops out of the studio acoustic with a spatial and timbral realism that is startling.

All these qualities culminated in *The Rite of Spring*, perhaps the great Keith Johnson's best-sounding orchestral recording. The MQA version of the *Rite*, played through the Meridian 808v6, Constellation monoblocks, and Magico Q7 Mk.II speakers at concert-hall levels, was the single most realistic reproduction of an orchestra I've heard in my life, from any format and on any playback system. I'd never before felt this composition's raw, primal power to nearly this degree. It was beyond thrilling.

I should mention here that my system is extremely resolving of spatial cues. I regularly use the EnigmAcoustics Sopranino self-biasing electrostatic supertweeter atop the Magico Q7 Mk.II for its ability to expand the sense of space and increase the resolution of low-level detail without changing the tonal balance. With MQA recordings, the Sopranino rendered an even greater contribution to the system's ability to accurately present spatial cues. In addition, the dedicated AC lines, extensive Shunyata power conditioning, and incomparable Critical Mass Systems racks all worked together to fully reveal the low-level spatial cues of which MQA is capable. These cues are fragile, and easily lost in the playback chain. MQA can deliver the spatial cues; it's up to the playback system to resolve them. This isn't to say that you need a mega-system to hear these qualities, only that MQA didn't appear to be the limiting factor when providing the source to a state-of-the-art playback system. I heard exactly the same qualities from MQA when listening through headphones or to my desktop system with the $299 Explorer² DAC.

As with soundstaging, MQA's rendering of timbre is unlike any other digital. MQA's most salient timbral quality is a lack of hardness and glare overlaying instrumental textures. Digital has always imparted a synthetic character to timbres that dilutes realism, decreases engagement, and fosters rapid listening fatigue. I've written in the past that analog's distortions are easier to overlook than are digital's distortions because digital's distortions are woven into the music's fabric while analog's distortions seem to exist separately from it. That observation referred to digital's pervasive glare and lack of tone-color richness. Conventional digital renders instruments as bright and glassy, while at the same time the overall presentation lacks air and treble extension. Today's very best digital recording and playback systems have minimized this phenomenon but have not eliminated it. This fact is made obvious by comparing MQA and conventional-digital recordings of the same title. The removal of this synthetic overlay, coupled with MQA's extraordinary resolution, allowed the beauty of the instrument's texture to be fully revealed. Comparing standard digital with the same recording in MQA was like looking at a very old painting or fresco that had been blackened by centuries of neglect before and after a meticulous restoration. And to top it all off, the MQA file is, astonishingly, much smaller than the original.



No matter what the instrument, the tone color was dense, rich, and saturated. I'm not sure if this improvement was the result of greater resolution of timbral information, or if MQA's ability to remove glare and hardness simply unmasked the instrument's natural tone color. Whatever the cause, I had the ability to hear, with vivid clarity, the mechanism by which the instrument makes sound. From air moving past a vibrating reed, to a string plucked or bowed over a resonant wooden body, to percussion being struck, there was a lifelike realism to the sound. This realism of timbre, coupled with the image focus and bloom already mentioned, gave instruments and voices palpability and vivid immediacy. In fact, this palpability and vividness was so pronounced that instrumental entrances were startling in the way they suddenly appeared in space. This was one of the first characteristics of MQA I noted in my initial audition more than two years ago. A curious effect of this quality was that I found myself visualizing the instruments, and the acoustic spaces they were in, without any conscious effort.

Removing the distortion overlaying timbres made for a decidedly relaxed and non-fatiguing experience. The sound's ease opened up my ears to be more receptive to the music's expression, as though I no longer had to hear through the sound to get to the music's meaning.

MQA's unparalleled timbral and transient fidelity was apparent on every instrument, but I'm going to single out three that sounded much more lifelike than their conventional-digital facsimiles—piano, the human voice, and cymbals. With piano, the MQA reproduction was completely lacking the glassy hardness that often accompanies the attack of notes, particularly in the higher registers. The texture of decaying chords was simply unparalleled, with a richness of tone color and harmonic complexity that has eluded any previous recording technology. In addition, left- and right-hand lines were much better differentiated, each note clearly heard on its own. The sound of the hammers hitting the strings was exactly as you hear it in life; conventional digital by contrast renders this component of the sound as a pale synthetic imitation. MQA conveyed much more of the pianist's "touch," and consequently, of human musical expression. Just compare Keith Jarrett's *The Köln Concert* in CD-quality and MQA versions; with MQA I could *feel* that expression so much more deeply. And then there's the spectacular sense of the instrument surrounded by, but distinct from, the acoustic space. Listen, for example to Costantino Catena playing Debussy [Fazioli] for an idea of the gorgeous harmonic richness and accurate dynamic and timbral portrayal of hammers hitting strings, and of the lush space that MQA is capable of conveying.

The human voice as rendered by MQA is particularly compelling. I was repeatedly startled by vocal entrances, as when a singer comes in for the first time after an instrumental introduction. The voice seems to suddenly appear from nowhere, with a compact image completely separate from the rest of the instruments and surrounded by air and bloom. More importantly, voices had a human quality, which had the effect of making the singer's job of "selling the lyric" that much easier. The realistic and natural rendering of timbre fostered an intimacy with the vocalist that engages the heart in the song's expression. I've been listening to Joni Mitchell's *Hejira* since its release but listening to the MQA version made me consider this well-worn record anew. Sonically, the most amazing vocal sound in the library that MQA sent to me was undoubtedly Frank Sinatra's 1957 *Close to You*. The idea that a hi-fi system is really a time machine struck home when listening to this record; Sinatra was palpably, believably right there, now, not in a studio nearly 60 years ago.

The third instrument that I'll single out for discussion is percussion, particularly cymbals. The very fine texture of the stick hitting the cymbal

(I'm talking about gently played ride cymbal rather than a hard-struck crash cymbal) was exquisitely resolved, and the following shimmer and decay was infused with density of detail and purity of timbre. These initial transients sound dull and muted on CD by contrast, and the shimmer lacks any textural nuance. It was as though a thick blanket had been removed, opening up the sound both tonally and spatially. The cymbals seemed to float in space, their decay hanging in air, just as we hear from the best analog. The texture of cymbals was much more organic; digital has always had a tendency to make cymbals sound like bursts from an aerosol spray can. All transient information was better portrayed by MQA, with a suddeness of attack but without etch, and a sense of an object striking another object, rich with all the timbral and spatial cues that tell you about the objects' material composition, size, and shape.

I suspect that I've chosen these three instruments as exemplars of MQA's qualities because they have always been the most colored by the distortion mechanisms that plague conventional digital. But I should stress that these examples extend to all the other instruments as well.

These impressions were garnered listening to full 44.1kHz or 48kHz/24-bit FLAC files, which had a bit-rate of about 1.3–1.5Mbs. As described in the technology sidebar, the 24-bit MQA files can be truncated to 16 bits to work with Apple AirPlay and other systems that are limited to 16 bits. Internet streaming services can truncate the 24-bit MQA file in increments of 2 bits if the service detects that your Internet connection lacks the bandwidth to stream 24-bit files. The file will still "unwrap" when it encounters the decoder, but not as well. In that situation, the information in the octave above 20kHz become progressively lossy rather than lossless.

How does MQA sound in its lowest-quality version, as 16-bit files? MQA sent me some CDs that had been truncated to 16-bits, which had a bit-rate of about 650kbs (half the bit rate of Red Book CD). Playing these CDs in the Meridian 808v6, I was surprised by how good they sounded given such a low bit-rate. Instrumental textures weren't as smooth and the treble didn't have the liquidity of 24-bit MQA, but the sound was still better than CD. Incidentally, when playing these special discs the 808v6's display showed the original sampling rate of the source used to create the MQA file. It will take some getting used to the idea that a 650kbs file, played on a CD player, can unfold to 88.2, 176.4 or 352kHz sampling rate on playback.

Conclusion

If MQA realized the sonic gains I've described, but did so with massive and impractical file sizes that only committed audiophiles could access at great effort, it would be judged a triumph. But, miraculously, MQA delivers this unprecedented sound quality at a bit rate of 0.8–1.3Mbs—less than that of CD. *Everyone* will have easy and convenient access to this level of source quality, not just audiophiles. That, in my view, is a game-changer for the music and audio industries. Moreover, MQA's ability to remove distortions from existing digital masters is an unprecedented development in the history of audio.

I've dissected MQA's sonic qualities to convey specifically how it improves upon existing technology. That's what reviewers are supposed to do. But missing from this analysis is the profound degree of musical communication, intimacy, and expression that MQA delivers. Music just sounds "right" with MQA, with an ease, naturalness, and engagement that makes conventional digital sound like a pale imitation of the music. It conveys so much more of the musicians' expression and artistry.

MQA is the most significant audio technology of my lifetime. The only question remaining is whether the record companies will release their catalogs in MQA so that all listeners can finally enjoy music as it was intended to be heard.

Feature Master Quality Authenticated (MQA)

MQA Technology at a Glance

Robert Harley

I described in some detail how MQA works in Issue 253 (also available on theabsolutesound.com), but will offer a short primer here, as well as a few new, recently revealed details.

MQA is a very sophisticated and clever process that is based on recent advances in neuroscience (particularly new brain-imaging techniques) that suggest humans are highly sensitive to distortion in the time domain—including the distortions that digital filters (essential to making digital audio work) add when they "ring" in response to musical signals. In effect, this ringing causes the musical signal to spread out, or smear, in time, with, usually, some of the smeared energy occurring *before* the musical signal. This so-called "pre-ringing" never occurs in nature, and is particularly detrimental to sound quality. The problem is compounded with cascaded digital filters, as in the recording and playback chains; each filter's smearing is cumulative. MQA calls this time smearing "temporal blur." Here's an analogy: Every single musical transient is like a hammer hitting a bell (the digital filter), setting it ringing.

Note that the filters required for high-sample rate digital audio, such as 192kHz, introduce less temporal blur than the filters required for 44.1kHz sample rate, which is the primary reason why high-sample-rate digital sounds better than lower sampling rates. Nonetheless, even the filters for 192kHz introduce audible blur (Fig.1).

Temporal blur can be expressed in milliseconds or microseconds—the time over which transient musical energy is smeared. MQA contends that if they had to quantify the performance of a digital-audio system with only a single metric, that metric wouldn't be frequency response (sample rate) or dynamic range (word length), but rather resolution in the time domain—how much temporal blur the system introduces.

CD-quality audio typically has temporal blur of 5ms. Audio sampled at 192kHz has temporal blur of about 500us, or one tenth the blur of CD. MQA has discovered techniques to reduce temporal blur to 10us throughout the entire digital encoding and decoding chain. MQA realizes this improvement in part with more a sophisticated approach to sampling and filtering, and by matching the encoder in the studio to the decoder in the consumer's DAC.

The design of these filters has been informed by the classic sampling theorem, developed in 1927 by telegraph engineer Harry Nyquist and advanced in 1949 by mathematician Claude Shannon and others. (It's sometimes called the "Nyquist-Shannon sampling theorem.") The audio engineering community accepted this thinking, and the filter designs that resulted, as inviolable givens. But it turns out that although Nyquist-Shannon is efficient for communication systems, the criteria are different for high-quality audio.

Sampling theory was developed in conjunction with information theory, which assumed that the signals being sampled were entirely unpredictable. But musical signals, much like visual images and other natural phenomena, are highly predictable (they have a "finite rate of innovation"), which is why lossless compression schemes such as FLAC work, delivering bit-for-bit accuracy at half the bit-rate. Researchers have advanced sampling theory beyond Nyquist-Shannon, and applied those advances to fields such as astronomy and medical imaging. There's a direct parallel between resolving time information in musical signals and distinguishing between closely spaced objects in astronomy and medical imaging. Bob Stuart and Peter Craven applied this new sampling approach to digital audio, guided by the latest neuroscientific research into human hearing.

For example, you may know that the "Nyquist frequency" is the highest frequency that can be correctly sampled for a given sample rate, and is half that sample rate. That's why 44.1kHz has an audio bandwidth of 20kHz—just under half the sampling frequency. But cutting-edge medical-imaging technology, by exploiting the signal statistics, can resolve up to *three times* the Nyquist frequency, allowing much finer resolution of visual detail. Other advances in sampling theory further improve the temporal resolution of musical signals. (For more technical detail, search on Google for "Sampling—50 Years After Shannon" by Michael Unser and "Sparse Sampling: Theory and Applications" by Pier Luigi Dragotti.)

These new sampling techniques allow an entirely different approach to filtering digital audio, which in part contribute to MQA's ability to reduce the filters' deleterious effects on music.

Now we get to the question of how these advances are realized at such low bit rates. As explained in my article in Issue 253, MQA "encapsulates" spectral

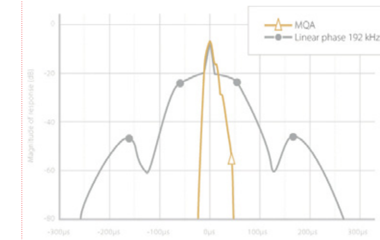


Fig.1 Temporal blur of MQA (narrow) and PCM (wide).

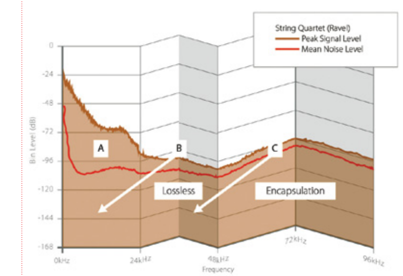


Fig.2 MQA encodes spectral information above 24kHz (B and C) and buries it beneath the noise floor.

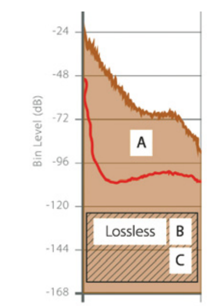


Fig.3 The encapsulated MQA signal at 48kHz/24-bit with spectral information above 24kHz hidden beneath the noise floor (shown as the red line).

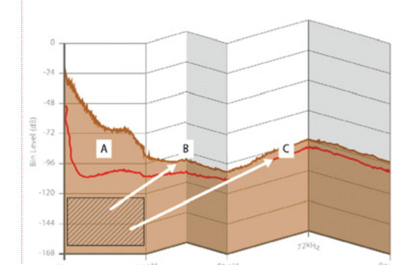


Fig.4 On playback the MQA decoder "unfolds" the spectral information above 24kHz, restoring the sample rate of the original source.

Feature Master Quality Authenticated (MQA)

information above 20kHz by "folding" it back into the 44.1kHz or 48kHz stream and burying it beneath the noise floor (Figs.2 and 3). If you don't have an MQA decoder, the DAC sees this 48kHz/24-bit FLAC file and plays it just as it would any other FLAC file, ignoring the buried information. But if the file finds an MQA decoder, that decoder will "unwrap" the buried information and restore the sample rate and bit depth of the original signal (Fig.4). On the Meridian 808v6, the sampling frequency indicated in the display changes with the recording, indicating the sampling frequency of the original master. (Encapsulation is more complicated that this. Interested readers are referred to Issue 253.)

MQA ties together the entire digital encoding and decoding chain into what is essentially a single system, with a method of authenticating the signal from end to end. When you see the MQA indicator on your DAC illuminate, you know that you are getting the version of the music created by the engineer, with no time smearing, and with no question about the file's provenance or corruption through the distribution channel. There are currently two levels of authentication, MQA and MQA Studio. MQA Studio indicates that the file provenance is assured by the artist or label.

This end-to-end connection of the A/D and D/A converters allows both ends of the chain to work together for optimum sound quality. For example, metadata carried in the MQA file includes information about the many parameters selected by the encoder. The MQA decoder can then, for example, apply a specific reconstruction filter on playback that has a complementary impulse response, improving the overall analog-to-analog performance. The MQA decoder also knows what DAC it is driving, and outputs different data depending on the DAC to compensate for that DAC's particular characteristics. That's why the "unfolded" file isn't available on a digital output; the decoder and DAC are irrevocably bound together.

So far I've described MQA distributed through a channel that can support a 44.1 or 48kHz/24-bit FLAC file. But what about playback systems that support only 16-bit data paths, such as Apple AirPlay, Bluetooth, or many automotive-audio systems? Or slower Internet connections? MQA has anticipated these conditions by making the technology hierarchically scalable. The streaming service knows what device it is streaming to and can truncate the 24-bit MQA file to 16 bits. Some high-resolution information is lost, but the listener still receives considerably more than CD-quality audio. When a streaming service detects that your Internet connection can't support the full 48kHz/24-bit signal, it can reduce the word length in increments of two bits at a time—22 bits, 20 bits, 18 bits, or 16 bits. Each reduction causes signals above 24kHz to be progressively more lossy, with some reduction in quality with each reduction. Note, however, that what the listener isn't getting with these truncated files is something he never had in the first place.

MQA doesn't tinker at the margins of digital audio with incremental improvements, but rather rethinks the entire encoding and decoding chain while maintaining compatibility with existing infrastructure and file formats.

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FAQ about MQA

MQA

How can I get MQA?
As of this writing, MQA is available through the streaming service onkyomusic.com and via download from the Norwegian label 2L (shop.klicktrack.com/2l), which is offering 130 titles at the time of this writing. The streaming service Tidal is poised to begin streaming MQA files, perhaps even by the time you read this. It remains to be seen if other streaming and download services, such as Spotify and iTunes, will offer MQA-encoded music. Streaming MQA in its highest quality requires that the streaming service offer 1.4Mbs streaming rate, which is higher than Spotify's and iTunes' current bit rates.

Will my existing music server handle MQA files?
Yes. MQA-encoded music is packaged as a standard FLAC file. I've seen imbedded Tidal applications within server music-management software; an easy software update will bring MQA-encoded Tidal streams to your server. To decode properly, however, the server must be bit-perfect (i.e., it doesn't corrupt the data).

Can my existing DAC be upgraded to decode MQA files?
It depends on the DAC. Software-based DACs (those that run on general-purpose DSP chips) could be made MQA-compatible through a software update, if the DAC manufacturer chooses to become an MQA licensee. DACs made in the future will have MQA decoding built into the DAC chip itself.

What happens if I play MQA-encoded files through a DAC that doesn't have an MQA decoder?
MQA is backward-compatible with all existing music-distribution infrastructures and consumer-playback hardware. If you don't have an MQA-capable DAC, the MQA file plays just like any other file, but with somewhat better sound quality than CD.

Does this mean that I have to buy my music library all over again?
No. With a service like Tidal, you'll have access to a library of MQA-encoded music. Just how much MQA-encoded music will become available is an open question. Watch theabsolutesound.com for updates on music availability.

When will more DACs with MQA launch?
In January 2016, MQA made available a "developer board" to help DAC manufacturers design MQA decoding into their products. Expect to see a proliferation of MQA-compatible DACs at all price points by mid-summer.

How many MQA titles will be available initially?
We'll have to wait until Tidal "throws the switch" on MQA to find out. However, Tidal stated last April that the company was working "furiously" to encode its music library in MQA.

Will MQA become ubiquitous?
Only time will tell, but MQA has many factors that suggest it will become the standard for music distribution. First, a single file works for all listeners, greatly simplifying life for the record companies. Second, it's backward-compatible; listeners don't need to make a conscious decision to buy into a new format. Third, it removes the question of provenance; the MQA light on the DAC indicates that the file has been signed off by the engineer or record company. Fourth, I believe that artists and producers will demand distribution in MQA once they hear how it conveys to fans more of their artistry. Fifth, MQA offers a protection to the record companies that allows them to deliver master-quality sound without actually distributing their masters.

Isn't MQA just another "lossy" compression system that ignores real musical information in an attempt to lower the bit rate?
MQA is the antithesis of lossy compression. It simply is a more efficient encoding system, coupled with significant advances in improving digital sound quality. —**Robert Harley**

Feature Master Quality Authenticated (MQA)

How Can MQA Improve the Sound of Original Masters?

Robert Harley

There's been some skepticism, and even hostility, toward MQA by posters on Internet forums, much of it engendered by MQA's claim that the technology can deliver sound that's *better* than the original master. Such a feat is impossible, according to the posters, and the claim constitutes *prima facia* evidence that MQA is a fraud. These armchair critics, it should be noted, have probably not read the Audio Engineering Society paper or the patent applications, nor have they listened to MQA for themselves.

But the question remains; just *how* is it possible that MQA can improve upon master recordings? Isn't delivering to the listener the master recording high-end audio's Holy Grail?

Before tackling that thorny question, we must first reconsider what we mean by the "master." In everyday language, the "master" is the physical media (such as analog tape) or digital file that first captured the musical performance or the "final" mix of that performance. But it's more correct to think of the "master" as one step further back in the chain—the signal fed *into* the analog tape recorder or analog-to-digital converter. Correctly storing, distributing, and recreating *that* signal is the true Holy Grail, not preserving the signal after it's been corrupted by a storage medium.

An MQA file created from a digital "master" contains extensive metadata, and uses information about the A/D converter used to create the MQA master. The MQA encoder can then remove the specific distortions introduced by that converter. It's different for every A/D (record companies keep detailed production notes about such things). Where the A/D converter is unknown, a generic algorithm analyzes the file and then removes typical artifacts. Specifically, MQA removes the "temporal blur" caused by filter ringing. (See the overview of how MQA works elsewhere in this article, as well as my in-depth technical feature on how MQA works in Issue 253.) This is one reason why MQA can sound better than the master, and take the listener one step further back in the chain to the signal *before* the master recording.

As I recounted in the March issue's From the Editor, at the most recent CES I heard an original 88.2kHz/24-bit recording made by Peter McGrath (a live performance of *Tosca*), along with an MQA version of that recording. The original had a bit-rate of 4.23Mbs. The MQA file had a bit rate of 1.3Mbs. The MQA version was so much better than the original in every possible way—timbre, dimensionality, ability to differentiate individual instruments, the realism of the applause, and most importantly, in musical communication. Peter was sitting next to me and was equally stunned by this seemingly impossible feat.

After receiving the Meridian 808v6 and MQA-encoded files, I made quite a few comparisons of digital recordings with their MQA counterparts by listening to the non-MQA versions from my library on the Aurender W20, from CD, or streamed from Tidal, all through the same DAC. On every paired comparison, the MQA was decidedly better sounding, in some cases dramatically so. One problem, however, with these comparisons is that the provenance of the conventional digital version is unknown; the MQA file could have been created from an entirely different mastering. That wasn't the case with the Peter McGrath recording; we were listening to his original file created during the recording.

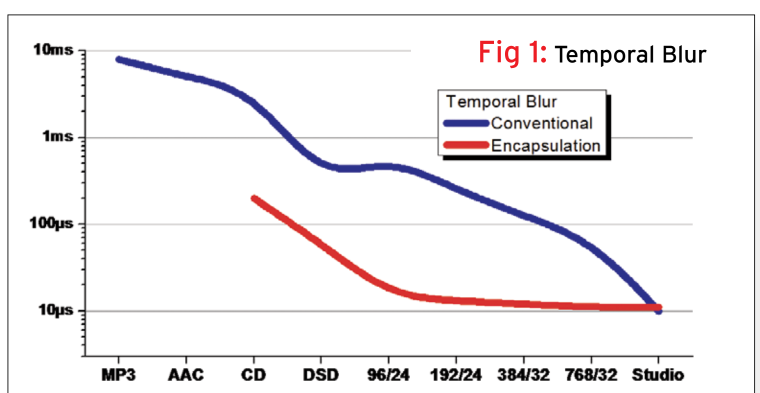

Fig 1: Temporal Blur

To further explore the claim that MQA can actually improve upon the original, I asked MQA to encode a file of a recording I had made live to DAT back in 1988. I ripped the file from the CD release (as WAV) and sent it to MQA with the information about the A/D converter (a Sony PCM-2500 professional DAT recorder). The recording was made at 44.1kHz/16-bit, which was the highest resolution available at the time. MQA returned the file to me, and I compared it with the original. I should mention that I've heard this recording of a straight-ahead jazz quintet hundreds of times on many systems over the past 28 years.

The MQA version was improved, and in myriad ways. On a macro level, MQA seemed to "de-homogenize" the soundstage, presenting each instrument in its own specific location, surrounded by air and bloom. In addition, the soundstage also had more air and dimensionality. The original sounded flatter and less dimensional. Timbres were more realistic; the flugelhorn had greater body, denser and richer tone color, and wasn't overlaid by a synthetic patina. The attack of drumsticks on cymbals was transformed by MQA; they now had the same transient life and immediacy they'd had in the studio. By comparison, the cymbal strikes on the original file were dulled and heavy rather than crisp and clean. This sonic improvement had the musical effect of heightening rhythmic intensity. Another significant sonic quality that affected musical perception was the MQA version's greater resolution of individual instrumental lines. Although, as mentioned, I've listened to this recording hundreds of times, listening to the MQA version I heard, for the first time, exactly what the pianist was playing during the flugelhorn and tenor sax solos.

A more dramatic example of MQA's ability to improve upon 44.1kHz digital recordings

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is Zappa's orchestral album *The Yellow Shark*. Compared to my CD version, the MQA presented much more complexity of instrumentation. This "de-homogenizing" mentioned earlier really benefited this densely orchestrated music. It was like really hearing the compositions for the first time as each instrument's contribution was fully revealed. For one example, some pieces are scored for a celesta, whose delicate but significant part tended to be fused into the music's fabric on the CD, but was clearly resolved spatially, timbrally, and musically on the MQA version.

Recording professionals are hearing the same things. Alan Silverman, engineer of the new Judy Collins record *Strangers Again*, said this about the track "When I Go" (with Willie Nelson): "I've just compared the MQA playback with my original 88.2k 24-bit master and find the MQA to be mystifyingly more satisfying, and not by just a subtle shade. Listening to Willie and Judy, their voices sound much more real, at the same time, they have a textural filigree and detail of tone that I am not hearing in the original master!The same holds for the banjo and the subtle electric guitar in the right channel. I am delighted and extremely enthusiastic about the MQA process."

Morten Lindberg of the Norwegian label 2L, which has had 23 Grammy nominations since 2006, 16 of them for Best Engineering, said, "I have spent many hours with Bob Stuart, listening to original recordings and [am] constantly amazed by the incredible sense of space and clarity brought by MQA." About his title *Magnificat*, Lindberg said, "The MQA sounds great! You can hear positive 'sweetening' of the top-end on both the strings and especially the sibilants of the soloist. MQA smoothes the recording—it was like removing the veil of a net curtain. This is what MQA does." I should note that *Magnificat* was nominated for a Grammy award this year for Best Engineering, and was originally recorded at 352kHz/32-bit. Lindberg was comparing his 27Mbs master to the 1.4Mbs MQA file.

This ability to scrub away digital artifacts and restore natural timbre, air and bloom, dynamic attack, and differentiation of individual instruments is nothing short of miraculous. I had long ago resigned myself to the fact that any recording that had ever been subjected to 44.1kHz/16-bit encoding was irreparably damaged. So much music was recorded at 44.1kHz or 48kHz in the 1980s and 1990s, before recorders with high sample rates and longer word lengths became available to professionals. Of course, the end result of MQA is much better if the original recording started off with a high sample rate, or was recorded on analog tape.

Some ill-informed critics have suggested that any music that had ever been on analog tape and converted to MQA doesn't deserve to be called "high-resolution" on the assumption that analog tape is a low-resolution medium. That view is blatantly false. Many of the MQA files I've been listening to were transferred from the original analog masters, some from the 1950s and 1960s, and I can tell you that they are every bit as high in resolution as any recording made with modern digital technology. Just listen to the MQA version of "With Every Breath I Take" from Frank Sinatra's *Close to You* and then suggest that the sound is in any way "low-resolution." tas

DACs

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Equipment Report

Mytek Brooklyn MQA-Compatible DAC

Game Over

Steven Stone

The Mytek Brooklyn is the first non-Meridian-branded DAC that supports MQA. Because of that, every time it's been shown, whether at a consumer or industry event, it has generated practically standing-room-only interest. I first laid eyes on the Mytek Brooklyn DAC at the 2015 Rocky Mountain Audio Fest when it was only a passive display. As I looked through its list of features and capabilities I thought to myself, "This is one heck of a fully-featured DAC even without MQA."

I reviewed the Mytek 192 Stereo DAC, which was priced at $1595 (now discontinued, remaining stock available at $1095), in the spring of 2013; I was impressed by its sonics, ergonomics, and overall value. The Brooklyn represents Mytek's next step in the evolution of its "entry-level" yet full-featured DACs. The $1995 Brooklyn is not only a DAC, but also a preamplifier for both analog and digital sources, a headphone amplifier that supports single-ended and balanced cans, and a phono preamplifier for both moving-coil and moving-magnet cartridges. The Brooklyn also comes with its own dedicated control app that allows you to operate all the Brooklyn's functions from your computer as well as perform software updates. The Brooklyn even has provisions for linking with an Apple remote.

Technical Tour

Given the Brooklyn's ergonomic flexibility, its front panel is a model of minimalism. The centrally located color LCD is flanked on the left by two buttons and a pair of ¼" headphone jacks while on the right side are two more buttons and a volume/selector knob. To access the Brooklyn's settings you merely push the knob in—the display will change. The top half of the new display furnishes current volume, peak, and average levels for your program material if anything is actively playing through the Brooklyn. Near the center is a small MQA logo, which will light up when MQA material is being played. The bottom half of the display offers four setting boxes, each accessed by the corresponding pushbutton below. Once you've pushed a button the box turns blue (meaning it is available for a change of its setting), then by turning the volume knob you can cycle through the options. Once you have chosen the option you prefer, simply push the button again to save your setting.

Among the adjustments available to the end user are the choices of either line-level analog or phono preamp on the analog input. You can also set the appropriate gain for either moving-coil or moving-magnet phono cartridges. Another option lets you route the output to the headphones, main outs, both, or to auto-sensing. You can also choose a digital or analog volume control as well as a full-output-level bypass option. Input options include the aforementioned single-ended RCA analog/phono, AES/EBU, two SPDIF inputs, TosLink, and USB 2.0. The two SPDIF inputs can also be used for professional DSD SPDIF electrical interface for direct connection to professional equipment such as the Tascam D3000 DSD recorder. The Brooklyn also has provisions for word-clock input and output, as well as an optional 12-volt DC/battery, so you can use the Brooklyn off the grid or with larger dedicated external (third-party) power supplies. Outputs include a pair of balanced XLRs and single-ended RCAs, as well as the two headphone connections on the front panel.

The pair of headphone outputs can be used several ways. You can connect one pair of single-ended headphones to either output, or connect two headphones simultaneously. In addition you can, via a cable adapter available from Mytek ($159), use 'phones with a balanced connection for a true balanced headphone output.

If you use a Windows computer for music playback you will need to install a dedicated driver that is available on Mytek's website. If you use a Mac no additional drivers are necessary for full functionality. I used the Brooklyn connected to a MacPro desktop unit running the latest version of El Capitan with no compatibility issues whatsoever.

Mytek has developed its own application, called Mytek Control Panel, which allows you to adjust and control all the Brooklyn's functions via your computer. Many users may find the app easier to navigate than the Brooklyn's front panel. The app can also perform firmware updates as they become available. Going from firmware version 2.00 to 2.05 took less than two minutes total.

One of the unique features of the Brooklyn is the ability to turn off MQA decoding if you wish.

Equipment Report Mytek Brooklyn MQA-Compatible DAC

Although I'm at a loss as to why you would want to do this on a regular basis, you can use the feature to compare any MQA-encoded file with what it sounds like with no MQA encoding. You can also compare an MQA-encoded file without MQA decoding against a non-MQA-encoded version of the same file. While this provision may be of value to recording engineers and record labels, for your average audiophile it's not a feature that needs to be used, except when he or she is driven by extreme boredom.

Another of the more exotic (but useful for professional engineers) options includes the ability either to use an external clock, or to use the Brooklyn's internal clock to sync with multiple Brooklyn DACs for multichannel playback, or to sync with digital video devices that rely on a master clock.

Sound

If you are a devotee of Internet audio sites, especially those that feature "reviews" by amateurs, you've probably come across reviewers who swear they can, after a brief listen, discern what DAC chip was being employed in a digital device merely by its "sound." My response is: "Good for you!" I'll readily admit to not having that ability. When I review a digital product like the Brooklyn I listen principally to my own recordings and compare what I hear with what I heard during the recording sessions and subsequently on other playback devices. Using that yardstick the Brooklyn ranks with the best DACs I've used regardless of price or internal DAC technology. Try as I might I was unable to hear any sonic personality that varied from what was on the recording. Having said that, I could hear differences between MQA and non-MQA versions of the same recordings quite clearly once I learned what to listen for.

When used as a DAC/pre in my nearfield system the Mytek Brooklyn's sonic signature was quite similar to that of the Grace M-9xx DAC/pre, but with more gain (and a lower noise floor), due in part to the Brooklyn's balanced outputs. Both DACs did an excellent job of allowing me to listen deeply into complex mixes, but on MQA-encoded material the Mytek had an obvious sonic advantage.

Harmonically the Mytek Brooklyn is as neutral as it gets, so any warming or cooling of your system's overall harmonic character will have to come from some other component in the signal chain. Bass extension was such that if there was deep bass I could immediately tell. The amount of bass energy and treble extension I heard during playback was primarily a function of the transducers I used, and not due to any audible sonic limitations imposed by the Brooklyn.

The Brooklyn's single-ended headphone outputs reminded me of those of the Grace M-9xx, but with greater ability to drive difficult 'phones, since I also had the option of using the Brooklyn's balanced output mode. With highly sensitive in-ears, such as the 117dB-sensitive Westone W-60s, I could hear a small amount of hiss but no hum. With standard-sensitivity headphones, such as the AudioQuest Nighthawk, the Brooklyn's headphone amplifier was dead quiet. I also tried the Nighthawk with a balanced Silver Dragon cable from Moon Audio. The balanced connection gave them a bit more dynamic verve and low-frequency

SPECS & PRICING

Conversion: Up to 384k/32-bit PCM, native DSD up to DSD256, and DXD
MQA hi-res decoder: Built-in, certified hardware MQA decoder
Digital inputs: USB2 Class2 (32-bit integer, OSX, Linux driverless, all formats), AES/EBU (PCM up to 192k, up to DSD64 DOP), 2 x SPDIF (PCM up to 192k, up to DSD64 DOP), TosLink/ADAT 2 x SPDIF (PCM up to 192k, up to DSD64 DOP), SDIF-3 DSD up to DSD256
Clock: "Mytek Femtoclock Generator" 0.82ps internal jitter
Analog outputs: RCA, balanced XLR, simultaneous, 50-ohm impedance
Headphone outputs: 500mA, 6W, dual headphone jacks
Built-in attenuator: Choice of 1dB-step analog attenuator, separate for main out and headphones, 1dB-step digital 32-bit attenuator, and purist relay bypass
Built-in analog preamp: Line level input or phono mm/mc input, relay controlled
Audio interface function: All digital inputs can be routed into computer via USB2
Weight: 4 lbs.

Mytek Digital
148 India Street, 1st floor
Brooklyn, NY 11222
(347) 384-2687
mytekdigital.com

Associated Equipment
Source devices: 2013 MacPro Desktop with a 3.7 GHz Quad-Core Intel Xeon E5 processor with 16GB of memory and OS 10.11.5, running iTunes 12.4 and Amarra Symphony 3.3, Pure Music 3.0.1, Audirvana+ 2.5, Roon 1.2, and Tidal 1.3
Analog sources: VPI TNT III w/Graham 1.1 tonearm, ClearAudio Victory II cart, VPI HW-19 with Souther SLA-3 'arm and Denon 103/van den Hul cartridge. Vendetta 2B and Rossi LIO phono preamps
DACs: PS Audio Direct Stream Jr. DAC, Cary Audio DMC-600SE Music Hub
Amplifiers: Pass Labs X150.3, April Music S-1 monoblocks, NuPrime ST-10
Speakers: Spatial M-3 Turbo SE with two JL Audio Fathom F-112 subwoofers. Audience 1+1, Role Audio Sampan FTL, Dali Opticon 1, ATC SCM-7 II, one Velodyne DD 10+ subwoofer
Cables and accessories: WireWorld Silver Starlight USB cable, WireWorld Eclipse 7 balanced interconnect, AudioQuest Carbon USB cable, J-Cat USB cable, AudioQuest Colorado single-ended RCA interconnect, Kimber KCAG single-ended and balanced interconnect, Audience Speaker AU24e speaker cables, PS Audio Quintet, Dectet, Octet, and Premier power conditioners

Equipment Report Mytek Brooklyn MQA-Compatible DAC

extension. I also noticed an improvement with the balanced connection over single-ended with the Mr. Speakers Ether C headphones. In balanced mode the Ether Cs had greater dynamic ease and punch. Using the toughest-to-drive headphones in my collection, the Beyerdynamic DT-990 600-ohm version with a single-ended termination, the Brooklyn never maxed out due to power limitations—19dB (–0 dB is max and –99 is the lowest level before mute) was the highest output level I used with any headphone, including the DT-990.

In order to see how well the Brooklyn's phonostage performed I pulled it out of my desktop system and installed it in my room-based setup. I set the analog input to mc phono, put the gain into bypass (full output), and connected it to my VPI HW-19 with a Souther linear-tracking 'arm and Denon 103/van den Hul moving-coil cartridge. This phono system had previously been attached to the $3875 Vinnie Rossi LIO, which I was using as a phonostage. As with the LIO I could adjust the gain levels of the Brooklyn via the volume control, but the Brooklyn also had the option of bypassing the volume control completely. When I compared the Brooklyn's analog volume control with bypass mode, bypass delivered a slightly more open top end and better-defined soundstage and imaging. I found the Brooklyn's phonostage to be as quiet as that of the LIO, and its overall performance was sonically comparable. The LIO had a slightly wider and deeper soundstage but the Brooklyn's focus was a bit more precise. I could listen to either for hours without any complaints.

MQA

Shortly before I began to write this review Warner Music announced an agreement that made it possible for MQA to encode the entire WMG catalog. That is a lot of music. So far I've heard and done critical A/B listening on several systems with MQA-encoded music files from more than a dozen sources including Warner. In every case the MQA file has been sonically superior to the un-MQA'd comparison music file. I even had five of my own recordings, which were predominantly DSD tracks, encoded into MQA. Much to my surprise the MQA files sourced from my own DSD128 masters sounded superior to the originals! In what specific ways do they sound better? They were all spatially more accurate with more decipherable low-level information. On one of my recordings, which was recorded at 44.1/16 with a Marantz PMD-671 field recorder, and featured Chris Thile, Gabe Witcher, and Chris Eldridge playing outdoors, the low-level sounds far in the background were easier to decipher than on the original recording. The sounds from another workshop going on simultaneously over 150 feet way were also easier to hear on the MQA file than on the original.

Another of my field recordings featuring Bryan Sutton and Chris Eldridge playing vintage Martin dreadnaughts that was originally done at DSD128 also sounded better on the MQA-encoded file than on my master recording. Once more the difference was the decipherability of low-level information. It was simply easier to listen into the mix, plus everything within the mix had better delineated dimensional cues. Magic? Voodoo? Not really, if you understand the basics and weaknesses of digital recording.

The weak link of all analog-to-digital recorders (and digital-to-analog decoders) is their ability to handle extremely low-level signals. According to Robert Stuart, "MQA's target for temporal blurring is to do no more harm to sound than passing through a couple of meters of air—it seems trite, but it is actually a profound concept. Simultaneously, but separately, MQA uses advanced sampling and playback methods that particularly stabilize low-level signals and the recording 'noise-floor.' This uses advanced insights from sampling theory and neuroscience." MQA removes the distortions that were added during the recording process.

If you have a digital recording device that uses an analog-to-digital converter, try this test: Record something at maximum levels that peak just below 0dB, and then record the same track at the lowest settings possible. The lower-level recording will have far more additive distortion than the higher-level one. Even when a recording is done at correct volume levels the quiet passages and accompanying background noise will inevitably have higher levels of distortion than the loudest sections. This is not debatable—it's science. If you can reduce these low-level additive distortions the results will be a better-sounding recording. It is really that simple.

Anyone who doesn't understand how digital recording functions may have problems comprehending why MQA works, but even if you don't get the tech, if you critically listen you will hear the audible superiority of an MQA-encoded file when compared with the PCM or DSD original.

Conclusion

As I learned from my mentor J. Gordon Holt, reviewers have a tendency when confronted by a new medium that reduces distortions to be over-enthusiastic in their praise. One of Gordon's regrets was that he wasn't more critical of the first CD player he heard, the Sony CDP-101. The Mytek Brooklyn and its MQA capabilities placed me in a similar situation. So far I've been unable to discern anything sonically negative while listening to MQA-encoded files through the Mytek Brooklyn. My natural tendency would be to write a spittle-flying gobsmacked rave, but that would be giving in to my baser instincts.

Even without MQA the Mytek Brooklyn offers exceptional value due to its versatility, flexibility, ergonomic elegance, and overall high level of sonic performance. Once you throw MQA into the equation, I have to say, "Game over" for any DAC or DAC manufacturer that can't keep up. tas

Equipment Report

Meridian Explorer² MQA DAC

Lowering the Price of Admission

Robert Harley

The $22k price for Meridian's flag-ship MQA CD player, the 808v6, is far beyond the means of all but a few listeners, but Meridian's $299 MQA-capable Explorer² makes MQA accessible to just about anyone. In fact, most people will experience MQA for the first time through the Explorer². I requested a review sample of this popular "pocket DAC" that has been on the market for some time (without MQA capability). A software update in early February added MQA decoding.

This small oval tube of a DAC is designed for personal listening, but with both line output and a headphone jack, can be used in a home system. The Explorer² has one mini-USB input at one end of the oval tube, and stereo 1/8" line-out and headphone-out jacks at the other end. One of the three LEDs indicates whether Explorer² is decoding a standard file (white), an MQA file (green), or an MQA Studio file (blue). The two other LEDs indicate sampling rates of 88k/96k and 176k/192k, respectively. Mac users can plug-and-play; Windows users need to download a driver from Meridian.

Inside the extruded aluminum case is an asynchronous USB interface, analog volume control, and Meridian's apodizing digital filter. The filter and MQA decoding run on an XMOS DSP chip with 1000MIPS of horsepower. For comparison, Meridian's 808v2 CD player that introduced Meridian's apodizing digital filter made do with 150MIPS of DSP power. The Explorer²'s filter upsamples incoming data to 176.4kHz, but passes 192kHz data natively to the TI PCM5102 DAC. Output impedance is 0.47 ohms.

I listened to the Explorer² primarily in my desktop system with the Audience 1+1 V2+ single-driver speakers, Audeze LCD-4 and PSM M4U 2 headphones (driven directly by the Explorer²), and the headphones powered by the Moon by Simaudio 430HA headphone amplifier with the Explorer²'s line output driving the headphone amp via Audience Au24 1/8"-to-RCA cable. I also listened to the Explorer² in my main system, fed by an Aurender W20 playing MQA-encoded files as well as conventional PCM files. Note that the Explorer² won't work with a Sooloos system, which doesn't support the USB interface (Sooloos is network-connected only).

Playing standard (non-MQA) files, the Explorer² proved itself to be a good $299 portable DAC. The treble was fairly clean, dynamics were wide, and the sound was reasonably resolved and transparent. It's a huge upgrade from the computer's audio output, boasting much smoother treble and more liquid midrange. The Explorer² was significantly more dimensional and spacious, with better differentiation among instruments. By comparison, the computer's audio output was grainy and flat. The Explorer² brought the sound quality up to an audiophile level. I would characterize it as a solid and competent performer in the category when compared with other products in the very competitive low-priced portable DAC category.

But the Explorer² morphed into an entirely different animal when decoding MQA files. The disparity in sound quality between standard files and MQA files was large, and far greater through the Explorer² than between those same files decoded by the Meridian 808v6. Meridian's flagship CD player/DAC's performance on standard material was significantly better than the Explorer² (as would be expected between a company's $299 entry-level product and $22k flagship), but less dramatically better when playing MQA files. The MQA decoder knows what DAC chip it is driving and can correct for certain DAC shortcomings. The Explorer²'s less-expensive DAC chip apparently benefited to a greater degree from this aspect of MQA than did the superior DAC chip in the 808v6.

When decoding MQA files on my desktop system, through headphones, or even at the front of a world-class reference system, the Explorer² sounded stunningly great. It delivers, to a surprising degree, the MQA experience I described elsewhere in this issue. Playing MQA files, the Explorer² has that sense of realism and presence that defines MQA. This was largely because of the increased dimensionality, along with the removal of the glassy hardness overlaying instrumental and vocal timbres. Even at the front of a massively resolving system of Constellation electronics and Magico Q7 Mk.II speakers, the MQA experience was unmistakable. Of course, it didn't have the sonic performance as Meridian's flagship 808v6, but it came closer than one would expect considering the 74x price disparity.

As I listened to MQA files through the Explorer² and PSB M4U 2 headphones ($299 and $395 respectively) on my PC, it struck me just how good this combination sounded for not a lot of money. This level of sound quality at this price would have been unimaginable not that long ago.

The Explorer² is great way for you to experience MQA for yourself, in a desktop, portable, or even home system. It's a good-sounding DAC with conventional digital files, but spectacular when decoding MQA. It's not the ultimate realization of MQA, but it delivers the technology's musical essence at an eminently reasonable price. tas

Equipment Report

Esoteric N-05 Network Audio Player

Bravo!

Vade Forrester • Photography by Dennis Burnett

You've probably seen the music industry data: Sales of physical music playback discs (CDs and SACDs) are dropping sharply. In their place have sprung up two alternatives: file-based music playback and streaming digital playback. The former, also known as computer audio playback, consists of music stored in digital file format on some sort of storage drive: a spinning hard drive, a solid-state drive (including USB flash drives), or a network drive. To enjoy file-based music playback, you must acquire music files, either by downloading them over the Internet or copying (ripping) them from your CDs. Music files can range from extremely high resolution to CD resolution to lossy compressed formats designed to minimize the space needed to store them. The distinguishing characteristic of file-based music playback is that you must possess the files you play back.

Streaming digital playback consists of music played over the Internet. You don't have to possess any files at all, just subscribe to a service that has access to the files and will send them to you over the Internet when and where you want to play them. Usually, streaming services have huge libraries of music files, which allow you to hear a wide variety of music at various levels of quality. Until recently, most streamed files were the lossy compressed variety (e.g., MP3), and even if sound quality was your area of interest, the best quality streaming service readily available was CD quality; however, the file encoding technology known as Master Quality Authenticated (MQA) promises to make streamed high-resolution file playback available. It also drastically reduces the size of downloaded files, making the downloading process much faster.

The music industry data also tells us that the popularity of file-based music playback is waning, as streaming digital playback improves the sound quality it delivers. That makes sense; it's far easier just to turn on your equipment, select the music you want to hear from a huge collection, and hit the Play button rather than have to download files and copy them to your storage medium first.

Which brings us (finally) to the subject of this review: Esoteric's new N-05 Network Audio Player. The N-05 supports both file-based playback

Equipment Report Esoteric N-05 Network Audio Player

and streaming playback from two major services. It's Esoteric's first venture into the file-player/streamer market, so that the disc drive that normally fronts its DACs has been replaced by a file renderer, an ugly-sounding term for the circuit that converts a stored file into a bitstream that the DAC can decode into an analog signal. Esoteric makes some of the best SACD/CD players and DACs on the planet, some of which sell at stratospheric prices. But the N-05 falls into the mid-range pricing area: $6500. For that price, you get a music file renderer (player) and a DAC capable of playing PCM files up to 384kHz/24-bit and DSD files up to DSD128. The USB input accepts DSD256, but the internal renderer plays only up to DSD128. There aren't many DSD256 files for sale yet, but their numbers are increasing. Not only capable of file-based playback, the N-05 also streams music from the Tidal online music service, for which a subscription is needed, and from the Qobuz music service, not available in the U.S. If $6500 seems expensive, keep in mind that Esoteric gear is made to the highest standard and $6500 is among the company's lowest prices for audio gear. As with all Esoteric products, the N-05 comes with a three-year parts and labor warranty.

Many file-based music players provide some sort of internal storage—a hard drive or a solid-state drive—but the N-05 does not; you must provide an external drive, either a USB drive or a network attached storage (NAS) drive. Which should you get? Here is a summary of pros and cons:

USB drive advantages

- The cheapest form of high-capacity storage; single drives with 8 terabytes of storage capacity cost $250 or less.
- Readily available at your local office supply store or Best Buy.
- Quiet.

USB drive disadvantage

- With the N-05, only the folder view of the remote-control ESS app (see below) is available.

NAS drive advantages

- Since NAS drives are installed on a network, files stored there can be played on any music playback device on the network—so you don't need a separate copy on each music player device.
- NAS storage is easy to expand so its capacity can be huge.
- NAS drives are relatively easy to back up; mine has a one-button backup feature.
- NAS drives can be located outside your listening room so their slightly noisy operation won't intrude on your listening environment and they won't need space on your equipment rack.

NAS drive disadvantage

- Depending on the number of bays and drive size, they can be very expensive. They are designed to be left on continuously and use rugged hard drives.

Construction-wise, the N-05 is a typical Esoteric product: a chassis constructed of heavy brushed silver aluminum plates with a sculpted faceplate that curves gracefully around into the side panels. The term "audio jewelry" could have been coined to describe Esoteric products. If that construction sounds heavy, it is: 24 3/8 pounds. In addition to the external hard drive, you will need a local area network (LAN), a WiFi router, and an Apple iPad to run the app, which remotely operates the N-05.

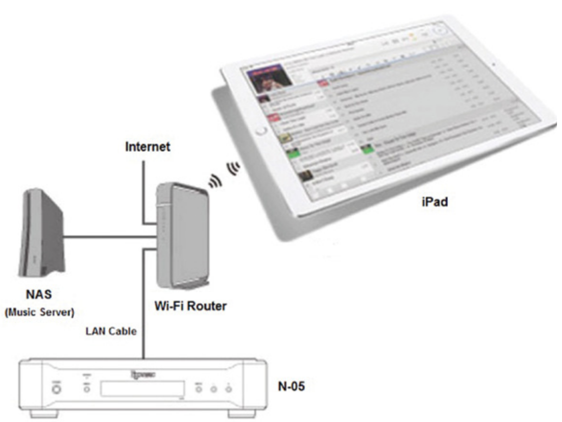
Like Esoteric's disc players, the N-05 can function as a stand-alone DAC, with SPDIF inputs on RCA and optical TosLink jacks, and a USB Type B input. There is also an Ethernet input to connect the N-05 to your LAN and a second USB Type A input into which you can plug an external USB drive (hard drive or flash drive). There is also an RCA digital output jack that enables you to use a DAC other than the one built into the N-05, but of course, it's limited to the highest speed of SPDIF output (192kHz sampling rate/24-bit word length), too slow for the highest-resolution music files.

I can't complain about having external digital inputs, but in my view, since the player part of the N-05 does exactly the same thing as a computer or external digital music player, those inputs seem less useful than those on Esoteric's disc players or DACs. Actually, the fact that the N-05 is *not* a computer (or at least doesn't look like one) is precisely why it will appeal to many users, who just want to turn it on and play mu-

Equipment Report Esoteric N-05 Network Audio Player

sic. The N-05 will play DSF, DFF, FLAC, Apple Lossless, WAV, AIFF, MP3, and AAC music files—virtually all commercially available files except MQA, a very new format with (so far) limited music file availability. Both balanced (XLR) and unbalanced (RCA) output jacks are provided to connect the N-05's analog output to your amplifier or preamplifier. The volume control is in the app, allowing you to drive a power amplifier directly. Esoteric recommends leaving this volume control at the maximum setting, and adjusting the volume with a preamplifier or integrated amplifier.

Esoteric recognizes the sonic superiority of a really high-quality digital clock, so the N-05 uses a high-precision VCXO (voltage-controlled crystal oscillator) to supply "a highly accurate reference clock signal to the digital circuitry. The N-05's large, custom-designed VCXO was jointly developed with Nihon Dempa Kogyo (NDK), a leading manufacturer of crystal oscillators, exclusively for high-quality audio playback. The cornerstone of quality sound, its large crystal element realizes both excellent center accuracy (±0.5ppm, as shipped from the factory) and extremely low levels of phase noise to ensure exceptional sound playback quality."



But that's just a start: Esoteric makes two even more accurate external clock units, the G-01 and G-02, both with styling identical to that of the N-05. The lower-priced $5000 G-02 would probably be matched to the N-05. The N-05 has input and output jacks for the external clock.

A music file player is only as good as its remote control app, and Esoteric's app is called Esoteric Sound Stream, which runs only on iPads and is available free from (where else?) Apple's App Store. I'll discuss Esoteric Sound Stream in detail in the next section.

Esoteric marketing specialist Scott Sefton said that "the N-05 uses most of the same designs and circuitry as the K-05X SACD Player/DAC." That means it uses dual Asahi Kasei Microdevices AK4490 DAC chips, which according to Esoteric's website "are configured in four parallel and differential circuits (8 outputs) driving each channel." Also, "technology developed for the Grandioso C1 (Esoteric's $40,000 top-of-the-line linestage) is also employed in the DAC's power supply, which features Electronic Double-Layer Capacitors super capacitors. This regulated power supply boasts a total capacity of 500,000μF per channel for exceptional low-frequency sound reproduction." That's more than many power amplifiers have!

Since the player and DAC share the same chassis, there's no need for a connecting cable; and the two sections can share the same clock through an I^{2}S connection, which is the least sonically degrading connection—it's used internally on most CD players.

The actual sound of the player is affected by the settings used, which include upconversion to multiples of the input sampling rate, as well as conversion of incoming PCM files to DSD. You can also set the output of the renderer to a specific sampling rate up to DSD256. Like many DACs, the N-05 has several available filters, as well as an off position which doesn't use a digital filter for PCM playback. There are two Finite Impulse Response (FIR) filters, with sharp and slow filter slopes, and two short-delay filters with sharp and slow filter slopes. There is also a filter that cuts off frequencies over 50kHz when playing DSD. If you're worried about ultrasonic noise produced by DSD, that filter should reduce it. Other settings let you select which analog output you want to use, if any; and whether you want to use the digital output (I didn't, so I turned it off). There are also a setting that automatically darkens the display after playback has stopped for 30 minutes, an automatic power saving setting, and a dimmer for the display. A very complete assortment of controls.

Setting Up and Using the N-05

You need to make three connections to get the N-05 up and running: a power cord, an Ethernet cable to your home network, and a connection of the output signal to your preamplifier or integrated amplifier. The network connection provides two functions: It connects the N-05 to the network storage where you store your music files, and it connects the N-05 to a WiFi router on your network which lets your iPad

SPECS & PRICING

Type: Combined music player and DAC
Formats supported: DSF, DFF, FLAC, Apple Lossless, WAV, AIFF, MP3, and AAC. PCM files up to 384/32, DSD files up to DSD128 via internal renderer and DSD256 via external USB input
Outputs: analog: RCA (unbalanced) and XLR (balanced); digital: SPDIF on RCA connector
Drive capacity: None
(requires external USB drive or NAS)
Streaming services: Tidal and Qobuz
Dimensions: 17 5/8" x 4 1/4" x 14 1/8"
Weight: 24 3/8 lbs.
Price: $6500

INTEGRA
(U.S. DISTRIBUTOR)
18 Park Way
Upper Saddle River, NJ 07458
(201) 818-9200
esoteric-usa.com

Associated Equipment
Speakers: Affirm Audio Lumination speakers; JL Audio fathom f110 subwoofer
Amplifier: David Berning ZH-230
Preamplifier: Audio Research SP20
Digital source: SOtM sMS-1000SQ network music player with sPS-1000 power supply, QNAP T-251 NAS, PS Audio DirectStream DAC with Torreys operating system

Equipment Report Esoteric N-05 Network Audio Player

running the Esoteric Sound Stream app control the N-05's operation. I connected the N-05 to my preamp using Audience Au24 SX balanced cables, and to my home network using an Ethernet cable. The drawing below shows how the N-05 connects to the home network.

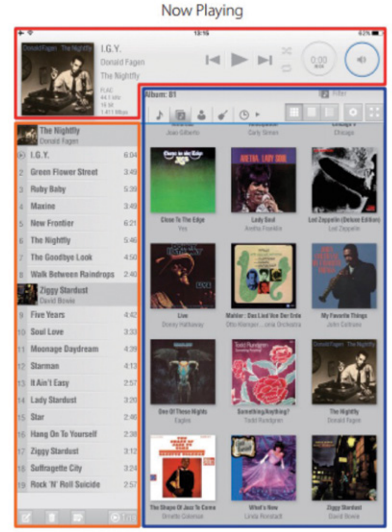
If you buy your N-05 from a dealer, you probably won't get to appreciate how well the unit is shipped: It's packed in three nested boxes and trucked in on a pallet. Outside of personal delivery by the manufacturer, that's the most carrier-proof shipping mode I've ever experienced.

The N-05's designation as a network audio player tells you that the preferred file storage medium is a network drive. The primary purpose of a network NAS drive is to store and retrieve files, but since it has an internal computer processor, it can also run programs on its own, and for use with the N-05, the NAS should run the MinimServer program. MinimServer is free, although donations are encouraged. Many NASes come with MinimServer already installed, or at least with an installation program for it already installed. My QNAP TS251 came with a MinimServer installation program. Often, review components come with throw-away power cords, but the Esoteric cord looked pretty robust, so I used it for the review.

I downloaded the free Esoteric Sound Stream app to my iPad Air2. The set-up process was easy; all I had to do to get started was select the N-05 as the music player and the Minimserver program running on the NAS as the music library. It was also fast, scanning my NAS and preparing a display of all the music there in about five minutes. Many apps have taken over an hour to accomplish that task.

A typical audio component's user interface consists of the knobs and switches on its front panel and a remote control, but a music player's user interface is its remote app running on a tablet computer, in the N-05's case, the aforementioned Esoteric Sound Stream app, which I'll call ESS for short. It was straightforward to use ESS; all you do to play an album is tap its cover art, and ESS shows you the songs in that album, and when you tap one of the songs, ESS starts playing the songs beginning with the one tapped. That's quite easy—if you've ever used a music playback app before. If not, it could be confusing. An icon that looks like a gear brought up the Setup menu, which is where I found the user guide for ESS. To reach it, tap Setup, then scroll down to the bottom of the Setup menu. Tap App, then About, then Help, then English Manual. An online user manual will be displayed. Just scroll down to see how to use ESS. You can also either read the manual online or if you'd like, perhaps print it out from Esoteric's website.

Now Playing



Playlist Library

ESS' main screen is divided into three sections: the top section shows information about the selection that's currently playing; the section on the bottom left shows the playlist of songs selected to play; and the bottom right section shows the library, the albums available to play.

The Now Playing section provided lots of useful information, such as the sampling rate of the song being played and its file type. Another thing I appreciated was ESS' "folder view" at the top of the Library section (not visible in the screenshot above), which lets you view the contents of your music storage folder as if it were a computer folder, which of course, it is. Sometimes that's a more reliable way to find an album you want to play.

Setting up Tidal was child's play. I tapped the Tidal button on the ESS screen, entered my Tidal user ID and password, and Tidal's menu was displayed on the screen. Playing a song or album on Tidal was similar to playing a song or album in your library: just tap the cover art thumbnail for the song or album and it will be added to the playlist. If you want to hear an entire album, touch and hold your finger to its cover art and you'll see several options for playing all the songs. The four buttons to the right of the cover art, from left to right, let you play a selection immediately, play the selection after the current selection finishes playing, add the selection to the end of the playlist, or clear the playlist and play the selection.

You can name and save a playlist if you want to play the same assortment of songs several times. Since I was playing the same list of selections using several different filters and upsampling, I created a review playlist of the selections I used so I wouldn't have to select them individually each time I played them. There's also an icon on the ESS screen that shows the newly installed albums in your library—very useful.

Sound

I started using no filter, then tried a couple of the available filters. I also tried an oversampling option. For most of the review I used the internal renderer, but to test the external inputs, I used an external server connected via USB cable.

In heavy rotation recently at Casa Forrester: Mari Kodama's recording of all the Beethoven Piano Sonatas, a DSD64 Pentatone recording purchased from primephonic.com. I've enjoyed Kodama's performances on CD and SACD (ripped to my hard drive, of course), but welcomed the opportunity to acquire all 32 sonatas in the superior-sounding DSD format. On Sonata No. 32 the piano sound was very powerful, yet detailed. I have always admired the way DSD piano recordings depicted hammer action on piano strings, and this recording is a good example. The N-05 projected the lower registers of Kodama's Steinway D with considerable power, and the well-defined microdynamics revealed Kodama's sensitive phrasing. Through the N-05, I could hear complete, accurate note production, beginning with the initial transient that occurred when a key was pressed and a hammer struck a string, followed by the sustain part of a note, where complete harmonics were portrayed, to the decay part, where the note dwindled off into silence. An amazing recording, reproduced in realistic detail.

Equipment Report Esoteric N-05 Network Audio Player

Next up was old favorite "Miserere" from The Tallis Scholars' album *Allegri's Miserere & Palestrina's Missa Papae Marcelli*, a 96/24 FLAC recording downloaded from gimell.com. This *a cappella* choral work, a setting of Psalm 51, was originally reserved for performance in the Sistine Chapel, and is performed here by a small choral group recorded in a church. The main group is located at the front of the soundstage, while a small solo group is located some distance behind the main group. The N-05 laid out the soundstage before me with considerable realism: The solo tenor appeared centered at the front of the soundstage, the rest of the main group was spread out between the speakers

The Esoteric N-05 Network Audio Player is an auspicious entry.

at the front of the soundstage. The solo group sounded appropriately distant, yet I was able to hear them in considerable detail as the soprano soared to a high "C." Some digital components impose a smear of distortion when the distant solo group enters, but not the N-05. Also, the solo tenor's voice was free from any edginess or distortion, which can be a problem with some components. Although he's located front and center, there was a feeling of spaciousness around the tenor, as though he was singing in a large space, which, in fact, he was.

Another old fave, "Folia Rodrigo Martinez," was ripped as an AIFF file from the CD *La Folia 1490-1701* (Alia Vox AFA 9805). On it, Jordi Savall and his band re-create a historically informed performance of a musical work dating from the year 1490. If that sounds like stodgy, boring music, it's actually one of the more rollicking fun pieces of any vintage I've heard. Through the N-05, the piece sounded quite detailed, beginning with the three opening whacks on the *cascabels* (sleigh bells), each of which sound slightly different, as they do on better components. The baroque guitar plays a tune that is echoed by a harp, and although they're playing the same tune, the two instruments sound somewhat different. I've heard the difference sound more pronounced with some other, more expensive gear, however. The main tune is played by leader Savall on his viola da gamba, a lively melody that constantly varies in loudness and speed. Percussion instruments consisting of castanets, a wood block, and a drum accompany the melody—the castanets sound somewhat distant, and have a slight tendency to get blurred into the background. The drum extends surprisingly deep—I think it goes into the mid-20Hz range—and provides a foundation for the piece. The N-05 reproduced the drum with power and impact, though perhaps a smidgen less of both than I've heard from a few other components. Still, it was a good workout for the subwoofer. The wood block was audible throughout the piece, and the N-05 accurately portrayed the transients of strikes on the block. The viola da gamba's harmonics sounded accurate, although I noted a slight brightness.

So I've used three different musical samples: a solo piano recorded in DSD format, a choral group recorded in high-resolution PCM format, and a small instrumental band, recorded from a CD: a variety of recordings. Now let's see how those selections sound when we switch in a couple of the N-05's filters. (The filters can be set through the use of the Menu button on the front panel.)

First, the FIR1 filter, described in the manual as "a steep roll-off...used to sharply cut signals outside the audio band." This type of filter is sometimes known as a "brickwall" filter because its action outside the audio band is very drastic. It only works on PCM recordings, so I didn't use the DSD recording here. On the "Miserere" track, I thought I heard two effects, both very subtle and hard to detect. First, I thought I detected a very slight hardness with the FIR1 filter switched in. Second, I thought the sound of distant solo group behind the main choral group was very slightly more diffuse, less defined. With the "Folia Rodrigo Martinez," the effect was also slight; I thought the slight brightness or edginess increased a small bit, and I thought the castanets sounded a little smeared. (In case you're wondering, yes, I did fully break in the FIR1 filter.)

OK, now let's switch to the SDLY2 filter, which the manual describes as a "short delay filter with a slow roll-off...used to gently cut signals outside the audio band." I picked this filter because I thought it would be the most different from the FIR1 filter. And it did sound different. On "Miserere," the slight hardness seemed gone, while the sound of the solo group sounded "groupy-er," or better defined. On "Folia Rodrigo Martinez," the edginess was reduced, though was not totally gone, and the castanets seemed better defined. Of all the filter settings, I preferred the SDLY2 filter. There's a DSD filter, too, which rolls off the response over 50kHz, to eliminate the extremely high frequency noise that DSD produces. I must confess I heard absolutely no change when I switched in the DSD filter; maybe those with more extended hearing and speakers with a super-tweeter could hear a difference.

The N-05 gives you the capability to upconvert PCM signals to higher sampling rates. One such option converts them to DSD256 signals, which I decided to try. I figured that if any upconversion were obvious, it would be the DSD upconversion, and I was right, it *was* obvious. On "Miserere," I wasn't smitten by the effect; the sound seemed somewhat homogenized, a bit rounded off. The sense of depth was good, but overall, the sound seemed a bit congested. If the N-05 were mine, I'd leave the upconversion turned off—a personal choice.

So there you have it. I didn't want to spend a lot of time with filters and especially not with upconversion, since the effects are subtle and subject to personal preference. But since the options are there, I felt I should sample some of them and tell you what I found. Your reaction to them might be different than mine.

To evaluate the N-05's external digital inputs, I suppose most people would plug in a computer, and that's what I did; however, rather than use a PC, I used an SOtM sMS-1000SQ network music player that performs most of the functions the N-05 does. It plays DSD and PCM music files and is controlled by an app running on an iPad. It also streams Tidal, although not as straightforwardly as ESS. It's even housed in an attractive chassis that looks nothing like a computer. It differs from the N-05 in that it doesn't have an internal DAC, so it's just what I needed to plug into an external digital input on the

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N-05 and use its internal DAC. I connected the SOtM network music player to the USB input on the back of the N-05, moved the network cable from the N-05 to the SOtM, turned it on, turned on the iPENG 9 app that controlled its functions, selected the USB input on the N-05's front panel, and was ready to play music. I confess my expectations were that the SOtM playing into the N-05 DAC would sound indistinguishable from the N-05 player/DAC combination. Ha!

Beethoven's Piano Sonata No. 32 had slightly weightier sound, with stronger upper bass. I heard a tad more detail and better definition of the overall formation of notes, with especially well-defined leading-edge transients. Great microdynamics made Kodama's interpretation more exciting. "Folia Rodrigo Martinez" sounded rather different—most noticeable was the more extended and powerful bass drum, which had considerably more extension and impact. The slight brightness I had heard from the N-05 was no longer present. Castanets were noticeably better defined and didn't merge into the background noise. Harp and guitar were more distinct and sounded clearly different—and more like a harp and guitar. "Miserere" also sounded different, though perhaps not as much. The main choral group was portrayed with more detail, so the individual voices were more distinctive. The solo tenor's voice sounded more expressive; I could better hear how he phrased the words. The distant solo group sounded further behind the main group, yet I heard more detail in their individual parts. No hint of brightness was evident, although the high frequencies were quite well defined.

"How can that be?" the bits-is-bits crowd will ask. "All the player does is produce a bitstream that is sent to the DAC." Doggone if I know why—all I do is report what I hear.

Comparison

I used my SOtM network music player ($4000 with its sPS-1000 power supply) and a $5995 PS Audio DirectStream DAC. I used the same Audience Au24 SX balanced cables to connect the DAC to the preamplifier, and a $980 Audience Au24 SE USB cable to connect the server to the DAC. Right away, I appreciated that the ESS app provided more information about the file being played, including sampling rate and if it's a DSD file, what speed DSD file it is—information missing from the iPENG 9 app. Otherwise, the iPENG 9 app performed essentially the same as the ESS app, including streaming music from Tidal, although setting up iPENG 9 to play Tidal was an exercise for computer geeks.

Beethoven's Piano Sonata No. 32 sounded even more dynamic—downright thunderous in the climaxes. Kodama uses continuous tempo changes in phrasing the piece, and the SOtM/PS Audio system seemed to make those more obvious. The piano's harmonic structure seemed just a bit more accurate. I've strained to differentiate differences between the N-05's internal DAC and my PS Audio DirectStream DAC; both were excellent.

If I mostly preferred the sound of the SOtM/PS Audio combination to the sound of the N-05, remember their combined cost ($10,945 including a USB cable) was considerably more. And the N-05 only needed one shelf on my rack, another cost factor that should be considered. While I've always thought the SOtM player

looked quite nice, next to the elegant N-05 it looked rather plain.

Bottom line

The Esoteric N-05 Network Audio Player is an auspicious entry into the world of file-based and streaming music playback. It sounds first-rate, looks gorgeous, is easy to set up and use—it's the type of component that makes it hard for a reviewer to pick nits. A wide and useful assortment of filters lets you tweak the sound to suit your taste. The internal DAC is a beauty; if you have a digital device besides the N-05, you can use the N-05's DAC to improve that device's sound.

If you want a music player combined with an advanced DAC, check out this new Esoteric. If you want to get into file-based or streaming music playback but don't want to deface your audio equipment rack with an ugly noisy computer, the Esoteric may be your ticket. Will the N-05 seriously encroach on the sales of Esoteric's SACD players? I think not; as the data showed, audiophiles are increasingly turning to file-based and streaming music playback and the N-05 makes Esoteric-quality equipment available for that purpose. With the N-05, Esoteric's first entry into the network audio player market is a winner. Bravo, Esoteric. tas

Equipment Report

T+A PA 3000 HV and MP 3000 HV

Swiss Sound for Less

Alan Taffel

I've been testing these two flagship T+A components for longer than any other review equipment in memory—over a year now, on and off. One reason is that they are so fascinating; in some ways they're downright unbelievable. Another is that they are so comprehensive (especially the MP3000 HV music player) that there are seemingly infinite modes to evaluate. Throughout this odyssey, T+A has been gracious, helpful, and patient. (I offer my profound thanks to them for indulging me for so long.)

During this extended evaluation period, my perspective on these two components has gone through several phases. It seems fitting to recount them to you, in the order they occurred, so that you can share my journey with these unusual and in many ways remarkable products.

Phase 1: Abject Lust

I've had the opportunity to test quite a lot of very expensive gear lately, but none of those has inspired more lust than the T+A HV series. To uncrate these components is to be smacked upside the head by their obvious top-drawer quality. They are weightier than you'd expect. Hoist one of these things and you know you're getting something for your money. Then there are the aesthetics. These are ruggedly handsome pieces that instantly telegraph "we mean business." Yet there are also stylishly extravagant touches, like the glass inset on top that lets you peek at the classy componentry within.

There are lust-worthy operational touches, too. Large informative screens with touch controls dominate the front panels. The PA3000 HV integrated amp's screen includes very cool, cassette-deck-like power output meters. The screens are flanked by enormous, positive-action knobs that imbue the user with a sense of complete command.

The FD100 remote, which is included with the MP3000 but treats all HV units as an integrated whole, is the most tricked-out device of its kind that I know of. A two-way system, it not only governs every imaginable function, but also displays status information such as the source selected, volume level, and album cover art. Although T+A also offers a nice tablet app, I never felt the need to use it. Meanwhile, HV units communicate with each other via an "H-Link" connection, making operations even simpler and more seamless.

Finally, lust springs from the no-compromise sonically-oriented features. Like *two* AC inputs—one for digital and one for analog—on the player. Want to tweak the digital sound to your liking? The music player's DAC lets you select from four available filters. You can make your choice on the fly from the listening position using that incredibly resourceful remote. For its part, the integrated amp sports an oversized AC input socket, massive heatsinks, and dual sets of binding posts made of rhodium-plated solid brass.

And these are just the visible signs of serious sonic design. The spec sheets and technical details read like audio porn. For instance, as is the case with such benchmark brands as Soulution, CH Precision, and Spectral, the HV-series is ultra-wide bandwidth. T+A employs additional top-tier touches like highly-regulated power supplies and dual-mono, symmetrical, discrete, fully balanced, zero-global-feedback circuitry. But these products are far from copy-cats; T+A has gone in some bold new directions. Most notably, the "HV" in its model names indicates that these pieces run at an unusually high voltage. Whereas most solid-state amp electronics operate at about 100 volts internally, T+A gooses its HV units to a whopping 360 volts—roughly the range of tube gear. As in valve equipment, these voltages ensure that the amplification devices are working well within their operating parameters. Indeed, the HV models utilize only about 20 percent of their amplification transistors' available range. This, in turn, greatly reduces non-linearities. The goal, says T+A, is to mate the naturalness of valves with the speed of solid-state.

With all these aesthetic, operational, and technological goodies, it's impossible to meet these HV components and not fall at least superficially in love with them. I certainly did. Ah, but would the promise be fulfilled? The need to hear what these HV Series components sounded like was becoming urgent.

Phase 2: Corporate Culture Envy

In the course of getting ready to do just that—what with setting everything, meeting company reps, and poring over manuals—I learned a few intriguing things about T+A. One is that those letters don't stand for what you thought they did. (And, by the way, shame on you!) Rather, they stand for Theory + Application. That's

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not hype. As its name implies, T+A has always prioritized pure theoretical research over technological ideology, marketing trends, or price points. As a result, the company's history is impressively replete with innovations that T+A either spawned or was among the first to adopt, including software-based digital filters; multiple speaker advances, like active amplification, transmission-line configuration, and digital room matching; and discrete, switching power supplies.

T+A's culture also includes a genuine commitment to social consciousness. The Herford, Germany, campus consists entirely of green buildings, and the production line avoids substances that are potentially damaging to the environment or worker health. That means no CFCs or even chlorine-based cleaning agents. Most plastics and PVCs are also shunned. Wherever possible, parts and casings are made of recyclable metals, an approach that serves the dual purposes of screening components from external electrical interference while protecting the atmosphere from electro-magnetic radiation.

As much as I admire these corporate touchstones, the element of T+A's culture that I most wish other companies would emulate is its dedication to fair pricing. Almost all high-end manufacturers give lip service to this principle, but T+A walks the walk. For example, as you may be aware, over the past two years the exchange rate between the euro and the dollar has undergone a seismic shift in favor of the greenback. This makes European goods sold in the U.S. cheaper—at least in theory. Yet, can you name any European audio company that has reduced its prices accordingly? I didn't think so. In contrast, when the rates shifted, T+A lowered U.S. prices across its entire lineup. That's just the kind of company T+A is, and I for one applaud it.

Phase 3: Value Incredulity

As I (finally!) embarked on the listening stage of my time with the HV components, the word "Value" with a capital "V" constantly swirled around my brain. Let me tell you why. As readers will know by now, I am a dedicated fan of what I call the Swiss Sound. At first this school was represented by Goldmund and Spectral; now there is Soulution and CH Precision as well. What makes them arguably the best electronics on earth is that their high-speed circuitry and power supplies deliver fast, virtually unlimited dynamics, well-defined transients, vanishingly low distortion, tremendous timbral detail, and near-perfect linearity in both frequency and time domains. The resulting sound is exciting, engaging, and true.

But these virtues come at a price. Circuitry bandwidth must be much wider than usual, power supplies have to be carefully regulated, and the builder is obliged to include extensive protection mechanisms. None of that is cheap. So the first miracle of the T+A HV-series is that it employs all of these design principles yet delivers them at a fraction of the price of the Swiss alternatives. The second miracle is that—significant price difference notwithstanding—the sonic result is a dead ringer for this school's more expensive gear.

How close is the sound? Let me start with the PA3000 HV. At $19,000, this 300-watt integrated amp costs about 15 percent of my reference CH Precision C1/2xA1 combo. Yet when I switch between them the most striking thing I hear is their utter similarity. Of course, I tried to find differences. On the Original Master Recording LP of Donald Fagen's *The Nightfly*, I queued up "The Goodbye Look" and carefully compared bass (identical), vocals (identical), the twang of the solo guitar (identical), and the snap of the xylophone (identical). Most importantly, both presentations preserved the percolating rhythm that make this—and many of the album's other songs—such an enduring pleasure. To be sure, the reference CH equipment creates a wider soundstage, and its tonality is a little more fleshed-out. But I seriously doubt I'd be aware of either of these without a back-to-back comparison.

The biggest difference between the T+A and the CH Precision is at the very top end, where the reference is more refined, though not any more extended. Bear in mind that even this difference, though audible as a touch of roughness, still falls into the subtle category.

SPECS & PRICING

PA3000 HV Integrated Amplifier
Power output: 300Wpc into 8 ohms
Inputs: 4 XLR, 2 RCA, H-Link (HV data bus), LAN (system control), trigger input
Outputs: 2 pairs speaker binding posts, XLR balanced line-level, RCA line-level, 3/8" headphone jack
Input impedance: 20k ohms single-ended, 5k ohms balanced
Gain: 38.6dB
THD: .001% (pre-amp stage), .03% (power amp stage)
Frequency response: .5Hz–450kHz (pre-amp stage), .5-150kHz (power amp stage)
Dimensions: 18" x 6.7" x 18"
Weight: 84 lbs.
Price: $19,000; optional phono module $1650

MP3000 HV Music Player
Inputs: FM antenna, remote antenna, 5 SPDIF (2 BNC, 1 coax, 2 TosLink), 1 AES-EBU, LAN, USB, USB Master-Mode (stick or HDD)
Outputs: USB, SPDIF, H-Link (HV data bus)
File formats: CD, UPnP 1.1 streaming, UPnP-AV streaming, DLNA streaming, WiFi streaming, FM, Internet radio, MP3, WMA, AAC, OGG, FLAC, WAV, AIFF, ALAC
Dimensions: 18" x 6.7" x 18"
Weight: 57.3 lbs.
Price: $15,500

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As evidence, consider that while trying my darndest to ferret out differences like this one, I frequently put down my pen and succumbed to the music. I listened to entire sides of even the most familiar albums. That's an indication of how little these scant distinctions matter, and how miraculously close the PA3000's sound and capacity to captivate come to the higher-buck Swiss Sound stalwarts.

As icing on the cake, T+A offers an optional phono module for this amp. I'm sure such an option, were it available from a Swiss brand, would run many thousands of dollars. But T+A's module costs just $1650. Eminently fair, as always. Naturally, I compared it to my Swiss reference, a Goldmund PH-01. Once again, the similarities vastly outweighed any differences. Speed! Dynamics! Nuance! As before, there were some disparities; however, in this case, they were not all in favor of the reference. For instance, the T+A phonostage is actually more linear and less euphonic than the Goldmund, with purer tonality. On the other hand, the HV's bass is less meaty. A tradeoff—and a tossup. Without question, if you don't already have a high-quality phonostage and are investing in a PA3000, the optional phono module is a no-brainer.

While the integrated's value proposition is based on sonic miracles, the MP3000 HV is attractive partially for the same and partially for different reasons. In the latter category, know that this is one of the most all-encompassing units of its kind you're likely to find. Let me count the ways in which this thing delivers music. First, naturally, there is the superb built-in CD player (more about that later). But that's merely the iceberg's tip. The MP3000 is also a full-fledged DAC that handles USB and SPDIF—the latter via coax, BNC, AES/EBU, and TosLink interfaces. You can also plug a USB hard drive or thumb drive directly into the unit. Then, too, the MP3000 will happily stream music from a NAS, and it will do so through either a wired or a wireless connection. As if all this weren't enough, the MP will play Internet radio and even pick up good old FM. You may be thinking that managing all these source options—and the content within each—must be a nightmare. The truth is that the remote (or the tablet app) makes it easy.

Of course, sonics matter too. In its CD mode, the MP3000 is every bit as impressive as the PA3000. This is one remarkably good CD player. Not only is it clean, open, richly detailed, and dynamic, but it gets completely out of the way of the music and imposes virtually no coloration or digital artifacts. While the CH Precision C1/D1 DAC/transport combo (about $80k) has certain advantages—greater scale, timbral density, and dynamic jump—when considered independently, in the context of a PA3000-based system, the MP3000 actually sounds better. This is not unheard of; the synergies reaped by staying within a given manufacturer's line can be surprisingly powerful. In any event, the HV combination plays music more organically than when mixing and matching, with greater rhythmic drive and coherence.

As an additional reference point, I compared the MP3000's CD playback with that of my trusty Bryston BCD-1. Although this great CD player is no longer in production, when it was available and selling at $3500, it punched well above its weight class. My goal in this comparison was to see if the T+A, even without all those other inputs, justified the extra money. So, did the Bryston come close to the MP3000? No, it did not. Not even a little. The MP3000 is far more open, larger in scale, deeper in dimensionality, more extended, and even more musically compelling than the Bryston.

Another of the MP3000's inputs that squarely hits the sonic mark is SPDIF. This input runs a bit mellower than the CD, but in every other way the two sources are very close. Of course, the SPDIF input has an advantage in that it can handle hi-res source material, and this sometimes gave it the edge over CD. All in all, listening to either of these two sources had me once more agog at what I was hearing.

Phase 4: Reality Check

As it turns out, the MP3000 is not perfect. Specifically, its other sources don't measure up to the benchmark set by its own CD and SPDIF prowess. Switch from either of these to NAS streaming, for instance, and the soundstage and instruments flatten. The sound isn't objectionable, mind you, but nor does it engage. If you must stream into this DAC, be sure to use a wired connection. That route will still be less dynamic, open, and extended than the CD, but not to the same extent as going wireless, which throws a thick soggy blanket over the proceedings. For all I know, this is no fault of the MP3000's and is instead endemic to wireless connections. More research is required, but I can say for sure that this particular instance of WiFi streaming isn't suitable for anything other than background music.

There is better news on the USB front. This interface, at its best, sounds way better than streaming. "At its best" means downloading T+A's custom USB2 driver rather than using the ones that self-install when you first connect the unit to your computer. T+A's research revealed sonic problems with kernel streaming drivers as well as ASIO drivers, so it developed its own approach. The MP benefits from the use of a good USB cable. You'll want to select the "Bezier"—as opposed to the "Bezier plus IIR" or any other—filter. Thus armed, the MP3000's USB sounds quite good. The only problem is that the CD and SPDIF sound *very* good.

The main knocks on USB compared to the MP3000's best inputs are that vocals are more recessed, dynamics are more restrained, and the presentation isn't as three-dimensional. None of these do major damage, so USB turns out to be quite enjoyable. As an illustration, consider Charles Mingus' *Ah Um*. Listen first to the album via USB, and you'll be tapping your feet and marveling at how realistic the brass sounds. The first track "Better Git It In Your Soul" can lose all sense of cohesion in the wrong hands. But the MP3000's USB DAC is fully up to the task. Yet when you switch to the CD, the sound suddenly bursts with more life, the stage opens up, and those tonally convincing instruments now take on three-dimensionality. The same contrast holds true when comparing CD with USB-tethered hard drives.

These discoveries tempered—but didn't eradicate—my original excitement about the MP3000. Naturally, I yearned for USB and streaming that sounded every bit as good as CD and SPDIF. I also found myself wishing that the MP3000's transport handled SACDs and that its DAC supported DSD files. It's worth noting, though, that T+A makes a more expensive music player, the PDP3000 HV Reference DAC/Transport ($22,500).

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That model includes everything the MP3000 HV does (except client streaming functions and an FM tuner), adds in the missing SACD and DSD capabilities, and utilizes a more sophisticated DAC.

Phase 5: Full Circle

After a Reality Check stage that, as noted, somewhat curbed my enthusiasm, I decided to set all that aside and listen afresh to the T+A combo playing either CDs or hi-res files via SPDIF. The sound, once more, just blew me away. I invited fellow TAS writer Karl Schuster to drop by and have a listen. He summed things up perfectly when he described the sound as "spooky good." That spook factor stems from how eerily close these units come to the sound of far costlier Swiss gear. And that, I realized anew, is really the bottom line here.

For $15,500, the MP3000 delivers tremendous versatility and, on its best sources, sound that rivals digital playback from components that cost six times as much. Not all of its sources are up to that standard, so consider your own listening habits and decide if the MP3000 is for you. Similarly, the $19,000 ($20,650 with phonostage) PA3000 not only competes directly with integrated amps that run all the way up to $50k, it holds its own against $120k worth of Switzerland's best separates. This is a component that's not to be missed.

But these HV models not only stand up to their Teutonic brethren, they sound just like them. What T+A has done is to make it possible for audiophiles of more modest (though still significant) resources to get in on the extraordinary build-quality, sonic merit and character, and sheer musical enjoyment of the Swiss School. And that is surely a promise fulfilled. tas

Equipment Report

Berkeley Audio Design Alpha DAC Reference Series 2

Ne Plus Ultra

Robert Harley

Berkeley Audio Design's Alpha DAC Reference broke new ground in digital-audio sound quality when it was introduced two years ago. But Berkeley didn't sit on its laurels and regard the problem of digital-to-analog conversion as solved. Rather, the Alpha Reference's unprecedented technical and sonic performance provided a platform for discovering previously unseen techniques for improving sound quality. Designer Michael "Pflash" Pflaumer spent nearly two years researching these techniques to create the new Series 2.

The Series 2 looks and operates identically to its progenitor. (You can find a full description of the Alpha DAC Reference in my review in Issue 246, or at theabsolutesound.com.) To summarize the salient features, the Alpha Reference will decode all PCM resolutions up to 192/24, has balanced outputs, sports a digital-domain volume control for driving a power amplifier directly, and offers selectable digital filters. The Alpha Reference is designed for all-out performance. That means no DSD decoding and no USB input. To accommodate USB sources you'll need Berkeley's Alpha USB ($1895), which converts USB to AES/EBU or SPDIF. This $1895 box is, by a wide margin, the state of the art in USB conversion. Berkeley contends that including the USB input in the same chassis as the D/A conversion circuitry degrades the sound. If you want to play DSD files you'll need to convert those files to PCM in a computer. The lack of a USB input and DSD decoding speaks volumes about Berkeley's ethos of no sonic compromises. I'm sure that it has lost some potential customers by omitting both, but it's not in Berkeley's DNA to add features that degrade sound quality.

The original Alpha DAC Reference was priced at $16,000; the Series 2 is $19,500. Owners of the original can upgrade for the $3500 difference. (Contact your dealer or Berkeley Audio Design for details.) Note that Berkeley Audio Design is an MQA licensee, and will offer a software update to the Alpha Reference and Alpha Reference Series 2 later this year. The units need not be returned to the factory for the MQA upgrade.

Berkeley is characteristically guarded when describing the Series 2's technical innovations. The company did, however, suggest that the updates include optimized filters and improvements to the analog stage. It's worth mentioning that Berkeley takes a different approach to filtering than do other DAC manufacturers. In the Alpha DAC ($4995) and the Alpha DAC Reference, the digital and analog filters are designed essentially as a single cascaded system, with a custom digital filter running on a DSP chip followed by a hand-tuned analog filter.

Listening

I was skeptical that the Series 2 could offer a significant sonic upgrade considering the performance of the original. How much room was left for improvement? A lot, it turns out. With the original and the Series 2 in my rack fed from the same source (an Aurender W20 and Berkeley Alpha USB USB-to-SPDIF converter) for side-to-side comparisons, it didn't take long to hear the startling advances wrought by the Series 2. In fact, I'm so familiar with the sound of the Alpha Reference DAC, and the Series 2 is so much better, that the differences were readily apparent without comparisons.

The first piece of music I played was "You're Driving Me Crazy" by the Dick Hyman All Stars on the Reference Recording HRx sampler (at 176.4kHz/24-bit). This amazing Keith Johnson recording is exquisitely revealing of DAC quality, from timbral realism, to dynamic expression, to low-level detail, to spatial dimensionality. Even in the short piano introduction, before the band joins in, I could immediately hear the Series 2's improvement in liquidity and dynamic agility. The piano reproduced by the Series 2 was smoother and less glassy, and the transient attacks of hammers hitting strings were reproduced with greater alacrity. I wouldn't have thought it until hearing the Series 2, but the original Alpha DAC Reference has a hint of hardness and glare in the upper midrange. By contrast, the Series 2 has a gentle, flowing ease that creates an instant sense of relaxing into the music—a quality that comes so easily to LP, incidentally. This improvement reminded me of the difference between the Magico Q7 and the Mk.II version of that speaker. The Mk.II sounded less bright and forward, but the two speakers had identical frequency responses. The difference with the Q7 Mk.II, and now with the Alpha Reference Series 2, is a reduction in artifacts that are perceived as brightness, glare, and forwardness. Sig-nificantly, this smoothness doesn't come at the expense of darkened tonal color, a reduction in transparency, loss of treble detail, or a diminution of the impression of air riding above the top octave. Rather, the upper midrange and treble through the Series 2 are full of light and verve, with a full measure of upper-harmonic brilliance and extension despite the apparent lack of brightness.

I've previously contended that there's not a linear relationship between the objective change in a reproduced sound and the musical significance of that change. That is, a "small" change in the signal can have a profound effect on the listening experience. Similarly, a fairly large objective change can have a minimal effect on musical engagement. It depends entirely on the nature of the change. The Series 2's reduction in hardness and glare, and concomitant increase in ease, liquidity, and timbral purity, is one of those differences that engenders a far greater musical in-volvement than the sonic difference would suggest. The powerful combination of fewer artifacts and more musical information puts the brain in a state of musical receptivity. Consequently, you need more than a cursory A/B comparison to fully appreciate the consequences of the Series 2's sonic advances. The Series 2's relaxed ease sneaks up on you during a listening session as you find yourself more deeply engaged in the musical expression. The Series 2's sound is self-effacing, not calling attention to itself but rather getting out of the way of the performance.

This greater timbral liquidity coupled with increased resolution is one of the twin pillars of the Series 2's achievement. The second pillar is the extraordinary dynamic expression realized by the Series 2. Notes just start faster with the Series 2, and do so with more lifelike attacks. Moreover, this transient speed and punch isn't accompanied by hyped edginess. Rather, the Series 2 more accurately renders the way the sounds of instruments start and stop in life. It's not just percussion, piano, a drum kit, and other instruments which produce sharp transients that are rendered more realistically—a wide range of other instruments benefit as well. One instrument that struck me as being reproduced with greater clarity, dynamic expressiveness, and realism by the Series 2 was acoustic bass. During Scott Colley's extended and expressive bass solo on the track "Never the Same Way" from Gary Burton's *Guided Tour*, the Series 2 portrayed the attack of each string pluck with greater speed, impact, and clarity, bringing the instrument to fuller life. Another vivid example is Stanley Clarke's playing at the beginning of "Song to John" from the acoustic trio album (with Jean-Luc Ponty and Al DiMeola) *The Rite of Strings*. He creates a sustained sound during the introduction by rapidly but gently plucking one of the bass strings. Through all other digital I've heard, the individual attacks of each note tend to get blurred together. The Series 2 revealed, for the first time in my experience, the dynamic detail of this passage. I use this as an example to illustrate a point, but the real benefit of the Series 2's dynamic clarity is in the DAC's ability to convey the full measure of a musician's dynamic expressions and nuances. You can simply hear much more of what each player is doing. It's interesting to hear intimately familiar music through a component that breaks new ground in some aspect of sound quality; through the Series 2 I had a newfound appreciation for subtleties of dynamic expression that were previously unresolved. It's impossible to overstate the role of dynamics in fostering a lifelike sense of music-making.

But in addition to that quality, the Series 2's dynamic alacrity infuses the entire presentation with a sense of life, vitality, air, and openness. The album *Live in America* by flamenco guitarist Paco De Lucia exemplifies how the Series 2's dynamic expression brought music to vibrant life. The Series 2 conveyed the speed and zip of the lightning-fast guitar runs, the handclaps, and the *zapeteo* (percussive footwork) with a thrilling vividness, yet the sound never became edgy or fatiguing. This track also revealed the Series 2's superior resolution of spatial cues, particularly image focus and the space between instruments. On this album, and so many others, the musicians just sounded more exuberant and energetic once the sound was liberated from its dynamic confines. There was an unfettered and joyous quality to some music that I simply hadn't fully experienced before.

Low-level detail was also better-resolved, particularly very fine treble textures. Cymbal strikes were more gentle, now sounding more burnished bronze and less "white." The cymbal

Equipment Report Berkeley Audio Design Alpha DAC Reference Series 2

strikes seemed to be surrounded by a larger and better-defined envelope of air, and hung in space longer as they decayed. Moreover, I could hear more deeply into the cymbal's harmonic structure; it was less like a burst of noise (a gross exaggeration) and more like a delicate shimmer. I'm sure that you've all heard that characteristic of low-quality digital in which the treble is bright yet lacking in top-end air and extension. The Series 2 is the antithesis of this sound; it is less bright in the upper-mids and treble than the original Berkeley, yet is more open, extended, and airy. The sound was "illuminated from within," to use Jonathan Valin's evocative phrase. Less bright but more open and extended may seem like a paradox, but it's the way live music sounds. And that's quite an achievement to realize in an audio component, particularly in a DAC.

After I listened to a wide range of file resolutions through the Series 2, it struck me that the most impressive aspect of this DAC isn't its all-out performance with 176.4kHz/24-bit files (which is spectacular), but what it can do with garden-variety CD files. The sound quality difference between CD and high-resolution sources is less stark through the Series 2 than through any other DAC I've heard. It's surprising how good CD can sound when played back through a state-of-the-art system (CD ripped to an Aurender W20 and decoded by the Series 2 with the Berkeley Alpha USB converter). Because so much of my favorite music is available only on CD, this quality of the Series 2 is particularly welcome. I'm surprised that improvements in digital playback continue to extract more and more information from digital recordings, particularly CD. To our great fortune, our CD libraries contain much more music just waiting to be uncovered by improvements in digital-to-analog conver-sion.

Conclusion

The Alpha DAC Reference Series 2 delivers significantly better sound quality than its predecessor, and in ways that matter the most to musical enjoyment. After listening to music through the Alpha DAC Reference nearly daily for the past two years, I'm shocked that the Series 2 can push the state of the art that much further. The fact that owners of the original can upgrade for the price difference between the two models, and that Berkeley will offer MQA capability as a software update later this year, is icing on the cake.

The Series 2 seems to have crossed an important threshold in digital's long slow march toward musical realism. This DAC's sound is open, airy, transparent, highly detailed, lively, and fast, yet at the same time smooth, liquid, relaxed, and non-fatiguing. Throw in a newfound dynamic fidelity, ultra-high resolution, and a stunning rendering of spatial cues, and you've got the recipe for maximum musical engagement. tas

SPECS & PRICING

Input sampling rate: 32kHz–192kHz
Input word length: 24-bit
Inputs: AES/EBU, SPDIF on BNC (x2), TosLink
Outputs: Balanced on XLR jacks, unbalanced on RCA jacks
Output level: Variable: 6.15Vrms at 0dBFS (balanced); 3.25Vrms at 0dBFS (unbalanced)
Digital volume control and balance: 0.1dB steps, 0.05dB L/R balance, 60dB range
Remote control: Volume, balance, input selection, absolute polarity reversal
Digital filter: Custom, user selectable
THD+N: <–110dBFS at maximum output
Firmware: Upgradable through signal inputs
Warranty: Three years parts and labor
Dimensions: 17.5" x 3.5" x 12.5"
Weight: 30 lbs.
Price: $19,500

BERKELEY AUDIO DESIGN
(510) 277-0512
berkeleyaudiodesign.com

ASSOCIATED EQUIPMENT
Loudspeakers: Magico Q7 Mk.II, EnigmAcoustics Sopranino self-biasing electrostatic super-tweeters
Preamplifier: Constellation Altair II
Power amplifiers: Berning 211/845
Digital sources: Aurender W20 music servers, Berkeley Alpha USB
Support: Critical Mass Systems Maxxum equipment racks (x2), Maxxum amplifier stands (x2)
Loudspeaker cables: MIT Oracle MA-X SHD
Interconnects: MIT Oracle, AudioQuest WEL Signature and AudioQuest Wild
Digital interconnects: Audience Au24 USB, AudioQuest Wild Digital AES/EBU
AC: Four dedicated AC lines; Shunyata Triton 2, Triton DP, Typhon (x3) conditioners, Shunyata Sigma power cords
Acoustics: ASC 16" Full-Round Tube Traps, ASC Tower Trap, Stillpoints Aperture Panels
Accessories: Shunyata cable lifters, Stillpoints UltraSS and Ultra6 isolation

Equipment Report

dCS Vivaldi Version 2.0

Evolution

Jacob Heilbrunn

Digital reproduction has long been the Achilles' heel of the high end. The promise has always been there. A supernaturally low noise floor. No pops and ticks and cracks. Ease and convenience of use. But the actual execution has been another matter, which is a roundabout way of saying that "perfect sound forever" has suffered from more than its share of imperfections. The digital nasties, as they have come to be known, scarcely require enumeration.

One of the leading high-end companies that has been striving to relegate those nasties to the dustbin of history, if that's not too grandiloquent a phrase, is dCS of England. For several decades, its team of engineers has been toiling in the digital vineyards to extract headier performance from CDs, servers, and streaming devices. My first real encounter with dCS came several years ago when its indefatigable North American representative John Quick dropped off its then-flagship Scarlatti system. Next came the dCS Vivaldi, which I first heard when it was premiered in 2012 at New York's EarsNova with Rockport loudspeakers. Even in a system I was unfamiliar with I could tell that the dCS system constituted a considerable upgrade, particularly in image stability and overall musical coherence. Fast-forward to today, and the Vivaldi has itself experienced a major upgrade called Version 2.0 that offers exceptionally refined performance. (See Robert Harley's extended review of the original Vivaldi in Issue 233.)

Three of the four units—Transport, DAC, and Upsampler—are the recipients of major software upgrades. In addition, the Upsampler receives a hardware upgrade. These updates are included in all new units. No, if you already own the Vivaldi, your units are not outmoded. Quite the contrary. Existing owners receive the software upgrades gratis if their system was purchased through authorized channels. There is, however, a $2000 fee if they want the hardware upgrade for the Upsampler, which includes new network and USB hardware that nearly doubles the processing power of the Upsampler and streamlines the connection between the rear panel and the primary board. I can report that I was shocked by the difference that this one upgrade alone made in dynamic power and sheer grunt. Among other things, the software changes in the DAC are said to create a better mapping algorithm than the one previously employed. This "better math," as Quick put it, allows the Ring DAC to fire 5 of its 48 current sources per channel approximately 6 million times a second, in a new way that lowers harmonic distortion. For its part, the transport now boasts improvements to its clocking architecture and double-speed DSD upsampling for CD.

It was easy to hear the difference in

Equipment Report dCS Vivaldi Version 2.0

performance after the units had been reprogrammed. Initially, I listened to software beta versions, then a few months later the final one. From the outset, I heard an increase in resolution, much of which I ascribed to a lower noise floor. On piano and orchestra I was wowed by the improved precision and slam of various instruments. Dynamic contrasts, from pianissimo to forte, were far more vivid. When dCS managing director David Steven visited me after the initial updates had been performed, he raised his eyebrows at the overall level of sound. I can only imagine what he would think now that the entire complement has been installed.

On the legendary 1969 Columbia recording, now available in SACD format, of the Philadelphia, Cleveland, and Chicago brass sections, I was almost literally blown away by the refulgence of the tuba on Gabrieli's *Canon a 12*. As with other big, dynamic recordings, the dCS conveyed it with a sense of weight and gravity, majesty and grandeur that I had not heretofore experienced, apart from a live performance. This is a recording that I and other students at Oberlin Conservatory would listen to time and again, a kind of Holy Grail of brass playing. To hear it with this kind of resounding fidelity is thrilling. On the Jimmy Cobb Quartet's Chesky recording *Jazz in the Key of Blue*, I was smitten by an entirely different quality—the sinuous ease with which the dCS rendered Roy Hargrove's plangent flugelhorn playing and Cobb's shimmering cymbals on cuts such as "If Ever I Would Leave." Once more, the size of the soundstage was immense. Indeed, one of the most notable aspects of the Vivaldi 2.0's ability to conjure up a wide, deep soundstage is that you not only win a more lifelike sense of the power of individual instruments but also an improved perception of where they are playing in relation to one another. In fact, I'm reminded of this very quality as I'm listening to a Harmonia Mundi recording by the Freiburg Baroque Orchestra of Bach's violin concertos—the sense of the various string sections combining to form an ensemble is much more intelligible, as are the very quiet passages where the solo violinist is barely touching bow to string. The pace of individual musical lines is more measured and stately, as though rhythm and timing have been subtly but perceptibly improved.

Much of the emphasis of these Version 2.0 upgrades is targeted at improving the ability of users to employ high-resolution sources. I personally continue to like playing CDs, but the industry is clearly moving with the introduction of servers. So when Quick proposed setting me up with a router and a NAS, I was eager to give it a go, especially because I have a cache of private recordings bestowed upon me by recording engineer Peter McGrath that I always enjoy playing.

I used my iPhone to set up a playlist and listened to a wide range of recordings, ranging from classical to jazz to rock. In those instances where I could directly compare a CD to a hi-res file, I have to say that the latter format often appeared to edge ahead by a nose, both in detail and ambience. It also appears as though dCS is going to offer support for Master Quality Authenticated (MQA)—a vivid sign of where the recording industry is headed now that CDs have largely become a remnant of the past.

If you possess the requisite ardor, not to mention the financial means, for what can only be called Himalayan heights of digital playback, I urge you to demo the Vivaldi 2.0. Its flexibility and high quality mark it out as a special product. I can't imagine anyone who wouldn't be happy with it. I know I am. tas

SPECS & PRICING

Vivaldi Transport CD/SACD transport
Outputs: Dual AES/EBU with proprietary dCS encrypted DSD (SACD), DXD or DSDx2 (Upsampled CD); 16/44.1 via AES/EBU, S/PDIF (one RCA, one BNC), or SDIF. Word-clock in, word-clock out
Dimensions: 17.5" x 7.8" x 17.2"
Weight: 51.1 lbs.
Price: $41,999

Vivaldi DAC
Inputs: AES/EBU x4 (each can be used independently or as dual pairs to accept DSD or DXD); SPDIF x3 (two RCA, one BNC); SDIF-2; USB Type B, word clock x3
Outputs: One stereo pair balanced on XLR jacks, one stereo pair unbalanced on RCA jacks
Output level: Variable (maximum of 2V or 6V output user selectable)
Digital filter: Selectable, up to six for PCM and five for DSD
Dimensions: 17.5" x 6" x 17.2"
Weight: 35.7 lbs.
Price: $35,999

Vivaldi Upsampler Plus Digital-to-Digital Converter
Inputs: Network (RJ45), USB (Type B connector), USB (Type A connector), AES/EBU, SPDIF (2 RCA, 1 BNC, 1 TosLink), SDIF-2
Outputs: AES/EBU (x2; can operate independently or as a dual pair to carry hi-res PCM or dCS-encrypted DSD), SPDIF on RCA and BNC
Dimensions: 17.5" x 6" x 17.2"
Weight: 31.3 lbs.
Price: $21,999

Vivaldi Master Clock
Dimensions: 17.5" x 5" x 17.2"
Outputs: Two groups of four independently buffered outputs on BNC connectors
Inputs: Reference Input for use with external clock sources and for software upgrades
Clock frequencies: 44.1, 48, 88.2, 176.4, 192kHz
Dimensions: 17.5" x 6" x 17.2"
Weight: 29.9 lbs.
Price: $14,999

Data Conversion
Systems, Ltd
Unit 1, Buckingway Business Park
Anderson Road, Swavesey, Cambridge, CB24 4AE
United Kingdom
dcsltd.co.uk

Equipment Report

Linn Klimax DSM Digital Streamer

Serotonin Burst

Alan Sircom

Back in 2007, Linn Products launched a game-changing product called the Klimax DS. It was the first post-physical digital streaming player to really take music seriously. It underwent a couple of changes in the intervening years, but the basic package remained essentially unchanged for one very obvious reason—it sounded bloody good. 2007 in streamer years is ancient, but the Klimax DS has stayed the course, and any Klimax player can be brought up to date to the latest standard.

But now there's the new Linn Klimax DSM, and everything has changed. And, of course, what changed with the DS also applies to the DSM, which adds HDMI and line-level preamp functionality to the standard DS streamer. As both of these products feature the Exakt RJ45 links for fully digital active (aktiv, in Linn-speak) connections to many Linn, B&W, KEF, and Kudos speakers, the preamp is superfluous unless you are adding a line-level source.

The core (kore?) of the latest upgrade is what Linn calls its Katalyst DAC architecture. In many digital systems, digital conversion takes place under fairly tightly constrained digital architectural limitations: The circuit itself is often a variation on a theme of the application notes or application board sent out by the chip designer. In fairness to some makers, there are not a lot of options open to an audio engineer faced with a chip that has very tightly specified demands; however, this leads to the somewhat erroneous but understandable concept that any digital product essentially "sounds like its chips."

Those who know their way around digital design don't follow so narrow a path. Some—like Chord Electronics and dCS—go as far as to design their own DACs from scratch. Linn went instead with the Katalyst architecture, and just as the Exakt system launched to the pithy "the source is in the speaker" sound-byte, so Katalyst and Klimax is all about "a DAC is more than just a chip." Katalyst involved scanning all the chip catalogs on the planet in search of devices flexible enough to accept not just a single voltage, but multiple power supply feeds—two for modulation and three for the conversion stage—all fed from an extremely stable and fully isolated voltage source. This is perhaps not unexpected from a company like Linn, which has a long history of making stable voltage power supplies for devices like the Radikal for the LP12 turntable, but the process required looking beyond the "usual suspect" DAC chips, all of which accept a limited voltage input to the chip, despite the fact that voltage is also being fed to a range of different sub-systems within the DAC.

Power feed alone makes a big difference to the performance of the DAC, but that's only part of the Katalyst architecture. The signal is fed through a data optimization process (a 16x/768kHz upsampler working at 35-bit precision, then to a 8x/6.144MHz modulator) before being passed to an array of bitstream DACs, and finally passed to a new analog output driver. The whole digital signal path from upsampler to the main conversion of the DAC array is governed by a high-precision master clock.

This data optimization system largely obviates the need for super-high-resolution files and DSD, because the upsampling process raises 16/44 to a high performance level (24/192 PCM) internally. Given Linn has been able to track what digital streaming users actually listen to (not individual listeners in some kind of Big Brother tracking, but the Linn DS users as a cohort), it seems that we are moving away from local collections of manicured super-high-resolution files and toward online services like Tidal. As a result, the company sees no need to break its own rules about "open, commonly used" formats. Moreover, Linn's Studio Master recordings are sold as 24/192 FLAC files, but are also sold as SACD discs, so I guess they would have a good track on what is and isn't important in high-resolution audio. This is at odds with the somewhat enforced DSD/MQA "acro-nym arms race," and I respect Linn's stance on this.

Linn Klimax DSM fits in the standard Klimax chassis from 2007 (very early Klimax cases need some internal surgery to fit), and a solid-aluminum chassis with internal cham-bering to physically separate digital, analog, logic, and power supply is still a very good way of making a digital device. Linn retained the chassis, and designed the latest architecture to be an almost direct replacement for the existing internals of the predecessor. From a manufacturing standing, that means no retooling or reworking the casework, which given the sophistication of the case is no bad thing. It also allows existing Klimax users to upgrade without losing out.

Linn retains a loyal following for good reason. And the Klimax demonstrates a major part of that good reason. If you are the owner of an existing Klimax, you don't end up consigning that expensive streamer to trade-in or eBay hell. Instead, if you want, your existing Klimax

Equipment Report Linn Klimax DSM Digital Streamer

gets the full upgrade treatment, and you get your old Klimax back in a basic "Renew DS" box. And now it's time to call on the hackneyed car analogy, because that's like driving your one- or two-generation-old Mercedes S Class into the showroom, asking the salesperson if they could turn your old S Class into a new S Class, then give you back the original drivetrain, electronics, safety features, and interior of that older S Class, in a new C Class body. What you do with your Renew DS is up to you: An initial comparison is obvious, but then you could use it to extend your system to another room, adding amp and speakers along the way; you could hand it down to a family member or friend (+500 brownie points guaranteed); or you will get very good money for it if you choose to sell it. Whatever you choose to do, Linn's "leave no Klimaxer behind" plan seems eminently sensible to me.

Because this was a very hush-hush review, with strict embargos and non-disclosure agreements that explained in graphic detail what would happen to my technical area if I even breathed a word about this product before the middle of September, I listened to the new Klimax in a top-spec Linn system in Scotland, and I used the previous-generation Klimax as comparison. This, however, is a decent place to start because the older Klimax is already among the best digital streamers out there, and many such units will be used in this system context. I had expected the comparison process to be a protracted, nuanced affair, trying to define subtle differences between products that really weren't that different. So, out came "Son of a Preacher Man" from Dusty Springfield's justly famous *Dusty in Memphis* album [Phillips], which sounded extremely good on the older product. Two bars into the same track on the new Klimax and it sounded like she was singing with a band, where the older model now sounded like she was singing to a backing track. It was as if a group of better and better-rehearsed musicians had turned up. In truth, it took longer to acclimate myself to the conditions than it did to parse the differences between the two models. In the context of a system you know, if you already have a Linn Klimax the amount of time you will need to audition the new model before realizing you have to buy the new one is about twice as long as it will take you to read this sentence.

Naturally, this hot Linn-on-Linn comparison action came with several Studio Master albums from the Linn label. Perhaps the most significant was the Largo from Beethoven's Piano Concerto No. 3 [Scottish Chamber Orchestra, Linn Records]. This is a wonderful piece of music, played beautifully, and on the older Klimax, listening was a therapeutic experience, as it felt as if your heart rate and blood pressure calmed in the listening. But the new unit took this to new levels. It felt like Beethoven was working on you at a synaptic level. This felt like a serotonin burst. I probably wasn't a smarter or nicer person for the playing of this track, but I felt a burning desire to work some differential calculus while rescuing a kitten. "Get Lucky" from *Random Access Memories* by Daft Punk [Columbia] sounded like "Get Lucky" on the previous model, but on the new one it sounded like "Get Lucky" on cocaine, in gold lamé hot pants, and with glittersprinkles.

Then there's the whole finding new music aspect, which comes as a result of that effortless Tidal connectivity. "String Trio—Continuity Theory" by the Janaki String Trio on its *Debut* album [Yarlung] is not something I would normally play, but I happened upon it almost at random and found it profound and powerful.

"Profound" is the watchword, here. The new Klimax simply makes music more profound. That sounds trite, but it holds throughout. Although the comparison between old and new is an instant one, the difference also has more staying power. With less "filter" in the way of the music, the new Klimax opens the listener up to so much greater depth to his music, and as a result, listening sessions get protracted. Like the best LP replay systems, you can follow every line of the music, without losing sight of the composition and intent of the musician. This was possible with the previous-generation Kli-

SPECS & PRICING

Type: Network music player (DSM with preamp functions)
Analog inputs (DSM only): 1x balanced XLR pair
Digital input: Ethernet RJ45 (DSM adds 3x HDMI Type A, 1x SPDIF RCA also configurable as output, 2x TosLink)
Analog output: 1x balanced XLR pair, 1x unbalanced RCA pair
Digital output: 2x RJ45 Exakt link (DSM adds 1x HDMI Type A)
Supported file types: FLAC, Apple Lossless (ALAC), WAV, MP3, WMA (except lossless), AIFF, AAC, OGG
Audio sample rates: Up to 192kHz
Word lengths: 16–24 bits
Control protocol: Compatible with UPnP media servers, UPnP AV 1.0 control points, OpenHome.org
THD+N (line output): <0.0007%
Dynamic range: >110dB
Gain range: -80dB to +20dB, 1dB steps
Finish: Black or silver
Dimensions: 35cm × 6cm × 35.5cm
Weight: 8.6 kg
Price: Klimax DS, $23,375; DSM, $27,500; Klimax DS upgrade, $5720; Klimax DSM upgrade, $6160

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Equipment Report Linn Klimax DSM Digital Streamer

max, because of that streamer's unfatiguing delivery and inherently "undigital" treble, but the level of musical insight the new model brings to the music just makes the process a lot more organic, in the way you might turn your attention from one musician to another, or from melody to harmony, when listening to live music.

I've not heard every single digital device, but I've heard a lot of them, and new Klimax is the best of the ones I've heard.

It's not just audiophile-approved pieces of music that have this kind of effect through the new Klimax. "The Hunter" by bizarre Icleandic space pixie Bjork, "Because" by the Beatles, "God Only Knows" by the Beach Boys, even "Satellite" by Nine Inch Nails, all captivate, all drag you into the music. This is music replay as orgiastic tribal stuff. The last time it got this atavistic, I'm sure there was a big black monolith and a thighbone involved.

The strange thing about the Klimax sound is you don't tend to talk about the sound, more about how the sound has an influence on you. It is, obviously, extremely detailed, very tonally accurate, dynamic, coherent, and possessed of the sort of ringing-free, effortless treble that when not present makes a lot of digital audio sound, well, digital. But where many other products focus on these aspects of performance, this one does that rare holistic thing that makes you reach deeper into your musical collection, whether locally streamed or on Tidal or Qobuz. If you have spent any time with the original or second-generation Klimax, you'll know what I am talking about here—and what the Klimax DS does, the new Klimax does an order of magnitudebetter.

I've not heard every single digital device, but I've heard a lot of them, and new Klimax is the best of the ones I've heard, or at least the best I've heard that don't cost as much as a decent luxury car. And even at the super-lofty end of high-end digital, the Klimax DSM stands with the best of them, and even shows a clean pair of heels to some of audio's upper echelon with ease. It might even be the best of all of them, and therefore comes profoundly recommended. tas

PORTABLE

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Equipment Report

Onkyo DP-X1 Portable Music Player

MQA Goes Portable

Steven Stone

When I reviewed Astell&Kern's first offering in early 2013, the AK100 ($699), the concept of a high-performance portable music player was new and the AK100 was unique. Flash forward three years—nowadays audiophiles have a plethora of options. Astell&Kern alone offers seven players, from the AK Jr ($499) to the AK380 ($3499). Other manufacturers including Sony, Questyle, Calyx, Colorfly, iBasso, and Fiio have all come out with high-resolution, high-performance players whose prices range from less than $300 to $1300. Obviously, the portable player market has "blown up" into a massive business driven by an increasingly mobile customer base. And the plethora of choices continues to grow.

One of the latest manufacturers to toss its portable player hat into the ring is Onkyo. Its DP-X1 ($799) offers a unique set of features and capabilities at a highly competitive price. The first headline on the DP-X1's web page leads with "Powerful, Portable, ~~Pricey~~." Onkyo's intent is clear: Release a high-value high-performance portable player whose features and performance rival "premium-priced" competition. Given the highly competitive nature of this particular market, Onkyo needed something more than merely "we sound better" to elbow its way in. So, what has the Onkyo DP-X1 got that the others haven't got? MQA. The Onkyo DP-X1 is the only portable player so far, besides its "cousin" the Pioneer XDP-100R ($699), to offer MQA capabilities. But, wait, of course there's more. The DP-X1 also includes a true balanced headphone output (and dual DAC chips) with the capability to drive difficult headphones that usually require beefy external amps to sound their best. Add to all this the ability to access and play popular streaming sources, including Tidal, Spotify, and Pandora, and you have a player that does indeed challenge competitors with much higher pricetags. Will the Onkyo succeed in displacing other competitors on the pedestal of best-priced high-performance player? That is a distinct possibility.

Technical Tour

The DP-X1 uses two amplifiers and two digital-to-analog converters, so it can deliver a true balanced signal. This is the primary difference between the DP-X1 and the Pioneer XDP-100R, which has one DAC and one amp and only supports single-ended headphone connections. With double ESS Sabre ES9018K2M DACs and double ESS Sabre 9601K amps, a balanced output is available via the DP-X1's 2.5mm connection, which is located to the right of a standard 3.5mm single-ended stereo connection. In addition the DP-X1 also has two types of balanced drives, ACG and BT. ACG is short for Active Control Ground drive, which according to Onkyo can deliver "greater stability, increased S/N ratio, and greater spatial dimensionality," as well as "greater delineation for lower frequencies in hi-res audio, and overall robust and taut sound." Inside the owner's manual, Onkyo has a slightly more detailed explanation of AGC. "The basic operating method is the same as the balanced mode, but AGC uses technology to even more forcefully fix grounding standards...output volume is the same as the regular single-ended operation, however."

The DP-X1's storage capacity currently maxes out at 432GB. To achieve this amount of storage you will have to use two 200G micro-SDXC cards. Internal memory is limited to only 32GB, and some of that will be occupied by the OS and whatever apps you choose to add to the DP-X1.

The DP-X1 supports a multitude of audio formats including 11.2MHz DSD, 384kHz/24-bit PCM, MP3, WAV, FLAC, ALAC, and AIFF, as well as MQA files. Basically if it's a music file, the DP-X1 will play it.

Ergonomics

The DP-X1 uses an Android 5.1.1 platform for its OS, which allows it to have all the functionality of a smartphone minus the annoying phone call part. You can access the Internet, send and receive email, and even keep your address book on the DP-X1 if you wish. Internet Access via WiFi also lets you use Google's Play Store to add any apps you wish to the DP-X1. I added Tidal and Onkyo's own "Onkyo Music" store to my review sample. Downloading and installing was quick and easy. The quick part was due to my WiFi's 5.0GHz connection speed, which speed-tested on the DP-X1 at over 100MBps! That throughput rate rivals my hardwired Ethernet connection. How come so fast? A month after I moved into my new home in Denver, CenturyLink offered my neighborhood fiber-optic connections. Since every time it rained I lost my Internet due to the old copper cable's lack of water-tightness, after the tenth service call I jumped at the opportunity, not so much for the speed (which has been nice) but for the reliability. Now even if I lose power my Internet still works for as long as the high-speed fiber-optic modem's battery back-up lasts.

If you already use an Android phone the DP-X1's pages will be quite familiar to you. Unlike some players with their own customized Android-based interface that can limit functionality, the DP-X1 is open to whatever you want including third-party music players and apps. While I didn't try out other player apps since I found Onkyo's supplied one did everything I needed, if you have a player that you're used to, or prefer to use, you can easily add it to the DP-X1. But since the DP-X1 has Android openness, you might download an untested program that could in extreme cases "brick" (make non-functional) your DP-X1, so I would advise some restraint.

Within the DP-X1's settings you have many options for general operations. In the music settings you can choose which form of amplification you wish to use (ACG or BT) as well as eq. The DP-X1's eq functions include five presets as well as 16-band user-selectable ones. Adjusting the 16-band eq requires a steady finger (or stylus) since the delineations are rather close together. The Onkyo also has something called "featured eq" which includes 18 different settings developed for different pop musicians including Buckcherry, Scott Ian, Tim Lopez, Steven MacMorran, Midi Matilda, Leo Nonventelli, Strange Talk, Chris Traynor, and Jim Ward. You can modify any of these eq settings and store up to 1000 custom eq curves.

The DP-X1 has three gain levels. But the differences between level settings aren't so great that you can't use "low" with low-sensitivity headphones. I know this because for the first couple of days I used the default "medium" with a wide variety of headphones before I found the gain adjustments, which are buried among the Sound & Notification settings. Perhaps seasoned Android users will find these nested menus old hat, but for new Android users the Onkyo's menu system will require a learning curve. The onboard owner's manual app is essential reading if you hope to become deft at navigating through the DP-X1's many features. Some adjustments, such as upsampling, digital filter, and DSD upsampling-conversion options, are found within the Onkyo music player app via a drop-down menu. While its settings are not as convoluted as those of some players, the DP-X1's more arcane controls are not intuitive in function or location.

The DP-X1 supports Bluetooth headphones or other playback devices via aptX. Once paired you can send an audio stream to any compatible BT device.

Battery life is listed at 16 hours using 96/24 FLAC files and a single-ended headphone connection. With balanced headphones, battery life will be quite a bit shorter. Also, if you leave the DP-X1 hooked up to a balanced headphone in pause mode overnight, the battery will be exhausted by morning and need a full recharge, which takes somewhere around three hours.

Populating the DP-X1 with music was as simple as connecting it to my MacPro's USB 3.0 inputs. Onkyo has its own file-transfer app called X-DAP Link (PC and Mac), which you can download from its site, but I used another app called Android File Transfer to move files into the DP-X1. This little app popped up every time I connected the DP-X1 to my Mac via the supplied USB cable. One further advantage of this method was that instead of appearing on my desktop as an external drive, which is what occurs with many portable players, the DP-X1 is recognized by the app, but not as a drive so you don't have to wait for it to un-mount before disconnecting it.

The DP-X1 can also be used as a "source device" to connect to other USB DACs. You will need a special cable to accomplish this, but Cables to Go, among other sites, has what you need to make the connection. Once hooked up you have a multitude of options to send files to an external DAC, including upsampling and different DoP (DSD over PCM) file protocols. And if your external DAC is MQA-compatible, the DP-X1 can even output MQA files to that device.

Sound

I've reviewed a fair number of portable players during the past couple of years. With most of them the primary limiting factor in overall fidelity has not been the player itself, but its synergy with the headphones or transducers connected to it. I used a plethora of headphones with the DP-X1 from hyper-efficient in-ears like the Westone W60 to the most power-hungry full-sized cans, such as the Beyerdynamic DT-990 600-ohm version. Even in single-ended mode the DP-X1 had no trouble driving the DT-990s to satisfying levels, and with the efficient ones the low-gain modes delivered sound without hiss or hum.

I used the DP-X1 via its single-ended output for

Equipment Report Onkyo DP-X1 Portable Music Player

several weeks before I received a Silver Dragon adapter cable to go from the 2.5 TRRS connection to a standard 4-connector XLR from Moon Audio. With the adapter installed I tried all the headphones in my collection that use balanced connections. These included the HiFiMan HE-560, Sennheiser HD700, Grado RS-1, Audeze LCD-2.2, AudioQuest Nighthawk, and Mr. Speakers Ether and Ether C. I also tried both of the DP-X1's balanced modes, Bal and AGC. I found the Bal had a slightly higher output level. With several phones, including the Mr. Speakers Ether-C, I preferred Bal overall due to its superior dynamic contrast and bass extension.

Comparing two different portable players is not easy. Making sure levels are the same is the first problem; the second is that switching from one player to another takes more than a couple of seconds, making direct comparisons even more difficult. I set up a test to compare the Onkyo DP-X1 against the Astell&Kern AK240. After listening to several of my own recordings via both players I was forced to conclude that at least with the three earphones I used, the Ultimate Ears RR, Jerry Harvey Laylas, and Empire Ears Zeus, I could not identify any differences between the two players when they were both playing back my own DSD5.6 recordings.

I could spend multiple paragraphs detailing the hows and whys of MQA, but it will be far more efficient for you to look at the video links at theabsolutesound.com ("MQA Explained in Short Videos"). For more information read Robert Harley's technical article about MQA ("Beyond High Resolution"), also on theabsolutesound.com. Finally, if you like questions and answers take a look at this interview with Robert Stuart

on the Computer Audiophile site. On the DP-X1 all my MQA files played without any issues. MQA-encoded files also loaded and played just as fast as regular non-MQA versions.

When I compared MQA conversions of my own recordings with the originals, on some headphones I could not discern any sonic differences, but on those headphones and in-ears that I currently use for reference, such as the Ultimate Ears RR and Mr. Speakers Ether C, I could hear the improved resolution. For me the improvements manifested by the MQA-encoded files were in soundstage specificity, image placement, and low-level details. It was easier to listen into the mix, and to differentiate between sounds that were more homogenized on the non-MQA files. On my recording of Bryan Sutton and Chris Eldridge playing "Church Street Blues" at a workshop outdoors, Eldridge's voice was better isolated from his guitar (whose sound hole was less than eight inches below his mouth). Instead of blending into one sonic entity the guitar and voice were separate and easily differentiated in space. Also some of the subtle variations in Bryan Sutton's picking were easier to discern on the MQA-encoded file.

Conclusion

Yes, there are plenty of options nowadays for anyone looking to acquire a high-resolution high-performance portable player. But if value-for-dollar and maximum flexibility and functionality are high on your list of must-haves, you can substantially narrow down the list.

Taking it further, if future-proofing is among your most-wanted attributes, I can think of only two players that qualify, and only one of those can provide a true balanced output—that's the Onkyo DP-X1.

While the DP-X1 may not be quite as disruptive a new technology as MQA, it does raise the question of why, except for aesthetics or ergonomics, anyone would choose another player if his budget maxed out at under $1000 (except perhaps for the Pioneer XDP-100R, if I were absolutely sure I would never, ever, need a balanced output). I predict that Onkyo will sell a lot of DP-X1 players because it is currently the best value out there in flexibility, functionality, and sound. Recommended? Is that even a question? Onkyo has hit a home run that deserves two trips around the bases. tas

SPECS & PRICING

Operating system: Android OS 5.1.1
Total (current) maximum storage: 432GB
Internal storage/RAM: 32GB including Android OS system area (RAM: 2 GB)
Extended storage: 400GB via two 200GB micro-SD card slots
DAC and HP amplifier: Two ESS SABRE DAC ES9018K2M and two headphone AMP SABRE 9601K
Wi-Fi specification: 802.11a/b/g/n or 802.11ac (Wi-Fi direct / WPS)
Bluetooth support: A2DP/ AVRCP/ HSP/ OPP/ HID/ PAN
Codec: SBC/aptX (Transmit only)
Playable audio formats: DSD/DSF/DSD-IFF/FLAC/ALAC/WAV/AIFF/Ogg-Vorbis/MP3/AAC/MQA
Sampling rates & bit rates: 11.2MHz/5.6MHz/2.8MHz 1-bit, 44.1k/48k/88.2k/96k/176.4k/192k/352.8k/384k 16-bit/24-bit (32-bit float/integer can be played down-converted to 24 bit)
Supported video formats: H.263/ H.264 AVC/H.265 HEVC/MPEG-4 SP/VP8/VP9
Balanced output spec: 150mW + 150mW
THD: Less than 0.006 %
S/N Ratio: 115dB
Frequency response: 20Hz–80kHz
Dimensions: 3" x 5" x 0.5"
Weight: 7.16 ounces
Price: $799

Equipment Report

Sony NW-ZX2 Digital Media Player

High Performance On The Go

Steven Stone

When Apple discontinued its 160GB iPod Classic portable music players, a funny thing happened: Their prices on eBay doubled overnight. And while many tech-pundits see dedicated portable players as an ergonomic dead end (supplanted by ubiquitous smartphones), crowd-funded sales in excess of $3 million for Neil Young's Pono player demonstrate that music lovers still have a healthy appetite for dedicated portable media players.

Sony, which created the first "Walkman" portable player, has been involved with portable audio since its inception, but recently has not been as dominant in the market as it was in the early days. That could change with its latest offering, the NW-ZX2. Priced at $1199, this Android-based player can handle any commercially available music file including DSD128, plus it also plays videos from YouTube, Hulu, and Facebook. The NW-ZX2 has WiFi and Bluetooth support. In short, the new NW-ZX2 does virtually everything an Android-based smartphone can do except make and receive phone calls. And it sounds much better than any smartphone I've heard.

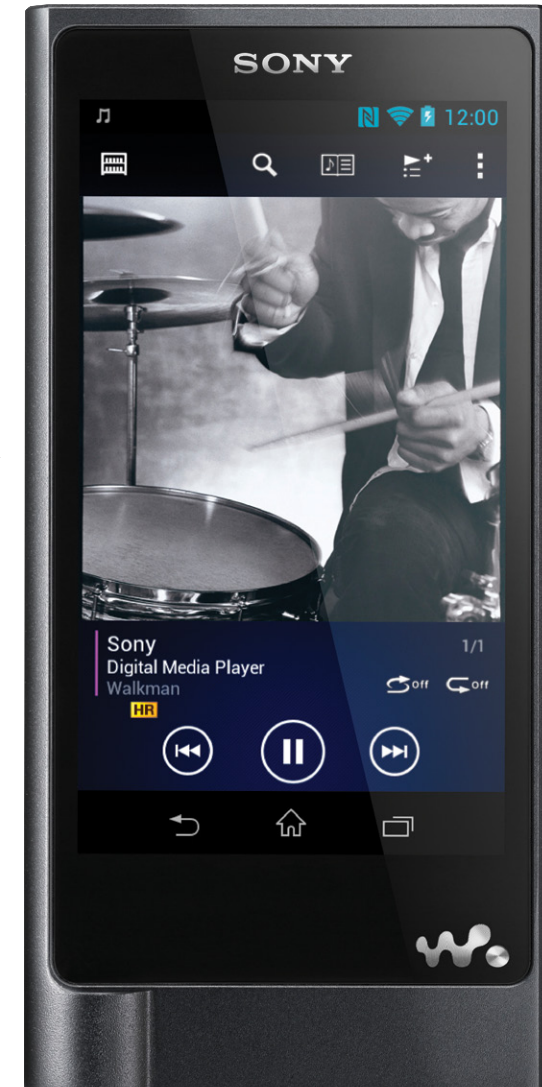
Tech Tour

Instead of an oddly shaped or "look at me, I'm different" case, the Sony NW-ZX2 is conventionally phone-shaped, measuring approximately 2 ½" by 5" by ½". Most of its front panel is a 2" by 3 ½" touchscreen. The NW-ZX2's enclosure has a matte-black anodized finish with just a hint of texture, making it easier to hold than early iPhones with a mirror finish. The back of the NW-ZX2 is inset with textured genuine leather that further enhances its grip-ability. Ever since my first iPod Touch lasted exactly 30 minutes before it jumped out of my shirt's breast pocket and into the toilet, I've valued players with less slippery surfaces that remain in pockets even when gravity nudges them in other directions. The Sony NW-ZX2 feels secure in my hands (or pockets) due to its shape and thickness. Weight-wise, it achieves a happy medium between being neither too heavy (like the Sony PHA-2 DAC/amp or Colorfly C4 portable player) nor too light and unsubstantial like an iPhone 5. No amount of time in your thigh pocket will bend or otherwise alter the NW-ZX2's case.

Sony has incorporated a number of new technologies into the NW-XZ2. First and foremost is its use of supercapacitors to enhance power output capabilities. According to a Sony technical paper, a supercapacitor can augment a Class D power amplifier's peak power output by over three times! This makes it possible for the NW-ZX2's headphone amplifier to produce quite a bit more power during dynamic peaks. Also, the supercapacitors increase battery life by relieving the battery of some of the peak-power demands that can reduce its reserves.

The NW-ZX2 employs two crystal clocks. Sony's previous (but not distributed in the U.S.) player, the NW-ZX1, could only do 44.1, 88.2, and 176.4kHz natively, but the ZX2 adds 48, 96, 192kHz native rates, as well as native DSD64 and DSD128.

The NW-ZX2's chassis is constructed of solid aluminum. The interior of the chassis is lined with gold-plated copper to reduce noise and improve isolation between electronic subsections. Other "tweaks" include use of high-purity solder and MELF capacitors in the analog output stage. These high-cost metal-electrode caps are usually only found in custom-tweaked or megabuck components, and are currently the best parts of their kind available. The NW-ZX2 also employs seven Os-Con caps, three in power filtering and four in the analog circuit.



Ergonomics

Anyone who has spent any time with an Android-based phone or tablet (such as the Sony Xperia) will find himself right at home with the NW-ZX2. Upon startup you will be greeted by that swoopy Android graphic and unlock screen. Once unlocked with an upward swipe (if you choose not to use the password lock), the NW-ZX2 will display its home screen, including whichever app you had open when you last used the device. The NW-ZX2 comes with "Play" as the primary music app. It looks very much like the music app on Sony's HAP-Z1ES full-sized digital player and includes many of the same features, including SenseMe mood channels, playlists, and multiple view options.

Through Google's "Play Store" you can acquire additional apps. I added Tidal as well as Oppo's HA-1 remote-control app. With the preloaded Google Chrome browser you can do anything that you would do with a web-enabled smartphone, including watching videos, logging into Facebook, or reading e-mail. You can also set up the NW-ZX2 so it can instantly access your Gmail account. The only limitation is that the NW-ZX2 needs access to a WiFi hotspot to enable all this space-age connectivity—it has no other way to directly access Web-based content.

If you use and like the Android operating system, you will be very comfortable with the NW-ZX2. But if you are an Android newbie, there will be a learning curve. My review unit arrived without any instructions (it was only the second one in the U.S.), so I had to fly blind during my initial listening sessions. Except for a minor panic attack when I managed to mute the NW-ZX2's outputs (I unmuted it somehow and haven't had the problem since), I had no operational issues with the NW-ZX2. I've gone back and forth, playing tracks from Tidal, then Sony's Play app, then YouTube vids via Chrome with no hang-ups or inordinately long delays between selections. Also I could field e-mail and surf the Web while listening to music with no hiccups. After several weeks of use, the NW-ZX2 and its Android OS have proven to be stable and reliable.

The only notable operational issue I experienced with the NW-ZX2 was when I disconnected it from my MacPro desktop computer. If you merely click "disconnect" from the NW-ZX2's screen, instead of first moving the NW-ZX2's icon from your Mac desktop to the trash, you can corrupt the contents of any micro-SD card mounted in the NW-ZX2. This happened to a 32GB card, and it took me almost two hours of copying to repopulate it fully. To avoid this catastrophe, I suggest following Apple's "best practices" and getting in the habit of always moving USB icons to the trash (or virtually "ejecting" them) before disconnecting the physical device itself.

If you use and like the Android operating system, you will be very comfortable with the NW-ZX2.

One particular ergonomic area where the iPod Classic long excelled was "blind" in-pocket use. As much as I love touchscreen-controlled devices, they are virtually impossible to use or navigate by touch alone. The iPod Classic's selector-clickwheel still rules when it comes to on-the-go use. And while the NW-ZX2 does have dedicated navigation buttons on one side, when it's in screen-save mode (with the screen blacked-out), only the pause and skip-forward and skip-back buttons are active. If you want to adjust the volume you will have to push the unlock/power button or pause button, and then you can adjust the volume by using the up/down buttons located below the power button. With some practice you can probably do this while the NW-ZX2 is still in your pocket and still in its leather case. If you use the shuffle mode and have a wide variety of music with differing "average" volumes, you may find yourself needing to adjust volume "on the fly"; here, the NW-ZX2's push-push scenario for volume adjustment is a less-than-optimal solution. Of course, you could activate the NW-ZX2's automatic volume "leveling," which will adjust all your tracks to have similar max volume levels. But for most listening situations where I could manually adjust my levels, I preferred leaving this auto-volume adjustment feature set to off.

Importing music from my music library from my Mac to the NW-ZX2 was as simple as drag-and-drop. According to the owner's manual, the NW-ZX2 will support up to eight layers of sub-folders, so you won't have to change your folder hierarchy to bring music into the device. The combination of the NW-ZX2's 128GB internal storage and its micro-SD card slot (which will currently support up to a 128GB card) gives the ZX2 a maximum storage capacity of 256GB (but remember some small part of this storage is delegated to the OS). Although not enough storage for an entire mature music library, it is certainly large enough to supply music for several long vacations. And if you insist on bringing your entire music library with you, no matter where you go, the NW-ZX2 supports any number of additional micro-SD cards, so nothing except your budget prevents you from acquiring a sufficient supply to hold all your music.

Although populating the Sony NW-ZX2 is simple and reliable, it's not exactly a hands-free operation. So it's a nice touch that the NW-ZX2 offers a desktop app called Sony Media Go that's similar to the HAP-1 app for the Sony HAP-Z1ES. Although it's only compatible with Windows-based PCs, it allows users to set up a system that regularly transfers new music to a NW-ZX2 device from their main library and manages these files automatically.

One of the best features of the iPod Classic (and other iPods) was the way it easily integrated with the entire iTunes ecosystem, both on- and off-line. The Sony NW-ZX2 has a leg up on many other manufacturers' current portable players because it also integrates into a larger digital eco-system, except from Google instead of Apple. Is Google's virtual world as slick and well integrated as Apple's iTunes, App store, and iTunes library match? Nope. But Sony's choice of Google's open Android operating system does allow for a potential level of flexibility, device customization, and functionality that can't be matched by any completely Apple-centric portable device.

Battery life ranks as one of the performance parameters that seems less important than sonics when shopping for a portable player, yet it often ends up being one of the principal complaints that new users have with their players. On paper the NW-ZX2 seems to have excellent battery life—60 hours playing MP3 files and 33 hours playing high-resolution files.

Equipment Report Sony NW-ZX2 Digital Media Player

However, in the real world my playing time was quite different. When I first received the player, I found that something was draining the battery even when the NW-ZX2 was in sleep mode—if the player sat for a day, the battery would be nearly dead when I turned the unit on. The only two apps I was using were "Play" and "Tidal," but for some reason when one was stopped, the other would begin playing. After a couple of days this problem vanished. Why? Neither Sony's engineers nor I could duplicate it again, so I have to chalk it up to "teething pains," rather than an acknowledged and repeatable bug.

Sound

Back in the early years of high-performance audio, when Harry Pearson was developing his yin/yang sound paradigm, audio source devices were more harmonically and dynamically colored than they are today. Identifying whether a source device was warmer or cooler than neutral was easier then. Today, finding any current-production portable player that is archetypically "tube-like" or "solid-state-like" is nearly impossible. It's not that every device sounds the same, but that the sonic differences among top-tier portable players are subtler. More often than not, the overriding sonic factor is how well their analog output stages interface with a pair of headphones rather than the "voicing" that the manufacturer has added to the player's sound. During the review I used the NW-ZX2 with a wide variety of headphones. Regardless of their type or price, the 'phones themselves varied from neutrality far more obviously than any of the top-tier portable players I compared with the NW-ZX2.

In my experience the principal reason that a portable music device doesn't perform up to its full sonic potential is a mismatch between the device's headphone amplifier and the headphones connected to it. With sensitive in-ear monitors, some portable devices have a continuous hiss or hum because the amplifier isn't quiet enough or delivers too much gain. Conversely, many portable players lack enough amplifier power to drive low-sensitivity, high-impedance, full-sized headphones to satisfying volume levels. The Sony NW-ZX2 proved to be among the more "headphone-friendly" portable players I've tried, in that it supported a wide range of headphone sensitivities.

Given that the NW-ZX2 is a portable device, I think it's safe to assume that more often than not it will be paired with in-ear monitors, which are generally higher in sensitivity than full-sized, over-ear cans. With my most sensitive in-ears, the 115dB/1mW Westone ES-5 custom monitors, I could hear only the very faintest midband hiss when no music was playing. With any live music track, including my own live classical concert recordings, room noise masked this low-level hiss completely. Switching to the only-slightly-less-sensitive Jerry Harvey Roxanne in-ear monitors I heard nothing but silence, even during the spaces between cuts. Other in-ear monitors I tried, including the Ultimate Ears In-Ear Reference Monitors, Cardas EM5813, and Etymotic 4Ps, were all dead quiet and able to play well above my maximum high-volume comfort zone.

With the far-less-sensitive 90dB/1mW HiFiMan HE560 full-sized earphones, the NW-ZX2 delivered enough power to drive them to satisfying volume levels on any commercial release I tried. With my own live recordings, which have on average -5dB lower volume levels, I would have preferred a bit higher output levels. Switching to a pair of even-less-efficient cans—the Beyerdynamic DT-990 600-ohm version—resulted in lower-than-ideal maximum volume levels, even on standard commercial releases. If you must use something with extremely low sensitivity and high impedance, you may need to employ an additional external headphone amplifier such as the new Sony PHA-3 (which can drive the DT-990s to well above satisfying levels even with my own lowest-level recordings).

With the U.S.-made Grado RS-1 headphones—longtime audiophile favorites—the NW-ZX2 delivered more than adequate volume with every music file in my library. Although the Grados aren't that difficult to drive, they do need an amplifier capable of some power to sound their best, especially when it comes to bass control. Through the NW-ZX2, the RS-1's bass took on a slightly woolly character that lacked some speed and definition compared to the Oppo HA-1 or Woo Audio WA-7 "Fireflies" desktop headphone amplifiers. The RS-1's upper midrange through the NW-ZX2 was also a trifle more prominent than with either of the two desktop headphone amplifiers.

One of the best full-sized headphone pairings with the NW-ZX2 was the new $199 group-buy AKG K7XX headphones from MassDrop. In addition, this version of the venerable K-701/702 design had a better fit due to slightly softer earpads, and its neutral sound signature mated nicely with the NW-ZX2. The

SPECS & PRICING

Memory capacity: 128GB
External memory: Micro-SD (card not included)
Key features: High-resolution audio playback, S-Master HX digital amplifier, DSEE HX (Digital Sound Enhancement Engine), ClearAudio+, Clear Bass, equalizer, VPT, one-touch listening via NFC and Bluetooth connectivity
Compatible audio formats: MP3, WMA, FLAC (192kHz/24bit), linear PCM (192kHz/24bit), WAV (192kHz/24bit), AAC-LC, HE-AAC, Apple Lossless (192kHz/24bit), AIFF (192kHz/24bit), DSD (2.8MHz, 5.6MHz)
Battery life: MP3 up to 60 hours
Charging time: Approximately 4.5 hours
Operating platform: Android 4.2
Display: 4-inch FWVGA (854 × 480) TRILUMINOS Display for mobile
Communication mode: WiFi (IEEE 802.11b/g/n/a)
Bluetooth: Bluetooth (A2DP/AVRCP/OPP/HID/SPP)
Accessories: USB cable, leather carrying case, spacer (headphones not included)
Dimensions: 65.1mm x 131.2mm x 18.5mm
Weight: Approximately 235g
Price: $1199

Sony Electronics Inc.
16530 Via Esprillo
San Diego, CA 92127

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without sounding grainy or hyper-articulated. The bass response of the K7XX, which is 3dB higher than that of the original K-701 according to AKG, mated well with the NW-ZX2, giving the K-7XX more warmth and musicality than the original version.

The other standout full-sized headphone pairing was the Oppo PM-1 fitted with PM-2 earpads. The PM-1 is among the easiest-to-drive and most universally device-friendly, full-sized, over-ear headphone currently available. As you might guess, the NW-ZX2 had no trouble driving the PM-1s well past most normal-humans' comfort level, and this combination resulted in a sense of dynamic effortlessness that is rare in portable players. Also, the NW-ZX2's built-in five-band eq allows users add a touch more high-frequency emphasis to the PM-1 at 2.5kHz and 6kHz.

I could clearly hear the increased fidelity from the NW-ZX2 compared to my iPhone 5.

Besides the five-band eq, the NW-ZX2 also has some additional "sound-shaping" controls. Although Old School audiophiles largely eschew eq adjustments, headphone enthusiasts often employ "frequency curves" to modify the sound of their cans. You may or may not find the NW-ZX2's additional sonic modifiers of value, depending on your tastes. Under "Sound Adjustment" there is "Surround Sound," which has five options: off, studio, club, concert hall, and matrix. While I'm sure there are some tracks that will benefit from these DSP modifiers, I used "off" 99.9 percent of the time. The NW-ZX2 also includes something called "Dynamic Normalizer," which reduces the differences between output levels of tracks. While I can see where this could be of value in certain situations, again I left this off for most of my listening.

One sound enhancement I did find valuable was Sony's DSEE HX, which I have previously experienced on the Sony HAP-Z1ES music player. It works on all MP3 and lossy formats to improve high-frequency extension.

Obviously the NW-ZX2 ($1,199) has some serious competition—principally from Astell&Kern's AK100 II ($899) and AK120 II ($1499), Calyx Audio's M player ($999), as well as Sony's own, more affordable A17 Walkman hi-res player ($299). Although I did not have the A17, AK100 II or AK120 II on hand, I did have the AK240 ($2495) and Calyx M. For sonic comparisons I used Tidal and my own high-resolution DSD128 recordings. Headphones for the comparison included both high-sensitivity in-ears, such as the Westone ES-5, Jerry Harvey Roxannes, and Ultimate Ears In-Ear Reference Monitors, as well as lower-sensitivity full-sized headphones, such as the HiFiMan HE560, Audeze LCD-2, Mr. Speakers Alpha Primes, Beyerdynamic DT-990 600-ohm version, and the Oppo PM-1.

With all three players—the Astell&Kern AK240, Calyx Audio M, and Sony NW-ZX2—the headphones had more pronounced colorations and sonic personalities than any of the players. Of the three, the Calyx M delivered the most drive for difficult headphones, such as the Beyerdynamic DT 990 600-ohm. Also when coupled to the Mr. Speakers Alpha Prime headphones, the Calyx had slightly more low-bass extension, giving electronic dance music a bit more throb. All three delivered hiss-free sonics with most 115dB sensitive in-ears (but the Sony did have a slight hiss with the Westone ES-5).

Listening to the same stream of Justin Townes Earl's latest album *Absent Fathers* from Tidal, I could clearly hear the increased fidelity from the NW-ZX2 compared to my iPhone 5. With my AKG K-7XX headphones the music had a larger soundstage through the NW-ZX2, greater three-dimensionality, and a more natural and organic harmonic balance. The upper midrange was less "splitchy," and lacked the slightly brittle character in the upper midrange that I heard through the iPhone.

Moving up the price ladder, when I compared my own live recordings played through the NW-ZX2 with the Calyx M, the differences were much less pronounced, and with some headphones the differences between these two players' sonic signatures were nil. As I mentioned earlier, the Calyx could and did drive my most difficult-to-drive headphones better than any other portable player I've used, so the Calyx had an edge there. But with less demanding headphones, the sonic differences were so slight that many times in my own A/B tests I could not reliably tell one from the other.

Pitting the Sony NW-ZX2 against the twice-as-expensive Astell&Kern AK240 player was, for me, a sonic dead-heat. Hard as I tried, using the most revealing headphones and in-ear monitors, I could not detect any readily identifiable sonic differences between the two

players when all the sonic shape-changing modifiers in both players were turned off. Once I activated any of the NW-ZX2 or AK240's filters, EQs, or soundfield settings, I could reliably tell which was which.

As for which is "better" or "the best" portable

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player...for most users it will probably come down to features, ergonomics, and which player's operating system is more in tune with a prospective buyer's own preferences. For some purchasers, Sony's Android OS may be a bit too busy or feature-laden; for others, the AK240's two-way streaming may be redundant. Obviously, Sony has an edge price-wise, but Astell&Kern also offers its less expensive (and comparably priced) AK120 II and AK100 II players.

The NW-ZX2 reestablishes Sony as one of the preeminent manufacturers of portable audio playback devices.

I compared the NW-ZX2 with my (now-discontinued) first-generation Astell&Kern AK100. With higher-impedance headphones the NW-ZX2 did a much better job of retaining dynamic drive and bass control. With higher-sensitivity in-ears the sonic differences between the two players were much less pronounced. I still preferred the NW-ZX2 with higher sensitivity in-ears, but its advantages were primarily in soundstaging and dimensionality. (The Sony consistently produced a larger soundstage with greater separation among instruments and a more pronounced sense of three-dimensionality.)

For a final A/B listening test, I compared the Sony NW-ZX2 with the Oppo HA-1 desktop DAC/headphone amplifier. With similar prices, the two have some overlapping capabilities such as being able to play Tidal streams. When I compared their sound on the Tidal stream of Joshua Radin's latest album, *Onward and Sideways,* I found the two produced equally excellent sonics, capturing the intimate quality of Radin's vocals and the gutty edge of his fingerpicked acoustic guitar. Both produced equally large soundstages with similar dimensional characteristics. I could live happily with either.

Wrap-Up

Looking at 2015 CES press coverage, I saw many tech journalists and bloggers writing that the NW-XZ2 was "the return of the Walkman," which is less than entirely true. Sony has been making portable players continually since its first Walkman cassette player appeared in 1979. What has changed is that the company now offers the A17 Walkman along with the NW-ZX2, Sony's first cost-no-object digital player designed to challenge premium players from other companies. The NW-ZX2 reestablishes Sony as one of the preeminent manufacturers of portable audio playback devices. And, yes, Sony has succeeded masterfully in achieving its design goals—the NW-ZX2 delivers excellent sound, plays any digital format thrown at it, and looks, feels, and responds like a high-performance product should. If you had any doubts about Sony's commitment to high-quality audio, the NW-ZX2 should put them to rest. tas

Equipment Report

Aurender Flow DAC/Headphone Amp

Desktop Delight

Steven Stone

When I first laid eyes on the Aurender Flow, I didn't get it. Taken from its form-fitting leather case it looked like another portable player, albeit big and sorta on the heavy side. It also looked 90s-ish with a big ol' center knob, a wiggly curve to its chassis resembling a logo for a hydro-spa, and one lone single-ended headphone output. Paging Forrest Gump: We got your portable player, right here. But I was completely wrong.

First, let me make one thing perfectly clear, the Aurender Flow is not a portable player. It is, in fact, a DAC and headphone amplifier capable of being used as a preamplifier and external drive (if a drive is installed in it), that makes it ideally suited for nearfield high-performance desktop use. That large knob I mocked earlier... well, its size and feel make it one of the most accommodating volume knobs I've ever had the occasion to fondle while hunting for that ideal SPL.

Tech Tour

With its footprint measuring only 5½" by 3½" by 1", Aurender packs a lot of technology into the Flow's one-pound chassis. The DAC uses an XMOS USB interface and Sabre ESS9018K2M chips, and has its own internal 4450mAh battery power supply. The Flow can handle any digital data stream up to 384/32 PCM and 128x DSD via USB and 192/24 PCM via its TosLink input. Although the Flow has only a single-ended ¼" 'phone-jack output, it can be configured in several ways. It can be variable output in 0.5dB increments up to 2 volts or you can configure the Flow for fixed output at either 2 or 5 volts. No, that was not a typo—5 volts. Output impedance is only 0.06 ohms.

The first time I saw the Flow I was confused by its m-SATA drive capability. You can add a drive to the Flow, and most people would assume it is for storing music to be played on it. They would be correct, but unlike a portable player where you could access the drive on the go, the Flow's drive can only be used when it is connected to a computer. But using an Apple camera connection cable, one can also access the contents of an iPad or iPhone.

In function, this is similar to the Auralic 2000 DAC/headphone stand that I reviewed in Issue 246. It, too, had provisions for tethering a drive that could only be accessed while the Auralic was connected to a computer. The difference is that the Flow holds the drive internally while the Auralic uses external drives.

The Flow is the first USB DAC I've seen that is USB 3.0-compatible. If your computer only supports USB 2.0, no worries, the Flow has provisions within its menu for several different "host modes" optimized for various computer systems. The options include USB2, USB3, Mac, IOS, and Android.

The Flow also has user-selectable digital filters. For PCM it has, by default, a PCM1 filter (which is a slow roll-off, in-band filter), and a PCM2 (which is a minimum-phase PCM filter). DSD users have the option of moving the DSD cut-off filter from the default, DSD at 47.7kHz, to 50, 60, or even 70kHz. There are three charging options: CHG+ is constant charging mode; CHG- turns off the charger; and CHGA- configures the Flow for automatic charging whenever music is not playing.

Setup and Ergonomics

Unless you intend always to use the Flow as a fixed-output device, its ideal location should be somewhere within arm's reach. Heck, even if you never intend to use its volume control, the Flow is much easier to operate when it's close to you, so you can see its display. Yes, the Flow has a display in the circular area inside its volume knob. Given the small area of this display, it is remarkably complete. Not only can you see the current volume level but also the USB mode, the current format being played, the battery condition, the output mode, and even whether a headphone is connected.

The Flow can be placed so it lays flat on its back (there are four small rubber bumpers to protect its rear surface), or you can lay it on its side so the control buttons are all located on the top. The only controls in addition to the large circular volume knob are along one side of the Flow. They consist of a power on/off, menu, move up, move down, and play buttons. The menu button has two modes, one for commonly changed settings and another push-and-hold mode for the settings that you will only need to alter occasionally.

Equipment Report Aurender Flow DAC/Headphone Amp

Upon initial installation you are supposed to designate which kind of computer or smartphone the Flow will be connected to via the push-and-hold menu button. But if you're the kind of person who doesn't read the owner's manual cover-to-cover and assumes that if you're using a Mac, the Flow will be plug-and-play, the Flow *will* work, although I found performance to be better if you do set it up optimally for the device it is going to be tethered to. On a Mac, once designated, I found that the play, pause, move forward, and move backward buttons will operate iTunes as well as Aurdirvana+, Pure Vinyl, Pure Music, and Amarra Symphony. Keyboard and mouse controls also remained fully operable with all these apps.

The review sample of the Flow came with a 250GB mSATA drive mounted in it (it is sold *sans* drive, which is easily user-installable). My MacPro recognized the drive immediately and mounted it on the desktop. As with any mounted desktop drive, if you remove the drive without first unmounting (or ejecting) it, you will get an error message, and if you turn off or disconnect the Flow you get that same error message. This error warning gets old. Because a 250GB drive was too small for my entire music library (the Flow holds up to a 1TB mSATA drive) and I didn't need another set of back-ups, I turned the drive off via Flow's menu—after ascertaining that it could be written to and read from successfully.

Manufacturers of battery-powered devices will always face the dilemma of figuring out how and when they should be recharged. The Flow gives you the three options that I noted earlier. For optimal sound, I recommend turning off the recharging completely. When used as a preamp I could hear some low-level noise generated by the Flow's charging circuits even in the "charge only when not playing" mode. When attached to an analog preamplifier the noise levels were the lowest in fixed-output mode with charging turned off.

I used the Flow with a wide variety of earphones from highly sensitive in-ear monitors to my least efficient full-sized headphones, and I was pleasantly surprised that they all worked well. Even with the most sensitive Westone ES-5 there was only the very faintest bit of low-level hiss. At the other end of the efficiency spectrum, the Flow had more than enough power to drive Beyerdynamic DT-990 600-ohm version well past *loud*. The Flow is the first headphone amplifier I've experienced that didn't need multiple gain settings to successfully accommodate a full range of headphone options.

One feature I've never given much thought to (but will in the future) is how a headphone amplifier interfaces with a new headphone. When you unplug and then plug in a new headphone, an amplifier can handle the new headphone in several ways: The amplifier can merely reproduce the previous volume settings. Or it can mute the output until the volume level is adjusted by hand, at which point the previous volume level manifests itself. Or it can mute the output and then reset the volume to maximum attenuation. After being blasted by more than my fair share of headphones, I much prefer the last method. Especially with the Flow's 0.5dB volume increments, matching levels when comparing two headphones—even allowing for the opportunity to linger over that wicked-cool volume knob—was rapid and repeatable, and I never had to worry about lowering the volume before installing a different pair of cans. A further nice ergonomic touch is that the Flow's display has an outer ring that shows you the volume level—when you remove a pair of earphones, you can watch the that volume ring drop, reassuringly, back to -90dB.

Sound

In the short time it's been around, Aurender has already garnered a reputation for making excellent-sounding gear. The Flow should enhance its already sterling character. I used a wide variety of headphones with it and couldn't find a mismatch. Unlike some headphone amplifiers that favor a particular set of headphones or type of 'phone, the Flow was very much an equal opportunity amplifier; everything I threw at it worked fine and sounded good. Also, the Flow allowed each headphone to produce its own unique sound signature. Grado RS1s still presented a different soundstage and imaging characteristics than Mr. Speaker's Alpha Dogs.

Flow users have several PCM digital filter options that I mentioned earlier. Listening to Sia's "Chandelier" off Tidal, I liked the PCM2 filter better than the PCM1 default. PCM2 produced better decipherability of her phrase "can't feel anything" and more precise imaging on the background singers located hard left and hard right. Also in this mode, the intentionally added distortion bed was a hair less aggressive. In the past I've found that many PCM filters are more software than hardware dependent, and this was true with the Flow. Some music will benefit more from one PCM filter setting than another, so it's not a question of which filter is overall the "best," but rather, which one suits the music better. Too bad the Flow can't remember and erable for a particular track, but as of now you still must change the filter settings manually via the menu.

I also used the Flow as a DAC/preamp by feeding its output to the analog input of the NuForce DAC-10H. Although it required using a ¼" headphone-to-female RCA adapter and

SPECS & PRICING

Sample rates: Up to 192kHz via SPDIF; up to 384kHz, DSD128x via USB
Compatible bit depths: 16–24 (SPDIF), 16–32 (USB)
Internal storage: Up to 1TB total via mSATA bus
Output impedance: 0.06 ohm
Output power (0.1 percent THD): 43mW/600 ohms, 87mW/300 ohms, 384mW/56 ohms, 570mW/32 ohms
THD+N: –114dB
THD (1kHz, 5.1V RMS output): 0.0002 percent
Dynamic range: 122dB
Damping factor: >130
Power supply: 4450mAh Li-ion rechargeable battery
Dimensions: 3.1" x 5.4" x 1.1"
Weight: 1 lb.
Price: $1295 without mSATA drive

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Equipment Report Aurender Flow DAC/Headphone Amp

then a 1 meter length of interconnect (I recommend something flexible such as the Kimber KCAG for this task), the setup worked nicely. I found the Flow's noise levels were lowest when I used the 2V fixed-output mode coupled with no battery charging. I used the NuForce ST-10 power amplifier tethered to a pair of Audience 1+1 speakers in my desktop system for these listening sessions. I also had a Velodyne DD10+ subwoofer tethered to the DAC-10H. I was impressed by how close the sound quality of the Flow was to the NuForce DAC-10H. Once levels were matched—which was pretty easy with the DAC-10H's numbered volume settings—the DAC-10H had a slightly wider soundstage, but the Flow's soundstage was deeper. The DAC-10H also had better low-level detail due to its somewhat quieter base noise level, but the Flow matched the DAC-10H's dynamics and pace.

I also compared the Flow with the Oppo HA-1, once more using the Oppo's analog inputs so I could compare the two in a matched-level A/B test. Again it was a close call with the Flow having better dimensionality and upper-midrange energy and the HA-1 having more relaxed transient response. The Flow produced a more three-dimensional soundstage, but the HA-1 produced better lateral delineation and separation between instruments in the soundstage.

Neither the Oppo HA-1 nor the NuForce DAC-10H could successfully handle as wide a range of different headphones as the Flow. Even with its different gain ranges, the NuForce DAC-10H could not go from high sensitivity to low with the same equanimity as the Flow. With the DAC-10H, you have to hunt and peck for the best combination of gain and volume; with the Flow, you merely turn the volume knob to the right point. And while both the Oppo HA-1 and the NuForce DAC-10H offer far more flexibility in input and output options, if your primary use will be with headphones and not as a preamplifier for a speaker-based system, the Flow's feature set and sound make it a better option than the other two.

Summary

I've heard there are some audiophiles who like an uncluttered desk. For someone who wants great sound, smooth ergonomics, and a compact footprint, the Aurender Flow offers an elegant solution for headphone and nearfield listening. Put a large mSATA hard drive in it and you have a clever rig for a traveling audiophile. Although the Flow will work in portable applications, in my view its one-pound weight and form factor make it more suitable for desktop service. Also, The Flow could easily find a place with music professionals, carrying it from studio to studio to ensure monitoring consistency.

Never before have audiophiles had so many fine options for DAC/preamps in the $1000 to $1500 range. I've mentioned several with which I'm familiar during this review. But the Flow's physical dimensions and its ability to drive everything I could throw at it headphone-wise make it special. Yes, my first impression of the Flow was wrong, but after giving it a chance to strut its stuff, I have to admit that it has become my current go-to headphone listening rig. If headphone listening from a computer source is your thing, you need to hear the Flow because it was made for you. tas

Equipment Report

Astell&Kern AK380 Portable Music Player

The Summit Just Got Higher

Alan Taffel

What began as a single, brilliant, high-end portable music player—one that created the category virtually singlehandedly—has morphed into a broad line of offerings. Astell&Kern wants to make its products accessible to a wide range of consumers, from high-end neophytes to grizzled, uncompromising audiophiles. Recently, the company expanded its already-broad model range in both directions.

At the entry level is the new AK Jr, which brings A&K just barely into the under-$500 range. On the other end of the spectrum, the AK380 takes its place as the brand's new flagship. The previous top dog, the AK240, remains in the lineup. The latter player raised eyebrows with its then unheard-of $2499 price tag. But Astell&Kern wasn't cowed and/or the AK240 met with strong market success, because the new AK380 goes for a whopping $3499. Pony up $3999 if you'd like yours in copper.

That's a whole lot of money for a portable player, but the more deeply you look at the AK380, the more its price seems justified. I'll get to that shortly, but first let's have a look at what the AK240 and AK380 have in common. Both have large touchscreens (the AK380's is slightly larger) encased in aircraft-grade Duralumin bodies clad in custom-fitted leather. Both are oversized compared to, say, an iPod Nano or a Sony Walkman, but both feel substantial and swanky in the hand. Inputs and outputs are identical: There are ports for micro-USB, standard headphones, and balanced headphones. Feature-wise, both units feature dual DACs for better channel separation, native DSD with no interim PCM conversion, MQS support, streaming over WiFi or (heaven forbid!) Bluetooth, a 20-band parametric equalizer, and 256GB of internal memory that's expandable by 128GB via a microSD chip.

Clearly the AK240 was already a richly featured device. Seems as if there wouldn't be much to add, doesn't it? But the AK380 goes the AK240 quite a bit further. Most immediately obvious is the new touchscreen. Aside from a bump in size from 3.31 to 4 inches, the new player trades an AMOLED display for WVGA. The difference is instantly apparent. The new flagship's screen is brighter, sharper, and more colorful. Meanwhile, in terms of connectivity, the AK380 adds aptX Bluetooth. Trust me, if you're going to use Bluetooth, aptX is the way to go.

But the most significant changes for the AK380 are deep inside. First and foremost is the switch from dual Cirrus Logic CS4398 chips to dual AKM AK4490's. Aside from any sonic benefits, which are evaluated below, the shift enables the AK380 to support resolutions all the way up to 384/32 (the AK240 topped out at 192/24) and DSD256 (versus the AK240's DSD128). Most users won't need the AK380's extra resolution now, but it's if and when broadly available source material evolves to that level.

The AK380 goes yet another step further by pairing the new chipset with a high-precision VCXO (Voltage-Controlled Crystal Oscillator) clock. The clock module has a jitter rating of just 200 femtoseconds (a femtosecond is a quadrillionth of a second, or 10^{-15}), which in turn reduces the AK380's overall jitter to just 30 picoseconds—roughly half that of the AK240. And just to make sure the new chip/clock combo can devote all its resources to sonics, peripheral functions like the parametric eq are offloaded to a new dedicated DSP chip.

If you're getting the impression the AK380 is a very serious piece of equipment, you're right. And bristling as it does with so much advanced technology, it would seem a shame to use the AK380 solely to play music through a set of cans.

Astell&Kern certainly thinks so. Which brings us to what I consider one of the AK380's most significant features: extensibility. Unlike nearly all others of its genre, the AK380 is not necessarily a standalone device. Rather, it can serve as the center of an entire ecosystem. If you need to drive low-sensitivity headphones, slap a module called AMP ($699) right onto the back of the unit. AMP has its own battery pack, so it doesn't shorten playing time, and seamlessly integrates with the AK380 both physically and functionally. No interconnect necessary, no dueling volume controls. Astell&Kern also offers the AK CD Ripper ($349). Although it's a physically separate unit, the AK CD Ripper, like AMP, is plug-and-play.

Equipment Report Astell&Kern AK380 Portable Music Player

Now comes my favorite of these functionality extenders: the AK Cradle ($349). When nestled into the aluminum-bodied Cradle, the AK380 suddenly becomes not a portable player but a high-end streamer/DAC front end for a high-end audio system. You know, the kind that is installed in a listening room and doesn't move. What the Cradle adds that self-contained portable players necessarily lack is a set of high-grade, balanced XLR outputs. The Cradle also powers the AK380, which means that if you run it in conjunction with the AK Connect app for your smartphone or tablet, you can leave the cradled player untouched and use it just as you would any other digital front end. The CD Ripper can also be directly linked to the Cradle, completing the scheme.

Needless to say, all the attention A&K has lavished on the AK380's build-quality, features, technology, parts, and extended functionality would be pretty much moot, especially at its elevated price, if the unit didn't sound the part. Admittedly, I was skeptical on this point. I thought the original AK100 was a landmark in both design and sonics. In my review of that unit, I compared it with a fifth-gen iPod Classic—the best of its breed—and there was simply no contest. I didn't see a lot of room for improvement—until I heard the AK120. The subsequent Mark II versions of these models sounded even better, much to my surprise. Then came the AK240, and I finally felt I'd reached the summit in personal player sound. Honestly, I had zero complaints.

But the AK380 has once again bushwhacked me. It stretches the boundaries of what's sonically possible from a personal player in a way I never imaged possible. For instance, with the AK380, instruments exist in a field of air, as they do on a well dialed-in dedicated system, and in real life. These air pillows are missing on the AK240. But the air around instruments is merely one example of the AK380's greater transparency. The ability to draw more detail from the bits also manifests itself as richer timbre—for instance on brass and string bass—that makes instruments more lifelike. Previously hidden details, such as the decay of reverb, become easy to hear.

The AK380 is also "faster"—that is, notes start and stop more quickly—than the AK240. This gives it the ability to trace rhythms more accurately. Consequently, beats are tighter and more infectious through the new flagship. This is true not only with rock but with material like chamber music, which relies on less overt sources of rhythm. Through the AK380, you can definitely tap your feet to a chamber quartet or octet (try the Dvóřak *Serenades* on Praga), and you can more easily pick up on rhythmic variations such as syncopation.

With each succeeding generation of AK players, Astell&Kern has managed to lower the noise floor. This has the obvious benefit of a more relaxed listening experience, but there is another, equally important advantage. With a lower noise floor, instruments not only stand out from the background, they stand out more clearly *from each other.* Even compared to the already quiet AK240, the AK380 is better at allowing each instrument to be heard more distinctly. Once more, it's not necessary to enlist complex music to hear the difference. Even on something as uncomplicated as a jazz trio, the AK380 better conveys what each player is up to, as well as the sound of his specific instrument.

The final distinction wrought by the AK380 is superior spatiality. The new flagship exhibits tight (but never edgy) imaging and an extremely wide soundstage. I suspect this is due to better channel separation. Whatever the cause, material such as the HDtracks 192/24 version of Led Zeppelin's "Whole Lotta Love" becomes mind-expanding.

Since the AK380 can also serve as a music streamer (in either portable or cradled mode), I wanted to test the quality of its sound in that regard. It's been my unfortunate experience that many otherwise excellent digital source components fall down when asked to stream. This is especially true for wireless streaming, which is the only type the AK380 supports. One encouraging sign, though, was that the AK380 supports DLNA (Digital Living Network Alliance), a rigorous protocol for exchanging media files between servers (such as a NAS drive) and clients (in this case the AK380).

So I proceeded to compare the sound of the AK380 when streaming over WiFi versus playing directly from its internal memory. What I found was that the difference is nearly impossible to hear—a welcome and astonishing result. If I listen *hard,* I can detect a slight veiling of voices when in streaming mode. Norah Jones, who is always recorded in such a way that her voice comes across intimately, sounds subtly less "there" when streaming. But that's about it. As it turns out, the AK380 sounds great regardless of how it's accessing music.

With the AK380, Astell&Kern has created a flagship that transcends the genre of audiophile-quality portable players. Never, to my knowledge, has such a device incorporated features and parts comparable to those found in the best high-end components. And never, in my experience, has a portable device delivered sound so uncannily similar to that of the best high-end systems. But the AK380 goes beyond delivering superb sound. Thanks to a set of clever peripherals, this player can serve as the digital front end in any audio system. Combine all that with generous storage, a large, bright display and next-generation resolution, and the AK380 becomes the portable player to beat. tas

SPECS & PRICING

Display: Four-inch WVGA touchscreen
Supported audio formats: WAV, FLAC, WMA, MP3, OGG, APE, AAC, ALAC, AIFF, DFF, DSF, integral Tidal streaming
Maximum input rate: 384/32, DSD256
Battery: 3400mAh 3.7V Li-Polymer
Outputs: Phones (3.5mm), optical (3.5mm), balanced (2.5mm, 4-pole)
Memory capacity: 256GB plus 128GB microSD
Wireless: 802.11 b/g/n (2.4GHz), Bluetooth V4.0 (A2DP, AVRCP, aptX)
Supported OS: Windows XP, 7, 8, and 10; MAC OSX 10.7 and up
Dimensions: 3.14" x 4.42" x .70"
Weight: 8.11 oz.
Price: $3499

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DISC PLAYERS

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Equipment Report

Oppo BDP-105D Audiophile Blu-ray Disc Player

Flexible and Unflagging Firepower

Neil Gader

Do you ever feel as if you've lost control of your audio system? I know the feeling. Yes, it was a simpler time back then in BC (before computers). Digital sources were mostly limited to compact-disc players, and perhaps a DVD player to liven things up during family movie night. But the dawn of high-resolution music and computer audio changed all that. New partnerships were formed as network audio and cloud storage and Internet streaming encroached on the territory normally reserved for traditional source components. They also brought with them controller apps via smart devices, and an expansion of formats and options perhaps unparalleled in audio's history. Ultimately, it has become a really deep dive trying to keep up with the blitz of gear and peripherals, and then making them all play nice together. That's where Oppo Digital comes to the rescue—providing hope for the hard to cope.

The actual idea of a universal-format player is not new, but in actuality most were known for their multi-format audio capability or their video chops but rarely for both. It was at this intersection of audio and video that Oppo made its name. The BDP-105D is Oppo's latest expression of its philosophy. The brand's top-line multi-format audio and Blu-ray disc player is, like they say in the car biz, "Loaded, baby." Like its lower-priced sibling, the BDP-103, the 105D offers hi-res stereo and multichannel audio, as well as a panoply of cinema surround-sound options like DTS-HD Master Audio and Dolby TrueHD. Its Blu-ray video side comes with 4k upscaling and 2D-to-3D conversion, and sports a plethora of inputs and outputs that includes dual HDMI.

However, it is at this point that the BDP-105D diverges from the rest of the Oppo line. Oppo has seen the writing on the wall regarding the decline of the optical disc. In response it has added computer-audio capability to its formidable battery of format firepower. Thus the BDP-105D is a true media player in the most widely expressive sense of the word. Its new, asynchronous USB input, feeding a new ESS Sabre32 Reference DAC, now supports PCM audio up to 24-bit/192kHz and DSD in standard rate (DSD64) or double rate (DSD128). Additionally the BDP-105D courts the server/cloud crowd with hi-res network capability, allowing users the option of streaming from an external NAS drive. There is also wireless capability and media support for Tidal, Netflix, Pandora, Vudu, and Rhapsody.

In keeping with its recent entry into the personal-listening market with acclaimed planar-magnetic headphones and headphone amps, Oppo has also included an internal headphone amp, accessible from the front panel via a ¼" input jack. The headphone amplifier is connected directly to the DAC, which offers a unique performance advantage over many stand-alone headphone amplifiers. There's a nicely graduated volume control on the remote for headphone control.

The Oppo's front panel is spacious but spare. The front-loading disc drawer in the center separates the player-control buttons on the right and the fluorescent display on the left. At the bottom right edge of the front panel resides a USB port, an HDMI port, and the headphone jack input.

True to its roots, the BDP-105D also remains a physical-media lover's dream. Beyond DVD, Blu-ray, and Red Book CD with HDCD decoding (quick show of hands—anyone remember Pacific Microsonics?), the Oppo continues to support hi-res formats like DVD-A and SACD. (Dang, I *knew* I shouldn't have sold my Donald Fagen *Nightfly* DVD-A.) Now, I won't belabor the fact that these two high-res formats didn't gain much market traction. (I have never met a non-audiophile who didn't look at me cross-eyed when I exclaimed, "Wow, did you know this is a hybrid-CD/SACD disc?") However, the simple fact remains that with a player like the Oppo we can all still enjoy the terrific audio quality of these moribund formats. I do, and in the case of SACD, I continue to add to my modest collection.

After connecting the Oppo directly to my flat panel using the HDMI 1 output, I found the on-screen menus extensive and well laid out. It's well worth the time getting to know the myriad set-up options and preferences on both the audio and video side. Connectivity on the whole was excellent. Once linked to my home network via the LAN input, I downloaded the Oppo MediaControl app to my iPad, and it promptly identified my external NAS drive, and wirelessly I was off to the races. The app itself was visually workmanlike, not very sexy, but stable and intuitive to navigate. Personalizing the GUI layout wasn't in the cards, however—it's a one-size-fits-all proposition. Operationally, the BDP-105D was a screaming-fast disc loader. SACDs,

Equipment Report Oppo BDP-105D Audiophile Blu-ray Disc Player

even data-heavy Blu-ray discs booted up to their respective home pages in well under ten seconds or less.

I'm embarrassed to admit that it crossed my mind that this player could be a jack-of-all-trades and a master-of-none. Thankfully, that thought couldn't have been further from the truth. The sonic character of the Oppo was persuasively musical with all media. It conveyed a slightly cooler sound—one that placed a greater priority on catching the leading transient edges of wind instruments—but was of a refinement that consistently wore well throughout my listening sessions. The extension of Edgar Meyer's acoustic bass was weighty with resonance and good pitch definition. During Copland's "Fanfare for the Common Man," there was a hint of added bloom to the kettledrums and bass drum but overall the player's control and grip remained solid. Its SACD performance maintained a clear sonic advantage over well-recorded Red Book discs by permitting micro-dynamics to emerge, enlivening tonal color, expanding ambient space, and breathing more timbral realism into bass reproduction. Compared to a top-notch player like the dCS Puccini, the Oppo narrowly missed the mark in terms of its noise floor and its ability to reproduce the physical dimensions of a venue. There was just a hint of digital sheen that glossed over inner detail and reduced the tactile elements of musical instruments—sonic cues like the skin of a kettledrum, or the resonance and decay signatures of an orchestra or chamber group. Top-flight players like the Puccini allow a symphony orchestra to fully inhabit the hall space, and for an instant recreate a reality that will actually fool the ear.

SPECS & PRICING

Inputs: Digital, one USB-B, two USB-A, two HDMI, one coaxial, one optical
Outputs: Digital, two HDMI; analog, 7.1-channel RCA, stereo RCA and XLR
Formats: PCM up to 24-bit/192kHz, SACD, DSD64/128, DTS-HD Master Audio, Dolby TrueHD, Dolby Digital
Dimensions: 16.8" x 12.2" x 4.8"
Weight: 17.3 lbs.
Price: $1299

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Funtioning as a media player over my home network, the BDP-105D stepped up its game considerably by digging into high-resolution PCM titles like The Eagles' *Hotel California* and Yes' *90125* with a harmonic ease and dynamic pop and explosiveness that CD playback couldn't match. During "Hotel California," it captured the personality of the flexing drumhead from Don Henley's loosely tuned tom fills and the ringing drone strings from the lead 12-string acoustic. On Yes' "Owner of a Lonely Heart," a track filled with ear-popping acoustic and electric contrasts and channel-pinging effects, the sound cues were pristinely defined and layered. A couple of other noteworthy sonic impressions would be the Oppo's ability to reproduce tiny volume gradations from the delicate high-hat figures that open Shelby Lynne's "Just A Little Lovin'" or the fragility of a concert harp or triangle or the pastel colors from a mark tree. Still, in comparison to the more costly dedicated media player like the Cary Audio DMS-500 (review forthcoming), the BDP-105D didn't capture the full spatiality of a recording, such as the air flaring outward from a crash cymbal, or fully differentiate the subtle timbral distinctions of an electric bass and a kick-drum beat.

The headphone amp was musically solid and satisfied the extended demands of the Audeze LCD-X and the less finicky HiFiMan Edition X planar-magnetic designs I had on hand. The sound was quick, extended and tonally neutral. During Vanessa Fernandez's cover of Led Zeppelin's "The Lemon Song," the Oppo slightly attenuated the bottom-end air and skin impact of Jim Keltner's anchoring kick-drum patterns, and the articulation of the acoustic steel vamp was less individuated. During the Rutter *Requiem* there was also a sense that the boundaries of the venue were closed in, and the vast chorus and hall ambience sounded somewhat congested as a result. Finally, the BDP-105D couldn't quite match the crystalline resolution and depth of focus of a stand-alone headphone amp like the formidable Pass Labs HPA-1, but as is the case with most of my observations about the flexible Oppo player, the unit performed far better than a typical pre-packaged headphone amp.

For the cineastes among us, Oppo equips the BDP-105D with Darbee Visual Enhancement or DVP. Although TAS doesn't cover video technology *per se*, as a movie enthusiast I would be remiss in failing to lend my own seat-of-the-pants impressions of the Oppo with and without DVP. While purists might think of this enhancement

in the same way that audiophiles fret about the use of tone controls, DVP, as I understand, is not to be confused with the broader brush-strokes typical of television's global sharpness or contrast controls. We've all experienced the pronounced digital artifacts these leave in their wake. DVP is said to be a great deal more measured and to operate narrowly on a pixel-by-pixel basis. Indeed it does sharpen picture detail where it appears to need sharpening, gently refocusing soft, mushy backgrounds and strengthening edge detail. Since it's easy to adjust or disengage using the on-screen menu, I tended to use it selectively. You can overdo it, but applied conservatively it's a nifty tool in the Oppo's set-up box.

For the devotee who likes his audio/video just like he likes his pizza—with everything on it—the BDP-105D makes quite an impression. Of course, depending on your tastes and listening habits, Oppo's "everything and the kitchen sink" approach may not appeal. However, if you're looking to regain control of an unruly system while adding A/V formats (old school and new), Oppo's affordable, one-box, crossover solution should get a lot of enthusiasts' mouths watering. tas

Equipment Report

AVM Evolution MP 5.2 Media Player

A Clean Machine

Neil Gader

Don't count the compact disc player out just yet, folks. The CD format may have lost ground to computer-based audio, but as I've been positing for the last few issues, the venerable CD drive will not be joining the ranks of the electronic-homeless anytime soon. Audiophiles, music lovers, and millennials alike are reintroducing themselves to the pleasures of experiencing tactile, physical media versus the ephemera of digital "storage"—media that includes not only vinyl, but also CDs. Joining the newly minted segment known as "media players" is the AVM Evolution MP 5.2, a component that aggregates the best of all digital worlds in a single chassis.

Celebrating its 30th anniversary, AVM might not be a household name in North American consumer hi-fi, but its extensive range of electronics is well respected in high-end circles. In TAS, for instance, Wayne Garcia covered the AVM Inspiration C8 CD-receiver back in Issue 224. However, the $8400 Evolution MP 5.2 is an enitrely different kettle of fish. As AVM's mid-tier media player, it features a tube linestage based on a pair of ECC83s—custom designed and built for AVM (which calls them AVM 83Ts)—and run in balanced mode. The plate-glass window in the top panel of the MP 5.2 offers users a nice interior view of the blood-red glow of those valves. (For the tube-averse, AVM offers an identical player in the $6600 MP 3.2, which has the same features only with solid-state circuitry.) The stylish, brushed-aluminum chassis reveals nary a screw or bolt to blemish its seamless look. On the front panel is a large blue-lit display centered above the disc slot. It shows source and track/time data, plus sampling and filter selection. (As for the small line of buttons, I'm not sure whether I'm more dismayed that its tiny and faintly lettered, or alarmed at my declining eyesight.) In any case, the clean, uncluttered look of the front panel disguises the fact that this player is bristling with capability. Foremost is the slot-loaded CD player, featuring a TEAC Pure-CD drive—supplied exclusively to AVM—that's spring-mounted and equipped with a mono-focal lens. A glance at the back panel reveals a bundle of digital inputs: four SPDIF (two coax, two optical) plus an AES/EBU and an asynchronous, galvanically isolated USB input with DSD64 (2.8MHz) capability. Significantly, the MP 5.2 is also a network player with both wireless and LAN over Ethernet. Digital-to-analog conversion is provided courtesy of a Wolfson 8471 chip, which offers both 24-bit/192kHz PCM (coax and USB) and DSD resolution (USB only) in double-balanced architecture. There is selectable upsampling of all incoming signals from native to 44.1, 48, 88.2, 96, 176.4, 192kHz. Plus, there are two digital filters, Smooth and Steep. Other welcome features include input renaming and auto-play when inserting a CD.

In an email exchange, AVM's owner Udo Besser offered some insights about the tube linestage. Implemented in both his Ovation and Evolution products, the design uses the regulated DC heating of the tube to compensate for the tube's aging process and to extend its life. This process required proprietary, custom alterations of the ECC 83 and 803. "Since we use our tube stages only as a linestage, we may run the tubes in the ideal area of its amplification curve; as a side effect this expands the [tube] life-expectancy very much. Overdriving the tubes is effectively avoided by adding a solid-state Class A output stage, which drives the connected cable and is tolerant to short circuits."

Remote controls are rapidly becoming legacy devices in the face of control apps for smartphones and tablets. AVM is no exception, and offers the RC S remote app, which is available as a free download for iOS and Android. Operationally it's a reasonably efficient interface but lacks the colorful pop and dashboard organization of today's better control apps, such as the Lumin app I'm using to drive the Lumin A1 Network Music Player I reviewed in Issue 248. For those who remain wedded to a traditional handheld RC, there's the optional RC 9 RF/IR controller that features a small color display and includes a nifty charging dock.

Network setup (an old nemesis) was an intuitive breeze—my network password was all that was required to get the player up and running. A distinct advantage of network players like the MP 5.2 is the ease with which the factory can update the unit. For example, toward the conclusion of the review process, I was alerted via the app that a free firmware upgrade was available with several features that are now bundled as standard equipment with current-production units, including variable output, pre-installed Tidal support, plus a new Internet radio service provider with additional features such as podcasts, favorites, and so forth. Very cool.

As I listened to the AVM, I moved back and forth between CDs and the identical material stored on my Synology NAS drive. Throughout, the MP 5.2's sonic performance was excellent. Tonally, it had a glassy-smooth character with intimations of midrange and top-end warmth that were very

Equipment Report AVM Evolution MP 5.2 Media Player

SPECS & PRICING

Digital inputs: Two coaxial, two optical, one AES/EBU, one USB input (up to 24-bit/192kHz)
Outputs: One stereo XLR, one stereo RCA, one coaxial, one optical
Dimensions: 16.9" x 5.12" x 14.6"
Weight: 20.25 lbs.
Price: $8450

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appealing to my ear. Low-level and background details were well resolved. The rattles of the tambourine during Holly Cole's "The Heart of Saturday Night" from *Temptation* were concise and cleanly represented, as was the ring of octave strings from the 12-string guitar during k.d. Lang's version of Jane Siberry's "Love is Everything" on *Hymns of the 49th Parallel*. With its high-output trumpet excursions, The Manhattan Jazz Quintet's cover of "Autumn Leaves" continues to leave me breathless. This version features a slow, low-key buildup that turns into an ambush of house-on-fire dynamic intensity. This track has a level of transparency that challenges the transient and dynamic alacrity of every component in a system.

The MP 5.2 traversed it with brio, imparting a vivid soundstage, nice depth cues, and a deep pocket of air for the piano. And as I turned to high-resolution Reference Recordings 24-bit/176kHz material, the MP 5.2 reached into the furthest recesses of an acoustic soundstage and furnished all levels of clues regarding venue scale and dimension.

The more I listened to orchestral string sections, the more I appreciated the denser timbre of the AVM. And by this I'm not referring to colorations *per se*, rather something more related to the physical, fleshy presence of musicians in performance. Similarly, string instruments such as cello and bass violin exuded the full-bodied, weighted voices and appropriately stout resonant foundations that I expect them to have. I can't say with any certainty to what extent my sonic impressions can be attributed to the AVM tube stage, but I can state that over the years I've found that tubes illuminate the soundstage a bit differently—as in a shift in color temperature from, let's say, a bluer "digital" cast to a slightly redder one. And this was the case here.

With the Esoteric K-03X SACD player/DAC still in-house (Issue 261), I was a little startled by just how closely the AVM approached the Big K's overall performance and resolving power. The AVM couldn't quite match the heavy and deeply rooted foundation that the K-03X establishes, but its timing and tonality didn't take much of a backseat. As I discovered during Diana Krall's cover of Joni Mitchell's "A Case of You" from *Live in Paris*, the AVM's personality was a little brighter overall, a trait that allowed transients to really pop and heightened immediacy. Although both players offered solid dimensional cues, the Esoteric was a bit chestier and darker on vocals, and my ears perceived a more convincing degree of felt padding on the piano hammers. The AVM only ceded ground to the Esoteric when attempting to resolve some lower-level imaging detail, such as those found in the background singers' shimmering vocals during Leonard Cohen's "Darkness" from *Old Ideas*. In this instance, their seductive voices were rendered a little less distinctly from one another and were not as palpably present.

When the Beatles' Paul McCartney sang in *Penny Lane* "...it's a clean machine," the line could easily have applied to AVM's neatly executed, smartly appointed, and musically virtuous MP 5.2. Call it the streaming future and the CD past rolled into one. AVM's small-footprint integrated solution offers the best of all digital worlds in a single, trim package. Easy on the eyes, easy on the ears, AVM's MP 5.2 is a very strong contender in the youthful media-player segment. Enthusiastically recommended. tas

A Closer Look at AVM

AVM or Audio Video Manufaktur was founded in 1986 and is headquartered in Malsch, Germany. Its current managing director is Udo Besser, formerly of Burmester. The company originally specialized only in amplifiers, but rapidly expanded into CD players, becoming a major player in the German market in the early 1990s. More then 75 different models have been introduced since then, including the ground-breaking integrated Amp A2, Preamp V2, and the mono power amp M3—several thousand pairs have been sold in many different versions over the years.

In 2010, Udo Besser assumed full ownership of the company and has been steering it in the direction of becoming a complete-system manufacturer that offers a wide variety of products: streaming and conventional analog, solid-state and tube, all-in-one systems, and separate components. Today's AVM product lineup has three distinct levels and includes CD receivers, standalone players, streaming CD receivers, plus integrated amps with tube or solid-state designs, preamps and stereo and monoblocks amps. The Inspiration 2.2 series represents the entry-level, followed by the all-solid-state Evolution Line 3.2 geared for audiophiles. The Evolution Line 5.2 adds a tube linestage built around AVM's 83T double triode. The flagship of the AVM offerings is the solid-state Ovation Line 6.2 and the top-tier Ovation Line 8.2 that has a tube linestage built around the AVM 803T double triode.

In its formative years, the company was focused primarily on its German domestic market, but today AVM is sold into more than 55 countries, with the U.S. market being central to AVM's current plans—a commitment underscored by the creation of its own subsidiary, AVM Audio USA. However, Udo Besser's design principle remains a clean, minimalist approach—as in no straight lines or visible screws, and front panels of elegant simplicity. This year represents AVM's 30th anniversary, and it's especially noteworthy that the original founders still form the backbone of the company: Günther Mania still heads R&D and Robert Winiarski is in charge of service and repair of vintage AVMs.

Equipment Report

Esoteric K-03X CD/SACD Player

X-Factor

Neil Gader

The K-03X is Esoteric's penultimate CD/SACD player. Similar in appearance to the significantly more costly Esoteric flagship K-01X, the K-03X represents a commanding presence. Built to a standard of quality and luxury that few players can match—from the exquisite three-point isolation footers to the buttery-smooth action of the aluminum disc tray, it's a mechanical delight to observe in operation. The player's vault-like construction and sixty-pound mass suggests a permanence of a millennial order. Seemingly immune to the ravages of time, it's a veritable monument of brushed aluminum and steel plate that neither mechanical nor acoustic resonances or vibrations can breach. The K-03X is much more than a single-box disc player however; the dual mono design houses an independent, fully fledged and equipped high-resolution DAC that supports hi-res files over USB, SPDIF, and TosLink. Outfitted with four DACs per channel operating in parallel, there are dual oversize toroidal transformers that supply independent power to the digital and analog circuits for each channel.

The new "X" designation isn't just for show either. It represents some serious upgrades over its predecessors, the K-03 and K-01 that colleague Alan Taffel reviewed in Issues 213 and 230 respectively. Chief among them is the new AK4495S DAC chipset, the same DAC that graces the über-flagship Grandioso series. Internal processing has also been upped to 34-bit from 32-bit. The DAC now supports DSD64 and DSD128 (double DSD) and 384kHz/32-bit PCM playback over USB and SPDIF. Additionally external clock sync is obtainable in USB DAC operation, a feature previously unavailable in the K-03. Esoteric's Scott Sefton said, "It also has a new USB board supported natively by Mac OS X, so in addition to more functionality (DSD) it is simpler, at least for Mac users." Finally, the variable-output level has been jettisoned, as Esoteric has deemed that the compromises inherent in digital volume controls were too restrictive on ultimate performance.

Back-panel connectivity includes both RCA and balanced XLR analog outputs along with accommodation for digital sources via USB, SPDIF, and TosLink inputs. These provide a level of flexibility that later permitted me to connect a trio of digital sources—a DVR, Apple TV, and a MacBook—without a hiccup. A remote control is included.

Internally the centerpiece of Esoteric's spinning disc technology is the latest incarnation of the company's famed VRDS drive mechanism. Purposefully designed to spin Super Audio CD discs, the VRDS-NEO VMK-3.5-10 is a high-precision, stable duralumin turntable that improves reading accuracy by mechanically correcting for surface run-out. This mechanism's massive architecture and the 10mm-thick turntable bridge ups the unit's weight to over nine pounds—and to more than 22 pounds including the rigid base. With the addition of its proprietary VS-DD spindle servo driver—previously offered only on Esoteric's flagship players—smoother high-precision servo control and a more stable drive is attained. Finally, a discrete three-channel amplifier circuit optimizes the waveform of the current supplied to the motor for quieter and smoother spindle operation. The whole package weighs in at a whopping 62 pounds.

During setup, attention *must* be paid to the K-03X's front-panel "Mode" button. Beyond basic configuration—display, power saving, etc.—pressing that button unleashes a bevy of sound optimization options, four up-conversion picks (2x, 4x, 8x, and DSD) and four filter selections: FIR1 (steep roll-off), FIR2 (slow roll-off), and apodizing variants, SDLY1 (short delay/steep

Equipment Report Esoteric K-03X CD/SACD Player

roll-off) and SDLY2 (slow roll-off). Out of the box, the default deactivates these settings and each must be applied individually for respective digital sources: disc, USB, and so on. AT discussed these in copious detail and his conclusions (which I urge readers to reference) were borne out by my own findings. Be prepared to spend an afternoon or two experimenting but the results are more than worth the effort. Remember there are no right or wrong answers here, and it only has to be done once as the settings are instantly saved in the K-03X software. In the end, my listening preferences resulted in up-sampling settings of 2x or 4x and apodizing set to SDLY1. In unfiltered, non-oversampled default mode, the sound was flat, thin, dry, and rhythmically uninvolving, but with preferred settings I observed greater focus, presence, dimension, and ambient energy.

For DSD file playback over USB including double DSD, I downloaded the latest free Esoteric HR Audio player software. This is basic, no-nonsense player software (no remote app for a smart device is currently available) to get up and running but many users will likely opt for advanced player software from the likes of Audirvana as they become acquainted with potential of the medium.

Before I turn to the K-03X's performance, I freely admit that as a dyed-in-the- wool LP listener, my early history with digital audio was a spotty one. Digital playback seemed to pack on a lot of generally positive sonic qualities: low noise, tight, extended bass, flat frequency response—a lot of calories if you will, but ultimately less filling musically. The spatial and dimensional components in particular seemed

SPECS & PRICING

Analog outputs: RCA single-ended, XLR balanced
Digital outputs: SPDIF on RCA jack, AES/EBU on XLR jack
Inputs: SPDIF, TosLink, and USB digital inputs; word clock input
Formats supported: CD and SACD discs; PCM up to 192kHz/24-bit, USB PCM up to 384Hz/32-bit; DSD64 and DSD128
Dimensions: 17.25" x 6.4" x 13.25"
Weight: 61.75 lbs.
Price: $12,000

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physical expression of the artists on the stage whether solo or in full orchestral song appeared flat, dry, almost wraith-like in its lack of texture and harmonic verisimilitude.

With these reservations in mind, the K-03X provided a level of sonic micro and macro thrills in more areas than I've ever encountered before in a digital component. Regardless of format—CD, SACD, and high-resolution over USB—recordings that I'd heard dozens of times before suddenly didn't sound quite as blandly familiar. My most common impression was an elevation of dynamics and image layering, elements that I more closely associate with analog playback. The satisfying crackle and pop of percussion dynamics during the Police's "Tea in the Sahara" and "Murder By Numbers" presented even wider gradations of expression and audacious impact. Listening to the latest Mobile Fidelity SACD transfer of *Kind of Blue* revealed heretofore-undiscovered nuances—the breezy rush of wet, textured air over a sax reed, or the metallic nasality of Miles' muted trumpet during "Blue In Green." On SACD discs like these the Esoteric is in its zone, delivering a more settled, dynamically unshackled performance.

Foremost among my listening impressions was the overall sonic atmosphere that the K-03X conveyed—dense, rich, and saturated, and awash with ambient information. A singer's lips were more supple, there was more rosin dusting on the cello bow, greater attack from a rim shot, ripple off a timpani; the splash of a cymbal was more finely wrought, brass bloomed with warmer intensity and weight, and more air emerged from the surrounding environment. Was that the rustle of cellist Pieter Wispelwey's shirt sleeves, or a creak from Evgeny Kissin's piano bench, and a soft exhalation of breath at the end of a delicately performed measure? You bet.

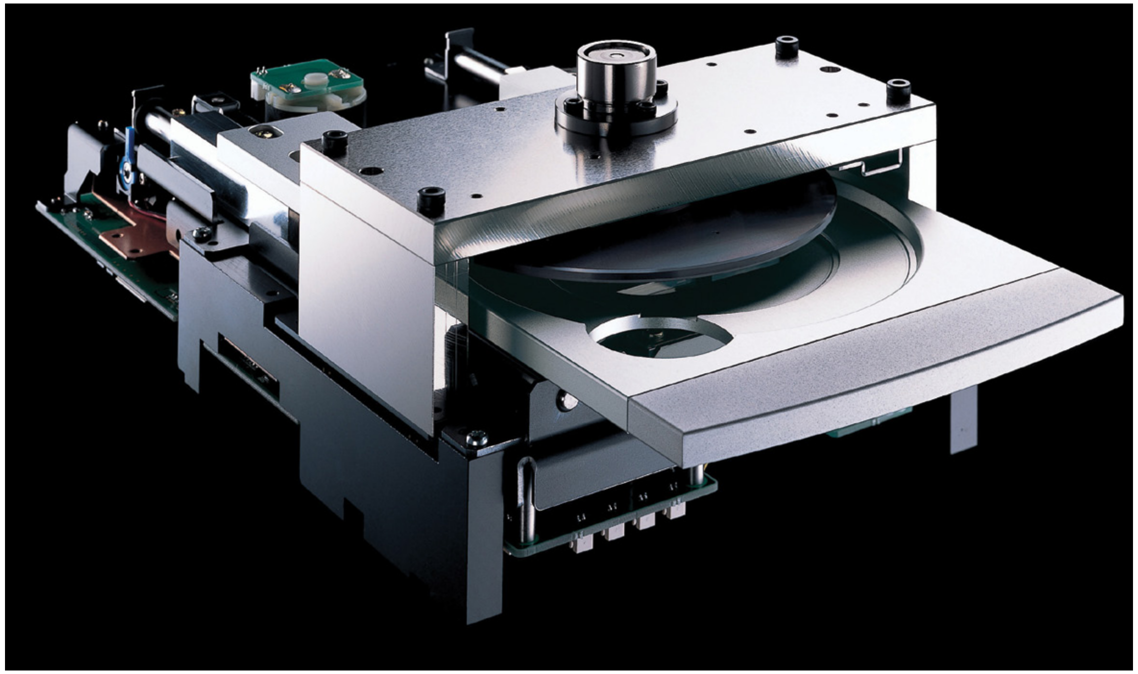
"Flat" is one four-letter word that you'll never utter about the K-03X. It shapes images like a sculptor, not just a sketch of frontal outlines, but with a singular sense of the front-to-back depth of the musicians. Symphonic depth of focus was unparalleled in my experience with digital playback. String section layering during "The Wasps Overture" had a clarity and transparency that bordered on the holographic [Reference Recordings]. In this crucial aspect, and keeping in mind that the audio signal exits the player in analog form, the K-03X transcends its digital roots at least as we've commonly associated the segment. If what I'm describing suggests that the K-03X is more analog; it is the case. Indeed it's the most persuasively "analog" disc player I've experienced.

Turning to vocals, even accounting for my familiarity with how artists like Norah Jones or Alison Krauss sound, I didn't recall these recordings having such levels of warmth and immediacy. They possessed a transparency and an openness that suggested I'd been listening through a thinly patterned veil all these years.

Equipment Report Esoteric K-03X CD/SACD Player

Vocal sibilances were more shapely—pointed, but not prickly. Low-level retrieval of piano cues during Jones' cover of "The Nearness of You" was beautifully articulated, backgrounds open and airy. Alison Krauss' "Slumber My Darling" had a newfound tranquility and lushness that replaced a cooler character that I had experienced from every previous player that had spun this *Appalachian Journey* disc.

In terms of flat-out performance and flexibility, the Esoteric K-03X runs the table. And while in recent years the rapid ascent of computer-based audio has stolen much of the digital thunder from the venerable CD player, take a moment to consider, where's the sex appeal in only downloading music files? Just how much of a high-end relationship can you have with that NAS drive, or music player app? Many of us have already rediscovered with the vinyl LP's latest comeback that there's something satisfying and seductive about physical media and the precision components that support that media. A top-tier player like the K-03X conveys that same message. Built to last and offering playback solutions that represent the best of both digital worlds, could the K-03X be the last player you'll ever buy? You won't get any argument from me. *A tour de force.* tas



Equipment Report

Meridian 808v6 Signature Reference CD Player/DAC with MQA Decoding

Milestone

Robert Harley

Way back in 1983, a small audio company from Huntingdon, England, created the world's first "audiophile" CD player by modifying a Philips machine. That company was Meridian Audio, and the product became the Meridian MCD. Meridian co-founder Bob Stuart took a look inside that first CD player and discovered many things that could be improved.

Thirty-three years later, Bob Stuart and Meridian Audio are still pursuing improvements in digital sound, but on a scale and with a technological sophistication that would have been inconceivable to the young engineer.

The culmination of everything Meridian has learned about digital-audio playback is realized in the new 808v6 Signature Reference Compact Disc Player reviewed here. Even considered as only a CD player, the 808v6's technology and performance would be beyond the ken of even the most advanced, mid-1980s engineering mind. It's interesting that in this age of streaming and downloads, some companies have abandoned the CD format as yesterday's news, while others have embraced it by introducing advanced new disc players (Meridian, Esoteric, dCS, for examples).

But the $22,000 808v6 is so much more than a CD player—it is the first disc player and DAC to offer decoding of Master Quality Authenticated (MQA) sources. As described in the accompanying feature, MQA delivers better-than-hi-res sound with a bit rate low enough to stream. This extremely sophisticated new technology was conceptualized and created by Bob Stuart with British mathematician Peter Craven. The 808v6 is thus not just the most sophisticated CD player in Meridian's history, but also the progenitor of an entirely new class of digital-playback systems based on the revolutionary MQA. The 808v6 is an even more significant milestone than was the MCD.

Let's first consider the 808v6's features and operation. The 808v6 looks and functions just like previous generations of this player have, with a large black chassis (colors are optional), generous-sized transport controls, card-cage construction with removable interface boards, extensive input and output options, and Meridian's two-handed MRC remote control. A flip-down panel opens to reveal additional controls, including volume setting, mute, and scan forward/backward. Since this is the "Signature" edition, the inside of the flip-down panel includes the signatures of Meridian founders Bob Stuart and Allen Boothroyd.

The 808v6 has six analog inputs (all on RCA jacks) along with source-switching and a volume control, making it a fully functional preamplifier as well as a disc player. (The volume control can by bypassed, with the output at a fixed level.) The gain of each analog input can be set independently. These inputs are named on the front panel with common source names, such as Disc, Radio, SLS (Sooloos), USB, etc. Digital inputs include two coaxial, two optical, USB, and Ethernet. This last input makes the 808v6 compatible with networked audio systems, including the Meridian Sooloos music server. In a nice touch, the remote control's track forward and back buttons, along with other transport functions, allow you to navigate a Sooloos playlist from the remote as well as from the Sooloos touchscreen. The 808v6 will output high-resolution PCM on a proprietary encrypted format, called Meridian High Resolution (MHR), for driving Meridian's digital active loudspeakers. Both balanced and unbalanced analog output jacks are provided. The substantial and handsome gloss-black case (colors are available at additional charge) has remained unchanged throughout the 808's long history. (I reviewed

Equipment Report Meridian 808v6 Signature Reference CD Player/DAC with MQA Decoding

the 808v2 in the August 2009 issue.)

The 808v6 is, like its predecessors, based on a CD-ROM drive for greater data integrity and reliability. Data from the disc are read into a buffer, and then clocked out with a high-precision clock. If uncorrectable disc errors are encountered, the drive can simply read that disc section repeatedly until the data are correct. In daily operation, the disc drive is smooth and quiet.

The 808v6 is, not surprisingly, software intensive. Filtering is performed on a general-purpose DSP chip. The filter is Meridian's "apodizing" type, which removes "pre-ringing" from the signal. Meridian's apodizing filter, first introduced in the 808v2, shifts the ringing energy so that it occurs after the transient rather than before *and* after the transient. This greatly improves the sound. The filter upsamples incoming data to 176.4kHz/24-bit. The powerful DSP chip is also put to work as bass and treble controls, accessible from the remote control. Unlike most manufacturers, Meridian doesn't tout the internal parts used in its products, suggesting that to do so would be like an automobile manufacturing promoting a car on the basis of its tires. They suggest that product design is more than a sum-of-the-parts list. Nonetheless, the 808v6 uses the Analog Devices AD1852 delta-sigma DAC. The output stage is entirely new for the v6. Earlier versions of the 808 featured a differential output stage to accommodate the differential DAC, and to drive the balanced output jacks. The new analog stage is a dual-differential circuit (the DACs are now in a dual-differential configuration), which reduces intrinsic noise by 3dB. The new analog stage also has wider bandwidth to take advantage of the improved time-domain performance realized by MQA.

The digital inputs have been upgraded in the new v6. The coaxial and USB inputs can accommodate up to 192kHz/24-bit data, and the player supports DoP (DSD over PCM) protocols. Meridian has long been at the forefront of reducing jitter, and the 808v6 features the company's most elaborate clocking and buffering system (with buffering and reclocking on the disc drive) on the control card, and on the digital inputs. A final FIFO (first-in, first-out) buffer before the DAC reduces any remaining jitter components and shifts the jitter frequency to below 0.5Hz. Each of these buffer/reclocking stages decorrelates jitter from the signal (jitter correlated with the audio signal is more sonically detrimental than decorrelated jitter).

The big news, of course, is that the 808v6 will decode MQA files. The player's large front-panel alphanumeric display indicates "MQA" when the 808v6 receives an encoded file. The display also shows the sample rate of the original source file that was used to make the MQA file. As explained in the accompanying article, MQA can encapsulate a wide range of sampling frequencies into a 44.1kHz or 48kHz file. The information above 20kHz is "unfolded" during playback to restore the original bandwidth of the signal used to make the MQA file. During my time with the 808v6 I saw the display indicate sampling frequencies of 44kHz, 48kHz, 88kHz, 96kHz, 176.4kHz 192kHz, and 384kHz. Again, these rates reflect the sampling frequency of the original signal encapsulated in the MQA file, and "unfolded" in playback to the signal's native rate. The display letters "MQA" indicate that the file has been "authenticated," meaning that the file is the same as that created by the engineer. A period after "MQA" signifies MQA "Studio" mode, indicating that the file being decoded has been approved by the artist or label.

My separate discussion of MQA accompanying this review includes an extensive sonic evaluation of the 808v6 playing back MQA files, so I won't reiterate those impressions here. Rather I will describe how the 808v6 sounds as a CD player and DAC with conventional (non-MQA) sources.

As with its superb predecessors, the 808v6 is in the upper echelon on the best-sounding digital playback devices extant. Its overall character was immediate and present, with a palpability that positioned lead vocals and lead instruments toward the front of the soundstage. The perspective was one of sitting close to the performers rather than hearing them from a distance. This quality fostered an intimacy with small-scale music and vocals, such as the Norah Jones album *Come Away With Me*. Gary Burton's vibes on his albums *Guided Tour* and *Common Ground* were right upfront, with vivid bell-like clarity. The Meridian is no shrinking violet, imbuing the music with a bold immediacy. By contrast, the Berkeley Alpha DAC Reference (my reference) was less upfront, with a farther-back-in-the-hall perspective. Which spatial perspective you prefer will be a matter of taste, as well as of how that perspective complements the rest of your system.

Transparency and resolution were in world-class territory. The 808v6's immediacy had the effect of fostering a sense of nothing between

you and the music, particularly with vocals. There was no veiling or opacity diminishing the impression of hearing deep into the acoustic space. Instruments toward the back of the hall maintained their clarity and timbre, and fine detail was well portrayed. The 808v6 also excelled at differentiating instruments from one another, a quality I enjoyed when listening to the modern big-band music of Gordon Goodwin, with its intricate arrangements. In these aspects I thought that the Berkeley DAC was a notch better, providing a deeper view into the soundstage and greater separation of instruments.

Equipment Report Meridian 808v6 Signature Reference CD Player/DAC with MQA Decoding

These comparisons were made, however, with both DACs driven via their USB interfaces (the outboard Berkeley Alpha USB, with the Berkeley DAC). But the Meridian sounds considerably better when connected via Ethernet rather than USB. (The Berkeley lacks a network connection, making comparisons in this mode impossible.) Compared with the USB connection, the 808v6 fed from Sooloos via Ethernet had smoother textures, particularly in the treble. The top end was cleaner and more refined. Detail resolution also increased, as did the sense of air and bloom around images. The Ethernet connection sounded more relaxed and natural, with more liquid midrange timbres. Even via USB, the 808v6 delivers world-class sound, but for the ultimate in performance Ethernet is the way to go. The Sooloos system, which connects to the 808v6 only by Ethernet, and which I had not used in many years, was superb functionally and sonically. The price has also come down considerably since the earlier versions.

The 808v6's bass was truly outstanding, with tremendous heft, weight, and dynamic impact. The generous bass fullness gave orchestral music a solid foundation and a great sense of power—the double-bass section of the orchestra had visceral body and authority. Rock was well served by the Merdian's dynamic and powerful low end, giving kickdrum a little more depth and impact than I'm used to hearing from my system. Bass lines were readily discernable. The Berkeley DAC was, by contrast, a little lighter in tonal balance, leaner and tighter, with less bottom-end fullness.

Conclusion

The 808v6 is a landmark product in that it introduces an entirely new digital technology, Master Quality Authenticated. That this new technology can be incorporated into an existing product (previous versions of the 808), yet elevate the sound quality to this degree, is remarkable. As a conventional CD player and DAC, the 808v6 is among the best digital front ends I've heard when driven via Ethernet. I have no experience with other MQA decoders to put the 808v6's MQA performance into perspective, but I can tell you that this Meridian playing MQA files sounds nothing like any digital anyone has ever heard.

If time travel were possible, I'd love to take the 808v6 back to 1983 and show it to the Bob Stuart who modified that first Philips player. The juxtaposition of these two machines is Bob Stuart's lifelong commitment to music and audio writ large.

SPECS & PRICING

Disc drive: CD-ROM
Analog outputs: Balanced on XLR jacks, unbalanced on RCA jacks
Digital outputs: Coaxial, Meridian SpeakerLink (optional MHR encryption on digital outputs at high sample rates)
Analog inputs: Six unbalanced
Digital inputs: Two coaxial, one Meridian SpeakerLink, two TosLink optical, one network (Ethernet) for connection to Meridian Sooloos, one USB (all inputs with nameable legends); coaxial and USB inputs compatible with up to 192kHz/24-bit; optical inputs compatible with up to 96kHz/24-bit
Digital filter: Meridian apodizing filter with upsampling and resolution enhancement
Decoding: MQA decoding and rendering
Dimensions: 18.9" x 6.9" x 16.2"
Weight: 38 lbs.
Price: $22,000 ($1100 additional for custom colors)

MERIDIAN AUDIO LTD.
Latham Road
Huntingdon
Cambridgeshire
PE29 6YE
England
meridian-audio.com

MERIDIAN AMERICA INC.
351 Thornton Road #108
Lithia Springs, GA 30122
(404) 344-7111

ASSOCIATED EQUIPMENT
Loudspeakers: Magico Q7 Mk.II, EnigmAcoustics Sopranino self-biasing electrostatic supertweeters
Headphones: Audeze LCD-4 driven by Moon by Simaudio 430HA headphone amplifier, Nordost Heimdall 2 balanced cable
Amplification: Constellation Audio Altair II preamplifier and Hercules II monoblock power amplifiers
Digital sources: Meridian Sooloos, Aurender W20 music servers
Support: Critical Mass Systems Maxxum equipment racks (x2), Maxxum amplifier stands (x2)
Cables: MIT Oracle MA-X SHD
Interconnects: MIT Oracle, AudioQuest WEL Signature and Wild
Digital interconnects: Audience Au24 USB, AudioQuest Wild Digital AES/EBU
AC: Four dedicated AC lines; Shunyata Triton 2, Triton DP, Typhon (x3) conditioners, Shunyata Sigma power cords
Acoustics: ASC 16" Full-Round Tube Traps, ASC Tower Trap, Stillpoints Aperture Panels
Accessories: Shunyata cable lifters, Stillpoints UltraSS, Ultra6 isolation

Equipment Report

T+A PDP 3000 HV CD/SACD Player and DAC

Master of All Trades

Robert Harley

Theory + Application Elektroakustik (T+A) may be the biggest and most technically innovative high-end audio company you've heard little or nothing about. The Germany company has been on a long-term growth trajectory but has intentionally kept a low profile in the U.S. so that it could focus on the European and Asian markets. That's unfortunate for those of us in North America because T+A makes an extensive range of technically innovative, beautifully built, forward-looking, great-sounding products that are fairly priced. Founded in 1978 by Siegfried Amft, T+A began life as a loudspeaker company with just two employees. Amft still heads the company, which has grown to a staff of more than a hundred. Fourteen of the employees are graduate-level engineers, many of them specialists in fields such as circuit-board layout, software development, and mechanical engineering. The company's history is one of fundamental technical research driving product development. T+A is as far from a marketing-driven "me-too" company as you'll find. For example, the company designs and builds its own disc-transport mechanisms from metal rather than buying off-the-shelf plastic mass-market drives. T+A also writes its own software, including the filters in its digital products. I was astonished to discover that T+A was creating its own software-based digital filters way back in 1989, a time when I thought that only Wadia and Theta Digital possessed that capability.

The PDP 3000 HV CD/SACD player and DAC reviewed here exemplifies T+A's engineering-driven approach. The company's flagship digital product is packed with sophisticated design and lavish execution. I got an inside look at the PDP 3000 HV during the Munich show where I sat down with T+A's lead designer, Lothar Wiemann. A physicist by education, Wiemann has been with T+A for more than 30 years. The 57-pound player's top panel features a round see-through window that shows off the internals. The massive chassis is made from aluminum, with isolated compartments for the digital and analog power supplies, and separate compartments for the digital and analog circuits. The transport is housed in its own aluminum chamber. This construction isolates the subsections magnetically and mechanically, and prevents coupling via RF. The exterior metalwork, which is available in dark grey or silver, is exemplary, as is the feel of the controls and the sophistication of their operation. Press the drawer-open button and the tray glides out with a smoothness and solidity that is unmatched in my experience. I would not have been surprised to learn that the PDP 3000 HV was priced at twice its U.S. retail of $22,500 (alas, up from $19,500 before the Euro-Dollar valuation swing this summer).

The front panel is dominated by two large knobs that select inputs and access the player's extensive menu. Those inputs include USB, AES/EBU, three SPDIF on RCA, two SPDIF on BNC jacks, and two TosLink optical. All these inputs can be named via the remote control's keypad. The remote control is a large, heavy unit machined from metal. The markings are a bit cryptic; you must refer to the table in the owner's manual to decipher their meanings. The disc drawer sits beneath a large display that shows the selected input, track number or time, and the set-up functions.

The rear panel has a couple of unusual twists. First, the PDP 3000 HV requires two AC cords, one for the player's digital power supply and another for the analog supply. The second twist is two sets of analog outputs, one for PCM sources and the other for DSD sources. Dual outputs are offered because the PDP 3000 HV employs completely separate signal paths for PCM and DSD decoding, all the way through the analog output stages and output jacks. Most DSD-capable DACs simply convert DSD to PCM for conversion to analog. T+A wanted to build a statement product without the compromise of designing a single DAC and analog output stage that would work for both DSD and PCM. This arrangement, however, requires two pairs of interconnects between the PDP 3000 HV and your preamplifier if you plan on listening to DSD downloads or SACDs. If you're not that hardcore, a menu setting will route all signals through the PCM output stage with a small penalty in DSD sound quality.

Much effort went into optimizing the performance with SACD and DSD sources. In addition to separate signal paths for PCM and

Equipment Report T+A PDP 3000 HV CD/SACD Player and DAC

DSD, the DSD DAC is a T+A custom design realized with discrete components rather than an off-the-shelf chip. In addition, the PDP 3000 HV allows you to select between two SACD and three DSD filters and noise-shaping algorithms to optimize the sound quality for different systems and DSD sample rates (see sidebar). I'm not aware of any other DSD DAC with either a discrete custom DSD DAC or selectable DSD filters.

I describe the PDP 3000 HV's technical details in the sidebar, but here's a synopsis: custom digital filters, completely separate signal paths for DSD and PCM, an all-discrete signal path including the current-to-voltage converter, dual-differential PCM DACs, custom discrete DSD DAC, isolated digital and analog power supplies including dual power cords, custom metal transport mechanism, an elaborate power supply, massive aluminum chassis, no op-amps in the signal path, and extensive jitter reduction. That's an impressive list of design features.

I should mention that if you find the PDP 3000 HV appealing but it's beyond your budget, and you don't need disc playback, consider T+A's $3995 DAC 8. It is based on the same design concepts as the PDP 3000 HV but in a less elaborate implementation. It still offers the discrete 1-bit DSD converter, selectable DSD filters, and many other T+A technologies. I haven't heard the DAC 8 but based on my experience with the PDP 3000 HV, I expect it to be outstanding.

Listening

In assessing the sound of the various inputs before beginning the evaluation, I discovered that I couldn't get the AES/EBU input to lock to the Aurender W20. Fortunately, the PDP 3000 HV offers coaxial inputs on BNC jacks, which don't suffer the technical and sonic compromises of RCA digital inputs. I slightly preferred the sound via the BNC inputs compared with USB, although the best-sounding configuration was with a Berkeley Alpha USB between the Aurender and the PDP 3000 HV. Note that you must use the USB input for decoding DSD files.

Starting with CD playback and PCM files of various resolutions, the PDP 3000 HV revealed itself to be a first-rate CD player and DAC. The player had certain qualities that were exceptional, particularly the sense of musical flow, rhythmic expression, and visceral involvement in the music. It's endlessly interesting to hear familiar music through a new playback device; the new product under evaluation sometimes pushes different musical buttons. That is, the way in which I engage with music is a little different with some products. With the PDP 3000 HV I found myself more immersed in music's rhythmic power and dynamic expression. Although the PDP 3000 HV hit all the critical listening checkmarks, it had something extra—a visceral energy that connected me with music on its most fundamental level.

Perhaps part of this character is due to the PDP 3000 HV's powerful, robust, and dynamically "tight" presentation that conveyed the physicality of music. It wasn't just that the bottom end was weighty and dynamic; there was more to why the player had such an energetic and upbeat quality that was deeply engaging. I think that the PDP 3000 HV more accurately reproduces the timing information in music, contributing to the sense of live music making. The PDP 3000 HV has a powerful rhythmic pull that connects on a deeper level than, for example, encouraging a dissection of the soundstage. This quality was

SPECS & PRICING

Type: CD/SACD player and DAC
Digital inputs: AES/EBU (x1), SPDIF on RCA (x3), SPDIF on BNC (x2), Tos-Link optical (x2), USB (x1)
Digital output: PCM on RCA jack (x1)
Formats supported: CD, SACD, DSD (up to DSD512), PCM up to 384/24
Analog outputs: Decoded PCM on RCA and XLR jacks, decoded DSD on RCA and XLR jacks
Conversion: Double-differential quadruple converter with four 32-bit sigma-delta DACs per channel (PCM); T+A True-1Bit DSD conveter (DSD)
Filtering and upsampling: Custom T+A PCM upsampling filter with four filter options; custom SACD filter with two filter options; custom DSD filter with three filter options
Disc mechanism: Custom T+A linear-tracking drive
Power: Dual IEC AC input jacks
Dimensions: 18" x 6.7" x 18"
Weight: 57.2 lbs.
Price: $22,500

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RUTHERFORD AUDIO
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rutherfordaudio.com

ASSOCIATED EQUIPMENT
Loudspeakers: Magico Q7 Mk.II, EnigmAcoustics Sopranino self-biasing electrostatic super-tweeters
Preamplifier: Constellation Altair II
Power amplifiers: Constellation Hercules II and Berning 211/845
Digital sources: Aurender W20 music server, Berkeley Alpha USB USB-to-SPDIF converter
Support: Critical Mass Systems Maxxum equipment racks (x2), Maxxum amplifier stands (x2)
Loudspeaker cables: MIT Oracle MA-X SHD and MIT ACC 268
Interconnects: MIT MA-X SHD, AudioQuest WEL Signature and AudioQuest Wild
Digital interconnects: Audience Au24 USB, AudioQuest Wild Digital AES/EBU, Audio-Quest BNC
AC: Four dedicated AC lines; Shunyata Denali conditioners, Shunyata Sigma power cords
Acoustics: ASC 16" Full-Round Tube Traps, ASC Tower Trap, Stillpoints Aperture Panels
Accessories: Shunyata cable lifters, Stillpoints UltraSS and Ultra6 isolation

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evident on the 96/24 download of Talking Heads' *Speaking in Tongues*. The PDP 3000 HV nailed the precise timing and transient pop of the exquisitely intricate percussion and James-Brown-esque rhythm guitar (and the way that they worked together), conveying this album's remarkable whole-body propulsive feel. Many DACs, by contrast, dilute this aspect of the music. The PDP 3000 HV had an uncanny ability to convey a sense of upbeat energy, as though the musicians were not just technically better but also more engaged in the performance. Of course, a music-reproduction component can't make musicians sound "better"; it can only more accurately convey their musicianship, which is what the PDP 3000 HV did consistently. (Incidentally, two days after listening to this album I cued it up again through the Aurender's iPad app and after about 30 seconds thought that something had gone wrong in my system. The sound was hard, bright, spitty in the treble, flat, and generally unpleasant by comparison. Three or four songs in I realized that I was listening to an old CD rip I had made, not to the 96/24 download, both of which were on the Aurender server.)

This rhythmic energy and involvement extended to all kinds of music. The great Freddie Hubbard-penned track "Byrdlike" from George Cables' 1979 album *Cables' Vision*, an album I've been listening to for decades, was rendered by the T+A with freshness, the be-bop melody and extended virtuoso solos coming alive, all driven by Peter Erskine's powerful drumming. The music just had an urgency and flow that were remarkable.

The overall spatial perspective was on the immediate and upfront side rather than relaxed. It was like sitting a few rows closer to the orchestra rather than mid-hall. This quality conveyed startling palpability and presence, yet the sound never crossed the line into sounding forward or dry. Moreover, the immediacy was accompanied by good soundstage depth, fine layering of images within that depth, and a wonderful bloom and air around image outlines.

The differences in sound between the four digital filters for PCM reproduction were greater than those of filter choices I've heard in other DACs. The outstanding dynamic alacrity, immediacy, and front-of-the-hall perspective described are in part the result of filter I chose for most of my daily listening (T+A calls the filter "Bezier"). Choose a different filter and the sound becomes a bit more relaxed. The precise sense of timing described is diluted somewhat with the other filters, but those filters have their own merits, including a bit smoother treble. Fortunately, it's easy to switch filters on the fly from the remote control.

With most CD/SACD players, the improvement in SACD over CD or CD-quality files is significant. But in the PDP 3000 HV, SACD and DSD files sounded massively better than CD-quality PCM. The T+A produced what was, by a wide margin, the best SACD and DSD sound I've ever heard. The clarity, transparency, dynamics, and resolution were simply stunning. On the terrific Chesky hybrid SACD *Jazz in the Key of Blue* by drummer Jimmy Cobb (the drummer on *Kind of Blue*, incidentally), the T+A player produced a stunningly realistic sound on Roy Hargrove's trumpet, in timbre, dynamics, resolution of fine detail, and that ineffable impression of the instrument being in the room with you. Cobb's gentle brush work was portrayed with tremendous resolution; this delicate sound was imbued with a dense filigreed texture that was utterly realistic. The background vocals on the track "Gaia" from James Taylor's *Hourglass* were also superbly rendered; I could hear the individual voices like never before, and the soprano sax floated in three-dimensional space.

In addition to enjoying and evaluating the sound of well-worn SACDs, I played some sample tracks from Blue Coast Music, an audiophile label that records exclusively in high-speed DSD, and makes those recordings available for download. Note that unlike many DSD recordings, Blue Coast's titles remain in the DSD format with no intermediate conversions to PCM. This approach, along with their purist recording techniques, result in some spectacular-sounding recordings. I'm not big on singer-songwriters, a staple of the Blue Coast catalog, but I found some music to enjoy. Guitarist Alex de Grassi performing "St. James Infirmary" on solo acoustic guitar, decoded by the PDP 3000 HV, wasn't just the most realistic guitar recording I've heard, but also one of the most realistic recordings of *any* instrument in my experience. So great was the PDP 3000 HV's transparency to sources that the sound had an almost "fool-you" realism.

Another wonderful DSD recording is the recently released album of James Matheson's compositions on Yarlung Records, downloaded in double-DSD from the website nativedsd.com. The album includes Matheson's String Quartet, recorded late last year at Samueli Theater, part of the Segerstrom Center for the Arts in Costa Mesa, California. I was fortunate to have attended a later Yarlung recording session in this hall. The recording was made in quad-DSD (DSD256) and is available for download in that format, but I was able to listen only to the DSD128 version because the Aurender W20 doesn't currently support DSD256. In DSD128, the string quartet was rendered by the PDP 3000 HV with exceptional vitality and timbral realism. The PDP 3000 HV's dynamic alacrity and visceral quality described earlier beautifully conveyed this music's unusual rhythmic flow. The album includes a piece for piano and soprano called *Times Alone* that was stunning in every way: the timbre and dynamics of the piano, the palpability and purity of the soprano, and the way that the instruments were presented spatially within the acoustic. It was all well served by the PDP 3000 HV.

As great as SACD and DSD sounded in the standard modes, the PDP 3000 HV has a trick up its sleeve that takes the sound quality to another level. The SACD and DSD filter options, mentioned earlier and described in detail in the sidebar, proved their worth. With Mode 1 engaged (what would be the only filter on other DSD DACs), DSD sound is as I've described—stunningly transparent, immediate, detailed, and present. Even if this setting represented the PDP 3000 HV's DSD performance, I would have still thought it the best DSD I've heard. But switch to Mode 2 (a filter with a higher cutoff frequency and gentler slope) and all those qualities stepped up a notch. The sense of air and detail increased, and with it the timbral, dynamic, and spatial realism. The final step is to remove any filtering (Mode 4, or what T+A calls "True DSD") and listen to the

Equipment Report T+A PDP 3000 HV CD/SACD Player

raw 1-bit datastream. The combination of no filter, T+A's custom DSD DAC built from discrete components, and an analog output stage that has been optimized for DSD playback elevated the performance to new heights. The wider filters allowed more very fine detail to be resolved and stripped away that last scrim between you and the instruments. On the Alex de Grassi track with Mode 1, you can sometimes hear him breathe or shift in his chair, but it's not very distinct. In Mode 2 these sounds are more clearly heard as what they are. But Mode 4 renders them so clearly that you get a goosebump-inducing impression of a person in front of you. In Mode 4, the illusion of hearing live music-making on the James Matheson album was absolutely startling. In fact, the third movement of the String Quartet was almost overwhelming in its intensity.

These filter modes are for decoding DSD files via the USB input. SACD playback also offers two filters, with the wider-bandwidth filter (Filter 2) offering a greater sense of palpability, fine detail, and sense of an instrument hanging in space surrounded by the recorded acoustic.

There's one big caveat, however; the wider filters won't work in all systems, and engaging those wider filters (or running with no filter, as in Mode 4) has the potential to damage tweeters. See the sidebar for more detail for why these filters are necessary and how the wider filters have the potential to damage your system.

Conclusion

The T+A PDP 3000 HV is an extremely sophisticated, versatile, and highly musical disc player. When playing CDs and decoding PCM files, the PDP 3000 HV was among the best DACs I've heard, particularly in its ability to convey the music's dynamic expression and rhythmic flow. It offers the kind of presentation that draws attention to the music rather than to the sound.

Had this been all there was to the PDP 3000 HV, it would have earned an enthusiastic recommendation. But the machine also offers what is far and away the best SACD and DSD playback I've heard. The various SACD and DSD filter options, unique to T+A, vault the performance to new levels of transparency, resolution, and realism. The PDP 3000 HV is an SACD and DSD lover's dream.

Although $22,500 is a lot of money for a disc player, the PDP 3000 HV nonetheless represents high value considering the performance, sophisticated technology, and battleship build-quality. Many companies charge this much or more for less. It took a long time for North American audiophiles (myself included) to discover this 38-year-old German company, but the PDP 3000 HV proves it was worth the wait.

Under the Hood

The DPD 3000 HV's transport mechanism is built around a dual-rail sled on which the disc tray slides, driven by a robust screw mechanism. The design and execution are worlds apart from the cheap plastic mechanisms found in even very expensive players. In addition to greater reliability, smoother operation, and potentially better sound, a custom transport provides the owner with the assurance that the player can be repaired in the future if necessary. I've heard horror stories from readers whose expensive disc players were rendered useless because the manufacturer could no longer source the transport mechanism. There's another advantage; the disc drawer's "cool factor" and luxurious feel exude precision and elegance. This mechanism was designed in-house by T+A, and they also machine all the metal parts that go into it.

As described in the body of the review, the PDP 3000 HV employs separate DACs for DSD and PCM decoding. The DSD DAC isn't an off-the-shelf chip, but a discrete custom circuit developed in-house. According to T+A, a discrete design allows them to clock the circuit with much greater precision and lower jitter than a DAC on a chip. One of the problems with a chip is that although the designer may clock the chip with a clean and precise clock, that clock signal can be degraded inside the chip. T+A's discrete DAC obviates that possibility. This discrete DAC is followed by its own analog output stage and output jacks that are not shared with the PCM DAC. Menu settings allow you to select different filters for DSD decoding.

The PCM filter is again of T+A's own design. Four different filters are available, selectable via the front-panel set-up controls or the remote control. As mentioned in the introduction, T+A has been writing its own digital filters for more than 25 years.

The PCM DAC is implemented with four Burr-Brown 32-bit DACs that are described as "advanced current-segment DACs" rather than as traditional R/2R ladder devices. The four DACs are arrayed in a double-differential configuration. An advantage of differential DACs is that a balanced signal is created in the digital domain before the DACs simply by creating an inverted version of the bitstream. When converted to analog by differential DACs, the signal appearing at the rear-panel XLR jacks is truly balanced. Some products without differential DACs but with "balanced" outputs simply add a phase splitter in the analog domain to create + and – signals, but this adds another active stage to the signal path.

Unusually, the current-to-voltage converter is a discrete design rather than the ubiquitous op-amp. This crucial component, which exerts a large influence on the sound quality, must convert the DAC's current output to a voltage that can drive the output buffer. The rest of the analog output stages (the stage for PCM as well as the stage for DSD) are also all discrete, fully balanced, dual mono, and with zero global feedback.

The "HV" in the product's name, and in the rest of T+A's "HV" series, stands for High Voltage. This refers to the fact that the rail voltages supplying the circuits are much higher than is traditionally employed. T+A says that the high voltage rails allow the transistors to operate over a more linear portion of their operating range. The power supply features multiple stages of cascaded discrete regulation, with the last regulation stages located right next to the circuits supplies.

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DSD Primer and the PDP 3000 HV's DSD Modes

Direct Stream Digital (DSD) is Sony's tradename for an encoding technique in which the audio signal is sampled very quickly (2.8224 million times per second), but with an amplitude resolution of just one step per sample. This encoding couldn't be simpler; if the signal is increasing in amplitude, a binary "1" is recorded. If the audio signal is decreasing in amplitude, a binary "0" is recorded at each of the 2,228,400 sample points per second.

This technique produces a signal with a very wide bandwidth but a very high level of quantization noise. In fact, the signal-to-noise ratio of raw DSD encoding is just 6dB. You may recall that a digital audio system's signal-to-noise ratio is determined by the number of bits in each sample. A 16-bit system gives us a signal-to-noise ratio of about 98dB, or 6.01dB per bit.

So how can a digital encoding system with such high noise work? A technique called "noise shaping" shifts the noise out of the audioband to a higher frequency. With standard-rate DSD encoding as found on SACD (called "DSD64" because the 2.8224MHz sampling frequency is 64 times that of CD's 44.1kHz rate) the noise begins rising just above 20kHz, with most of its energy between 40kHz and 100kHz. All SACD players and DSD DACs incorporate a low-pass filter (typically at 50kHz) to prevent this noise from appearing at the product's analog output jacks. If left unfiltered, this ultrasonic noise could cause your power amplifier to become unstable or oscillate, potentially overdriving your tweeters. The amplifier can also create intermodulation products by this combination of high noise level and the audioband musical signal. Even if the amplifier is well behaved when driven by a high level of ultrasonic noise, that noise puts a burden on tweeters. You don't hear the noise, but the tweeter's voice coil must still dissipate that energy. The wider the amplifier's bandwidth, and the more stable and linear it is at those ultrasonic frequencies, the less problem DSD's ultrasonic noise poses. Moreover, today's tweeters are better at handling higher power levels that tweeters of twenty years ago.

As the DSD rate is doubled ("double-DSD" or "DSD128"), the noise energy can be spread out over a wider bandwidth than with standard-rate DSD. The filter requirements are thus relaxed for DSD128 compared with DSD64. Quad-rate DSD (DSD256) allows the noise to be spread out over an even wider frequency band, again changing the optimal filter characteristics.

Almost every other DSD playback device I'm aware of employs a single filter and noise-shaping algorithm for all DSD rates, compromising the performance of higher-rate DSD files. But the PDP 3000 HV allows you to select between two different SACD filters and noise-shaping algorithms when playing SACD, and four different DSD "Modes" when decoding DSD files via the USB input. These Modes correspond to three different DSD filter characteristics.

The SACD 1 mode is a standard filter just as you'd find in any SACD player. The owner's manual doesn't specify the cutoff frequency, but my guess is it's around 50kHz. SACD 2 has a higher cutoff frequency and a more gentle slope. T+A recommends this filter for systems with amplifiers that can handle a higher level of ultrasonic noise.

Moving to the DSD DAC Modes, when playing files through the USB input, Mode 1 is a standard low-pass filter that is suitable for DSD64 and any playback system. It removes all the ultrasonic noise. With Mode 1 engaged, the PDP 3000 HV sounds as I described it: the best DSD64 playback I've heard. But switch to Mode 2 (a higher cutoff frequency and gentler filter slope) and the sound takes on an increased transparency, resolution, and timbral realism.

Mode 4 is what T+A calls "True DSD." This mode imposes no filter whatsoever, letting the DSD DAC run wide open. T+A recommends this mode only for higher-speed DSD, and if your amplifier has a wide bandwidth with very low transient intermodulation distortion. As described in the review, Mode 4 is absolutely startling in its transparency. Very fine detail, such as the sound of fingers moving on strings on the Alex de Grassi track, for example, took on a greater vividness.

Mode 3 isn't a fourth filter option. Rather, Mode 3 automatically engages the Mode 2 filter for DSD64 files, and switches to Mode 4 (no filter) for DSD128 and higher sources.

The PDP 3000 HV's manual cautions against using the True DSD mode (no filter) unless your amplifier and loudspeaker can handle the high levels of ultrasonic noise. I listened in Mode 4 with DSD64 files without a problem, although I kept the volume level moderate. As a safety measure, you must allow the PDP 3000 HV to engage True DSD in the basic system configuration menu in additional to selecting it from the DSD Mode menu. tas

MUSIC SERVERS

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Equipment Report

Sony HAP-Z1ES HDD Audio Player

Game Changer

Steven Stone

As the flagship model for its "High Resolution Audio Initiative," the new Sony HAP-Z1ES defines what Sony sees as the future of two-channel audio. It attempts to be easy for a naïve user to operate, yet capable of the highest audio quality. And while it's relatively simple to make an audio product that is easy to use, very few ergonomically elegant mass-market audio devices also produce state-of-the-art sonics. Conversely, there are quite a few state-of-the-art computer audio rigs that sound superb, but require at least a bachelor's degree in electronics with a minor in computer sciences to set up and use. Bridging the gap between these two extremes is exactly what the Sony HAP-Z1ES is all about.

The Grand Tour

What *is* an HDD audio player? In the case of the HAP-Z1ES, it is a local network-aware device that plays digital music files. It hooks up via Ethernet or Wi-Fi to your local network and the Internet. The HAP-Z1ES contains a 1TB hard drive for storing music files; it also has the ability to use external USB drives for additional storage. And what can the HAP-Z1ES store and play? It supports virtually any format audio file, including: DSD (WSF and DSDIFF), WAV, AIFF, FLAC, ALAC, ATRAC, MP3, AAC, and WMA files.

Since it is a local-network-aware device, any music file on any computer hard-drive in your home network can be imported into the HAP-Z1ES via a proprietary application program called "HAP Music Transfer." The HAP Music Transfer app can run on almost every PC that supports 32-bit versions of Windows or Mac OS. Besides the initial transfer of music files, the HAP Music Transfer app can also automatically and periodically transfer any new music files on designated hard drives in your home network to your HAP-Z1ES player's HD storage.

Don't look for SPDIF, USB, or AES/EBU digital inputs on the HAP-Z1ES player, or any digital outputs. The only hard-wired input is the aforementioned Ethernet connection, and the only outputs from the HAP-Z1ES player are analog. Located on the rear panel you'll find a pair of balanced XLR and a pair of single-ended RCA outputs. If you are in need of digital outputs to connect to your DAC or AV receiver, the HPA-Z1ES won't help you.

The front panel of the HAP-Z1ES is almost as Spartan as its rear panel. It has an on/off button on the extreme right, a large 3 7/8" by 2 1/4" full-color display panel in the center, and four buttons and one large knob on the left side—the four buttons are menu, back, enter, and play. The HAP-Z1ES also comes with a small wand remote that supports basic functions including play, pause, jump forward, jump back, and select tracks for play. But most users will probably want to use Sony's new dedicated app with the HAP-Z1ES. My review sample came with a Sony Xperia tablet that had the HAP app already installed. By the time you read this review Sony will have versions available for IOS and Android devices. I'll tell you more about the app later in the review.

While the outside of the HAP-Z1ES may be simple, its inside is full of new, sophisticated circuitry. For compressed music files Sony has developed DSEE (Digital Sound Enhancement Engine) technology, which restores upper frequencies and the "tail" of waveforms that were truncated by lossy compression schemes. The HAP-Z1Es also includes Sony's new "DSD Remastering Engine," which according to Sony "combines a high-performance DSP (digital signal processing) and FPGA (field-programmable gate array) to convert *any signal* (my emphasis) into DSD128 signals. It was designed based on the know-how garnered from Sony's 8-times oversampling and Extended SBM (Super Bit Mapping) technology for professional recorders." Yes, you read that right: the remastering engine can convert any and all PCM music files into DSD128 format, regardless of their original sample- or bit-rate. You can, if you wish, turn off the DSD Remastering engine via the main settings menu so the HAP-Z1ES will not convert PCM to DSD.

Once a digital file has been converted into DSD128, the final step is to convert that DSD file into analog for playback. The HAP-Z1ES does this step with an analog FIR (finite impulse response) filter. Along with reducing the extreme high frequency noise inherent in DSD signals, the FIR filter system has independent right and left channels with four separate filters per channel.

A low-phase-noise liquid-crystal oscillator

Equipment Report Sony HAP-Z1ES HDD Audio Player

handles internal digital timing in the HAP-Z1ES, which acts as the master clock for all digital signals. According to Sony's measurements, the low-noise liquid-crystal oscillator delivers 20–30dB lower noise than conventional clocks.

The HAP-Z1ES has two separate large-capacity transformers, one for the analog power supply and one for the digital supply. Both receive a special vacuum impregnation pretreatment so all the winding coils are uniformly coated with varnish. By using separate transformers for analog and digital power supplies, the HAP-Z1ES achieves separation of analog and digital signals at the circuit board level. This reduces the adverse effects of digital noise to a minimum.

Unlike many digital products, where the chassis is merely a big metal box, the HPS-Z1Es uses "Frame Beam Chassis" construction, which Sony has used on all its ES-level products in the past. The HP-Z1ES's base is composed of two metal plates of different thicknesses that support the main chassis. There are two additional base plates under each power transformer. Along with these metal plates, Sony employs structural beams than run crosswise to reinforce the overall rigidity and improve resonance control.

To further improve overall vibration control the HAP-Z1ES uses a new foot design that employs ribs combined with an offset connection that isolates sound pressure from external sources. Inside the HAP-Z1ES Sony uses special mounting methodologies—an example is the analog connection terminal, which is mounted separately on its own isolated board to minimize the effects of vibration. An internal cooling fan is mounted via a damping system to minimize any vibration it might generate. It is also specifically angled so that it can operate with maximum efficiency and minimum noise.

Sony's attention to detail on the HAP-Z1Es extends even to the main dial on the front panel. It is attached to an iron plate to prevent twisting or lateral movement. Although priced at only $1999, the HAP-Z1ES' fit and finish certainly rivals preamps and network players costing a lot more.

The Setup

The original set-up plan was for a Sony technical expert to fly into Denver from San Diego and set up the HAP-Z1ES for me. An especially vigorous snowstorm curtailed his visit. He got as far as the outskirts of Boulder before he had to give up. Undaunted, I set up the HAP-Z1ES by myself without any outside technical assistance. I found that even an audiophile with limited computer savvy could install a HAP-Z1ES with little difficulty.

After unpacking the HAP-Z1ES, I placed it on an equipment rack shelf and attached its analog outputs to my preamp and connected its Ethernet input to my home network via a 100 feet of Cat 5 Ethernet cable. I could have used the HAP-Z1ES' built-in Wi-Fi (I got a signal strength reading of 61 from the HAP-Z1ES's built-in Wi-Fi signal strength meter), but I wanted to make sure the HAP-Z1Es was receiving the most robust signal I could supply.

After connecting the HAP-Z1ES I turned it on and went to the "Network Settings" section of the main menu. There I selected "wired set-up" and "Auto" from the IP address page. After that, the HAP-Z1ES linked to my network and I saved the configuration. For users who like reassurance, the HAP-Z1ES lets you check and confirm that the settings are "OK" before closing the network settings pages. The procedure is much the same for wireless Wi-Fi, except you have a page that lets you select your access points. If you live in a Wi-Fi-intensive environment you can pick the correct Wi-Fi network and enter your password. Near the end of the review period I switched over to Wi-Fi access and had no issues with changes to the installation or impaired Internet performance.

Once the HAP-Z1ES is connected to your home network, either via Ethernet cable or via Wi-Fi, you can transfer music files to its internal hard drive. Unlike many music servers that employ a closed system (see AHC's review of the Olive player), the Sony HAP-Z1ES permits you to add, store, and backup your music files onto standard USB hard drives as well as its internal drive. Although created so those new to music servers can easily use it, the HAP-Z1ES can fit into a fairly complex computer music eco-system. Sony expects the average HAP-Z1ES owner already has a library or even multiple libraries of music. With the Sony HAP Music Transfer application owners can not only transfer current music files over to the HAP-Z1ES, but also periodically and automatically copy over any new music to their HAP-Z1ES.

Initially I had some problems using the HAP Music Transfer application on my ancient Dell D620 laptop, which runs Windows XP. Even though I was running the last version of XP, the D620 did not recognize the HAP-Z1ES. After a couple of e-mails, Sony determined that the D620 was not running XP in the 32-bit mode that is needed for the program to run successfully. Any PC running a more current version of XP, Windows 7, or Windows 8 won't have this issue. Since my ancient laptop proved to be better suited for doing firmware upgrades than running current software, I asked to see the Mac version of the HAP Music Transfer application. Sony then sent me a Beta copy of the Mac version which had just become available. It worked flawlessly.

When first used the HAP Music Transfer application has a default location for your Mac's music library that may or may not be correct for your system. If you don't keep your music on your primary drive you will have to change the app's default location for your music folders. You must change the music library default or nothing will be transferred because the app won't be able to find your music files.

The HAP Music Transfer app supports multiple music folder locations. This means that if you and your family have separate music libraries on different computers in your home, as long as they are attached to your home network via Ethernet or Wi-Fi, the HAP Music Transfer app can move them over to the HAP-Z1ES after you've selected and added them to the HAP Music Transfer's music library folder list.

Once your music folder locations have been entered into the HAP Music Transfer app, you can specify what kind of files you would like to transfer. The HAP-Z1ES supports 3GP, AA3, AIF, AIFF, DFF, DSF, FLA, FLAC, M4A, MP3, MP4, OMA, WAV, and WMA file types. And while you can transfer any and all of these formats over to the HAP-Z1ES, you might want to restrict its library to higher-quality lossless file formats. For users who've generated MP3 versions of their full-resolution files for their portable devices, being able to exclude MP3 files is a useful feature. By checking or unchecking the format boxes on the "Contents Settings" page of the HAP Music Transfer app, you can specify exactly which formats will be transferred. Once you've specified file types, pushing the "Start" button will initiate file transfers. My initial transfer involved 5697 music files and required almost 20 hours to complete. You can expect the first transfer to take a while, which is why a wired Ethernet connection with its faster transfer rates is the best option.

After all your music files are transferred to the HAP-Z1ES by the HAP Music Transfer app, the HAP-Z1ES connects to Gracenote's database to acquire artwork for any files that may not have artwork. A majority of my music files already had artwork, but for some of my own recorded tracks the HAP-Z1ES found some interesting, if not entirely correct, art and attributions. On one particular track, which was a recording by my acoustic band, Knapweed, of the Bill Monroe/ Peter Rowan song, "Walls of Time," the song was incorrectly attributed to Emmylou Harris and the Nash Ramblers from their *Live at the Ryman* album. I was quite surprised when I selected it; instead of Emmylou's superb vocals I heard my own pitiful croaking.

If you select "auto update" from the HAP Music Transfer program's options, during each launch it will immediately look for any new tracks in your designated music library locations and automatically transfer any new files onto the HAP-Z1ES.

In addition to playing music from your music library, the HAP-Z1ES also has a built-in Internet radio tuner. Called the "V-Tuner," this feature includes the ability to search for Internet radio stations by genre or location. It also lists the bit rate of each station so you can see exactly what quality level a station can deliver. I quickly found the local stations that I listen to regularly and designated them as "favorites" via a heart symbol icon, which added them to a special list that I could access more easily.

Sony also added a special AI feature to the HAP-Z1ES called SenseMe channels. According to Sony, SenseMe channels is a function that analyzes and automatically categorizes music tracks according to their mood and tempo using the 12-tone analysis technology developed by Sony. SenseMe has twelve categories of music—morning, daytime, evening, midnight, energetic, relax, upbeat, mellow, lounge, emotional, dance, and extreme. These could be handy, especially if you'd like something a bit more selective than good old-fashioned shuffle mode. In my music library of almost 6000 songs, selecting "extreme" brought up 34 tracks. I guess I'm just not an extreme kinda guy.

The HAP App and HAP-Z1ES Remote

The HAP-Z1ES comes with a silver wand-shaped remote control. It also has its own dedicated free downloadable app. The remote control duplicates all the buttons on the HAP-Z1ES front panel. It also adds jump forward, jump reverse, as well as mute and volume controls. Although the HAP-Z1ES has a fixed output level, both the volume and muting can be controlled by compatible Sony receivers and integrated amplifiers, or even assigned to products from other manufacturers, using the HAP-Z1ES's "Amp Control Setting."

The HAP control application will be available for Android phones, iPhones, iPads, and Sony Xperia, and other Android tablets. At the time of the review, only the Android app had been finalized, so Sony included an Xperia tablet with the app installed on it. Once the app located the HAP-Z1ES on my network it worked flawlessly with no crashes or delayed responses. The app lets you choose music, make playlists, and find particular tracks in your music library. Among its extra features is a "new music" list that shows you the latest additions to your HAP-Z1ES's music library and the most popular tracks called "favorites" (in case you really enjoy playing the same tracks over and over.) One nice, yet completely superfluous feature is that the background colors of the app change in response to the primary colors in the cover art of any currently playing track.

Day-to-Day Use

While I'm pretty sure there's a computer in there somewhere, its lack of computer-based issues has made living with and using the HAP-Z1ES on a day-to-day basis a joy. I just turn it on and it works. Whether controlled from the front panel, the remote control, or the app, the HAP-Z1ES responded to commands quickly, and except in the case of hooking up with Internet radio stations via its V-Tuner, where it sometimes took as much as ten seconds for some stations to start to play, any music on the internal HD began playing almost instantly after being selected.

While I didn't find Sony's SenseMe feature of particular value, I'm sure most users will find some use for it, if only to annoy significant others by selecting "lounge." One feature I did enjoy was the "Favorites" selection feature in the V-Tuner. I was able to assemble a very nice list of higher-bit-rate Internet radio stations in a short time by using V-Tuner's search features.

The Sound

As someone who has felt that the best digital reproduction comes from files that have not had their native rate changed, reading that PCM files can be converted into DSD by the HAP-Z1ES raised some red flags. But after comparing the HAP-Z1ES's DSD Remastering Engine's rendition of PCM recordings with those same files played back at their native rate through the HAP-Z1ES, I can only conclude that whatever Sony is doing in the conversion process doesn't appear to have any signature negative sonic effects. And while I wouldn't go so far as to write that the Sony HAP-Z1ES does a better job of reproducing PCM than PCM-centric DACs or HD players, it certainly is on sonic par with the best I've heard.

After an initial break-in period I did a number of A/B comparisons between the HAP-Z1ES and two streaming audio/computer based sources. The first source was a Sonos ZP100 feeding a Mytek Stereo192 DAC via a coaxial digital connection. The second source was a Mac Mini running Pure Music into the Mytek Stereo192 via its USB 2.0 connection. It took me several sessions of comparing these three systems before I could consistently recognize the HAP-Z1ES from the other sources in a blind A/B.

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The primary and telling difference was that the Mytek had slightly more energy in the upper midrange into the lower treble. In my system I felt the HAP-Z1ES was slightly more natural sounding with less edge. On *Ella Fitzgerald and Oscar Peterson*, Ella's voice had more air through the MyTek, but it had a more natural and organic tonality through the HAP-Z1ES.

In many respects the HAP-Z1ES and the Mytek DAC were very similar in their sonic presentations. Both recreated a soundstage with convincing three-dimensionality. Both also had the same level of dynamic contrast on the micro- and macro-levels. Bass extension was also a virtual dead heat with both quite capable of full low-frequency extension and subtle inner detail.

Which sound is more neutral or preferable will very likely depend on the rest of your system. If your system is on the darker side of neutral, the Mytek's extra bit of forwardness would match quite well, while the HAP-Z1ES could sound a bit subdued and perhaps even hooded. But if your system has any tendency toward brightness, the HAP-Z1Es will probably be better received than the Mytek. There's also something quite seductive in the HAP-Z1ES' midrange presentation that is hard to resist.

The most difficult and least conclusive A/B test I performed during the review was comparing the DSD Remastering Engine's DSD conversion of PCM files with those same files played back without the DSD Remastering Engine engaged. When switched back and forth there was a pause followed by about a two seconds of playback of the last snippet of music before the switchover. During that two seconds the sound was slightly different, seemingly warmer and rounder, but after that initial two seconds the sound reverted, and in blind A/Bs I could not tell whether I was listening to Remastering Engine or native output. I used both 16/44.1 and 24/96 PCM files for this test and didn't hear any differences when I switched between DSD and PCM on standard Red Book or higher-definition digital files.

During the A/B listening sessions I had ample opportunity to compare the HAP-Z1ES app with the "Remote" app for iTunes. I much preferred Sony's App to Apple's. The HAP app was easier to use and navigate. It also provided more information about tracks including the original sample and bit rates.

One final aspect of the HAP-Z1ES' performance that deserves attention is its prowess as an Internet radio tuner. It was easily the best-sounding Internet radio I've heard to-date from any device. And while I didn't hear any changes when I switched in Sony's DSEE (Digital Sound Enhancement Engine) on my uncompressed music files, when it was activated for Internet radio the overall sound quality improved dramatically. For some prospective owners the HAP-Z1ES' stellar Internet radio performance could be a primary reason for ownership.

The High Value HAP-Z1ES

In overall sonics and build-value for the dollar, the Sony HAP-Z1ES sets new standards. A Mac Mini with monitor, keyboard, mouse, and external drives attached to the MyTek Stereo192 DAC runs over $2500, and if you use better quality cables the price could go substantially higher. Even the Sonos ZP100/Mytek Stereo192 front end costs around $2300 when you include a NAS drive. For $1999 the Sony HAP-Z1ES supplies the computer, hard drive, DAC, and app to run it all. While this is a bit of a stretch, the HAP-Z1ES could be considered the iMac of HD music players—everything you need to acquire, store, and reproduce HD music files, regardless of format, in one carefully thought out and powerful box.

For audiophiles and music lovers who want to listen to high-quality digital music files without the hassles of keeping another computer working optimally, the HAP-Z1ES is an attractively priced, yet fully featured option. It also doesn't hurt that its control interfaces are easy to use and unintimidating even for non-techy users.

Sonically, it's difficult to fault the HAP-Z1ES. Its sound quality was such that it rivals comparably priced standalone DACs, yet delivers more functionality and won't be made obsolete by the latest USB, FireWire, or Thunderbolt interfaces since it uses Ethernet and Wi-Fi as input connections.

Throughout the review period as I put the HAP-Z1ES through its paces, I looked for reasons the player might be not be considered a true high-performance component and found none. If you plan to spend more than $2000 on any digital front end, whether it be an audio-computer, CD player, DAC, network player, or any other front end that uses digital files as a source, and you don't audition a HAP-Z1ES, you are failing to consider what may well be *the* benchmark digital product of 2014. tas

SPECS & PRICING

Frequency response: 2Hz–80kHz +/-3dB
Dynamic range: 105dB or higher
THD: 0.0015% or less
HDD capacity: 1TB
Supported playback formats: DSD (DSF, DSDIFF), LPCM (WAV, AIFF), FLAC, ALAC, ATRAC Advanced Lossless, ATRAC, MP3, AAC, WMA (2 channels)
Outputs: Unbalanced 2.0V RMS (50k ohms); balanced 2.0V RMS (50k ohms), 600 ohms
External ports: Type A USB for hard drive, IR Remote-Out jack for IR blaster
Power consumption: 35W (on), 0.3W (off), 2.8W (standby)
Dimensions: 17" x 5 1/8" x 15 3/8"
Weight: 32 lbs.
Price: $1999

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Equipment Report

NAD Masters Series M12 Digital Preamplifier DAC and M22 Stereo Amplifier

Sensible but Serious

Neil Gader

NAD electronics has been marching down audio's red carpet for years, picking up awards and accolades from high-end journalists and customers alike. Its classic BEE line—all buttons and knobs, and blue collar to the core—is still being turned out in sturdy but sensible, olive-gray stamped enclosures. Outward frills are kept to a minimum, and that's always been NAD's point. It's the sonic thrills rather than the visual bells and whistles that customers have come to expect, and that keep them coming back.

The Masters Series represents the more uptown side of NAD. Stylish and sophisticated, this is a company flexing its technical muscles while preserving the underlying value it is famous for.

In case your experience with NAD electronics ended with the original, circa-1978 3020—the modest integrated amp that addicted many a young audiophile to this hobby—you'll be in for a big surprise. The latest generation Masters Series (the originals were introduced in 2005), as embodied in the M12/22, is unreservedly gorgeous. The aluminum casework is elegantly crafted. Its shiny black, accented front panel and vented top plate are sumptuous to the eye and the touch. Both units are powered-up via a recessed top-mounted switch. The M12's large, readable touchscreen display easily handles functions normally left to a small army of buttons and toggles. The only vestige of a bygone era is the nicely weighted rotary volume knob. For that, let us all give thanks.

The M12 is a preamplifier/DAC—a high-end segment that has been growing in leaps and bounds. Like the M2 Direct Digital Amplifier, and more recently the C 390DD, the M12 employs its own "Direct Digital" 35-bit processing technology, thereby circumventing all analog stages in the signal path. Music remains in the digital domain through the preamp. By NAD's reckoning this eliminates the phase shift, noise, and distortion of many analog designs.

Before I describe back-panel connectivity, a word or two is needed about a unique feature of the M12 architecture. NAD calls it "Modular Design Construction" or MDC, and it looms large in the Masters Series. MDC uses replaceable cards that fit into slots on the M12's back panel, essentially making the unit future-proof as upgraded features become available. The M12 back panel has slots for three additional modules. Currently one such MDC option is the DD HDM-1 HDMI module with three inputs and one output (3D video pass-through). My M12 review sample, however, was outfitted with the optional DD BluOS network-audio module, which permits streaming of various music services like Tidal and TuneIn radio, plus high-resolution PCM files (no DSD yet) from a NAS device or local USB HDD/SSD. The NAD Controller App (from the iTunes App Store) manages a music library and can be controlled with an iOS or Android device. The card includes integrated WiFi/Ethernet and aptX Bluetooth connections for hi-res streaming from a smartphone or tablet. For Millennials this module is likely a must.

Even without the DD BluOS module, standard M12 connectivity is excellent. The back panel is densely populated with inputs, including AES/EBU, asynchronous 24-bit/192kHz USB Type B, coaxial digital, and optical digital, along with balanced and single-ended line-level. There are also front and rear USB-A inputs capable of 24-bit/48kHz resolution. Additionally there's a pure Class A buffer using the newest generation of "Super OP Amps" to provide low-impedance balanced and single-ended connections to power amplifiers or active loudspeakers. Also resident is an mc/mm phonostage module with settings for both moving-magnet (mm) and moving-coil (mc) cartridges. Gain is set automatically. Vinyl lovers should keep in mind that with Direct Digital processing a 24/192 ADC will convert the analog signal to digital, automatically setting the gain for the best resolution and lowest noise. The included remote control handles all functions quite capably.

Master Stroke

NAD describes the M22 as a hybrid digital amplifier. Output is rated at 250Wpc into 8 ohms, and >650W into 2 ohms. In NAD's words, the M22 "uses the latest nCore amp technology licensed from Hypex." Further refinement of the UcD concept (Bruno Putzeys was its inventor) has yielded distortion that in NAD's words is now "below measurement, [providing] an ultra-high damping factor and unconditional stability with any speaker." Additionally, the M22 is DC coupled throughout, from input to output. There is no capacitor in the signal path. The power supply is a custom switch-mode design, while secondary supplies are individually regulated and decoupled at each op-amp to maximize dynamic range and lower noise. The M22 employs NAD's latest generation of digital Power Drive, which automatically senses the speaker's impedance and adjusts and controls the amp's power envelope to more efficiently drive that particular load.

Getting Up and Running

The touchscreen menus are well-organized. The main screen lists the input and volume settings in large script, while smaller shaded boxes indicate current preferences. Settings include: Mode for polarity, reverse, and mono and stereo options; EQ for treble, bass, and balance; and a disable option. Setup is divided into four subsections: Source for renaming inputs; Digital Output for selectable sample rate; Control Setup for IR and auto-standby options; and Speaker for adding and optimizing a subwoofer in a 2.1-channel configuration (including a second-order high-pass and low-pass crossover with selectable frequency).

Activating the DD BluOS module was as simple as connecting an Ethernet wire to the LAN connection. The NAD Controller App instantly recognized my WiFi network from my iPad Air 2. Logging in to TuneIn was easy, but finding my Synology NAS was another matter.

Fortunately NAD is sympathetic to the plight of the computer-phobic, and aware of the fact that hooking up an existing NAS with a DAC/renderer can be a bit of a nail-biter. It offers a couple of helpful options if you get stuck (as I did). First there's online help at support.bluesound.com, where I searched for instructions and found (Eureka!) the document titled "Synology NAS and Network Discovery Configurations." Alternatively you can email (as I did) to support@bluesound.com. I got a swift response, and within about five minutes, was up and running. In my particular situation the fix was easy—I needed to enable "Guest Access" from the Synology in order for it to broadcast its shared folders to BluOS. Not a big deal, but NAD's help probably saved me hours of fruitless fiddling.

Sonically the M12/M22 system will remind NAD followers that there is a bloodline here. These components remain true to NAD values in the way they prize midrange neutrality and integrity, yet also throw hints of warmth and richness into the mix. However, the M12/22 begins to depart from NAD tradition in subtle but important areas. Particularly rewarding is its broader, more crisply defined sound at the frequency extremes, where earlier NAD amps often softened up just a little bit. As I listened to *Nojima Plays Lizst* [Reference Recordings], the greater extension and air that the Masters separates brought to this recording were striking. Harmonics seemed to radiate and rise into the soundspace without a ceiling hanging over them. There was a stronger bell-like quality to hard stabs of the keyboard, and greater fluidity to the lighter touch of Nojima's *arpeggios*. Importantly, the top-end and the midrange sounded seamlessly co-joined. There was no sense of treble information "kicking in" suddenly. This benefitted resolution in countless ways. Low-level details were conveyed with superior clarity, which enhanced my ability to locate instruments within the mix. The M12/M22 also provided a level of immediacy that further enlivened recordings. Familiar singers such as Holly Cole and Norah Jones were reproduced with an intimacy that made me feel like a fly on the wall of the recording studio. Perhaps because of the Masters Series' improved micro-dynamics, transient speed, distortion, or all three in combination, I found a jaunty playfulness in the 24-bit/96kHz version of Malcolm Arnold's *Sussex* Overture that I haven't always heard in the past. This well-known bon-bon, overflowing with orchestral humor, color, and contrast, needs a similarly acrobatic system to let it blossom.

While the NAD duo maintains a solid grip on the midrange, imaging and soundstaging have also firmed up. I've been listening a lot to tracks from the LP *VHS* from the alternative band X-Ambassadors (including the hit "Renegades"—an alt chart-topper). The quartet is led by lead singer, co-writer, and multi-instrumentalist Sam Harris. His brother Casey adds keyboards. Fronted by Harris' full-throated baritone, which can leap between a rich, chesty timbre and an angelic falsetto, the band serves up jousting polyrhythmic tracks with chant-like backgrounds. Soundspace plays a large role in many of these arrangements. Like the negative space in a photograph, the songs live within their ambient minimalism, with low-level contrasts and pauses followed by strong, explosive, colorful hooks and choruses.

The Masters Series' bass response was startling. It's bold and extended with an iron-fisted grip that not only sounded deep during Copland's *Fanfare for the Common Man* [Reference Recordings], but which also provided superior pitch definition and tunefulness. In a large measure, this system represents the maturity of Class D bass. Many will remember that even in its earlier iterations, Class D bass response, though its key strength, produced textures and timbre that were often homogenized and overly controlled. The NAD system allowed me to hear more of the dynamic contrasts and harmonic complexities that define the timbre of a bass drum or a tympani or a tom-tom. A prime example of M12/22 capability was the seemingly infinite amount of expression and color that five-string double-bassist Renaud Garcia-Fons [*Solo: The Marcevol Concert*] manages to snap, pluck, or bow from his instrument—from rattling percussiveness to soothing, lullaby-like fluidity.

So how does the NAD duo stack up in the

Equipment Report NAD Masters Series M12 and M22

company of a couple of integrated amps from the "old guard," such as the Pass Labs INT-250 (review forthcoming) or my current reference, the MBL C51? An amp like the Pass will reproduce ambient and reverberant space just a bit more vividly. It edges the M12/22 in the specificity of individual images, too—chorus members achieved an added level of focus, for example, during Rutter's "Praise Ye the Lord." The MBL, on the other hand, is all about the sweetness of upper-octave piano and violin, and the air riding above the harmonics. During Peter Gabriel's "Don't Give Up," it has superior resolution of the delicate percussion accents, the ride cymbal, the Kate Bush vocal (of course), and the bouncing bass patterns and synth pads. The NAD often matched the Pass and MBL in bass performance, particularly in grip and pitch definition. The Pass had the darker, weightier signature; the MBL was a bit warmer overall. But all three integrated amps graced the music with convincing timbre and bloom. In sum, the NAD held its head high in this exalted company. Very high.

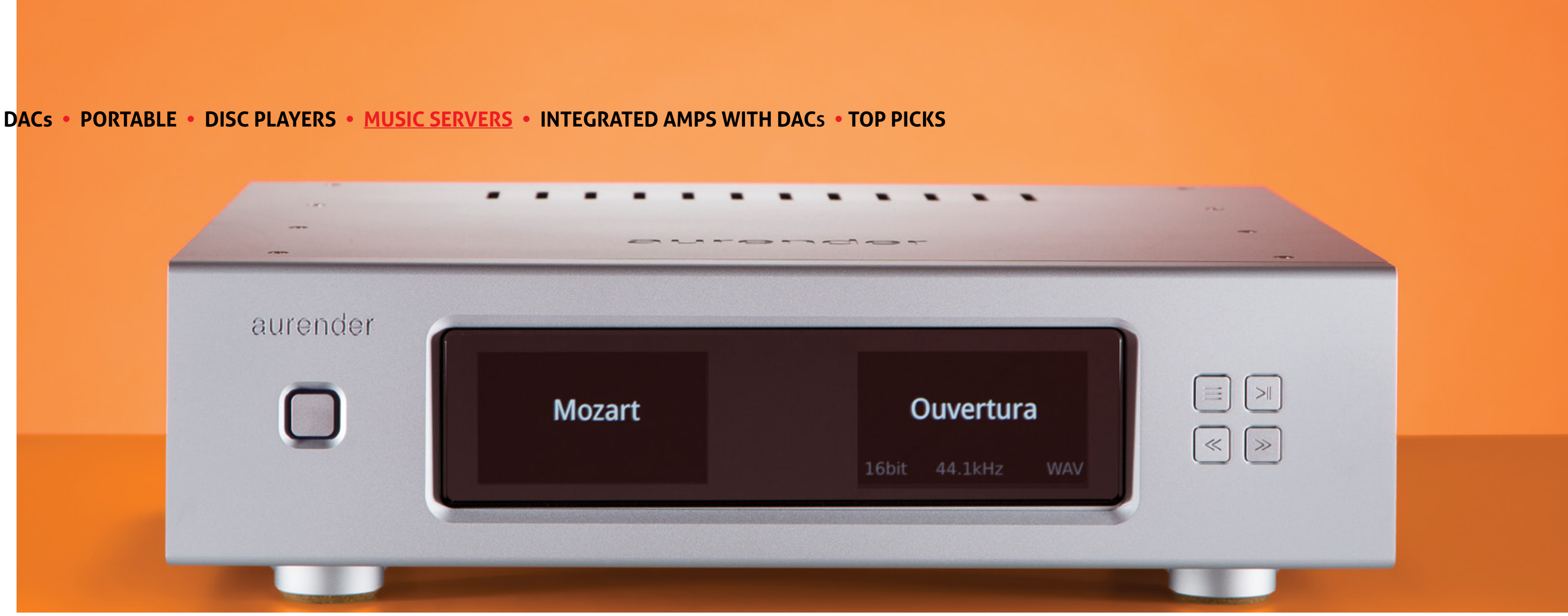
Once upon a time, NAD gear was thought to be great entry-level componentry but ultimately a stepping stone to something more fabulous and more refined from some other maker. Not so fast, and not this time. The NAD Masters Series M12/M22 combo proudly holds its own in pretty much any company. And it scores points at all levels—refinement of sound, classy execution, and cool, cutting-edge modularity that gives it attractive and sensible "have-it-your-way" appeal for the old guard and the network-savvy alike. Sensible, serious, masterful. tas

SPECS & PRICING

M12 Preamp/DAC
Inputs: SPDIF (x2), TosLink (x2), AES/EBU; analog RCA, analog XLR; phono and BluOS optional Outputs: Two digital, SPDIF/TosLink; analog RCA and XLR
Outputs: Two digital, SPDIF/TosLink; analog RCA and XLR
Dimensions: 17.1" x 5.25" x 15.1"
Weight: 32 lbs.
Price: $3499

M22 Stereo Power Amp
Power output: 250Wpc into 8 ohms
Inputs: RCA and XLR
Dimensions: 17.1" x 4.1" x 14.9"
Weight: 33 lbs.
Price: $2999

NAD ELECTRONICS INTL
633 Granite Court
Pickering, Ontario
Canada, L1W 3K1
(905) 831-6555
nadelectronics.com



Equipment Report

Aurender W20 Music Server

Not Just a Pretty Interface

Robert Harley

As I describe in *The Absolute Sound* Issue 258's From The Editor, beneath every music server's audio-component-like exterior lurks a computer. Do-it-yourself servers based on a Mac or Windows PC make no attempt to hide this fact from their users. But the *raison d'être* of "turnkey" music servers is to provide listeners with the benefits of file-based music without the hassles of computers. If you want instant access to thousands of albums with a couple of finger taps, but abhor the thought of drilling down through multiple sub-menus of arcane software settings, a turnkey server is for you. Our reviewer Steven Stone once wrote, only half jokingly, that DIY servers are for the ultra-geeky and turnkey servers are for the ultra-rich.

But there's another reason beyond convenience to buy a turnkey music server: sound quality. Building a server from the ground up allows the designer to incorporate techniques that optimize sonic performance—techniques that are unavailable in general-purpose computers. Most turnkey-server manufacturers, however, build their systems around a stock commodity-grade computer-motherboard to which they add custom digital outputs with more precise clocking, improved power supplies, and some measure of electrical isolation between the motherboard and the audio output. Although these are steps in the right direction, creating the state of the art in music servers requires designing and building an entire computer from a blank sheet of paper. This approach obviously requires a much greater investment of time and money, as well as considerable technical expertise.

That's what the Korean firm Aurender has done in creating the flagship W20 reviewed here. Nothing in the W20 is based on stock computer subsystems. Rather, every aspect of the W20's design is aimed solely at delivering state-of-the-art sound quality. As you'll see, the company has gone to extraordinary lengths in the pursuit of better sound.

The W20 is designed to do one thing and do it well: store music, allow you to access that music, and then present the highest possible quality digital-output signal to your DAC. The W20 has no integral DAC and no native CD ripping capability, and offers no metadata editing. The product's ambition is reflected in the substantial $17,600 price.

The W20 is housed in a handsome, robust chassis machined from aluminum plate with extruded aluminum heatsinks along the sides. The front panel houses two displays, a power button, and four buttons that provide rudimentary control over playback, as well as certain housekeeping functions. The display can switch between showing the name of the music track in play, the playlist menu, or signal-level meters (with a blue or brown background). In practice, you'll rarely interact with the W20 through these front-panel buttons and displays; instead, you'll use Aurender's Conductor app for the iPad to control the system. (More on this later.)

The rear panel showcases the W20's manifold capabilities. The two AES/EBU outputs can be configured either as two separate single-wire outputs or one dual-wire AES/EBU. This latter format is provided for those few DACs that require dual-wire inputs for accepting sampling frequencies above 96kHz. A clock input appears on a BNC jack, allowing the W20 to lock to DACs with a clock output, or to an external clock that sends a master clock to the W20 and a DAC. In addition to the two AES/EBU jacks, the W20 provides digital outputs via coaxial-on-RCA, coaxial-on-BNC, TosLink optical, and a dedicated audio USB connector. Two additional USB jacks are provided, but these are strictly data ports for connecting external drives. Finally, an Ethernet port connects the W20 to your network. I know of no other server with an array of features this extensive.

The W20 connects wirelessly to the iPad, but adding the server to your network and enabling Tidal streaming is best realized with an Ethernet connection. You can rip CDs directly to the W20's drives (by specifying those drives as the target in a ripping program such as XLD AutoRip) or download hi-res files directly to the W20. A better method is to rip CDs and download hi-res files to a network attached storage drive (NAS) that's on the same network as your PC or Mac and the W20. You then drag and drop music files from the NAS to the W20's disk drives. It's a bit of a hassle to go through this procedure

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if you just want to listen to a single, recently purchased CD. Transferring music to the W20 is best done in batches. (Just as I was finishing this review, Aurender announced a software update, available by the time you read this, that allows you to transfer music from the NAS to the W20 directly without a PC or Mac via the Conductor app for the iPad, but the update wasn't ready in time for me to try it.)

Adding a NAS should be considered mandatory because it provides a backup of your music library. Note that although the W20 has dual disk drives (6TB x 2), the second drive isn't a redundant backup; if a drive fails you'll need to reconstruct as much of your library as was on that drive. The Synology DS214 NAS I use ($557 with two 3TB drives) comes with software that performs automatic backup of any other drive on the network, including the W20's drives. A single NAS, however, shouldn't be your only backup. For true security, you should have a second NAS stored in a remote location that is periodically backed up. This may seem like an extreme measure, but not when you consider how much time, effort, and money your stored files represent, particularly if you've edited the metadata.

Technical Tour

Removing the thick, heavy aluminum top plate reveals a chassis compartmentalized into several aluminum sub-chassis. The fanless switching-mode that powers the computer motherboard is encased in an isolated block just behind the display. The dual 6TB disk drives are mounted on compliant platforms to reduce noise and vibration. I never heard the sound of drives spinning during my entire time with the W20. The audio-output board is separated from the other circuitry by an aluminum plate. The critical audio-output circuits are powered by two of the three separate banks of lithium-ion-phosphate rechargeable batteries that consume a big chunk of the interior real estate. By powering the audio-output electronics with batteries, the digital audio signal is made completely immune to power-supply noise or fluctuation. The two banks are redundant: One set is being charged as the other is being used. The batteries also protect the computer from sudden loss of power; when the W20 detects that the AC power has been disconnected, it safely powers down the system, protecting the stored data.

Another design feature aimed at delivering a pristine digital output to your DAC is a 240GB cache memory, which serves as a buffer between the disk drives and the audio output. When you select music and create a playlist, the W20 reads the audio data from the spinning disk drive into this cache memory, after which the disk drive is spun down to sleep, eliminating noise and vibration. This also minimizes wear and tear on the hard drives. The audio data are then clocked out of the cache with a high-precision, oven-controlled crystal oscillator. An oven-controlled crystal oscillator is encased in a small heated chamber that maintains a precise and optimal temperature for the crystal. These expensive devices are much more precise than the ubiquitous crystal oscillators found in virtually all digital products. Both the clocking circuit and the cache memory feature proprietary techniques for reducing noise and jitter on the output signal feeding your DAC.

Unlike many computer-audio products, the W20 comes with an excellent and well-illustrated "Quick Start Guide." A full owner's manual is offered on the Aurender website. Should you encounter problems with any current Aurender model server, you can request Remote Internet Support right from the app. An Aurender technician can then access your network and probably diagnose and resolve any problems with the W20 or its setup.

The Conductor App

A server lives or dies by its control app. The app can be a constant source of frustration or a joy to live with on a daily basis. I'm happy to report that the Aurender Conductor app is by far the best I've used. It's fast, visually appealing, stable, intuitive, capable, and uncluttered, and its features have been clearly refined through actual use. The app runs much faster on a 64-bit iPad (I tried it on an older iPad 2 as well as a new iPad Air 2.) The 64-bit iPads are required for rendering album art in high resolution. If you spend ten minutes with someone who knows the app (your dealer, for example) and then begin using it yourself, you'll feel like an expert half an hour later.

The majority of the app screen shows your library by album, artist, or song, with a smaller playlist section on the left. Tapping a track from the main display moves the track to the playlist. Entire albums can be moved to the playlist with one tap. A nice feature allows you to slide back the main display to show the playlist in greater detail. Another thoughtful design element is the way tapping a button brings up the additional controls you need in the context associated with that button. For example, I just mentioned that you can add an entire album with one tap. When you tap the album name, a menu appears that offers you the option of adding the entire album, and where in the playlist to do so. This structure keeps the interface clean and simple, presenting you with additional choices only when you need them. Moreover, the interface's colors, shapes, and organization are easy on the eyes. You can filter your library view by sample rate (showing you only hi-res titles, for example), DSD files, recently added titles, and those albums you've marked as favorites.

With two finger-taps the view switches from your music library to the Tidal streaming service. (A Tidal subscription is required: $19.95 per

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Storage capacity: 12TB (6TB x 2)
Formats supported: DSD (DSF, DFF), WAV, FLAC, AIFF, ALAC, M4A, APE, and others
Outputs: AES/EBU (x 2, single-wire or dual-wire mode), USB 2.0 (dedicated audio output), USB data ports (x 2), TosLink optical, Ethernet, coaxial (RCA), coaxial (BNC)
Inputs: Clock on BNC
Network: Ethernet
Dimensions: 16.93" x 4.17" x 14.57"
Weight: 41.9 lbs
Price: $17,600

AURENDER (Division of TVLOGIC AMERICA CO. LTD.)
209 N. Victory Blvd.
Burbank, CA 91502
aurender.com

month for unlimited lossless streaming.) You can create playlists with tracks mixed from your library and Tidal. Its integration is so seamless that it's easy to forget where your library ends and Tidal starts. The Aurender Conductor app's Tidal interface is better than Tidal's own app. The best software is powerful yet simple to use, and that is a good way to describe the Aurender Conductor.

As I mentioned, the W20 offers no way to edit metadata directly. You can, however, edit metadata with a program such as JRiver Media Center. Speaking of metadata errors, I discovered a couple of funny and interesting ones. I ripped a CD by the Western swing band Asleep at the Wheel and the band's name showed up in my library as "A Sheep at the Wheel." The double CD of John Mayall's 70th birthday concert appeared as two separate albums, one by John Mayall and one by John Mayall & the Bluesbreakers.

Setup

Setting up the W20 in my system wasn't without glitches. After using the system for a couple of weeks I powered it down to rearrange my equipment rack, and when I powered it back up the W20 wouldn't connect to my iPad Air 2. Oddly, it *would* connect to an older iPad 2. Aurender had not encountered this issue before, but I figured out the solution. (The W20 and iPad Air 2 weren't on the same network; resetting the router fixed the problem.)

On another occasion, after the W20 was turned back on, it wouldn't boot up. Previously unbeknownst to me (or to Aurender), the W20 won't boot up when certain DACs are connected to it. (I was using the DAC in the Hegel H160 at the time.) Aurender had not seen this problem with any other DACs.

I should add, however, how wonderful it was to connect different DACs to the W20 and have them instantly recognized, with their names shown in the W20's display. Anyone with a PC- or Mac-based server who has struggled to get his software to recognize the DAC will appreciate the W20's ease and reliability in this regard.

Listening

Does the W20's $17,600 price tag buy you merely the convenience of a turnkey server and a nice interface, or does it sound considerably better than, say, a fully tricked-out MacBook Pro? (There's no question that the user experience is vastly better with the Aurender than with the Mac-based server. In fact, the comparison's not even close.)

To answer that question I first assessed the W20's sound quality by auditioning its various digital outputs to find the best interface. I found that the best-sounding configuration was with the W20's USB output driving a Berkeley Alpha USB (a USB-to-AES/EBU or SPDIF converter), which in turn fed the Berkeley Alpha DAC Reference via a 1m run of AudioQuest Wild Digital AES/EBU cable.

The W20 was put under the extremely powerful microscope of the state-of-the-art in digital conversion, the Berkeley Alpha DAC Reference connected to some of the most transparent and resolving electronics extant—the Constellation Audio Altair 2 and Hercules 2, or the Soulution 725 and 701 combos. These, in turn, drove the Magico Q7 Mk2 and MartinLogan Neolith loudspeakers, all connected with MIT's best cables. The listening room's AC power, supplied via four dedicated 20-amp AC circuits, was conditioned by an all-out Shunyata system with the new Shunyata Sigma AC cords. This system's resolution immediately revealed exactly what was happening at the digital source. (I'm reviewing the new Constellation electronics and the Neoliths in the next issue. Shortly thereafter, I'll write a feature article on building this entire system and what I learned along the way.)

Listening to the W20 on a daily basis, after living with a MacBook Pro as a server for the past year, I was immediately aware that Aurender's extraordinary efforts in clocking, buffering, and lowering noise paid off in the musical experience. The W20's "sound" was characterized by a natural and organic quality that came closer to the "feel" of analog than any digital source I've experienced. The presentation had a dimensionality, life, bloom, and illumination that one doesn't associate with digital. I was repeatedly amazed by just how much space and depth were encoded on 44.1kHz/16-bit sources, just waiting to be revealed by playback hardware of this quality. I thought that we had long ago bumped up against the limits of standard-resolution digital sources, but the W20 feeding the Berkeley Alpha Reference DAC showed that the flatness, hard timbres, lack of air and depth, and absence of fine detail were not purely attributable to the standard-resolution digital format. Of course, there are many inferior-sounding CDs, but the W20 still managed to get the most music out of them. The W20 not only revealed new depth and dimensionality on well-recorded CDs I had ripped (in AIFF), but also rendered instruments as separate objects in the mix. The W20 "de-homogenized" the soundstage, allowing me to hear each instrumental line with startling clarity and focus. Reverberation decay was longer and deeper, adding to the impression of space and dimensionality. The recording *Live in America* by flamenco guitarist Paco de Lucia was a particularly vivid example: Paco's guitar was focused in the center of the stage, surrounded by the hall's dense reverberation, with the thrilling *zapeteo* (percussive footwork) and handclaps at the far left and right boundaries "lighting up" the acoustic space with each sharp transient. I've listened to this track many times, but never before felt I was hearing the recorded acoustic this clearly. The experience was mesmerizing.

In addition to greatly increased dimensionality, another salient characteristic of the W20 was its very quiet background. It was as though the W20 cleaned up a bit of low-level hash that was diminishing the impression of hearing instruments in space. Presented against a dead-silent backdrop, instruments took on more palpability, realism, and life. This low-level hash had also set a noise floor below which no information was being resolved. The Aurender's deeper silence allowed very fine details of timbre, micro-dynamics, and ambience to emerge. The W20 was so adept at resolving the lowest levels of information that I consistently heard new musical nuances on albums I'd been listening to for decades. Treble through the W20 was cleaner and purer, with less grit, hardness, and unnatural sheen. The top end had greater

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delicacy, air, and detail—qualities that were rewarding on Jack DeJohnette's wonderful cymbal work on the live Keith Jarrett recording *My Foolish Heart*. Cymbals seemed to float in air rather than being painted on a flat canvas; their airiness and decay approached that of analog with some recordings.

There was a sense of precision and order that made the music tight, defined, detailed, and dynamic. Bass was tauter and more tuneful, with greater pitch articulation and dynamic impact. This precision was particularly impressive on 44.1kHz/16-bit sources. Although the best hi-res material sounded spectacular, what impressed me even more was how the W20 reproduced standard-res material day after day. It was as though the Aurender had remastered my entire CD library, giving me new and improved versions of old favorites. In some respects, realizing great sound from standard-resolution sources is a greater technical challenge than performing the same trick with hi-res ones; on Red Book CD, the digital source and DAC have much less information to work with, leaving no room for error.

The W20 was significantly better sounding than the MacBook, even though both sources' USB outputs were being buffered, isolated, and reclocked by the same Berkeley Alpha USB. (I suspect that the difference in sound quality between the MacBook and the W20 would be even greater if the latter were driving DACs directly, without the Berkeley Alpha USB interface.) It did not take hours of back-and-forth comparisons to hear the W20's superiority. I started by listening to "Back Row Politics" from *Act Your Age* by Gordon Goodwin's Big Phat Band, first on the W20 and then through the MacBook Pro. The tune starts with few bars of piano intro. Switching to the MacBook Pro was almost like hearing a different piece of music. Through the MacBook the left- and right-hand piano lines were blurred into a single musical statement. Through the W20, the two lines were clearly distinct, and much more musically involving, the interesting meter generated by the left and right hands setting the stage for the rest of the tune. When the band came in, I heard a much tighter and deeper bottom end, a more open and spacious soundstage, and far more detail. Small percussive details smeared by the MacBook were rendered with pristine clarity by the W20. I had been listening to the MacBook Pro for about a year, and was surprised by just how much better my system sounded with the W20 as the source.

Conclusion

The Aurender W20 is in my experience the current state of the art in music servers. It excels in every parameter; its array of features is unmatched, the 12TB of available storage will accommodate virtually any library; its interface is wonderful and intuitive; and most importantly, it delivers sound quality unmatched by any other digital source I've heard. The W20 brought out the best in my system, delivering the greatest dimensionality, timbral purity, resolution, and freedom from hash I've heard from digital sources. Of course, a great digital-to-analog converter is required for realizing the sound quality I've described, but I can say that the combination of the W20 and the Berkeley Alpha DAC Reference sets a benchmark in performance.

Although many listeners will be drawn to the Aurender W20 by its capabilities and outstanding iPad control app, it's really the sound quality that makes the W20 special. The Aurender W20 is not just a pretty interface. tas

N100H: A More Affordable Alternative

I'm happy to report that much of the Aurender experience is available in Aurender's far less expensive N100H server, priced at $2695 with 2TB of integral storage. You don't get features such as dual-wire AES and the clock input, but most users don't need those capabilities anyway. Although the audio output circuitry is powered by a conventional power supply rather than by batteries, the circuit is based on the design developed for the W20. The N100H's much smaller chassis provides less isolation between subsections than that of the W20, but the N100H's chassis-work is still first-rate. It's like a miniature version of the W20. The N100H's cache memory is 120GB rather than the W20's 240GB. Nonetheless, you still get the same outstanding Conductor app, Tidal integration, and Remote Support.

I was able to compare the W20 with the N100H by creating playlists of the same music on both servers, and then switching between them by selecting the server I wanted to hear from the Conductor app (and moving the USB cable). I also compared the two Aurenders to the MacBook Pro described in the review.

The N100H was much closer in sound to the W20 than it was to the MacBook Pro, which came in a distant third. The N100H had much of the W20's expansive soundstage, dimensionality, purity of timbre, separation of individual instrumental lines, bass definition and dynamics, and resolution of low-level detail. In absolute terms, the N100H was not quite as clean in the treble as the W20, slightly less resolving of low-level information, and not as dimensional. I should stress that the differences between the W20 and the N100H were ones of degree, not of fundamental character. The N100H was significantly superior to the MacBook Pro in every sonic criterion, and inferior in none. I suspect that the N100H would be a major sonic upgrade to just about any digital front end. And the user experience is absolutely identical. The N100H strikes me as a compelling solution for many listeners—and a terrific value. tas

Equipment Report

Channel D Pure Music Software

Pure Heaven

Steven Stone

In Issue 202 I concluded my review of the Amarra software program with, "If you want to hear how good a quality Mac-based system can really sound, you *have* to use Amarra. In the end, it's that simple." Time and the latest version of Channel D's Pure Music software may make me eat those words. Priced at only $129, Pure Music promises to improve not only iTunes' sonics, but also adds high-resolution capabilities along with a host of other advanced sonic and ergonomic features.

Pure Music is such a powerful program that reading its "User Guide" is a must. I daresay that you will be reading this informative tome more than once. I recommend keeping Pure Music's User Guide PDF open on your desktop for the first week or so of operation, especially during initial setup. While nothing in Pure Music's preference panels is completely inscrutable, without the User Guide anyone not familiar with Pure Music's many options could screw up its settings in a myriad of ways. Don't say I didn't warn you.

Pure Features

Like Amarra, Pure Music's principal function is to bypass iTunes' signal processing and substitute a more direct and powerful 64-bit processing program. In addition Pure Music offers automatic rate-switching from 44.1/16 all the way up to 192/24, gapless playback of files that have been designated as gapless files, memory play, real-time high-resolution upsampling of CD tracks, a 64-bit internal signal path, dithered volume control, phase inversion, a subwoofer crossover, multichannel support, support for audio-processing plug-ins, Core Audio HOG mode playback, high-resolution audio streaming, precision signal metering, reverse play, and more. Some of these features, such as HOG mode and memory play, may sound like gibberish to the uninitiated, but these two features alone make Pure Music capable of elevating even a lowly Mac Mini into a formable music-delivery device.

I could easily fill many pages with a detailed description of individual preference panes and the various options these panes offer, but you can download the User Guide along with a demo version of the software from Channel D's Web site.The free demo offers 15 days of full-featured usage, and I daresay that once you've used Pure Music going back to ordinary ol' iTunes will be tough, unless you're listening through a Dixie cup.

Although a novice user, the sort of person who feels intimidated by anything labeled "preferences," can simply download and run Pure Music, to hear its full potential does require optimizing it for your particular system's capabilities. But even when it is used "plain vanilla" without any system optimization, I could hear differences between iTunes and Pure Music.

Among Pure Music's "must use" features is memory play. This loads your music file's stream into an adjustable RAM buffer before it's sent to your rendering device or DAC. It usually takes a few seconds for the buffer to fill and music to begin playing, but you can select a "Hybrid buffer" setting which will play the first

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couple of seconds of a track without buffering while the data is loaded into the buffer and then automatically switches to buffered mode once the buffer is filled.

Pure Music's upsampling capabilities allow it to turn a 44.1/16-bit file into a higher-res file. Among the options are "power-of-two" upsampling. According to Pure Music's User Guide, "this operation is more efficient than factored upsampling, and in the case of Red Book CD, 88.2kHz is, all things considered, a better target than 96kHz." If your DAC will support it, a Red Book CD can be upsampled all the way to 192kHz. With the Weiss DAC 202 I was able to set up Pure Music so it upsampled 44.1/16 files to 192/24 before sending them to the DAC.

Another unique feature of Pure Music is the HOG mode. According to the User Guide, "this option reserves the audio device for Pure Music's exclusive use while Pure Music is running. To use this feature, the audio device selected in Audio MIDI Setup should be set to a different device than the one used by Pure Music, to allow iTunes to fully access an audio device if necessary. Accordingly, by default, HOG Mode cannot be used for the audio device selected in Audio MIDI setup." This feature is best used on a dedicated music system. On a full-service computer it means that any time you want to use any program that requires an audio stream it will have to go to an alternative audio device, such as your internal speakers or a second DAC.

My final preferred HOG setup was pretty clever, if I do say so myself: I used the Weiss DAC 202 in FireWire mode for my Pure Music feed and the Empirical Audio Off-Ramp 3 for all other audio tasks. To change from Pure Music to other audio sources I only needed to select the DAC 202's RCA-S/PDIF input.

Pure Music also allows the use of third-party plug-ins, and comes with 18 plug-ins already installed and waiting for activation. A plug-in is a small application program that runs within Pure Music. Installed plug-ins include a peak limiter, graphic EQ, high-pass and low-pass filters, compressors, reverb, and shelf filters. My favorite plug-in is the Roger Nichols Digital Inspector, which shows clip incidents, consecutive clips, overall headroom, and master levels in real time for any music file being played through Pure Music. (Digital Inspector isn't included in Pure Music.) Since each plug-in takes up processor time, Pure Music monitors the total CPU load so that you don't overload your computer by using too many plug-ins all at once. On my Mac Pro with 12 gigs of memory I was able to run quite a few plug-ins simultaneously. But the best way to use plug-ins is with restraint. You can, if you're so inclined, use up to 14 plug-ins at the same time, but that would be a wee bit excessive.

With the right hardware you can even have Pure Music handle crossover settings for a multi-amped speaker system. To utilize this feature you will need a multichannel output device such as a Lynx AES-16 or Apogee Ensemble. Each channel can be selected and modified by Pure Music. For a two-way speaker system, channel one could be right tweeter, channel two the right woofer, channel three the left tweeter, and channel four the left woofer. You can choose either 6-, 12-, 18-, or 24dB-per-octave slopes for both the high pass and low pass. You can also adjust individual levels for each channel and the delay for each channel, making this a very powerful and flexible way to configure your crossovers.

While earlier versions of Pure Music had some small ergonomic quirks such as reading out "Paused" while it was playing, the current version, 1.6.3, proved to be exceedingly well-behaved. The only problem I experienced was with the Wyred4Sound DAC 2. During the silences between cuts I heard low-level crackling. Since this DAC uses its own proprietary driver, I suspect that was the culprit. I alerted Wyred4Sound of the problem and they added it to their bug-fix list for the next version of the driver.

One ergonomic issue I was glad to see Pure Music doesn't have is Amarra's death-grip on the computer's CD/DVD drive. If you rip a CD via your internal ROM drive while Amarra is running it won't let you eject the disc. You have to shut down Amarra (which shuts down iTunes) before you can remove the disc from your drive. That gets old pretty fast.

Pure Sonics

How does Pure Music sound? Better than iTunes alone, that's for sure. Compared to iTunes Pure Music is more dimensional, dynamic, detailed, and involving. ITunes sounds flat, not pitch-wise, but in its overall presentation. It is like going from a 128kbs MP3 file to a 320kbs file. Pure Music delivered substantially more musicality and more information than iTunes did.

I found I got the best sound from Pure Music when I used both memory play and HOG mode. This combination delivered a subtle improvement in both overall soundstage depth and dimensionality. The spaces around and behind individual instruments were better defined. The amount of improvement will vary depending on your particular hardware configuration. Although I heard the improvements through the Wyred4Sound DAC 2, the improvement was more pronounced through the Weiss DAC 202.

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Hardware platform: Apple Macintosh OS 10.5 or later with iTunes
Price: $129 (free 15-day trial with all features available)

CHANNEL D SOFTWARE
(609) 393-3600 (live support available 9-5 EST)
channld.com

ASSOCIATED EQUIPMENT
Source device: MacPro model 1.1 Intel Xeon 2.66 GHz computer with 12 GB of memory with OS 10.6.4, running iTunes 10.0.1 and Amarra 1.2 music playing software
DAC: Weiss DAC 202, Empirical Audio Off-Ramp 3
Preamp: none
Amplifier: Bel Canto S-300 stereo amplifier, Edge Electronics AV-6, Accuphase P-300 power amplifier
Speakers: ATC SCM7s, Paradigm S1s, Aerial Acoustics 5Bs, Role Audio Kayaks, Earthquake Supernova mk IV 10 subwoofer
Cables and accessories: Locus Design Polestar USB cable, Locus Design Nucleus USB cable, PS Audio Quintet, AudioQuest CV 4.2 speaker cable, AudioQuest Colorado interconnect, Empirical Audio Coax digital cable

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Naturally, I compared Pure Music with Amarra. Unfortunately, because you must shut down each program *and* iTunes when you switch from one program to the other, I couldn't do the kind of direct real-time A/B tests that I usually employ. On the longer, slower A/B comparisons I couldn't hear any differences between Pure Music and Amarra. Both were clearly better than iTunes, a fact I could easily ascertain via matched-level instant A/B comparisons.

Given that I found sonic differences between Pure Music and Amarra negligible, and Pure Music costs approximately 25% of Amarra's price, does that make Amarra obsolete? For budget-conscious audiophiles the answer is yes, but for those who are using one of the professional DACs that Amarra supports, Amarra still may be a better option. Also, given Sonic Studio's rapid rate of innovation, it's possible that by the time this review sees print Amarra may have seen improvements of its own.

I'm sure many readers would like to know how a Pure Music-enabled Mac system stacks up against a top-flight transport. Sorry, but you won't find any answers here. To be completely forthright, I don't listen to CDs through CD players or transports anymore. For me a CD is merely a way to get digital files. When I receive a new CD, I "play" it exactly once, when I add it to my digital library. Then it goes onto a shelf to collect dust. Transports are as useful in my world as a capo on a mandolin.

Pure Pleasure

Pure Music is a great piece of software at a price that even a flea-market-scrounging audiophile hobbyist can afford. Combine Pure Music with any recent Mac computer and you have a front end that will play back any digital file (except FLAC) from lowly MP3s up to 192/24 high-resolution with ease. Mate this front end with a top-flight DAC such as the Weiss DAC 202 and you have a digital playback system that will catapult you to the forefront of the new computer-playback revolution. Dare I say it? If you want to hear how good a quality Mac-based system can really sound, you *have* to use Pure Music, at least for now. tas

INTEGRATED AMPS WITH DACs

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Reinventing a Classic

Neil Gader

In NAD lore, "3020" are hallowed numerals. The long-ago integrated amplifier that bore that designation might have been a barebones affair, but it marked a departure from the budget norm when it first debuted in 1980. Built solidly, without extraneous signal-robbing bells and whistles, the 3020 offered musical truth in its tonal balance, lack of coloration, and dynamism in spite of its conservative 20Wpc specification. Music lovers responded *en masse*; more than one million 3020s have been sold—an astounding number for a high-end product.

Now, the 3020 is back with a "D" prefix for clarification. A capital "D." As compared with the all-analog original, the new D 3020 is a digital animal designed primarily for computer/USB sources. Power output is a solid 30Wpc thanks to NAD's ultra-compact Class D topology. True to NAD tradition the amp's power rating is deceptive in that it can output bursts up to 100W (into 4 ohms) during dynamic peaks. In digital connectivity, it offers aptX Bluetooth music streaming—an efficient alternative to Wi-Fi—plus a USB input that plays back computer-based music in up to 24-bit/96kHz resolution, and operates in asynchronous mode to ensure low jitter.

Nothing can prepare you for just how compact the D 3020 is when you first encounter it up close and personal. Truly a design for our times, it's improbably small and portable with a vertical form factor that lends it the visual profile of a network router. And I hasten to add, portable enough to be drafted into service as a headphone amp. Note that where space requires, it can also be positioned horizontally.

A top-panel touch control powers the D 3020 on, and the vertical front panel of inputs and volume indicators blinks to life for a few seconds. The gradations of the large volume control are indicated in 20dB numerical steps, the display fading or intensifying as the user makes changes. The look is nifty but I didn't get much of a sense of precision as I navigated up and down—only a rough idea of where the volume was actually set.

The back panel hosts a trio of digital inputs which includes USB, SPDIF, and TosLink plus a subwoofer output and a single, lonely analog input. Additionally there's a bass-equalization toggle and a multi-purpose auxiliary input that can be used either as a headphone jack from a MacBook Pro or, with the supplied TosLink mini-adapter, as an extra optical input. In a nod toward energy efficiency, when the amp doesn't sense a signal for about fifteen minutes it reverts back to a 0.5W standby mode.

Operationally I've only got a couple of nitpicks. The lack of a mute button seems a weird oversight. Also the iPod-style IR remote is all flat-black, including the navigation buttons. The only way to see what you're doing is to angle the remote so that it catches a glint of light to illuminate the markers. Most of us will memorize the six key buttons (on/off, volume +/-, and source select arrows), but *really*!

Sourcing my hard-drive-based music collection via USB was a snap; however, I was more impressed by how easy it was to get Bluetooth (BT) up and running—an area where I've occasionally run into snags in the past. Here, I simply selected Bluetooth from my Mac's System Preferences and made certain BT sharing was selected within the Sharing submenu. This made the D 3020 discoverable as a device. A simple click to connect and, after opening iTunes, I was instantly listening to one of my own "stations" on iTunes Radio. While the sonics of Bluetooth are more geared to convenience than to our inner audio connoisseur, I'd be lying if I didn't admit that it sounded darn good—not as open and dynamically sophisticated as the high-res USB connection but far better than I remembered from previous BT experiences.

Speaking of sonic performance, the D 3020 for all its humble appearance is pure NAD. It's firmly midrange-centered in its balance and never over-reaches in the sense of growing shrill on top or tubby on the bottom. Yes, it's lighter in overall sound due to some bottom-octave attenuation, but the D3020 retains an essential *presence*, a midrange integrity, that sculpts the body of a performance and makes it live in the listening space. It also maintains a solid grip in the midbass, resolving Lee Sklar's mellow bass lines with good pace and precision during James Taylor's "Fire and Rain" [Warner]. Its response softens and loses definition only

Equipment Report NAD D 3020

SPECS & PRICING

Power output: 30Wpc into 8 ohms
Inputs: Three digital (USB, SPDIF, TosLink); one analog
Dimensions: 2.3" x 7.5" x 8.7"
Weight: 4.6 lbs.
Price: $499

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slightly when confronted with hard-charging electric bass pulses or the double-kick-drum rhythm figures flying off the feet of Metallica's Lars Ulrich.

Vocals tended to sound a bit dry at times, an issue that affected female singers a little more than male ones. But multiple vocal images were generally very good. For example, during Jackson Browne's "Colors of the Sun" [Asylum] the D 3020 reproduced a significant amount of the detail and interplay between the vocals of Browne and Don Henley.

While the specs and form factor of the D3020 suggest that it is ideally suited for desktop duty, I wanted to throw a wrench in the gears by giving the NAD a real shake-down with a highly esteemed compact loudspeaker, the Franco Serblin Accordo, a two-way compact of impeccable craftsmanship and provenance, and one of the last speakers authored by Serblin, who passed on in 2013. At 87dB the Accordo's a medium-sensitivity loudspeaker with midrange and top-end response that are truly world-class. The D3020 never hiccupped at the challenge.

One of the liveliest recordings I have is the electrifying Jacques Loussier Trio playing *The Best of Play Bach*—a smile-inducing collection of jazz/classical bon-bons. The D 3020 handled the dynamics and harmonic and ambient density of this recording quite faithfully. There was some dynamic constriction and low-frequency pitch instability at moments, but overall performance from a sub-$1k 30Wpc amp has rarely been more impressive. And I admired the grip of this amp once again when confronted with the midbass tom-toms during Blood, Sweat & Tears' "More and More" [Columbia]. Though piano timbre during "Sometimes in Winter" was a little cool, there was still a suggestion of the felt on the hammers damping the strings.

Perhaps the biggest surprise I encountered during my listening sessions was the quality and smoothness of the amp's top end. This was a region where the Accordo tweeter would easily expose deficiencies, but the D 3020 met the challenge. As I listened to pianist Janne Mertanen play the Chopin *Nocturnes* [Alba], transient speed and harmonic openness were truly enthralling. Although there was a little bit of a ceiling over the performance—at least compared with pricier, wider-band amps that operate with more dynamic headroom—the D 3020 had little else to apologize for.

Although I'm an infrequent headphone user, whenever I don my AKG K501 cans (still terrific after all these years) I am always impressed by the gorgeous midrange tonality and intimacy these 120-ohm 'phones produce.

As a headphone amp, the D 3020 does its job noiselessly and is musically satisfying. The tonal characteristics that make it so appealing with conventional loudspeakers translate fully to the more intimate world of earspeakers. Frankly I haven't ever appreciated headphone listening as much I did during the time I spent with the D 3020.

If computer audio is your primary source for music, and Blue Tooth capability is a must, then the D 3020 makes a compelling argument. The other argument is, hello, its price tag of $499, making it by most standards a small miracle of packaging and portability, and with few exceptions a delight to use and listen to. *Too small for you?* NAD has you covered with a bigger cousin in the new D 7050—a streaming integrated with more power, advanced topologies, plus AirPlay wireless at $999. For many, however, the D 3020 will be just what the digital doctor ordered. Faithful to the original 3020 but totally dialed in to our times. tas

Equipment Report

Micromega MyAmp DAC/ Integrated Amplifier

Small Footprint, Big Performance

Neil Gader

Some clichés in the high-end die hard. One of the oldest chestnuts says that size matters. It's the notion that one's status as an audiophile is somehow tied to the weight and girth of your components, your sagging equipment racks, and the thick ropes of cabling that feed each product. There was a time when I was guilty of falling for this nonsense, as well. After all, as an audio writer I've gotten pretty used to receiving some pretty intimidating components. We all know them, and in some sense are still seduced by their presence—those big amps and preamps, glowering, un-liftable hunks of metal laden with aggressive displays of exposed heat sinks sharp enough to shave truffles.

However, a funny thing has happened. Two funny things actually. First, computer-audio playback has revolutionized high-resolution listening, from the living room to the desktop. This has coincided with a renaissance in "personal" listening—that is, *headphones*, in-ear, over-the-ear, closed-back, or open-back, take your pick. The upshot is that the "bigger-is-better" cliché has been unceremoniously turned on its, well… ear. Today it's hip to be small. It's relevant and credible and high-res. In fact a tiny footprint has almost become a mantra, particularly among younger audiophiles.

Micromega has been in on this trend for some time now. The French company has been a purveyor of full-scale electronics and streamers as well as the "My Range" of modest mighty-mites like the MyGroov and MyZic and TAS' 2012 Product Of The Year, the MyDac. Inevitably, an amp would appear to fill the void, so please welcome MyAmp. More than an integrated amplifier, MyAmp is a complete digital hub with wireless streaming, analog and digital source switching, and a headphone amplifier. Impossibly little, it's the teacup poodle of DAC/integrated amps. At a mere 5.5-inch square it also leaves plenty of room on the desk for a nice pair of speakers. I can literally palm it and fit it in our Volvo's glovebox. Try that with your Soulution 701. MyAmp is enclosed in an all-business, textured ABS casing—translation, plastic. What? You were expecting the CNC-machined aluminum of a Rowland? Calm down. Micromega chose ABS for its non-conductive properties and lack of eddy currents.

More important are features like the healthy output—30Wpc into 8 ohms, which commendably doubles into 4 ohms. The unit's small size suggests that the amplifier is based on Class D switching modules, but the amp is actually a Class AB design. The efficiency comes from a newly devised and highly unusual "LLC" power supply that reportedly delivers more power, tighter regulation, and a lower impedance than a conventional supply. Another factor in the unit's small size is the unusual forced-convection cooling system in which the power supply and amplifier output stage are cooled with a magnetic-levitation fan (no bearings) moving air through a tunnel. A thermal protection system shuts down the unit if it overheats, and also continually adjusts the fan speed.

The back-to-basics front panel houses a bevy of teeny buttons for source selection, plus a headphone mini-plug socket. Volume is indicated by a red-lit ladder display. Micromega states that the control is good to 256 steps in 0.5dB increments, but the indicator is so vague that it's virtually impossible to make precision, repeatable adjustments. Numerals would have been better. The DAC is the ESS Sabre Hyperstream DAC, the same chip found in many expensive units. Source switching is via FET-buffered relays—impressive in a $649 product. The coaxial digital input is transformer coupled, and the USB input employs an isolation circuit to keep the computer's noise out of MyAmp.

Jam-packed is the only way to describe the back panel. It hosts three analog inputs plus three digital, a 96kHz/24-bit USB, and 192kHz/24-bit optical and coaxial/SPDIF inputs. There's also a direct analog output, a sub output, and full-sized multiway speaker posts. The three-pin 10-amp cord is removable. The MyAmp streams conventional audio via the hugely popular Bluetooth aptX module, an efficient and more user-friendly (I've found) alternative to WiFi. This particular codec is also popular because it minimizes latency while improving bandwidth. Throw distance is always a consideration with Bluetooth, and depending on your home you can't really figure much more than twenty to thirty feet from the transmitting smart device. Pairing Bluetooth devices with the MyAmp was a breeze, and up to eight devices could join up. However, remember

Equipment Report Micromega MyAmp DAC/Integrated Amplifier

SPECS & PRICING

Power Output: 30Wpc into 8 ohms
Inputs: Three analog, three digital
Dimensions: 5.5" x 5.5" x 3"
Price: $649

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that when streaming from a device like an iPad/iPhone you'll need to disable any audio e-mail and push notifications as these bleeps, burps, and buzzes will temporarily mute the volume of the music. Of course, you can also connect a USB cable between your computer and MyAmp. Overall, MyAmp is designed to be an affordable, single-box solution for music lovers on a budget.

The MyAmp exemplifies what high-end audio should be about—solid sonics blended with flexibility and adaptation. It can spend the day in the desktop environment, and then just as easily be reassigned for small system duties in a den or cozy family room. In fact, in my small room, the MyAmp handily drove the expressive Epos K1 loudspeakers, the foot-tall two-way reflex compacts I reviewed in Issue 148. Although the Epos is nominally a 4-ohm speaker that's rated at 88dB sensitivity, the Micromega seems to have plenty of power to drive it, even when subjected to the kind of evil hijinks that I put every review sample through. It impressively preserved the key strengths of the K1 including its open full-throated midrange, general poise under dynamic pressure, and solid imaging. Sonically I couldn't extract anything bad from the MyAmp, save that it was a bit subtractive in the most benign sense. For example, the airiness riding atop Norah Jones' cover of "Cold, Cold Heart" was nicely preserved, if slightly shaded. Only the deeper bass pulses or power of orchestral percussion seemed slightly diminished in output and transient impact. For larger room applications, however, you'll likely want a solid 8-ohm speaker with higher sensitivity in the 90dB+ range. And there are plenty of them out there.

The MyAmp assumes an even greater comfort level with headphones. Of the models I had on hand it especially favored the higher-sensitivity models like the Cardas EM5813 (32-ohm/104dB), Audio-Technica ANC7b (300-ohm/109dB), and the B&W P3 (34-ohm/111dB). A few short minutes with one of these and MyAmp led me to understand why they call it personal listening. As I took in the opening verse of Bruce Springsteen's "Jungleland" from the 1975 concert at Hammersmith Odeon in London I could plainly hear Bruce leaning into the mike during some softly sung moments, and cupping it intimately between his hands for added effect. To gauge low-level resolving power I often turn to the backing harmonies of Fleetwood Mac's Lindsey Buckingham and Christie McVie from "Gold Dust Woman," a wonderful transfer from the 96/24 file. Behind Stevie Nicks' lead, these two distinctive voices emerged in stunning detail. And during Muddy Waters' "My Home Is On The Delta," a recording known for its terrific ambience and envelopment, everything was there, filling your ears with air and ambient cues. Waters' vocal was as lively and tonally accurate as I've experienced, although there was still that slight veiling on top and just a little speed-bumping of transients. Mind you, there are limits to the Micromega's transparency and dynamic slam. Soundstaging is not epic, and imaging lacks the sort of pinpoint focus that locks each instrument down within an acoustic space. Quick aside: My personal pair of AKG K501s are cans of notoriously low sensitivity (120-ohm and 94dB) and they couldn't be driven effectively by the MyAmp—a reminder that headphone/amp matchups matter. Remember that Micromega makes a matching dedicated headphone amplifier (MyZic) for difficult-to-drive headphones.

The whole point of entry level is to pare away the extraneous and cut to the chase—performance. In this sense, the MyAmp flat out gets down to business. It's not alone, however, in this tough segment—it goes right up against the NAD D 3020 (Issue 239), an equally excellent competitor with comparable sonics, better looks, and the edge on price. But the Micromega offers more inputs and overall flexibility. The MyAmp is confirmation that *serious* comes in all sizes. It's a desktop dynamo to be reckoned with. tas

Equipment Report

Yamaha A-S801 Integrated Amplifier and YBA-11Bluetooth Wireless Adapter

Full Featured & Good Sounding

Vade Forrester

Sometimes reviews of expensive, advancing-the-art gear lead us to think that's where all the interesting developments take place. While I can appreciate the ultra-high-priced spread, I can't afford it; I think it's more interesting when a manufacturer offers a component with tons of capabilities at a bargain price. And that's just what we have here: a 100Wpc integrated amplifier with an built-in DAC, priced at $899.

The Yamaha A-S801 may be inexpensive but its feature set is amazingly rich. Its internal DAC uses anESS Technology 32-bit ES9010K2M chip to play PCM files up to 384kHz/32-bit and DSD files up to DSD128. That range encompasses most of the computer-audio files available today. Status lights on the front panel indicate the sampling rate and DSD speed of a digital file being played. There are three digital inputs: asynchronous USB 2.0 on a USB Type B connector, coaxial on an RCA connector, and optical on a TosLink connector, which together will accommodate most digital sources. And you can add aptX Bluetooth connectivity by plugging Yamaha's $49.95 YBA-11 Bluetooth wireless adapter into the digital coaxial input jack. That lets devices such as smartphones and tablets (except iOS devices that don't support aptX) connect wirelessly—a shrewd design feature, since many music lovers have large collections of music files on their portable devices. But wait! There's also a USB Type A jack, the type you find on computers. What's that for? It powers the YBA-11. A separate cord is provided, with a USB Type A connector on one end and a small coaxial connector like you see on many power supply cords on the other. So you don't need a separate power supply for the YBA-11—clever.

In addition to the digital inputs, the A-S801 has five analog line inputs (labeled Line 1, 2, 3, CD, and Tuner), and even a moving-magnet phono input. And it has a single line-level output jack for connecting a subwoofer. Lots of folks today have 2.1 speaker setups (two satellite speakers and a subwoofer), and the A-S801 supports that arrangement—smart again. But wait, there's even more! The Line 2 and 3 inputs have both record and playback jacks, so you can connect a tape deck. There's a blast from the past—when a respectable audio system had open-reel and cassette tape machines.

The parade of features continues with bass, treble, and loudness controls. Loudness controls used to be common, but like tone controls, have since become scarce. (If you're wondering, a loudness control boosts the bass and treble as the volume level decreases.) There's even a balance control! There are also two settings you can use for maximum signal purity: Pure Direct, which bypasses unused audio inputs and turns off power to those inputs; and CD Direct Amp, which switches directly to the CD player, bypassing all other inputs and turning off the power supply to unused inputs. Both of those settings bypass the tone, balance, and loudness controls. There is also a two-position speaker switch that lets you select two sets of speakers, or turn the speakers off entirely so you can use the headphone jack. You can also activate both sets of speaker terminals simultaneously if you want to bi-amp your speakers. And there's a full-featured remote, too, which also controls all six of the CD players Yamaha makes. Did I mention that the A-S801 is feature-rich?

Styling is traditional Yamaha, which I've always admired. That means it comes in a full-sized case (17.125" x 6" x 15.25") with lots of controls on the front panel. Both black and silver faceplates are available; the silver review unit looked quite stylish, and all its labels were readable. The amplifier carries a two-year parts-and-labor warranty, reasonable for an $899 product.

Yamaha emphasizes that it uses premium parts in constructing the A-S801, though no specific examples are cited. The chassis is double-layered to suppress vibrations. Lots of manufacturers are turning to Class D output sections to achieve high power output at low cost, but I was pleased to find the A-S801 uses a Class AB output circuit. Input impedance for all analog inputs, including the mm phono input, is 47k ohms, which should pose no problems with most sources. The single, summed subwoofer output on an RCA jack has an output impedance of 1.2k ohms and a built-in high-frequency cutoff at 90Hz. The relatively high output impedance for the subwoofer jack could cause problems with some subwoofers; it's higher than

Equipment Report Yamaha A-S801 Integrated Amplifier and YBA-11Bluetooth Wireless Adapter

SPECS & PRICING

Minimum output power: 100Wpc RMS, 0.019% THD, 8 ohms, 20Hz–20kHz
Dynamic power (per channel): 140/170/220/290W (8/6/4/2 ohms)
Damping factor: 240
Frequency response: 10Hz–100kHz +/– 1.0dB
THD: 0.019% (50W/8 ohms)
SNR: 99dB (input shorted, 200mV)
Inputs: Eight
Outputs: Two
Digital inputs: Optical, coaxial, USB (type B)
Dimensions: 17.125" x 6" x 15.25"
Weight: 26.7 lbs.
Price: $899

Yamaha YBA-11 Bluetooth Wireless Adapter
Price: $49

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ideal for my JL Audio Fathom f110 subwoofer, for example. An impedance selector switch on the back panel lets you choose the output impedance of your speakers: low for 4-ohm speakers, high for others. The amplifier should be turned off before changing the position of the impedance selector switch. An auto power standby switch, if turned on, automatically puts the amplifier in standby mode if it's not operated for 8 hours. A two-prong IEC jack is provided for the power cord connection. There are no standard line-out jacks, so you can't use an additional power amplifier for bi-amplifying, or use separate left- and right-channel subwoofers. Those omissions seemed well-chosen for an amplifier at this price point.

Since headphones have become such an important part of the connected listener's experience, it was important to provide a headphone jack. The headphone amp is rated at 54 milliwatts into 16 ohms, and 400mW into 300 ohms.

A well-illustrated owner's manual is provided with sections in English, French, and Spanish. Each section is 19 pages long. The manual is also available on Yamaha's website. It has an exhaustively thorough list of specifications.

Setting Up and Using the A-S801

The full-width amplifier slid easily onto a shelf of my equipment rack with plenty of clearance for ventilation. It ran barely warm to the touch when playing. I used the power cord furnished with the amplifier, reasoning that at this price level users would probably not spring for an aftermarket cord. Since the amplifier had facilities for a 2.1 speaker system, that's what I used: KEF LS50 satellite speakers with a JL Audio Fathom f110 subwoofer. To evaluate the A-S801's bass performance, it was necessary to let the amplifier run full-range, so I turned the subwoofer off for that. Blue Marble Audio speaker cable, with banana plugs on the amplifier end, made it easy to connect to the A-S801's speaker terminals. An HP Envy laptop computer running 64-bit Windows 7 and Roon server software comprised a computer-based server. It was connected to A-S801's USB input using Audience Au24 SE USB cables. I remotely controlled Roon from both the iPad version of Roon Remote and from a second copy of Roon installed on a Toshiba laptop computer. It was cool to be able to use the Toshiba laptop to write this review and then switch over to use it as a remote for Roon. I also tried a dedicated server, the Linux-based Aurender N100H, connected to the USB input with a Wireworld Platinum Starlight 6 cable. The servers were connected to my home network, where I store my collection of music files on a QNAP TS-251 network attached storage drive.

I downloaded a copy of Yamaha's Steinberg USB Windows driver Version 1.9.5 for the A-S801 and installed it so the computer would recognize the A-S801's DAC, then changed the settings in Roon to play through the A-S801—a very straightforward procedure. Usually, a Windows driver installation actually installs several drivers, such as a Windows Audio Session API (WASAPI) driver and an Audio Stream Input/Output (ASIO) driver and possibly others. Normally, I use the WASAPI driver; however, Yamaha's WASAPI driver was buggy; it crashed a few times, and when it did work, it would not play DSD files, converting them instead to PCM, even though I had set Roon's DSD playback strategy to play DSD files using DoP mode. Fortunately, the Yamaha ASIO driver worked just fine. To explore this problem, I also tried the drivers in the JRiver Media Center version 21 server program; again, the WASAPI driver crashed occasionally, while the ASIO driver worked OK. The Linux-based Aurender server worked flawlessly. Grrr—whereas I found dealing with such a problem an interesting challenge, it would have been very frustrating for a newbie.

The A-S801 epitomized the need for amplifier break-in. Yamaha recommended 100 hours, so I let the amplifier play 24/7 until it reached (and passed) that elapsed time before I started listening critically. When I first connected the A-S801, I feared the review was going to be an ordeal; the amplifier sounded brittle and raw; but after 100 hours of play, it smoothed out dramatically. A couple of audio dealer friends who heard the amplifier before and after break-in (but not in-between) agreed there was a big improvement. Since I'm a headphone fan, I made sure the headphone jack got plenty of break-in time, too, using $299 NAD VISO PM50 headphones, a good match for the Yamaha.

Lots of people have large music collections on their smartphones, so it's handy to be able to play that music by connecting the phones to an amplifier using a wireless Bluetooth connection. I plugged Yamaha's YBA-11 Bluetooth wireless adapter into the digital coaxial input jack to provide Bluetooth connectivity. As noted above, the A-S801 provides a Type A USB port on the rear panel that powers the adapter. Both a power cable and a skinny SPDIF cable are also provided. They aren't very long, so you'll need to place the adapter near the amplifier—not much farther away than the front panel. Connecting the YBA-11 to my iPhone 6 was quite easy. It was one of the best-sounding Bluetooth connections I've heard, too—very enjoyable.

The remote control was easy to use, with buttons to raise or lower the volume separated slightly from the other buttons for easier access. A mute button, just below the volume buttons, partially muted the output of the amplifier, let-

Equipment Report Yamaha A-S801 Integrated Amplifier and YBA-11Bluetooth Wireless Adapter

ting the music play at a much reduced level. The muting was quiet enough to permit a telephone call or conversation, but loud enough to remind you music was still on. I was pleased to see that each input had a separate button on the remote, so you could access any input directly without having to scroll through intermediate ones. The source selector knob on the A-S801 was freely rotating, but had a detent at each input's position, and a small LED lighted up at each position to indicate which input was selected. The volume control had a small index line inscribed on it, indicating its position, but it was a bit difficult to see from my listening position ten feet away. The volume control was a continuously variable unit with a motor drive that enabled the remote to set the level. The power amp section turned off when headphones were plugged in. I wondered if the relatively low headphone output into low-impedance 'phones would pose a problem, but it drove the 32-ohm NAD PM50s to higher SPLs than I cared to hear. The NAD headphone is quite an easy load, so I challenged the A-S801 with the harder-to-drive 35-ohm HiFiMan HE400, which has a fairly low 92.5dB sensitivity. The A-S801 drove it louder than I could stand to listen to, so any worries I might have had about wimpiness from the headphone amplifier were put to rest. Similarly, in my largish room (23' by 20' by 12') the A-S801 effortlessly drove the 85dB-sensitive KEF speakers much louder than I wanted to hear them play. Bass without the subwoofer was punchy and dynamic, although it understandably didn't plumb the depths; after all, the KEF speakers only have a 5¼-inch mid/woofer—although sometimes the bass performance compelled me to check to make sure I hadn't left the subwoofer on. The A-S801 had no problem driving my sub, though I had to turn it down considerably from the setting I normally use with my preamp.

Unfortunately, I didn't have a moving-magnet cartridge available, so was unable to try the phono section. Nor did I have a CD player, which prevented using the CD Direct feature, but I did try the Pure Direct setting, which sounded a bit cleaner. I used the Pure Direct setting for the review.

Sound

The Yamaha A-S801 amplifier sounded sweet and smooth, particularly with vocals. Not a trace of the glare that plagued the unit before break-in was present, until the volume was advanced to a louder level than I ever cared to listen; then, some glare and coarseness set in.

It's useful to begin a review with a very familiar musical piece, so I queued up old favorite "Folia Rodrigo Martinez" ripped as an AIFF file from the CD *La Folia 1490-1701* [Alia Vox]. It's an information-rich recording of a musical piece written in 1490 and realized by Jordi Savall and his band. On the A-S801, the cascabels which open the piece were very clearly delineated, though without as much detail as I've heard on the best systems. The bass, which descends into the mid-20Hz range, was, of course, not fully developed on the small KEF speakers (with subwoofer off), but had plenty of impact, and the upper bass was quite detailed. When I switched on the subwoofer, the low bass was reproduced with impact, though it lacked the resolution I'm used to hearing. Percussion instruments sounded harmonically accurate, but blurred into the background a bit more than they do with top-of-the-line systems. Savall's viola da gamba sounded harmonically rich, and the baroque guitar and harp were distinctive. Sometimes the last two instruments seem to sound quite similar, making it hard to tell them apart. The A-S801 had plenty of microdynamic verve, so the music sounded quite lively. I could tell Savall and his forces were having lots of fun playing the piece, and I had just as much fun listening to it.

With "Folia Rodrigo Martinez" fresh in mind, I played the same piece from my iPhone 6 through the YBA-11 Bluetooth connection. My expectations weren't very high, but I was pleasantly surprised at the quality of reproduction: plenty of sparkle and dynamics, maybe less detail than via the computer connection, but thoroughly enjoyable. If you have a lot of music on your smartphone, as many people do, it's well worthwhile to invest in the YBA-11 Bluetooth adapter.

To see how the A-S801 handled a solo instrumental, I queued up Alex de Grassi playing "Shenandoah" from his album *Special Event 19* [Blue Coast Records]. The A-S801 did a fine job reproducing the guitar: the initial transient of the plucked strings, the sustain as the string sounded its note, and the decay as the note slipped into silence. Treble was extended but not peaky. The A-S801 accurately captured the sound of the drone strings from de Grassi's unusual instrument.

To evaluate how well a component handles soundstaging, I often turn to the piece "Miserere" from the Tallis Scholars *Allegri's Miserere & Palestrina's Missa Papae Marcelli* album [Gimell]. This *a cappella* work has two vocal groups: The main one is at the front of the soundstage, while a smaller solo grouping is located well behind them in the church where the piece was recorded. The main group was reproduced with plenty of detail and clarity, without any trace of the distortion that some components impose on the piece. I've heard the main (front) choir distributed more widely across the soundstage, but singers within the group were well localized. The rearward solo group was reproduced in a wash of reverberation, but the singers there were still understandable. I've heard this piece reproduced better, but by systems costing multiples of what the review system cost.

You can't write an audio review without playing a Girl with Guitar piece, so I queued up Shelby Lynne's *Just a Little Lovin'* [Acoustic Sounds]. Although this album was recorded at quite a low level, the A-S801 had no trouble playing it. Even though the subwoofer was turned off on the title song, the A-S801 reproduced the bass (mid-bass, actually) with a lot of impact and punch, so that the song was quite enjoyable. With the subwoofer back on, the bass extended quite a bit deeper—the advantage of a 2.1 speaker system. As in other pieces, treble was once again quite extended but not peaky. Lynne's voice was reproduced with just a little hoarseness, which I think is how it actually sounds. I could hear how she phrased the words quite clearly.

Finally, I challenged the A-S801 with full orchestra, playing Manfred Honeck and the Pitts-

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burgh Symphony's recording of a favorite piece: Beethoven's Symphony No. 7 [Reference Recordings/NativeDSD]. (If you're unfamiliar with NativeDSD.com, its rapidly growing catalog specializes in downloads of high-resolution music recorded in DSD and DXD formats, including DSD256.) The opening bars nearly blew me off the couch, since I had forgotten to reduce the listening level after playing *Just a Little Lovin'*. But even at that level, there was no strain to the sound. The DSD128 indicator light on the A-S801's front panel came on verifying that the A-S801's DAC was indeed playing that extra-high-resolution format, and the orchestral sound was gloriously natural—tons of detail, harmonically rich, with distinct but continuous dynamic levels.

Comparison

Ideally, I'd compare an item being reviewed to a similar component; however, I had no similar integrated amplifier on hand, and there doesn't seem much point in comparing it to my much more expensive reference system, with electronics alone costing over $23,000. I could sum up such a comparison like this: The reference system sounded better in virtually all respects, but so what? It doggone should sound better, given the considerable difference in price. It surely doesn't sound 23 times better! Instead, let me draw on my memory of other integrated amplifiers I've reviewed fairly recently and compare them to the A-S801.

Most closely resembling the A-S801 was the $2600 NuPrime IDA-16 amplifier I reviewed in Issue 252. It included an equally versatile DAC and had even more power: 200Wpc from a Class D amplifier section. The amplifier section sounded smoother and never showed any inclination to sound ragged, even at obscenely loud levels, as you'd expect, given the power difference. Class D amplifiers have gained a reputation for sounding a bit odd, but the NuPrime had no such problem. It had several more sophisticated design features than the A-S801, such as ultra-low noise JFETs employed in its input stage, and a volume control with ninety-nine ½-dB steps implemented via an advanced, thin-film, switched-resistor ladder where only a single resistor is in the signal path at any volume setting. Its digital volume-level display was easy to read from a distance. It also had a full stereo line output which could be used to drive subwoofers or another external amplifier. This differed from the A-S801's summed subwoofer output, and the NuPrime also had no built-in crossover. I don't view that omission as a problem; most subwoofers have built-in lowpass crossovers. However, the NuPrime amplifier lacked a phono section, and had no headphone amplifier.

Another recently reviewed (Issue 255) integrated amplifier I'll mention was the $599 Denon PMA-50. It's a smaller switching amplifier designed for a different environment—a bedroom, office, or just as a headphone amplifier. I think of the A-S801 as an integrated amplifier that will also drive headphones, and the PMA-50 as a headphone amp that will also drive speakers. In the latter application, it has a more powerful headphone amplifier. I didn't even try using the 25Wpc Denon to drive the 85dB-sensitive

KEF speakers; I imagine the maximum volume attainable would be fairly limited. The Denon had only a single analog input, and no phono section or subwoofer output. But it did have a very versatile DAC and remote control. Although it looks very minimalist, a lot of controls are accessible through the tiny remote, including tone controls and three-level gain for the headphone amp. It also had an internal Bluetooth section, so you didn't need an adapter like the YBA-11. However, I wasn't nearly as impressed by the sound of the Denon's Bluetooth connection as I was the YBA-11's. Befitting its intended purpose, the Denon was less than half the size of the A-S801. If you have pretty efficient speakers and don't want to play them loudly, or if headphones are your primary means of listening, the Denon amplifier could be a better buy than the A-S801.

Bottom Line

Is there any other audio component with as many features as the A-S801 amplifier? And it's not like the features were just thrown in to impress; the A-S801 surprised me by how good it sounded driving the low-sensitivity KEF LS50 speakers in my largish room. No, it didn't equal my far more expensive reference system, but during a listening session, several of my audio buddies said they derived genuine musical enjoyment from the system anchored by the Yamaha A-S801 amplifier, and could happily live with it. Coming from a group of lifelong audiophiles, that's high praise indeed.

The Yamaha A-S801 looks good, sounds splendid, and has a long list of useful features at a price that makes it a flaming bargain! I suspect many readers are lifelong audiophiles like me, for whom system upgrades are a way of life, possibly even the purpose of life. But for lots of people who just want a good hi-fi to play their music on, a hi-fi may be a once-in-a-lifetime purchase. For those people, or for anyone who wants good sound with lots of flexibility at a reasonable price, the Yamaha A-S801 integrated amplifier would be my top recommendation. It may be the only hi-fi electronics purchase they will ever need. tas

Equipment Report

NuPrime DAC-10H DAC/Pre and ST-10 Power Amplifier

True High Performance for Less

Steven Stone

In 2014, NuForce's cofounder, Jason Lim, with backing from the OEM factory, bought the assets of NuForce's high-end division, obtained the rights to NuForce technologies, and formed NuPrime Audio, Inc. Shortly afterward the NuForce company was sold to Optoma.

NuPrime's first offering, the IDA-16 integrated amplifier, was reviewed by Vade Forrester (Issue 252). He concluded that, "I wouldn't be ashamed to put it on a shelf next to the fanciest component." NuPrime's latest, the $1795 DAC-10H DAC/Pre and the $1595 ST-10 basic power amplifier, are slightly more expensive than the $2600 IDA-16 integrated amplifier, but promise an even greater level of sonic refinement and flexibility. How do they stack up in this highly competitive price range? Let's see.

The DAC-10H

Although the DAC-10H is only 2.4" high by 8" wide by 14" deep, which corresponds to roughly half the width of a "full-sized" component, it packs a lot of features and performance into a small package. The DAC section is built around the ESS Sabre Reference ES9018 32-bit DAC chip. According to NuPrime this DAC chip can deliver 135dB signal-to-noise with -120dB total harmonic distortion levels. To reduce time-domain errors the DAC 10H utilizes symmetrical signal processing combined with asynchronous data transfer. It supports PCM up to 384/32 and DSD up to 256.

On the analog side, the DAC-10H has borrowed from the NuForce P-20 preamplifier the stepped, thin-film switched-resistor ladder network for controlling volume. This device uses a MUSES chip combined with a proprietary look-up table to ensure that only a single resistor is in the signal path at any given volume setting. The volume adjustment is in 0.5dB increments and is displayed via a 0-to-99-numbered system on the front panel. Comparing different sources using these precise and repeatable volume adjustments was a pleasure.

In addition to the 99-step volume control, the DAC-10H also has dual gain settings for its outputs. The single-ended RCA output can have a maximum voltage of either 2 or 4 volts, while the balanced XLR outputs have 4 and 8 volt levels. The headphone amp also has two levels for its balanced and unbalanced output to allow for different headphone sensitivities and impedances.

In its input stages the DAC-10H uses ultra-low-noise JFETS with independent left and right power supplies that come from a multi-rail toroidal transformer coupled to a linear power supply. This helps achieve a crosstalk attenuation specification of at least 93dB at 1kHz.

The DAC-10H has two headphone outputs: a single-ended and a balanced connection. Both have the same output impedance of less than 10 ohms. The balanced headphone circuit uses an OPA2134 op-amp as a buffer for the pair of NuPrime-branded IC chips used to drive the output.

Setup and Ergonomics

The DAC-10H front panel has some stylistic similarity with earlier NuForce designs that lean toward a modernist aesthetic of understated minimalism. On the upper left side of the front panel, you will find a single-ended headphone connection; the balanced connection is on the right. Between them is a discrete set of LEDs that display the source and the bit-rate (if any) being generated by that source. Under the display and headphone connections is a single row of rectangular buttons. From right to left, they include the low/high output switch, down volume, power on/off, volume up, mute, and headphone volume selector. The only labeling on these buttons are small graphic symbols.

On the back panel of the DAC-10H, you'll find

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SPECS & PRICING

DAC-10H DAC/Pre
Inputs: One USB digital, two coaxial digital SPDIF, two optical digital SPDIF, two analog stereo RCA
Outputs: Optical (up to 24-bit/192kHz), stereo RCA (line out), stereo balanced (XLR-3 socket pre-out), balanced headphone amplifier (XLR-4 socket), unbalanced headphone amplifier (6.3mm jack socket)
USB sampling rates: 44.1kHz–384kHz and DSD 2.8MHz, 5.6MHz, 11.2MHz
Max. output power: 680mW @ 1kHz and 600-ohm load at the XLR-4 output
Dimensions: 8" x 2.4" x 15"
Weight: 10.5 lbs. (4.8 kg)
Price: $1795

ST-10 Power Amplifier
Input: Two RCA
Output: Five-way binding posts
Power output: 150Wpc at 8 ohms
Gain: 28dB
Input impedance: 23.5k ohms
Sensitivity: 0.89V to rated power
S/N ratio: 110dB at 1W, 10W, 100W
Dimensions: 215.4mm x 59mm x 394mm
Weight: 13.4 lbs. (6 kg)
Price: $1595

two pairs (one single-ended RCA and one balanced XLR) of variable-output analog connections, two pairs of single-ended analog inputs, two coaxial SPDIF inputs, two TosLink digital inputs, one USB 2.0 input, and an IEC AC power connection. While that sounds like a lot of connections to fit into a relatively tight space, the layout on the DAC-10H allows for easy access to all I/O's.

The overall fit and finish of the DAC-10H is commensurate with its technical specifications. All surfaces are impeccably finished. The little flourishes, such as the thin chrome bands around the two headphone outputs, give the DAC-10 an unmistakable touch of panache.

For most of the review the DAC-10H's balanced outputs were tethered to the NuPrime ST-10 power amplifier. The unbalanced outputs were split, one leg routed to a Velodyne DD10+ subwoofer, the other connected to an outboard headphone amplifier.

The DAC-10H comes with a unique-looking remote that is eight inches long and hexagonally sided. It's the same remote that NuPrime uses with its IDA-16 integrated amplifier. It duplicates all the controls on the DAC-10H, which is fortunate because if the DAC-10 is located beneath your desk—as it usually is in my nearfield system—it's very difficult to operate it "by feel" since all the buttons feel the same. To ensure that you are pushing the correct button requires counting across from right or left. Ninety-nine percent of the time I used the remote. I found its angle of acceptance to be quite wide, even more so than most units I've used. My only complaint is that all the buttons rattle; in fact, they rattled so much that the DAC-10 remote is suitable for use as a percussion instrument.

During the review period I tried all manner of digital sources, from lowly 128kbs MP3s to 128x DSD and 192/24 PCM. In every case, the DAC-10H played the files without incident. I'm also happy to report that during the review period the DAC-10 proved to be an extremely trouble-free component. Unlike many devices, the DAC-10H was absolutely silent during turn-on and turn-off with no thumps, clicks, or buzzes. Also, when you change inputs or un-mute the DAC-10H, it does a gradual volume ramp-up instead of giving you the full volume setting immediately; this allows a user time to lower the volume if it was set too high from the previous input.

Considering its plethora of input options, I see no reason why, despite its diminutive footprint, the DAC-10H would not be up to the task of serving as the control center of a highly evolved audio system—it even has a home-theater-bypass mode so you can use it in conjunction with a multichannel AV processor.

I tried a wide variety of headphones with the DAC-10H. With my most sensitive custom in-ears, the Westone ES-5, there was a slight amount of low-level hiss. On the other extreme, using the single-ended outputs, the DAC-10H had no trouble driving a pair of Beyer Dynamic DT-990 600-ohm headphones well past satisfying levels. The balanced outputs worked splendidly with both the Mr. Speakers Alpha Prime and HiFiMan HE-560 headphones. My original Grado RS1 headphones also had excellent bass extension and drive when connected to the DAC-10H's balanced output.

The Sound of the DAC-10H

For me, the most outstanding aspect of the DAC-10H's sonic performance was its silence. Even with DAC/preamps that have almost the same signal-to-noise specs, I can usually hear differences between the "silences" at full output compared with fully attenuated outputs (bear in mind that in my nearfield system the speakers are only three feet away from my listening position and my room is very quiet). With the DAC-10H/ST-10 combination I could hear only the very faintest added hiss at full levels when I moved my ears within a few inches of a tweeter, but at the listening position I heard nothing. And why should this be such a good thing? Because the DAC-10H's excellent signal-to-noise ratio lets the music emerge from silence with a level of delicacy and subtlety that more closely approaches what I hear from a live musical event than noisier DAC/preamps which don't have the same signal-to-noise capabilities.

Inner detail and low-level resolution through the DAC-10 are as good as I've heard through any DAC including the Antelope Audio Platinum DSD DAC. The differences in depth recreation and soundstaging precision between my original 128x DSD recordings and 44.1 down-sampled versions were immediately obvious when comparing them through the DAC-10H.

Depending on the recording, the sense of three-dimensionality portrayed through the DAC-10 can be nothing short of remarkable. Listening to B. B. King's classic album *Live at the Regal* over the TIDAL app, combined with the latest Amarra SQ+ 2.1 on my Mac Mini connected to the PS Audio DSD DAC, it was easy to hear how the audience sound comes from a point well behind the lateral plane of the band. Also, the clarity and tightness of the electric bass were exemplary.

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Since I also have an early stereo LP pressing of the same recording as well as a CD version, I was able to do some A/B/C listening, comparing the Tidal stream with the ripped CD played back through Amarra Symphony, and then the LP played back via my VPI TNT III turntable with Graham 1.5 tonearm, Clearaudio Victory II cartridge, and Vendetta 2B phono preamp.

While the differences in soundstaging, depth, and frequency extension were essentially nonexistent between the CD and the Tidal stream, the LP had noticeably superior dimensionality—instead of a wall of audience there was an individualization of each voice within that audience. Also B.B.'s vocals on the LP had more immediacy and dynamic energy. A friend who was present during the comparisons said to me, "I wish I could have the top end, midrange, and spatial characteristics of the LP in the digital copy, and the low-frequency clarity and punch of the digital on the LP." Yes, the DAC-10H's analog section and stepped volume control are capable of passing through even the subtlest of audible information in both the analog and digital domains.

Using the DAC-10H's headphone output I was impressed by the solidity of the image, the delicacy of upper frequencies, and the control of lower frequencies. Compared with the built-in amplifier in the Oppo HA-1, which was the DAC/pre I had in the system previously, the DAC-10H was a step up, both in its ability to drive difficult headphones via its balanced connections, and in its portrayal of low-level detail. I also compared the DAC-10H's headphone outputs with a dedicated single-ended tube headphone amplifier. The DAC-10 was its equal for midrange purity and upper frequency extension. In the bass, the DAC-10H was more controlled with better inner detail and dynamic punch. My conclusion: The DAC-10H's headphone outputs are good enough to make the need for an external, dedicated headphone amplifier optional.

The NuPrime ST-10 Power Amplifier

The NuPrime ST-10 amplifier is what NuPrime calls "near-reference class." Why only near-reference? As far as I can tell it's thus named because this stereo amplifier only puts out 150 watts per side into an 8-ohm load. The ST-10 utilizes NuPrime's proprietary, fourth-generation V4 amplifier module. According to NuPrime, this latest version offers substantial improvements including a 20dB reduction in the noise floor, a shortened circuit pathway, increased output current, and a 600kHz switching frequency. Other improvements over earlier designs include a new linear power supply that employs a high-efficiency toroidal transformer; superior reliability when not under a load, and an enhanced even-order harmonic circuitry that according to NuPrime, "resembles the most attractive features of tube-amp sound."

Although the ST-10 has a switching output stage, it is not a standard Class D switching amplifier. According to NuPrime's owner's manual, "Instead of the conventional saw-tooth configuration, NuPrime's patented circuit design uses an analog-modulating signal that adds neither noise nor jitter. Rather than reverting to off-the-shelf solutions, NuPrime's in-house advances have further unlocked the switching amp's potential without the difficulties pure switching amplifiers simply cannot avoid." The cliché that should follow would be, of course, "Not your father's Class D amplifier."

Among its technical advantages, the ST-10 has a damping factor of 400, which means it should be able to control any excess diaphragm movement better than an amplifier with a lower damping factor. The ST-10 also has far lower amounts of phase shift than most amplifiers, due to its unique closed-loop circuit.

The front panel of the ST-10 closely resembles that of the DAC-10H except it has fewer buttons and lights. Actually the ST-10 front panel has exactly one button, on the left side of the faceplate, and one light on the right side of the faceplate. That's it, apart from the NuPrime logo in the center.

Setup and Ergonomics

Although the ST-10 provides 28dB of gain rather than the standard 26dB, for most systems this won't be an issue, and many systems will benefit from that extra 2dB of gain. The ST-10's rear panel has all the connections that you would expect on a basic power amplifier: one pair of balanced XLR inputs, one pair of single-ended RCA inputs, one set of stereo outputs using five-way binding posts, a 12-volt trigger connector, an IEC AC connector, and a toggle switch for balanced or unbalanced input selection. However, unlike many stereo power amplifiers, the ST-10 doesn't have provisions for bridging it into a mono mode.

When you push the on/off button on the front panel you will hear a soft click from the amp's relays after a second, and then it's good to go. When you turn the ST-10 off, it has a delay of approximately ten seconds before it shuts down completely.

The Sound of an ST-10

Over the years I've reviewed and used plenty of swtiching power amplifiers from Bel Canto, Wyred 4 Sound, April Music, and others, and I appreciate what a well-designed model can bring to a system. And it happens that the ST-10 is the best switching power amplifier I've heard to date.

As you might have gathered from its specifications, the ST-10 is a very quiet, extremely low-noise power amplifier that, as long as it isn't pushed into clipping, sounds exceedingly neutral and uncolored. I tried the ST-10 with a variety of speakers from the 84dB-sensitivity Aerial Acoustics 5B to the 95dB-sensitivity Audience 1+1, as well as the ATC SC7 II, Dunlavy SC-1AV, and Mirage OM3. In every case the amplifier did a superb job of driving the speakers with authority and control.

I was especially impressed by the ST-10's performance at the top and bottom of its range. The bass was taut and tuneful. Conversely, the upper midrange and treble were airy yet accurate. On recordings with exaggerated upper midrange or treble energy I was aware of the additional musical information, but it was never emphasized to the point of harshness. After living with the ST-10 for a while I can understand why NuPrime draws attention in its sales literature to the ST-10's "tube-like" upper-frequency characteristics. While the ST-10 certainly

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doesn't soften or roll those off in the manner of classic tube designs, it brings to its upper frequencies the kind of ease and sweetness that are usually found in power amplifiers that employ tubes somewhere in their circuitry.

Depth recreation, dimensionality, and image specificity were also exemplary through the ST-10. On my live 128x DSD recordings of the Boulder Philharmonic, the soundstage was accurately portrayed with the spaces between the instruments elucidated with a level of specificity that was equal to the best I've heard from any amplifier.

Final Thoughts

Within their product categories the DAC-10H DAC/preamp and ST-10 basic power amplifier are priced at the lower mid-level, yet they both deliver performance that could be considered exemplary regardless of cost. The DAC-10H has the capabilities, sound, and feature set that should keep it current for years, while the ST-10 offers sonic quality that, unless you absolutely must have more power output, will make "upgrading" to anything but a far pricier amp more of a sideways proposition than an upward one.

As it is a relatively new firm, NuPrime has yet to develop the reputation of more venerable audio companies. But given the quality of its first three products, the IDA-16, DAC-10H, and ST-10, it's hard not to predict that NuPrime will be a force to be reckoned with now and in the future. Even if you have far more in your equipment budget than what the DAC-10H and ST-10 cost, I recommend giving these NuPrime products a listen, if you can. They deliver true high performance for far less money than you might expect. tas

Equipment Report

Lyngdorf Audio TDAI-2170

The Way of the Future

Robert E. Greene

Ever since the advent of CD in the early 1980s, when digital audio became the common method of providing music to consumers, the possibility of using digital as more than a passive substitute for analog in a small part of the audio chain—as more than a sort of stop-gap way to get the audio signal from the mixing board output to the preamp input with analog portions of the chain on either side—has been in the offing. From the beginning, there was the option of converting the signal to digital as early as possible. In addition there was the visionary thought that one could keep the signal digital all the way to the loudspeakers—to drive them directly with a digital signal. The part analog/part digital chain was always at best a compromise. In the end, the natural aim was digital all the way.

In addition, there was also the possibility of making changes in the signal to improve the final result, not just in the mastering of the recordings but also in the correction of speakers and their interaction with the room around them.

Peter Lyngdorf and his companies—originally TacT, and today Steinway Lyngdorf and Lyngdorf Audio—have played an important, one might well say central, role in these visionary developments for twenty years. The original TacT room-correction system of the early 1990s and its refinements later on were among the first successful devices for the correction of the interaction between speakers and the rooms around them. Then came the TacT Millennium amplifier (my TAS review is on-line), which was truly a landmark on the way to the Digital Age. At this point, Peter Lyngdorf started Steinway Lyngdorf in association with the celebrated piano manufacturer, and began a systematic effort to use digital technology to make the best possible audio system. This was a price-no-object effort and the first products were very expensive, though there was a small system that was both extraordinarily good and also at least plausible in price (review in TAS, Issue 219). Indeed as audiophile prices go nowadays, it was and is something of a bargain for what it offers sonically.

In any case, part of Peter Lyngdorf's vision was that as the Steinway Lyngdorf aim-for-the-stars products clarified the technology, it would become possible to offer the same benefits to audiophiles who do not have unlimited financial resources. This is the kind of "trickle-down" economics that can work!

The Lyngdorf Audio TDAI-2170 integrated amplifier/DAC is one of the fruits of this overall program. And an extraordinary device it is.

What It Does

The short answer would be: Everything that an integrated amplifier that accepts digital inputs could do, with correction of the room/speaker interface included. Everything, that is, except function as a phono preamp or keep an analog input at line-level analog for longer than it takes to convert it to digital. This is pre-eminently a digital device. The whole idea is that keeping everything digital all the way through is the way to go. This is not to suggest that the unit does not do a nice job of handling analog inputs. It does do a good job, sounding superb in this context, and I do not think that anyone listening to its being used as an analog linestage and amplifier would say instantly: "Well, it is nice enough but it sounds digital." It does not. Still, if you think for some unknown reason that analog is white magic and digital is the work of the Devil, then you will have to summon up some impartiality to discover that you are kidding yourself.

In any case, the Lyngdorf, as I shall call it now and hereafter, is philosophically intended for digital. And it will accept every kind of digital input extant. You can find details on the website link below. (Listing everything the Lyngdorf can do feature by feature would be an easy way to fill up my word count, as it's all available on-line—and since there is a lot of it—I am not going into it all in this review, though I shall list a bit of the specifications at the end.)

One might suppose it would be impractical to make a direct-digital-drive output stage work with all the different kinds of digital streams there are. And one would be right. Everything is converted to the same standard for output, namely pulse width modulation at 390kHz. If this disturbs you, again a little impartial listening ought to solve your problem. The Lyngdorf does not sound digital in any negative sense.

Where it does sound digital is in it clarity and background silence. The method of controlling volume is to adjust the power supply voltage. This means that the intrinsic digital signal-to-noise ratio of 110dB is independent of volume setting. The proverbial and much sought after black background is very much in evidence here.

Room Correction and Voicing Controls

The glories of the Lyngdorf are two-fold: the digital amplification and the RoomPerfect room-correction system. Let me discuss RoomPerfect first. This is an automated system. One measures the speakers' behavior in the room—a microphone is supplied and there is an easy "walk through" menu to follow, with instructions appearing on a small screen—and the system decides what correction to apply.

The system is very much plug and (pretty soon) play. The user does not have to have any particular experience with room-correction systems nor make too many decisions. And the result is very good—a definite improvement—even if one does not do everything absolutely ideally, "everything" in this case referring to the choices of where to put the microphone for successive measurements. Bass smoothness is improved, channel matching is much better in the frequencies affected most by the room, and stereo imaging as well as the presentation of the original recorded acoustics are significantly better. This is all what should happen with a room-correction system, and happen it does. It is hard to imagine that people won't like this and like it a lot.

RoomPerfect operates not just by measuring at the listening position. After one has done a listening position measurement, one is then instructed to move the microphone to some quite different position and measure once more. Then yet another position. (The owner's manual offers suggestions as to where one might place the mike each time, but except for the first measurement at the listening position, there are free choices.) The idea is that the system begins to accumulate what is called on screen "room knowledge." After each measurement, an update percentage figure for "room knowledge" is shown on the screen and somewhere in the 90% range one can stop.

This is impressive if somewhat magical in appearance, since no explanation is offered of exactly what room knowledge might be. Also, even if one achieves a high percentage of measurements, the exact result in the final correction may be slightly different for different paths to that room knowledge, for different choices of microphone position along the way.

The system generates two corrections: "local" and "global." (Interestingly, these are the words mathematicians use for the two kinds of differential geometry—local is for what happens right around a point, global for what happens overall.) The local setting is optimal for the listener at the first measurement position. The global makes the sound good over a larger area around that position, at the price of not making it quite so nearly perfect at the one chosen spot for the first measurement.

It is a bit of a black box system—it does not show you any measurement results, and one can only judge by listening or measuring with an external measurement system once the correction is applied. But there are indirect ways to control things, if not to know exactly in advance what their effects will be. For example, if one does all the measurements right around the listening position, then one can get something more like what a single-point measurement correction system would generate. Or one can measure over a larger area, or over the whole room (as is recommended). There is a lot of experimenting one can do.

The system does not attempt to change the basic sound of the speaker and it does successively less as frequency rises. The goal is not so much to change your speaker to some abstract ideal of speaker-dom—one is supposed to like the speakers one bought!—but rather to change the sound so that the speaker sounds as it would if the room and the speaker's interaction with it were ideal. Hence "room perfect"—the room interaction is perfect, the speaker remains whatever it was.

This seems like a sensible approach. And the results are gratifying, as noted. But they do vary somewhat with how the measuring is done so you may want to experiment quite a bit. Moreover, if you are the kind of person who really wants micro-control and micro-information, you might want to add an external measuring system and perhaps an outboard digital EQ to make small changes in the sound of your speaker itself, not just in its relationship to the room. If you use a computer as a music source, you can, of course, use EQ programs in your computer, of which many are available at nominal cost. I might mention here that on occasion RoomPerfect made some (measurable as well as audible) changes—a push up of the mids here or there for example—that seemed to me to have no particular reason, though the results were not overall displeasing.

It is not easy to make a truly automatic room-correction system, since the psychoacoustics of how one hears sound in rooms is very complex. One could argue that the idea that there is a perfect adjustment is not really well founded in the first place, given the complexity of the situation (the ear hears first arrivals strongly but also hears later arrivals, and a single signal correction cannot deal with the two or more items independently). RoomPerfect is a good system with reasonable goals in this context, one of the best ones available.

Moreover, the system has another degree of freedom so to speak. Once you have done the correction—or indeed, even if you do not do the correction but just bypass RoomPerfect—there

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are "voicing curves" that are user selectable. These include Neutral, Music 1, Music 2, Relaxed, Soft, Open, Open Air (these latter two cut bass, where as the former all reduce treble in one way or another), and various other settings. These are all very useful not only in getting one's speakers to sound what one considers ideal, but also in adjusting things according to recordings. The general shape of the equalization applied is shown on the little screen in the front of the TDAI. The specifics-exact turnover points, number of dB—is left up in the air (I could not find it on Lyngdorf Audio's website, either). For what it is worth, I ran a few quick before-and-after measurements. The things that look small on the screen measured a couple of dB more or less, and the things that look a good bit larger (e.g., Movie) are on the order of 5dB shifts.

It is a regrettable and often ignored truth about recordings that they are not often balanced ideally. Microphones tend to be peaky, and in the past were so to an extreme; in addition, the microphones are placed too close almost always, and generally the sound of actual acoustic music in a suitable venue is seldom achieved exactly. The "voicings" of the Lyngdorf are conveniently chosen to make it possible to improve recordings effectively. However different this may be from traditional warts-and-all audiophilia, it is still well worth having. And, of course, the supposed truth of most non-adjustable and non-corrected systems is largely illusory in any case. (Look at largish collections of in-room response curves from magazines or elsewhere, and the truth will shine out at you that measuring one that is really good without DSP correction is a rare event, though not unheard of.)

In addition, the voicing curves will be an educational experience for people who have not worked at all with EQ (which rather amazingly seems to be the majority of audiophiles—talk about going into the ring with one hand tied behind your back). You will find the sound of your system surprisingly variable, since overall frequency balance is the dominant determiner of what a system sounds like. (Once you get into this, you will be amazed to find how much of audio reviews is actually commentary on exactly such "voicings" as built into the equipment under review, just not described as such explicitly. "Those who cannot recognize frequency response are doomed to review it," to paraphrase Santayana.)

The Amplification Part

The amplification part of the Lyngdorf is an unalloyed triumph. Let me get this off my chest right away. Now I know that there are people who are skeptical of digital amplification even for digital signal inputs. And I am in the slightly embarrassing position of having expressed in the past the view that analog electronics have gotten so good that they are no longer a central issue. But that, of course, does not mean that they are all exactly alike!

In fact, what is wrong with analog amplifiers is that they vary. The point is that analog circuitry is by nature full of things that matter. Parts-quality matters. Connecting wires matter. Putting "bricks" on top to cancel electromagnetic fields matters. Vibration isolation matters. Even power cords can matter. (This is a sign that something is wrong with the design, that the power cord should matter, and often enough something apparently is wrong.)

Lyngdorf's thought at Steinway Lyngdorf and its descendants was to rationalize all this, to approach every issue so thoroughly that everything would be clearly explained and made predictable. Digital, of course, is the key to this. Correctly executed digital is above all predictable. But making digital that is correctly executed is not so easy to do.

However, the Lyngdorf TDAI-2170 does precisely that, or so it seems to me. One finds nothing really to fault, and the sound gives the combined impression of delicacy and solidity that is characteristic of real music. The impression of reality is enhanced by the extremely silent background. Music emerges without any apparent electronic artifice.

At this point, it would be traditional to start describing how I heard this that or the other thing on some recording or another, telling about details discovered, felicities revealed, and so on. It would be easy to do this. My old standby recordings, familiar, perhaps all too familiar, to long-time TAS readers did indeed sound extraordinarily good. But the real point here is that I found myself completely convinced that what I was hearing were the recordings as they really are. This is not an easy point to establish in description. People are accustomed to explicit description of new and better things heard. But my feeling was rather one of correctness, of lack of electronic additions of any kind, of a quietness and control and lack of splashiness (for lack of a better word) that was providing the speakers with an input that was truly representative of the recordings.

This point is worth expanding upon. It is a universal observation that perfection in apparently simple things is always of the highest difficulty and artistically of the highest significance as well. There is a school of thought in audio that suggests simple circuitry

SPECS & PRICING

Type: Two-channel digital integrated amplifier with RoomPerfect room-correction system; pulse width modulation output; 390kHz switching rate; level adjusted by control of power supply voltage
Power output: 85Wpc, 8 ohms; 170Wpc, 4 ohms
Output filter: Second order, 50kHz
Inputs: Two analog, single-ended, two coaxial digital (up to 192kHz/24-bit) four optical digital (up to 96kHz/24-bit)
Outputs: One coaxial digital (96kHz/24-bit), one single ended analog
Optional modules: HDMI modules with four inputs (up to 19 kHz/24-bit plus DSD64/DSD128) and one output /CEC & ARC compatible; streaming USB input module (up to 384kHz/32-bit and DXD/DSD64/DSD128); high-end analog input module (three single-ended, one balanced)
Dimensions: 17 3/4" x 4" x 14 1/4"
Weight: 17.6 lbs.
Price: approx. $3999 depending on choice of optional modules

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will achieve the corresponding audio goal of complete naturalness. But this is an illusion. Just as it takes a greatly sophisticated pianist to play Schumann's *Scenes from Childhood* with perfectly natural simplicity, the corresponding audio goal is attained here not by simple-minded circuits expensively executed (which always end up with a sonic signature of their own) but by very sophisticated digital circuitry, where one comes through to the other side of perfect simplicity and naturalness. It may seem like a paradox, but it is the reality of the situation.

Some years ago, when the TacT Millennium first came out, I was present at Peter Lyngdorf's CES exhibit when some other manufacturer came by to invite him to come to his exhibit where they were showing a speaker design based on modifying a DALI speaker (DALI is a Lyngdorf company). Lyngdorf declined at first—eventually he relented out of politeness—on the grounds that he no longer liked "listening to electronics." This seemingly peculiar view becomes quite comprehensible in listening to the Lyngdorf TDAI-2170. It is in literal terms electronic with a vengeance. But the outcome, the final result, is that one feels that there are no electronics there at all.

I would not want to suggest that analog electronics cannot get close to this goal as well. The Sanders Magtech, with its carefully regulated power supply, has a similar non-electronic character—if its input does. But analog electronics. no matter how superb. are subject to the vagaries of wires and analog devices further up the chain and so on. The Lyngdorf is free of all that. Nothing happens except numbers until the final conversion to drive the speakers. And the result is in effect no electronics at all.

It remains true that the most obvious difficulties with audio are acoustical, and for all that can be accomplished with DSP correction, the largest difficulty is acoustical—the interaction between room and speaker and the behavior of the speaker itself. The RoomPerfect system deals with this issue convincingly though one should still arrange to have as acoustically good a listening room as one can. And to a surprising extent, one hears through all the acoustical issues, fundamental though they are, to the non-electronic nature of the Lyngdorf TDAI-2170. One really understands why Peter Lyngdorf said that he is no longer willing to listen to electronics. I think the Lyngdorf TDAI-2170 represents the way of the future. And it not only is here now; it is even affordable. The TDAI-2170 is a sonic and practical triumph. tas

Equipment Report

Hegel H360 Integrated Amplifier

Powerhouse

Kirk Midtskog

Hegel Music Systems, of Oslo, Norway, has developed yet another fantastic-sounding integrated amplifier/DAC. Hegel also makes preamps, power amps, and digital products, but it is its continually evolving line of integrated amps that, in a way, represents the heart of the company. Bent Holter, the founder and chief engineer behind all things Hegel, truly believes in bringing as much sonic performance, versatility, and reliability to the music-appreciating public as possible for a reasonable price. He applies his considerable engineering skills—he holds a Master's Degree in Semiconductor Physics from Norway's principal technical institute Trondheim University—to designing high-performing audio products that will work in real-world situations and can be purchased by ordinary citizens, not just well-heeled aficionados.

Background and Description

I have reviewed three other Hegel integrated amps over the past few years, so I can understand that it may seem like I am "Mr. Hegel" at the TAS table. Although other TAS writers (including Robert Harley, Neil Gader, and Jacob Heilbrunn) have also reported on Hegel gear—all positively—I am happy to review yet another Hegel integrated amp because, among other things, Hegel makes good products in general, and the company has really pulled out all the stops with the H360 in particular. It is, to give you my overall assessment upfront, a truly excellent amp. I believe it can readily compete with separates costing more than its $5700 asking price.

With 250Wpc into eight ohms (420Wpc into four) and a damping factor of 4000, the H360 will drive a wide range of speakers with ease. The H360 is equipped with two line-level inputs, one RCA and one XLR, although a home-theater bypass can be configured to function as a third unbalanced (RCA) line-level input. In addition, the H360 has a very good, on-board DAC, capable of supporting 24/192 PCM files and native mode DSD64 and DSD128. The unit also supports Apple's wireless AirPlay, and can function as a DLNA digital-media streamer/renderer so you can connect a UPnP/DLNA-compatible Network Attached Storage device (NAS) through your local router and, *voîlà*, you have an amplifier that will play a lot of different sources.

To my mind, the most important aspects of the H360's performance come from the analog sections of its preamp and power amp. After all, a fantastic DAC can fall completely short if the analog amplification is less than first-rate. For this reason, I put the H360 through its paces primarily as a standard line-level integrated amp, and only evaluated its very capable DAC once I had established what the analog sections could do. (Fortunately for me, it was through my listening to the H360's NAS streaming capability that I began to reevaluate my previously less-than-stellar impressions of digital-file playback. The DAC can do more tricks, but I will cover them further on.)

The H360 represents some of the latest engineering and manufacturing acumen at Hegel. The company's patented SoundEngine technology has been further updated, and some of the rigorous parts-matching protocols, once only applied to Hegel's top power amp (H30), are now also apparently applied to the H360. To recap, one of the main aspects of SoundEngine is a feed-forward technique that reduces noise and also specifically addresses the crossover distortion commonly found in typical Class AB amplifiers when one half of the output section hands off the waveform to the other. SoundEngine adjusts the output transistors' biasing to accommodate ever-changing temperature conditions—depending on signal fluctuations—

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SPECS & PRICING

Power output: 250Wpc into 8 ohms, 420Wpc into 4 ohms
Analog inputs: Two RCA (one switchable to HT bypass), one XLR
Digital inputs: One coaxial, three optical, one USB, one Ethernet (RJ45)
Outputs: One fixed line level (RCA), one variable line level (RCA); one digital coax (from digital inputs only); speaker terminals
Frequency response: 5Hz–180kHz
Damping factor: More than 4000 (main power output stage)
Dimensions: 16.93" x 5.9" x 16.93"
Weight: 45.2 lbs.
Price: $5700

HEGEL MUSIC SYSTEMS USA
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ASSOCIATED EQUIPMENT
Analog source: Basis Debut V turntable & Vector 4 tonearm, Benz-Micro LP-S cartridge
Digital sources: Ayre C-5xeMP universal player, HP Envy 15t /JRiver MC-20, Hegel HD12 DAC
Phonostage: Ayre P-5xe
Linestages: Ayre K-1xe, Hegel P30
Power amplifiers: Gamut M250i, Hegel H30
Speakers: Dynaudio Confidence C1 Signature, GamuT RS3, YG Sonja 1.2
Cables: Shunyata Anaconda ZiTron signal cables, Cardas Clear Reflection, Nordost Heimdall 2 USB, Audioquest Coffee USB and Hawk Eye S/PDIF, Shunyata Anaconda SPDIF, Shunyata Anaconda and Alpha ZiTron power cords
A/C power: Two 20-amp dedicated lines, Shunyata SR-Z1 receptacles, Shunyata Triton/Typhon power conditioners
Accessories: Stillpoints Ultra SS and Mini footers, Shunyata Research DFE V2 cable elevators
Room treatments: PrimeAcoustic Z-foam panels and DIY panels

rather than setting a fixed bias for average conditions. The H360's preamp section has its own transformer to keep power-supply noise in the current-supplying power amp section from interfering with the more delicate signals in the voltage-gain preamp section. The DAC has also been completely updated from the on-board DAC in the H360's predecessor, the well-regarded H300 (reviewed by Neil Gader in Issue 233). I will compare the newer H360 to the older H300 in greater detail later. While the H360 does not run hot, it uses no switching power supplies or any mix of Class D technology. It is a 45-pound Class AB amplifier all the way. The cosmetics remain classic Hegel: simple, pleasant, subtle, functionally proficient...Scandinavian.

Listening

The commanding, clean 250W output and variety of analog and digital inputs would almost be enough to recommend the H360 from the start, but Hegel offers much more than mere competency. The real boon here comes from the H360's revealing, refined, and—best of all—musically compelling character. I could hear more deeply into recordings than I had any reason to expect from a $5700 solid-state integrated amplifier. Details like singers' lip sounds, guitarists' fingers on strings, or drummers' sticks on cymbals came through with clarity, and did so without sounding hyped or forced. The ease with which these sorts of musical cues flowed, coupled with stable solidity of imaging, lent the sound a liquidity and body reminiscent of a well-balanced tube amp. Likewise, the H360's dynamic sure-footed rhythmic drive underpinned the music in a way that propelled it along and made all sorts of music interesting—also somewhat like a good tube amplifier.

The H360's tonal balance is *not*, however, traditionally tube-like (as in a bit more weighted toward the midbass and midrange with a softening of the extreme upper frequencies and perhaps a slight reduction of definition and control in the low end). On the contrary, another strong suit of the H360 is its apparent neutral tonal balance—achieved without the price of sounding clinical or characterless, as too many products with neutral ambitions do. Hegel has a talent for delivering both tonal accuracy and musicality; all four integrated amps, as well as its top P30 preamp and H30 power amp combo with which I have direct experience, have this satisfying combination of fundamentally correct tonal balance and musical verve. Hegel's VP of Sales and Marketing Anders Ertzeid told me, when I visited Hegel in Oslo in 2012, that Hegel does not "voice" its products as such; rather, it pursues accuracy and noise-reduction through engineering and leaves tonal-shaping out of the design process. Of course, designer Bent Holter and his colleagues also listen carefully to various iterations of a given design, but they seek technology-improvement solutions rather than tonal adjustments. The results reveal a recording's own character as well as the music's inherent thrust—a confluence of positive attributes I more readily find in much more expensive electronics.

The H360's midrange and treble openness really help flesh out the leading edges and trailing tails of notes, as well as their overall timbral character. This fine resolution and accurate timbre, taken together, help make images properly positioned and proportioned in the soundscape. Spatial cues add up to a reasonable approximation of 3-D imaging and soundstaging—in as much as this is possible for solid-state electronics under $10,000. For example, instrumental images do not sound recessed; indeed, leading-edge sounds indicate a distinctly closer perspective, without making instruments seem disassociated from the ensemble and the hall. Other Hegel integrateds have this pleasant "greater context" presentation as well, although the H360 portrays images better than any of the others I have listened to extensively in my system (H80, H100, H200, and H300). The H360's apparent listener perspective is basically mid-hall, and the overall soundstage is quite wide, tall, and deep. Soundstaging is one of the areas of audiophilia where separate amplification components—especially monoblock power amps—seem to hold sway. An integrated amp can match or surpass some separates in areas of resolution, tonal and timbral truthfulness, power, and dynamic control, but the expansiveness of the outer reach of the

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soundscape seems to be aided by the separation of the primary amplification blocks—all other things being similar. I will say, the H360 portrays images and a soundstage better than any other sub-$10,000 solid-state integrated I have heard in a familiar system.

Owing to robust power supplies and—as I believe Hegel would suggest—other aspects of its designs, Hegel amps tend to sound more powerful than their nominal power ratings would suggest. The H360 did not disappoint. It drove all speakers I had on hand with ease: YG Sonja 1.2, GamuT RS3, and Dynaudio C1 II. (I would hazard a guess that the H360 will even match up well with power-hungry Maggies.) Like other powerful amplifiers, the H360 conferred serenity to music listening, perhaps because it doesn't distort or strain on crescendos, as is often the case with less powerful and clean-sounding amplifiers. Bass and dynamics are well served, too. The H360's bass always sounded deep-reaching and articulate, never weak or flabby. Macro-dynamic swings could, in fact, be startlingly powerful and the power region had plenty of slam.

Even though the H360 is powerful, with lots of commanding grip and control, it still sounds beguilingly delicate and detailed. An example of this "play big" and "play refined" ability came through when I listened to the second movement of "*Three Meditations from* Mass" on *Bernstein* [Oue, Minnesota, RR]. The opening cello solo was rendered with fine detail and emotional intensity, but when the orchestra joined in and welled up, the weight and force of the ensemble was reproduced realistically and with dimensional verisimilitude. No raggedness crept in, and the soundscape did not congeal.

Comparisons

So how does the H360 compare to its progenitor, the award-winning H300? Both are rated at 250Wpc, but H360 has a damping factor of 4000 where the H300's is 1000. Thus the H360 will, theoretically, offer even greater control over difficult speaker loads. The newer model also boasts 50 percent higher current capacity. The computer-controlled analog volume attenuators remain the same, but Hegel says its new individual voltage regulators reduce high-frequency noise. The new DAC has been extensively re-designed, and much of it is actually based on Hegel's top HD30 DAC. The USB input, according to Hegel, has a new receiver chip, which supports DSD128, has better voltage regulators, and has a superior "first-level" jitter-reducing layout. The new DAC chipset is the AKM 4490 instead of the 4399 in the H300. Both models sound very similar overall, but two performance areas add up to significant improvements in the newer model: First, the H360 sounds smoother and more transparent, especially in the treble; and second, the H360 is just plain more musically enjoyable. The boogie or sadness or tension in the music registered more easily—especially when the amp was mated to the wonderfully revealing and involving GamuT RS3 speakers (review forthcoming).

What about going up in the Hegel line? The top-level P30 preamp and H30 power amp (reviewed by Robert Harley in Issue 223) sounded even more solid and commanding, and the soundscape expanded in all directions. The pre/power amp combo also sounded more revealing, direct, and immediate—quicker, so to speak. The H360 did, however,

Nordost Heimdall 2 USB Cable

Hegel's Anders Ertzeid provided a two-meter run of Nordost Blue Heaven USB 2.0 cable (starting at $249/1m) to use with both the HD12 DAC and H360 integrated amp. Because of a greater distance between my computer and the DAC, I asked about getting a longer run. Accordingly, Nordost's affable and knowledgeable Jon Baker very kindly sent along a three-meter run of Heimdall 2 USB cable (starting at $499/1m). Not only did I then have a longer length of cable to work with, but the sound quality also improved substantially. I experienced, in my own system, what others have been pointing out: USB cable can greatly impact sonic performance. The Blue Heaven USB cable was quite good, but I was impressed by how much more detail, texture, body, and spatial information came through with the Heimdall 2 USB cable in place. It all added up to a more lifelike and enjoyable musical experience.

I had heard a demo of Nordost's complete line of USB cables at Rocky Mountain Audio Fest 2011. A Nordost representative started at the bottom of the line and worked up the product offerings with ever-improving sonic performance (and higher prices) at every cable swap. (At that time, Nordost had carried more than three USB models.) The source material remained the same, as did the volume setting and the rest of the system. Only the USB cables were changed. Every cable upgrade yielded more detail, less grain, better spatial cues, and greater musical involvement. My recent experiences in my own system with this critical link in the digital chain confirmed my impressions at the RMAF demo.

Bits are supposed to just be bits in the computing world. If the digital stream makes it intact from output to the desired input with the proper interface "hand-shaking," the cable is not supposed to matter, right? Well, what constitutes "intact" on the audio side of digital signal processing may be more involved than other common computing tasks. In high-performance audio, the USB cable matters a lot. In a way, it bothers me that the USB cable turns out to matter as much as it does because it then becomes yet another factor we need to pay close attention to—as if we don't obsess over enough already. On the other hand, better sound is better, and if we know how to improve it, then why not pursue it? Such is the nature of our hobby.

The sonic improvements brought about by the Heimdall 2 USB cable were highly instructive. Other writers—TAS' Robert Harley, Steven Stone, Alan Taffel, and Neil Gader, to name a few—have been commenting on the importance of the USB cable, and I concur. Considering how much we already spend on analog signal cables, $699 for a three-meter run of Heimdall 2 USB cable seems to be in line with current industry pricing practices. **KM**

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have a more liquid and musically enticing presentation—at least when it was paired with either the Gamut RS3 or Dynaudio C1 II speakers. To my mind, the H360's ability to perform so well when stacked up against Hegel's own $21k combo is highly commendable. Hegel will probably cringe, here, but I am not at all sure the roughly additional $15k for the P30/H30 would be worth it to a lot of customers, even though the combo is technically more accomplished from an audiophile perspective.

The DAC

I compared the H360's DAC to Hegel's very nice sounding HD12 DAC ($1200) on its respective USB ports, and also listened to the H360's renderer/NAS functionality. On USB, I don't believe I could consistently tell which DAC was engaged if someone else were operating the system. If I had to really seek out (or *project*, some might assert) sonic differences, I would favor the sound of the H360. It seemed to have a little less grain and sounded a bit more natural overall. Mind you, the HD12 had compared favorably against an Oppo HA-1 DAC (also $1200) in my system; so, one could think of the H360's DAC as equaling or surpassing a $1200 separate DAC. BTW, since I have been listening to more digital audio files in the last few months, I've discovered—like many others have—that the quality of the USB cable can make a substantial difference in sound. (Please see the sidebar about Nordost's excellent Heimdall 2 USB cable.)

The H360 also supports Apple's wireless AirPlay, but the user has to supply the wireless router. Hegel did not include an onboard wireless receiver because it claims that would introduce too much noise. Besides—from my own perspective—as wireless technology advances, consumers can more easily advance with it by upgrading the external wireless router. AirPlay works but is probably more appropriate for casual listening than serious audiophile sessions at this point, sounding, in my opinion, a bit muffled and thin. It will most likely appeal to many consumers, though, because they can easily stream their music from familiar Apple devices to their home system with the H360 as the main hub.

As I mentioned earlier, the real surprise on the digital side was the H360's streamer/renderer functionality. Using BubbleUPnP software on an Android tablet, I could control the H360's renderer to play the files on the attached QNAP TS-251 dual drive (configured and pre-loaded by Hegel). Digital files sounded much more lifelike through the H360/NAS than through my HP Envy 15t laptop running JRiver MC-20 and a HD12 DAC—even when this setup was tricked out with a good power cord, power conditioning, and aftermarket footers. The H360/NAS playback was truly musically rewarding. It sounded like a hybrid between my turntable rig and my regular universal-format disc player, and all in good ways: clarity, musical fluidity, focus, and lack of underlying graininess. Soundstaging and imaging also were more fleshed out, and timbres sounded more natural. The renderer/NAS method has the potential to turn this reluctant computer-audio guy into a more receptive digital explorer. Hegel has yet another trick in its digital repertoire, though.

If you already own a good stand-alone DAC (with a coax input), and you want to make use of

Robert Harley Listens to the H360

I've long admired Hegel's electronics for their fundamental sense of musical communication and involvement. These are amplifiers that go a long way toward making you forget the playback system and just enjoy the music. I listened to an H160 for about two months recently while the Magico Q7 was being updated, and the Soulution electronics made the rounds of some hi-fi shows. I found the H160 to be a superb performer, and spent many enjoyable hours with it.

As good an integrated as the H160 is, the H360, which I auditioned in my reference system with the Magico Q7 Mk.II, is in another league. Hegel's new integrated has beautifully rendered timbre, with a smoothness and lack of grain and glare that you expect from very expensive separates. The lack of electronic artifacts overlaying instrumental and vocal textures went a long way toward engendering the relaxed engagement I felt when listening to the H360. I was also impressed by the H360's dimensionality and totally natural rendering of a recording's spatial information. Again, this level of performance isn't expected from an integrated amplifier.

But it was the H360's bass extension, weight, dynamic authority, and visceral drive that put this integrated amplifier over the top. The H360 took iron-fisted control over the Q7's dual 12" woofers and 10" mid/woofer, delivering a huge dose of physical involvement on rock and blues. Bass lines were crystal clear and dynamic, with no hint of strain from the amplifier. There was a sense of unlimited power and dynamics, even on the most demanding orchestral climaxes.

Finally, the H360's DAC is exceptional. I drove the H360 alternately with the analog output of the Berkeley Alpha Reference DAC, and with a USB source. Although not the equal of the Alpha Reference (nothing is), the H360's DAC showed that it's a big step up from the H160's DAC and a worthy addition to this outstanding integrated. In short, the H360 is a terrific-sounding amplifier/DAC, as well as an amazing bargain.

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it to improve performance, Hegel offers a neat DAC-loop feature on both the H300 and H360. You can route any digital input's signal (up to 24/192, no DSD) on the H360 through its coax output to your outboard DAC's coax input, and then route the converted analog signal from the external DAC back to the H360 through its balanced analog XLR inputs. A couple of activation button selections on the remote, and you now have cleaner, re-clocked, jitter-reduced digital-file playback. I used it with both my computer and with the NAS drive as sources, and it worked with both like a charm. Everything sounded cleaner and more continuous through the DAC-loop, with less interstitial haze, greater transparency, and more 3-D depth.

Improvements

Could the H360 be better? Sure, at least one more analog input would be nice. The home-theater bypass input should probably be left as a single-purpose input, rather than allowing it to be configured as another line-level analog input. The display doesn't bother me, but some folks might like an improved screen, in which characters are nicer to look at, rather than the mix of somewhat crude upper- and lower-case characters Hegel currently offers. I realize there are probably good reasons why Hegel has not done this already—increased cost, possibly lower reliability, and maybe added noise. (I can almost hear designer Bent Holter grumbling.)

Conclusion

The Hegel H360 is simply a marvelous piece of audio kit. Its neutral tonal balance, articulate and lovely rendering of details, commanding power reserves, spacious soundstaging, and natural imaging are laudable. At $5700, as solely a linestage integrated amp of its quality and power output, it is a bargain; the included nice-sounding and versatile DAC makes it a real winner. I absolutely loved listening to the H360. I never tired of its low noise, dynamic liveliness, and winning musicality. A very easy recommendation. tas

Equipment Report

Audio Alchemy DDP-1 Preamplifier/ DAC/Headphone Amplifier, DPA-1 Stereo Power Amplifier, and DPA-1M Monoblock Power Amplifiers

Value City

Robert Harley

Audio Alchemy blazed a trail in the 1990s with a range of ultra-low-priced products housed in utilitarian cases with no cosmetic frills. The products were almost toy-like in appearance and name—the $199 DAC-in-the-Box, for example—but contained solid engineering inside. If you could overlook the Spartan casework, Audio Alchemy products delivered exceptional performance for the money. I reviewed quite a number of these components in the mid-1990s and found them to be excellent. Audio Alchemy folded in the late 1990s, probably because it didn't build enough profit into the products' retail prices.

But that was then and this is now. The company is back, headed by industry veteran Peter Madnick, the design talent behind the original Audio Alchemy (and many products from other companies). Audio Alchemy has retained the same value orientation as before, but this first wave of products from the new company is a far cry from the black stamped-metal chassis and faceplates of the original. Instead, the new company's first offerings boast upscale casework, an extensive and modern feature set, and more ambitious engineering.

The products reviewed here are the $1995 DDP-1 linestage preamplifier/DAC/headphone amplifier, along with the $1995 DPA-1 stereo power amplifier and $1995-each DPA-1M monoblock amplifiers. All are housed in compact chassis of the same size and shape, their rounded edges and satin-silver finish exuding a decidedly upscale vibe.

The DDP-1's front panel is dominated by two large knobs, one for volume and another for input selection as well as navigating the menus. The oval display shows the input selected, the volume setting, whether the unit is locked to a digital source, the digital filter selected, and whether "resolution enhancement" is engaged (more on these features later). Four small buttons provide additional controls, including mute, selecting between headphone output and preamplifier output, and back/enter buttons that are used in conjunction with the menu/input selector knob. An 1/8" headphone jack, a feature that for many years all but disappeared from preamps but is now mandatory, adorns the front panel. The power button just below the display rounds out the controls. A well-laid-out remote handles nearly all the DDP-1's functions.

The outboard power supply, a little larger than a "wall wart," can be upgraded to a more sophisticated supply, the $595 PS-5 Power Station. The PS-5 is housed in a chassis that matches aesthetically with the DDP-1, "nesting" into that unit's curved side panel. It offers independent supplies for the DDP-1's analog and digital circuits, more elaborate voltage regulation, and more filter capacitance. Audio Alchemy claims that the PS-5 offers lower noise and wider dynamics than the stock supply.

The DDP-1's sensible array of controls and buttons, its feel, and the display itself are all superb—this is one well-thought-out user interface. The display's source-selection is unique; as you scroll through the list of inputs, the one selected becomes larger in type size. The remote is also outstanding; your index finger naturally falls on the volume up/down buttons. Even the volume-control ballistics are perfectly dialed-in; I could quickly make large volume changes, yet had fine control once I was in the ballpark. Moreover, the chassis' industrial design and metalwork are far above what's expected at this price. The compact package, with the rounded edges and satin-silver finish, is extremely attractive, and a welcome departure from the less inspired chassis work of competing products. My only complaint is that the front-panel markings are white against a silver panel, with almost no contrast. Between the low contrast and the small type, the text is difficult to read. There are, however, so few con-

Equipment Report Audio Alchemy DDP-1

trols that it doesn't take long before you're operating the DDP-1 without need for the legends. Audio Alchemy reports that they are increasing the contrast of the lettering, which, incidentally, is laser-etched in the front panel. No channel-balance control is provided.

The DDP-1 offers two unbalanced inputs on RCA jacks, one balanced input on XLR jacks, and an extensive array of digital inputs. These include AES/EBU, two TosLink optical, two coaxial, USB, and even I^2S. The USB input accepts PCM up to 216kHz/32-bit along with DSD64. The other digital inputs accept PCM only (also up to 216kHz/32-bit). Mac users can connect to the USB input and start playing music. Windows users must download a driver. You can select from four digital filters, including an apodizing filter. (To recap, an apodizing filter shifts the filter ringing in time so that the ringing occurs after the transient, rather than before and after the transient. This is an important distinction, because in nature we never hear part of a transient signal's energy *before* the transient itself. This filter "pre-ringing" is particularly deleterious to music, and contributes to the glassy hardness of textures and flat soundstaging of most digital. In my experience, there's a slight penalty in bass tautness and definition with apodizing filters, but it's a worthwhile tradeoff.)

Through the front-panel display and controls, you can select any one of the filters as the default for a particular input. Similarly, resolution enhancement can be turned on and off for the individual inputs. The front-panel "Enh" legend turns green when resolution enhancement is on, red when off (see sidebar for more detail on resolution enhancement).

An important consideration when buying a DAC today is whether the its software can be updated to decode Master Quality Authenticated (MQA). I've written extensively about this new technology (Issues 253 and 261) that greatly improves digital sound quality. Because the DDP-1 is a purely software-driven product that runs on two XMOS general-purpose DSP chips and a field-programmable gate array, it may be possible that the DDP-1 can up updated to offer MQA decoding. Although Audio Alchemy hasn't committed to this possibility, it's worth noting that the demonstration board MQA has provided to manufacturers runs on the same XMOS chip used in the Audio Alchemy DAC, and that the Alchemy's software can be updated via the read-panel micro-USB port.

Overall, the DDP-1 is a highly capable and versatile centerpiece of a system that's a pleasure to use on a daily basis.

Looking next at the DPA-1, this stereo power amplifier delivers 125Wpc into 8 ohms and 200Wpc into 4 ohms. The front panel offers more features than are traditionally found on power amplifiers, including selectable gain (a +6dB button), clipping indicators, a mute button, and soft-start warm-up. Both balanced and unbalanced inputs are provided, and the binding posts are of high quality. A 12V trigger input allows connection to the DDP-1 (or other product with 12V trigger output) so that powering on the DDP-1 automatically powers on the amplifier as well. The DPA-1M is simply a monaural version of the same amplifier, delivering 325W into 8 ohms and 400W into 4 ohms. At the most recent CES, Alchemy announced the DPA-2 stereo amplifier with 250Wpc ($2995). The company also showed the matching PPA-1 phonostage and the Roon-ready DMP-1 Media Player, both of which are $1795.

The amplifier features a Class A input stage built from discrete FETs, the same topology found in expensive amplifiers. Most amplifiers at this price rely on op-amps rather than discrete circuits. The output stage is Class D, which explains the DPA-1's compact size and light weight—the amplifier weighs just 16 pounds. Specifically, the output stage is a Hypex UcD module, designed by Bruno Putzeys. The DPA-1M monoblock simply bridges two of these modules for greater output power.

From first impressions, these new products from Audio Alchemy appear to be quite a step up from those of the company's first incarnation.

Listening

I was eager to review the new generation of Audio Alchemy products for several reasons: I was a fan of the company's earlier offerings; I have great respect for the design talents of Peter Madnick; and most importantly, I heard the DDP-1 and DPA-1M sound amazingly great in very-high-end systems at several shows. One of those show systems (Munich) featured TAD CR-1 loudspeakers (perhaps the best stand-mount speaker extant) and another (Rocky Mountain) showcased the Alchemy products with the outstanding Wil-

SPECS & PRICING

DDP-1 Linestage Preamplifier/DAC and Headphone Amplifier
Analog inputs: One balanced, two unbalanced
Analog outputs: Balanced on XLR jacks, unbalanced on RCA jacks, 1/8" headphone jack (plus 12V trigger)
Digital inputs: Coaxial (x2), TosLink (x2), USB, I^2S (additional micro-USB for software updates only)
Digital format supported: Up to 192kHz/24-bit on all inputs, plus DSD64 on USB input
Digital filtering: Custom, with four user-selectable filters
Outputs: Balanced and unbalanced
Headphone amplifier power: 1W into 32 ohms
Input impedance: 50k ohms
Output impedance: 75 ohms
Channel separation: 100dB (digital input), 130dB (analog input)
Dimensions: 10.5" x 3" x 11.6"
Weight: 8 lbs.
Price: $1995

PS-5 Power Station (for DDP-1)
Dimensions: 5.5" x 3.5" x 11.6"
Weight: 9 lbs.
Price: $595

DPA-1 Stereo Amplifier
Output power: 125Wpc into 8 ohms, 200Wpc into 4 ohms
THD: 0.05%, 1W into 8 ohms
Input impedance: 100k ohms
Output impedance: 0.06 ohms
Gain: 20dB or 26dB (switchable)
Channel separation: 80dB
Dimensions: 10.5" x 3" x 11.6"
Weight: 16 lbs.
Price: $1995

DPA-1M Monaural Power Amplifier
Output power: 325W into 8 ohms, 400W into 4 ohms
THD: 0.05%, 1W into 8 ohms
Input impedance: 100k ohms
Output impedance: 0.06 ohms
Gain: 20dB or 26dB (switchable)
Dimensions: 10.5" x 3" x 11.6"
Weight: 16 lbs. each
Price: $1995 each

son Sabrina speakers. The Alchemy gear more than acquitted itself in this illustrious company.

Speaking of illustrious company...I dropped the DDP-1 (with the PS-5 supply) and a pair of the DPA-1M monoblocks into my reference system. After three days of warm-up, I began by listening to LPs, driving the DDP-1's balanced analog input, with the DPA-1M monoblocks powering Magico Q7 Mk.IIs. I was immediately impressed by the Alchemy's sonic virtues and ability to communicate the music. The sound was remarkably transparent, clean, dynamic, and resolved by any measure, and even more so considering the components' reasonable price.

The Alchemy products threw a large and well-defined soundstage, with outstanding depth, dimensionality, and separation of individual instrumental lines. On "Mars" from *The Planets* (Mehta, LA Philharmonic, Decca), the insistent snare drum that drives the rhythm was well back in the stage, with a real sense of air and space around it. The call-and-response lines of the tenor tuba and trumpet were well differentiated from each other and from the rest of the orchestra. The sense of size and scale was outstanding. Other hallmarks of the products were clarity and transparency—the sense of nothing between you and the music. The soundstage lacked the veiling that diminishes the sense of realism of instruments at the back of the stage.

With smaller-scale music, the Alchemy electronics showed that they were transparent enough to reflect a recording's spatial character. Intimate music, like Joni Mitchell's *Blue* (LP reissue), was rendered with the appropriate sense of presence and immediacy.

Perhaps the most salient characteristics of the DDP-1 and DPA-1M, however, were powerful rhythmic drive, wide dynamic expression, and rock-solid visceral grip in the bottom end. The timpani in "Mars" was taut, powerful, deep, and dynamic. Bass guitar had a solid feel that was simultaneously full and tight, combining timbral warmth and body with outstanding pitch definition and articulation. Kick-drum cut through the mix with a solid impact. Switching to the less powerful DPA-1 stereo amplifier, I heard no reduction in dynamic range, bass control, or bottom-end extension, at least driving the 94dB-sensitive Magico loudspeakers. (Less sensitive speakers may benefit from the monoblocks' greater output power.) Both the stereo and the mono versions of this amplifier sounded like indefatigable powerhouses, with plenty of dynamic headroom. I never heard the amplifier soften the bass, harden textures, or congeal the soundstage, no matter what the playback level or how demanding the music.

This powerful rhythmic expression wasn't just the result of terrific bass grip and definition. The DDP-1 and DPA-1 excelled at portraying transient information, such as drums and percussion. The Alchemy electronics were fast and dynamic, qualities that brought to the fore subtle rhythmic nuances by great drummers, allowing their kits to take on a lifelike quality. The contribution from the great Roy Haynes on the track "Windows" from the album *Like Minds* (Gary Burton, Chick Corea, Pat Metheny, Dave Holland, and Haynes) was highlighted by the Alchemy electronics. On the track "Helena" from Gary Burton's *Guided Tour*, drummer Antonio Sanchez (who, incidentally, composed and performed the soundtrack for the film *Birdman*, for which he won the Academy Award in 2015) lets loose with a *tour de force* solo that was well served by the Alchemy's outstanding speed and immediacy. Similarly, the timbales on the outstanding Mobile Fidelity reissue of Santana's *Abraxis* fairly jumped from the soundstage as though they were recorded yesterday.

When listening to LPs, I thought the overall sound was a bit laid-back in the midrange to the lower treble, with vocals slightly recessed in the mix. The DDP-1 and DPA-1Ms were at the other end of the sonic spectrum of electronics that are bright and forward in this region. This was a good sign, because I've selected for these qualities in my LP front end (Basis Inspiration turntable with Basis Superarm 9 and Air-Tight PC-1 Supreme cartridge), which leans toward a less incisive rendering than many vinyl playback systems. I'm no fan of moving-coil cartridges that are tipped up in the treble or that hype detail. In other words, the DDP-1's linestage section and the DPA-1M sounded like my LP front-end sounds; the Alchemy electronics managed to pass along the LP playback system's character with very little editorializing. This level of transparency to sources in a product of this price is remarkable, particularly when considering the quality of the LP front-end and the resolution of the Magico Q7 Mk.II speakers. These reference-grade components would have laid bare any added brightness, hardness, opacity, or reduction in dynamic expression.

When I switched to a digital source (the Aurender W20 via USB) and was listening to the DDP-1 as a DAC and preamplifier, all the virtues mentioned were present, but now the music had greater verve and illumination. The sound was a bit more immediate and upfront, reflecting the DAC's character compared with that of my turntable. It didn't take a lot of careful listening to realize that the DDP-1's DAC is spectacular—highly resolved, open, transparent, and extremely dynamic. The DAC is very lively and incisive, with a full measure of detail. As with the DPA-1 amplifier, the DDP-1's DAC excels at reproducing transient information, from the micro to the macro. The DAC's sound can be fine-tuned through filter selection; I opted for Filter 4, which has a more "gentle" sound than the other three.

The DAC's sound could be improved by engaging the resolution enhancement feature described earlier (and in the sidebar). Turning on resolution enhancement seemed to make the overall perspective a little less immediate and upfront, as though the entire stage moved back slightly. Put another way, engaging resolution enhancement was like moving from Row G to Row M. Resolution enhancement better resolved the space around individual instruments, and soundstage width and depth expanded. Reverberation tails were longer and better defined. On the 44.1kHz/16-bit recording *Aras* by the band Curandero, the first track begins with some sharp percussion work. Engaging resolution enhancement not only expanded the space around the percussion, but I could hear more detail and texture in the drumhead's decay, and more resonance of the air within the bodies of the drums. On the track "Switchback" from Jesse Cook's *Free Fall*, the multiple rhythm acoustic guitars behind the lead guitar were more clearly distinguishable as individual instruments, and they had a more immersive sound. That is, the soundstage was more continuous horizontally, with less impression of sound coming from two loudspeakers. The background guitars were also farther back in the

Equipment Report Audio Alchemy DDP-1

mix, increasing soundstage depth. The intricate horn and woodwind lines in the contemporary big-band music of Gordon Goodwin were more clearly resolved. Resolution enhancement also benefited the Alchemy's rendering of timbre, which was a little smoother, particularly in the upper midrange. Overall, resolution enhancement contributed significantly to my view that the DDP-1's DAC section is not only terrific in an absolute sense, but nothing short of amazing in a $1999 full-featured preamplifier.

Finally, I'll comment on the PS-5 power supply and the differences between the stereo and mono amplifiers. Compared with the stock power supply, the PS-5 vaults the DDP-1 into a different league. The sound with the PS-5 is more refined, spacious, and detailed. Instrumental textures are more liquid and natural. The upgraded supply also gives the sound much greater dimensionality, with a heightened sense of layering and depth, along with more air between instrumental images. I auditioned the DDP-1 only briefly with the stock supply because the sound was so much better with the PS-5. My description of the DDP-1's sound is with the PS-5. It's a worthwhile upgrade.

The DPA-1 stereo amp gives up nothing in sound quality to the monoblocks, except output power. The DPA-1's 200W into 4 ohms was plenty of power for the 94dB-sensitive Magico Q7 Mk.II. In fact, I never saw the clipping LEDs illuminate, even at high listening levels. Of course, if you're driving loudspeakers of lower sensitivity the additional power provided by the monoblocks will come in handy, but don't jump to the conclusion that you need the monoblocks. The cost difference between the complete package (a DDP-1 with its power supply) with the stereo and mono amps is $4600 vs. $6600—quite a jump. The best way to tell if the DPA-1's output power is enough for your loudspeakers, room size, and listening levels is to borrow one from your dealer and try it. There's simply no substitute for auditioning an amplifier in your own system.

Conclusion

These new products are a far cry from the Alchemy of yore, with much more advanced engineering, upscale casework, and a superb user interface. The DDP-1 and DPA-1 bring terrific sound and stunning value to the category. As a linestage, the DDP-1 is amazingly clean and transparent. Unlike most electronics of this price, the DDP-1 doesn't add a patina of electronic hardness over instrumental timbres. Nor does it add opacity to the soundstage or compress dynamics. The DDP-1's DAC section is simply sensational; this level of sound quality would be outstanding in a $4000 stand-alone DAC. Clarity, openness, detail, and exceptional dynamics define the DAC's performance.

The DPA-1 stereo amplifier and DPA-1M mono amplifiers are no less impressive. Their wide dynamics, terrific grip in the bass, and upbeat sonics made them a joy to listen to. Moreover, the amplifiers possess the same level of clarity and resolution as the DDP-1. Significantly, the amplifiers don't exhibit the shortcomings I've heard in previous Class D designs. Even in the context of reference-quality sources and loudspeakers, it was easy to forget that I was listening to electronics that aren't stratospherically priced.

The return of Audio Alchemy is welcome news for those seeking the highest possible price-to-performance ratio in electronics today. tas

DDP-1 Tech Tour

The DDP-1 incorporates a number of advanced technologies and circuit topologies that reveal its ambitions as a high-end product. First, the entire analog signal path is based on discrete Class A circuits rather than op-amps. On the digital side, the DDP-1 features dual AKM DAC chips in a proprietary configuration that reportedly increases dynamic range. The filtering and digital processing is performed on a pair of XMOS general-purpose DSP chips, followed by a field-programmable gate array. These DSPs perform the digital filtering and resolution enhancement.

The digital input stage is built around a dual phase-locked loop (PLL) architecture, a technique pioneered by Alchemy more than 20 years ago. The first PLL locks to the incoming data; the second PLL locks to the first PLL and generates the clock. This technique isolates jitter in the incoming data stream and creates a low-jitter clock that serves as the timing reference for the digital-to-analog converters.

One of the original Audio Alchemy's most ambitious and successful products in the mid-1990s was the DTI-Pro (and later, the DTI-Pro 32) that offered a "resolution enhancement" technology. The DTI-Pro was a purely digital device that was inserted between a CD transport and a DAC, allowing the user to selectively increase the DTI-Pro's output word length to 18 bits, 20 bits, or 24 bits to match your DAC's capability. When the DTI-Pro was introduced, digital-to-analog converters varied in how many bits they could handle. DACs with the Yamaha input receiver truncated incoming data to 16 bits, which introduces significant distortion. Those with the NPC digital filter truncated to 18 bits. DACs with the Pacific Microsonics PMD100 filter could handle up to 24 bits, but in some implementations, the DAC's architecture provided a data path of only 16 or 18 bits. The DTI-Pro thus allowed you to select the appropriate output word length for your particular DAC.

But in today's world, 24-bit (or wider) data paths and DACs are standard. The DDP-1's data path is 32 bits wide, and the AKM DAC can accept 32-bit input words (this doesn't mean that it has 32-bit resolution). The DDP-1's resolution enhancement algorithm knows this and redithers the data to 32 bits for input to the DACs no matter what the word length of the incoming data.

The resolution enhancement is most effective on data coming in on the USB input, and less so on the other digital inputs. Audio Alchemy is working on a software update that will apply resolution enhancement equally across all digital inputs.

Incidentally, the resolution enhancement in the DDP-1 was designed by Keith Allsop, who created the original resolution-enhancement algorithm for the DTI-Pro more than 20 years ago. He and Peter Madnick have worked together continuously since that time. Finally, it's worth noting that the DDP-1's DSP horsepower is greater than ten times that of the DTI-Pro.

Our Top Picks Disc Players

Rotel RCD-1570

$999

Long ago Rotel demonstrated that high-end sound need not come at a high-end price. Exhibit A was its now-legendary CD player—costing a mere $400—that outperformed units ten times its price. Rotel's new stack carries on that tradition, with three components that—aesthetically and functionally—were obviously designed to be deployed in tandem. First in line is the Wolfson DAC-powered RCD-1570 CD player. This slot-loaded unit has both single-ended and balanced analog outs. (There are also RS-232C and Rotel Link connections for external control.) A nice additional touch: The RCD-1570 has a digital out so it can be used as a transport in the event its owner decides to spring for a more expensive DAC. But even when used as a stock player, the RCD, like its now-famous forebears, makes few sonic compromises. rotel.com (242)

Oppo BDP-105D

$1299

For enthusiasts who like their audio how they like their pizza—with everything on it—the BDP-105D is quite the earful. It's a physical media lover's dream yet it also offers all the media player/renderer connectivity and Oppo's own media app control for the computer music aficionado, plus media support for Tidal and Netflix among others. Also unique to this Oppo model is an internal headphone amp to power a variety of cans (such as the maker's own planar-magnetic line). Driven by the ESS Sabre32 Reference DAC, sound quality was very good to excellent across all formats. Only SACD seemed to lack the micro-detailing, continuity, and immersiveness of the top-tier dedicated players. Cinemaphiles and multichannel mavens, take note: Blu-ray disc performance was stunning, and the Darbee Visual Presence video processing was an intriguing addition for optimizing picture quality. Well built and attractive, the BDP-105D is a true, one-box crossover solution for both physical and optical media—and for computer-audio fans. oppodigital.com (reviewed in this Guide.)

Moon by Simaudio Neo 260D

$2000 ($3000 w/DAC)

The Moon Neo 260D continues a tradition of fine CD players from Canada's Simaudio. However, unless you are a CD-only loyalist, you really need to consider adding Simaudio's $1000 high-resolution DAC section to the 260D. With a 32-bitasynchronous convertor and four rear-panel digital inputs (dual SPDIF, a TosLink, and a USB), this optional DAC effectively opens up a whole new world of digital connectivity. Standard CD playback, though expectedly excellent, pales next to the level of refinement that the DAC brings to the table on high-resolution material —an added complexity of dimensionality that almost seemed to re-inflate the soundstage. The DAC's superior reproduction of micro-dynamic gradations also more convincingly recreates the distances among the players in a symphony orchestra. With or without the optional DAC, the 260D offers natural sonics elegantly mated with resilient build-quality and good ergonomics. simaudio.com (244)

Aesthetix Romulus

$7000

This all-tubed CD player and DAC is one of the great bargains in high-end audio today. What makes the Romulus special is that it sounds so "non-digital." Rather than sounding flat and congealed, it opens up the spatial presentation, giving instruments and voices room to breathe. The Romulus couples this expansiveness with an unusual (for digital) sense of top-octave air and openness. The tonal balance is rich and warm in the bass, which, when added to its treble smoothness, results in an immediately engaging and fatigue-free presentation. The Romulus doesn't sound "tubey" in the classic sense, but neither does it sound like solid-state. The design and build-quality are beyond what's expected at this price. If you have no analog sources, the Romulus can serve as a preamplifier and DAC with multiple digital inputs, provided you purchase the variable-output option ($1000). Thanks to an innovative hybrid analog/digital volume control, there's no loss of resolution no matter the volume setting. aesthetix.net (243)

Our Top Picks Disc Players

Esoteric K-03X

$12,000

A brilliant concept beautifully executed, the Esoteric K-03X is much more than a CD/SACD player. It is also a full-fledged DAC, with ample inputs, multiple upsampling and filtering options, and even provisions for an external word clock. Its USB interface is state of the art, supporting the highest resolutions and asynchronous clock control. As a DAC, the K-03X has few peers. Both the SPDIF and USB interfaces are among the best AT has heard. In either case, rhythms are unflagging, details emerge clearly and naturally, and listener fatigue is non-existent. Dynamics are superb as well, and the sound is always open and airy. The K-03X also excels as a disc player, especially when playing SACDs. CD sound is not quite up to the K-03X's benchmark in other modes, but it is ravishing nonetheless. Though it is not cheap, the K-03X delivers a level of versatility, build quality, sound, and operational smoothness that fully justifies its price. esoteric-usa.com (261)

Meridian 808v6

$22,000

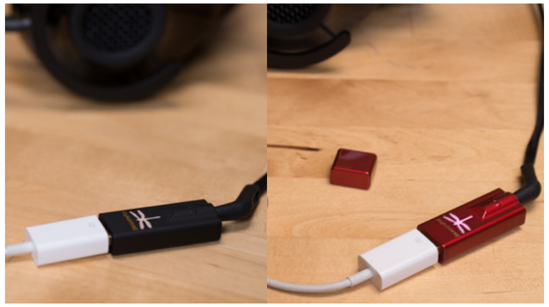
This update to Meridian's flagship CD player/DAC incorporates several performance improvements, but more significantly, adds decoding of Master Quality Authenticated (MQA) files. Even when decoding conventional digital, the 808v6 is in the top echelon of digital playback, with a smooth tonal balance, superb dynamics, and absolutely rock-solid and extended bass. But feed it an MQA-encoded file and the 808v6 takes on a whole new life, with tremendous dimensionality, tangible air between images, utter liquidity of timbre, and more realistic transient reproduction. meridian-audio.com (263)

T+A PDP 3000 HV

$22,500

This CD/SACD player and DAC from Germany's T+A may be the best all-around value in digital today. Solidly built and a joy to use, the PDP 3000 HV, features a custom-made transport mechanism made mostly from metal, rather than plastic, parts. As part of its no-compromise approach, the PDP 3000 HV features completely separate signal paths, DACs, and even analog-output stages for PCM and DSD sources. When playing DSD, the PDP 3000 HV uses different filters depending on the DSD rate. Sonically, the T+A performs with the best of them when decoding PCM sources, and offers the finest SACD playback RH has heard. (reviewed in this Guide.)

Our Top Picks DACs



AudioQuest DragonFly Black and DragonFly Red

$99/$199

AudioQuest practically invented the low-cost, high-performance USB DAC in stick form with the original DragonFly. It was a massive success. But these two new models greatly improve on the sound of the original, and the $99 Black version comes at a lower price, to boot. The Black is smoother than the original, with more extended bass. Although both sound superb and are amazing values, the Red at $199 delivers striking sound quality, with exceptional transparency, resolution, timbral realism, and wide dynamics. Add AudioQuest's $49 JitterBug USB isolation device to either and take the performance up another notch. The Red with a JitterBug is good enough to use as a front end in a budget high-end home-based system. audioquest.com (review forthcoming)

Resonessence Labs Herus

$350

The Canadian-made Resonessence Labs Herus is one of the most flexible USB-powered DACs in sample- and bit-rate capabilities. This lipstick-sized unit supports PCM up to 352.8/24 as well as DSD64x, DSD 128x, and DXD files. So, regardless of how you like your high-resolution files, the Herus will play them. Machined out of a solid block of aluminum, the Herus measures 2.5" x 1.25" by .75" and weighs less than a pair of CD jewel cases. On native 128X DSD sources it offered a level of sound quality that rivaled that of any DSD DAC SS has heard, regardless of price. resonessencelabs.com (245)

Wyred 4 Sound DAC-2

$1499 ($100 to add DSD, $1000 for SE boards)

The Wyred 4 Sound DAC-2 combines a rich feature set with remarkable performance at a price that makes it hard to beat. Its overall sound has a solidity and weight that are both arresting and involving. While SS hasn't heard every available DAC in its price range, he has yet to hear any USB DAC under $1500 that outperforms the Wyred4Sound. Factor in the basic DAC-2's 192kHz high-resolution capabilities, small upcharge for DSD support, and the ability to convert to SE anytime you wish via built-in circuit-board upgradability, and you have a DAC that will remain *au courant* long enough to make it a savvy and satisfying purchase, regardless of how much more you can afford to spend. wyred4sound.com (239)

Mytek Digital Brooklyn

$2000

The Mytek Brooklyn is the first non-Meridian-branded DAC that supports MQA. Because of that, every time it's been shown, whether at a consumer or an industry event, it has generated practically standing-room-only interest. The Brooklyn is not only a DAC, but also a preamplifier for both analog and digital sources, a headphone amplifier that supports single-ended and balanced cans, and a phono preamplifier for both moving-coil and moving-magnet cartridges. So far SS has been unable to discern anything sonically negative while listening to MQA encoded files though the Mytek Brooklyn. Even without MQA the Mytek Brooklyn offers exceptional value due to its versatility, flexibility, ergonomic elegance, and overall high level of sonic performance. Once you throw MQA into the equation I have to say "game over" for any DAC or DAC manufacturer who can't keep up. mytekdigital.com (265)

Our Top Picks DACs

Auralic Vega

$3495

If you are looking to take the plunge into the world of DSD and need a high-quality DAC/preamp capable of handling all your digital sources, look no further than the Auralic Vega digital/pre DAC. With AES/EBU, two coax, optical, and USB inputs, the Vega is highly versatile. Because it is also capable of acting as a preamp, all you have to do is add an amp and speakers and you're ready to start rocking. The Vega supports all PCM-based audio up to 384kHz/24-bit and DSD up to DSD128. As good as it gets for the price. auralic.com (240)

Berkeley Audio Design Alpha DAC Series 2

$4995

The Product of the Year Award-winning Alpha DAC is not only one of the best-sounding digital-to-analog converters, it's also an amazing bargain. In addition to world-class decoding of CD sources, the Alpha DAC can handle any sampling rate to 192kHz and word lengths to 24 bits. Its robust analog output stage and variable output level allow it to drive a power amplifier directly. This feature is significant, because the Alpha DAC is capable of such resolution, timbral purity, and dynamics you'll want to hear it without the limitations of a preamp in the signal path. When used at its best—fed by true high-res sources from a music server, and driving an amplifier directly—the Alpha DAC delivers stunning resolution of the finest musical detail, throws a spectacularly large and well-defined soundstage, and plays back music with gorgeous tone color and purity. It lacks a USB input, but you can add Berkeley's Alpha USB converter for the capability. berkeleyaudiodesign.com (189)

Esoteric N-05

$6500

The vaunted Japanese brand Esoteric has entered the network-player category in high style with the N-05 Network Audio Player. Just connect the N-05 to your network, add a USB or NAS drive, connect an iPad to your wireless network, and you're ready to access your music library via Esoteric's iPad app. Integral Tidal streaming expands musical offerings beyond your file-based library. Of course, the N-05 will also function as a conventional DAC, providing standard SPDIF, TosLink, and USB inputs. Sonically, the N-05 renders an expansive soundstage of considerable realism, projecting instruments in space with air around them and no smearing of images. Dynamic contrasts are well captured, particularly microdynamic nuances. Timbres are free from edginess and distortion. The build-quality and chassis work are typical Esoteric—that is, drop-dead gorgeous. With the N-05, Esoteric's first entry into the network audio player market is a winner. esoteric-usa.com (268)

Berkeley Audio Design Alpha DAC Reference Series 2

$19,500

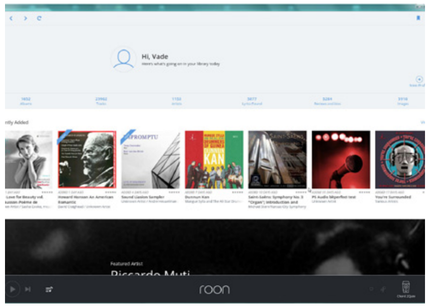
Berkeley took what was the finest-sounding DAC extant, its Alpha DAC Reference, and significantly improved it with the new Series 2 version. The Series 2 has smoother and more natural rendering of timbre, finer resolution of detail, a more transparent presentation, and, perhaps most importantly, a dynamic openness that greatly increases musical engagement. This new DAC's smoothness doesn't come at the expense of liveliness or tone color through the brilliance range; it somehow manages to combine liquidity with resolution, transient speed with lack of etch, and information density without fatigue. And it does this even with CD-quality files. Note that the Alpha Reference lacks a USB input; you'll need Berkeley's Alpha USB converter ($1895). The original Alpha DAC Reference was priced at $16,000; the Series 2 is $19,500. Owners of the original can upgrade for the $3500 difference. Digital doesn't get any better than this. berkeleyaudiodesign.com (266)

Linn Klimax DS and Klimax DSM

$23,375 (Klimax DS), $27,500 (Klimax DSM)

Best known for its venerable LP12 turntable, Linn Products was one of the first brands to abandon CD and SACD for file-based digital streaming, and the company's first Klimax DS was one of the reasons for the switch. Now in its third iteration, the Klimax DS streamer and Klimax DSM streamer with built-in preamplifier raise the performance bar to new heights. Central to the latest, and fully retrofit-able, Klimax is the new Katalyst DAC architecture, which uses a sophisticated circuit to create better power supply feeds to individual subsystems in the DAC chip, and greater clock precision. Existing Klimax owners can update their streamer, with the option of getting their original Klimax back in a basic "Renew DS" case. Although the existing Klimax was one of the best digital streamers in production, the latest Katalyst or "DS/3" model towers over its predecessor in virtually all aspects of performance. The change is so significant you can hear the difference in comparisons in only a few seconds. linn.co.uk (268)

Our Top Picks Music Servers



Roon Labs 1.1 Computer Audio Playback Software

$119/yr. ($499, lifetime subscription)

The Roon music management program offers a rich interface with way more information about the music in your collection than any other program reviewer Vade Forrester has seen. Its flexible and easy metadata-editing tools make it easy to fix the inevitable errors that creep into Roon's (and any other playback program's) graphic display of your albums. VF found playing music on Tidal easier through Roon than through Tidal's own playback program. Most importantly, Roon just sounded good—a little different from, and in some ways better, than J. River. roonlabs.com (258)

Channel D Pure Music2

$129

Pure Music is a great piece of software at a price that even a flea market-scrounging hobbyist audiophile can afford. Combine Pure Music with any recent Mac computer and you have a front end that will play back any digital file from FLACs to lowly MP3s on up to 192/24 high-resolution files with ease. Mate this front end with a top-flight DAC and you have a digital playback system that will catapult you to the forefront of the new computer-playback revolution. channld.com (211)

Sony HAP-Z1ES

$1999

As the poster boy for Sony's "High Definition Music Initiative" the new Sony HAP-Z1ES defines what Sony sees as the future of two-channel audio. It attempts to be easy for a naïve user to operate, yet capable of the highest audio quality. As SS put the HAP-Z1ES through its paces he looked for reasons it might be not be considered a true high-performance component—and found none. If you plan to spend more than $2000 on any digital front end—be it an audio-computer, CD player, DAC, network player, or any other front end that uses digital files as a source—and you don't audition a HAP-Z1ES, you are ignoring what may well be a benchmark digital product. sony.com (242)

Aurender N100H

$2695

Aurender's N100H brings you a surprising amount of the technology, sound quality, and outstanding user experience of the flagship W20 for a fraction of the price. You don't get features such as dual-wire AES and clock input, but most users don't need those capabilities anyway. The internal storage is 2TB rather than 12TB, but you can always add a NAS drive for more capacity. What you do get is the same outstanding Conductor app, Tidal integration, and Remote Support. Aurender's Conductor app for iPad is by far the best RH has used—fast, visually appealing, stable, intuitive, capable, and uncluttered, with features that have been clearly refined through actual use. Sonically, the N100H comes pretty close to the W20's state-of-the-art performance, particularly considering the cost differential. aurender.com (258)

AVM Evolution MP 5.2

$5999

AVM's mid-range media player is an elegant solution that embodies the streaming network future and CD playback past in one finely crafted component. With a tube linestage based on a pair of ECC 83 valves, the sonic performance of the MP 5.2 exuded a glassy smooth tonal character with appealing intimations of midrange and top-end warmth. The more one listens to orchestral music, the greater the appreciation for the AVM's personality, which offers something akin to the physical presence of musicians in performance. String instruments like cello and bass violin have the -bodied, weighty voices, appropriately stout resonant foundations, and timing and tonality that just don't quit. With wireless capability and DSD via USB and a downloadable control app, the MP 5.2 offers the best of all digital worlds in a well-honed package. avm-audio.com (263)

Our Top Picks Music Servers

Baetis Audio Reference
$13,995

Baetis vociferously rejects the standard deployment of the universal serial bus (USB) as the default digital interface between a music computer's motherboard and a DAC, maintaining that transporting audio data within a USB signal generates deleterious digital noise. Earlier Baetis designs achieved notable sonic results with a coaxial SPDIF output terminated with a BNC connector; now the company has determined that a higher-voltage SPDIF takes the playback of high-resolution stereo audiophiles to an even higher level of fidelity. To AQ, the Baetis Reference achieves the closest approach yet to musical realism in digital's long-frustrating history. baetisaudio.com (258)

Aurender W20
$17,600

Aurender's top-of-the-line W20 is one of the most feature-laden and capable turnkey music servers on the market. It also happens to have the best music-management app, an important consideration when choosing a server. Load the W20's internal hard drive (up to a whopping 12TB) with music, connect one of its many digital outputs to a DAC, link a tablet to your wireless network, and you've got virtually unlimited music. Seamless integration with streaming service Tidal greatly expands the W20's functionality. You'll need, however, a Mac or PC with an optical drive to rip to the W20. The W20's sound quality is outstanding, perhaps in part due to its 240GB internal cache memory, battery power supply for critical circuitry, and other performance-oriented design tricks. aurender.com (258)

Burmester MC151
$25,000

This elegant and capable music server beautifully blends high technology with luxury. It combines in one chrome-plated chassis a CD ripper and 2TB of storage, and has the ability to play music from a streaming service, USB stick, or external drive. What's more, the MC151 has variable output levels and source switching, making it a fully capable preamplifier. Burmester's iOS app is outstanding; it is easy to find and play back music. Music stored on the MC151 can be accessed by any other UPnP device on the network. The crowning glory of the Burmester MC151, however, is not its features, but its sound quality. It brings out exceptional upper-octave life and air, but still keeps the midrange warm and natural. Bass is equally excellent. burmester.de/en (255)

Our Top Picks Portable Players

Astell&Kern AK Jr

$299

The AK Jr is only 4.5" by 2" and 3/8". It will slip easily into almost any pocket you choose, except for the change pocket of your jeans. Unlike many portable players that include a balanced headphone output and claim to be able to drive every transducer that anyone has ever placed on his head, the Astell&Kern AK Jr is designed to power reasonably efficient headphones. Configured around a single Wolfson WM8740 DAC, it supports up to 192/24 PCM as well as DSD64 via conversion to PCM. The beauty of the AK Jr is that it can work with a wide variety of headphones without needing additional gear. Couple it with one of the many headphone options available in the $300 to $500 range, and for under $1000 you, too, can have a wonderful portable rig that delivers superb fidelity and simply slides in your pocket without any unsightly bulges. Indeed, the only things that are really junior about the AK Jr are is its size and its price. astellnkern.com (review forthcoming)

Astell&Kern AK100 II and AK120 II

$699, $1499

These portable players are best thought of as iPods on steroids. With their finely-brushed black aluminum cases and intuitive controls, they give up nothing to Apple in industrial engineering. But iPods max out at a tepid 48/16 resolution, whereas the AKs go to 192/24. The AK120 will even play DSD files! Sonically, these players simply stomp modern-day iPods and iPhones, which sound dull and dreary by comparison. Even on moderate-resolution material, the AKs deliver high-end qualities like timbral richness, airiness, detail, and pace. And once you have held hi-res in your hands, you will never settle for less. The AK120 boasts dual Wolfson DACs and twice the memory capacity (a precious resource when storing hi-res material) of the AK100. The flagship also has marginally more air, a smidge less grain, and stronger bass. Both players constitute wild successes, bringing true high-end sensibility and performance to portable music. astellnkern.com (236)

Onkyo DP-X1

$799

One of the latest manufacturers to toss its portable player's hat into the ring is Onkyo. Its DP-X1 offers a unique set of features and capabilities at a highly competitive price. The DP-X1 uses two amplifiers and two digital/analog converters to deliver a true balanced signal. While the DP-X1 may not be quite as disruptive a new technology as MQA, it does call into question why, except for aesthetics or ergonomics, anyone would choose another player if his budget maxed out at under $1000 (except perhaps the Pioneer XDP-100R, if he were absolutely sure he would never, ever, need a balanced output). Reviewer Steven Stone predicts that Onkyo will sell a lot of DP-X1 players because it is currently the best value out there in flexibility, functionality, and sound. Recommended? Is that even a question? Onkyo has hit a home run that deserves two trips around the bases. onkyousa.com (266)

Sony NW-ZX2

$1199

Sony, which created the first "Walkman" portable player, has been involved with portable audio since its inception, but lately has not been the dominant player it was in the early days. This could change with the NW-ZX2. This Android-based player can reproduce any commercially available music file including 128x DSD; plus, it also plays videos from YouTube, Hulu, and Facebook. It also comes with WiFi and Bluetooth support. The NW-ZX2 reestablishes Sony as one of the preeminent manufacturers of portable audio playback devices. And, yes, Sony has succeeded masterfully in achieving its design goals—the NW-ZX2 delivers excellent sound and looks, and it feels and responds like a high-performance product should. If you had any doubts about Sony's commitment to high-quality audio, the NW-ZX2 will put them to rest. sony.com (252)

Our Top Picks Portable Players

Astell&Kern AK240

$1999

Hard to believe, but the Astell&Kern AK240 improves upon the already brilliant performance of its highly regarded predecessors. Like them, it brings true high-end performance to portable music. Finally, audiophiles can enjoy music at the sonic level they're used to at home—without being anywhere near a reference system. Unlike iPods or iPhones, the AK240 can play high-res and even native DSD files, which can either be local or streamed across a network. That's a distinct sonic advantage. Even with lower-res material the AK240 delivers resolution, timbral nuance, dynamic inflection, ease, and authority unheard of in other portable players. Compared to the AK100 and AK120, the AK240 boasts a significantly quieter background, greater purity, and even greater resolution.
astellnkern.com (248)

Astell&Kern AK380

$3499

The AK380 is an expensive personal player, but the closer you look the more its price seems justified. A large, bright WVGA touchscreen is encased in an aircraft-grade Duralumin body clad in custom-fitted leather. There are dual AKM AK4490 DACs that deliver superior channel separation, native DSD with no interim PCM conversion, and resolution up to 384/32. Other features include MQS support, streaming over WiFi or Bluetooth aptX, a 20-band parametric equalizer, and 256GB of internal memory (expandable by 128GB via a microSD card). Yet this player is not necessarily a standalone device. When nestled into an AK Cradle ($349), which provides power and balanced XLR output jacks, the AK380 transforms into a streamer/DAC front end worthy of any high-end system. Sonically, the AK380 stretches the boundaries of what's possible from a personal player. Through the AK380, instruments exist in a field of air, and their rich timbres are exceedingly lifelike. The AK380 is also "fast"; its ability to trace rhythms accurately results in tight, infectious beats. Happily, all this is true regardless of how the unit is accessing music, including streaming. With the AK380, Astell&Kern has created a flagship that transcends the genre of audiophile-quality portable players.
astellnkern.com (263)