BUYER'S GUIDE TO
DIGITAL SOURCE COMPONENTS

Contents

Click on one of these links to jump to that section, feature or review.

Click here to turn the page.

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BUYER'S GUIDE TO
DIGITAL SOURCE COMPONENTS

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FROM THE Editor

The Computer-Audio Revolution

Welcome to the 2011 edition of our Buyer's Guide to Digital Source Components, brought to you by *The Absolute Sound* and *Hi-Fi Plus*. In this guide you'll find reviews of 41 digital components, ranging from disc players to DACs to music servers. It's no surprise that, compared with last year's Guide, this edition has many fewer disc player reviews and many more music-server reviews. The move away from packaged media such as CD and toward computer-based music deliver has reached juggernaut status in the past year.

My own recent experience confirms this powerful revolution. A loudspeaker manufacturer visited my listening room last week to set up a pair of speakers for review. After the speakers were roughly in place and we were about to listen to music for the first time, I showed the speaker manufacturer my disc player, opened the drawer, handed him the remote control, and invited him to play his reference discs so that he could dial-in the speakers' positions. "Discs?" he snorted. "I don't play *discs*." He proceeded to pull out his tiny laptop computer and ask me for a USB connection to my DAC so that he could play his music from iTunes.

In another sign of the changing times, during a visit to France the week before to tour speaker manufacturer Focal and electronics company Micromega, virtually of the listening we did was via wireless music streaming. This was true at both companies' multiple listening rooms, at two private homes, and at a French retailer's showroom. At Focal, we heard a complete stereo system, called Bird, consisting of amplification, a woofer built into the amplifier chassis, and two satellite speakers. The source component is an iPad, iPhone, or similar device that streams music to Bird wirelessly at full CD resolution. (Incidentally, the $999 Focal Bird sounded amazingly great; watch for a review in *The Absolute Sound*.)

In Paris to tour Micromega, I heard wireless streaming in three different venues; France's premier high-end audio retailer, Présence Audio Conseil, the listening room in Micromega's factory, and in the home of Micromega's CEO. With Micromega's forthcoming Aria streaming DAC, you simply stream music from your iPhone, iPad, or WiFi-enabled computer to the stereo system via the Aria. Listeners took turns playing music without ever getting out of their seats or pulling out physical media. We passed around our "nomadic" devices among each other, encouraging musical exploration and sharing. Not only was this amazingly convenient, but the sound quality was uncompromised. The Aria was at the front end of one of a handful of the best stereos I've heard, a mega-system with a price tag approaching seven figures. Even in this demanding context, the streaming music and the Aria's DACs delivered stunningly great sound.

I heard eight different systems during my week-long visit, and if it hadn't been for me bringing my own reference CDs, we would have never inserted a single disc into a player in any of the eight systems.

If that's not a revolution, I don't know what is.

Robert Harley

Click here to turn the page.

NuForce DAC-9

Reference Class 24/192 DAC, Pre-Amp and Headphone Amp All-in-One Solution

Engineered to meet the needs of the most demanding audio purists while also providing the greatest degree of flexibility, the NuForce DAC-9 is a true reference-class digital-to-analog stereo converter, pre-amplifier and headphone amplifier in one supremely sophisticated package. As to versatility, with its six digital inputs, the DAC-9 offers compatibility with all existing digital formats.

And, in keeping with NuForce's commitment to its customers, plug-in USB module/interfaces will always provide for upgradeability to higher bit rates, thereby securing the end-user's investment as digital technology continues to advance.

Features

* 6 inputs: USB, AES/EBU, RCA/BNC, RCA, Toslink, 3.5mm Optical*
* Outputs: XLR, RCA, 3.5mm and 6.3mm headphone output*
* Touch panel control
* Remote control: volume, input select, dim, mute
* Volume and sample rate display
* The headphone output and optical input share the 3.5mm Optical Input

Color Available

Recommended by *AVGuide, HiFi World, 6Moons, Secrets of Home Theater and High Fidelity, The New York Times' Technology Page* and more...

" ★★★★★ "

—*average customer rating,* **Amazon.com**

NuForce Icon HDP

High performance headphone Amp, USB (24bit/96KHz) and SPDIF (24bit/192KHz) DAC and Preamp

ON THE HORIZON

HOT NEW GEAR COMING YOUR WAY

Neil Gader

NuForce DAC 9
Engineered to meet the needs of the most demanding audio purists while also providing superb flexibility, the NuForce DAC-9 is a reference-class digital-to-analog stereo converter, preamplifier, and headphone amplifier in one sophisticated package. Six digital inputs cover the full range of today's digital-audio transmission formats. Each input channel feeds a pair of high-performance, non-over-sampling DACs, offering up to 24-bit/192kHz (96kHz for USB) performance. The output of these two DACs is then summed, thereby increasing the signal-to-noise ratio to near theoretical limits, and then input to a proprietary, no-negative-feedback passive current-to-voltage converter to maintain phase coherence. Dedicated headphone listeners have not been forgotten either. A fully independent headphone stage is provided, with volume adjusted by its own discrete analog control. Both 6.3mm and 3.5mm headphone jacks are included. And the plug-in USB module/interface will handle future upgrades to higher bit-rate USB performance.
Price: $1695. nuforce.com

Onkyo C-7000R Player
The C-7000R brings together features and refinements to take the full measure of any collection of audio CDs and CD-R/RWs, even MP3- or WMA-encoded discs burned on a computer. To ensure minimal signal interference, the C-7000R employs a thermally regulated super-precision clock, Phase Locked Loop (PLL) ultra-low-jitter technology, and a new circuit board construction. All of the highly sensitive componentry is housed in anti-vibration casing, with separate aluminum panels for the top, front, and sides. In keeping with the zero-tolerance approach to interference, digital circuitry and analog circuitry are physically separated, and there's a choice of operating the C-7000R in either digital-only or analog-only mode. The silent disc mechanism and solid die-cast aluminum tray epitomize the unit's audiophile build-quality. Equipped with separate Burr-Brown 24-Bit/192kHz DACs (PCM1792 x 2) for the L/R channels, it also sports the connectivity of AES/EBU balanced digital outputs.
Price: $1499. onkyousa.com

ON THE HORIZON

M2TECH: "E-Young" 32-bit /384kHz DAC

Encased in a stylish aluminum case, E-Young is a high-performance, high-value D/A converter capable of handling up to 384kHz sampling rates and a full 32-bits of resolution (USB input). The E-Young features a high-speed, asynchronous USB input based on "HiFace" technology, further developed to allow for higher sampling rates. A comprehensive input set provides great connection versatility. The E-Young encompasses a leading-edge technology for data handling and processing. A high-performance oversampling filter, custom-made on a programmable logic device (FPGA), implements minimum-phase filters optimized for sound quality, providing a very low noise floor. A 32-bit D/A integrated circuit is used in a non-conventional mode allowing 768kHz internal operation, while the synergy between the DAC IC and the custom over-sampler, makes for exceptionally low noise and THD. And the small footprint allows for easy placement in almost any living space.
Price: $1799. m2tech.biz

Cary Audio CD-500 player

Cary Audio Design's new CD-500 CD player features top-of-the-line Burr Brown 1792u DAC chips (24-bit/192kHz) in a fully differential-balanced configuration, and audio-grade Elna capacitors. Sound familiar? It's because the CD-500 shares its audio quality with its award-winning big brothers the CD 303T SACD Pro and CD 306 SACD Pro, but with a more modest entry fee. The CD-500 will upsample standard Red Book CDs at 96, 192, 384, 512, or 768kHz, and it offers HDCD decoding as well. It includes discrete RS-232 and IR command codes to make it custom-control friendly. Robustly built with a well-isolated chassis, the CD-500, a 23-pound heavyweight, is available with a choice of either black or silver anodized aluminum faceplate.
Price: $2995. caryaudio.com

Micromega Aria AirDream Wireless Streamer DAC

Micromega's new Aria AirDream accesses music stored on computer hard drives wirelessly and converts digital files to analog output. Based on the French company's third-generation AirDream wireless module, the Aria leverages Apple's iTunes software integration and AirTunes wireless transmission protocol for a seamless user experience. In addition, the Aria is designed as an ultra-high-performance digital-to-analog converter, employing three separate sophisticated power supplies, extensive RF and vibration isolation between sub-sections, a custom master oscillator, and a discrete pure Class-A JFET buffer following the Cirrus Logic 192kHz/24-bit DAC.
Price: TBA. audioplusservices.com

ON THE HORIZON

MSB Universal Media Transport

The Universal Media Transport (UMT) is MSB's all-in-one solution that offers universal disc playback (from Red Book PCM to Blu-ray, SACD, or DVD-Audio, two-channel or multichannel) from a network music streamer, a USB memory, or an HDD player. Not only does it come with all the digital outputs you could want, but it has a special mode for optimum use with MSB DACs or non-MSB DACs. The UMT is designed to be a two-channel playback system, with all standard digital outputs using MSB's proprietary jitter-removal system. The UMT plays back from solid-state memory using MSB-designed purist DSP-processing for preserving and ensuring bit-perfect accuracy. The MSB Network connects to MSB DACs at 32-bit resolution on the network. The UMT can be powered from an optional outboard supply ($595) or one of MSB's DAC IV Power Bases ($2495).

Price: $3995. msbtech.com

B.M.C. BDCD1 Transport/Player

The BDCD1 utilizes cutting-edge modular design and is available as a stand-alone transport or as a complete player. The top loader uses a belt-drive transport and boasts a 50-micron-tolerance precision-bearing plus an advanced switching power supply, with active primary voltage filtering and separate transformers for display, motor, logic and audio circuitry in the digital and analog domain. By adding the digital-to-analog converter module, the belt-drive CD transport becomes a complete CD player. The conversion is made by two 24-bit/192kHz Burr-Brown PCM1792 chips; output current is then filtered and converted to an output voltage by discrete, fully balanced I/V converters, which operate feedback-free. For transmitting digital audio signals it includes established S/PDIF-compatible interfaces (AES/EBU, coaxial 75 ohm, and optical TosLink), and also B.M.C.'s proprietary high-resolution "Superlink."

Price: Transport only, $4790; player, $5790. aaudioimports.com

MBL Corona Line C31 Player

The latest Corona line of electronics from MBL combines an elegant minimalist design with the German company's latest cutting-edge technology. The C31 Player features a state-of-the-art quality slot-drive and digital RCA and TosLink inputs as well as a USB dock (24-bit/ 96kHz native input) for CD audio data. Cirrus Logic DACs handle the D/A conversion, while high-precision buffer-data-reading is attained by the combination of a digital frequency synthesizer and nested analog PLL. Build-quality is unique with an outer solid aluminum housing enclosing a second casing of reinforced steel, thus all power supplies units and transformers are not only screened off by magnetically shielded partitions but also fully electrically isolated. When controlled by a Corona Line preamplifier or integrated amplifier, MBL's SmartLink network enables devices to communicate with one another with all associated input signals. It's also flash-memory upgradable via the integrated SD card reader.

Price: $9200. mbl-northamerica.com

Want to go back to the previous page? Click here.

ON THE HORIZON

Meridian Audio Media Core 600 Sooloos Digital Media System.
The Meridian Audio Media Core 600 is a complete Meridian Sooloos multi-zone solution that provides system core, storage, and multi-zone rendering in a single, rack-mountable case. Incorporating a pair of 2TB hard disk drives configured as a RAID 1 array, Media Core 600 provides enough primary storage for around 5000 CD albums, and always maintains an automatic mirrored copy of your complete collection. Housed in a brand new case, Media Core 600 incorporates an advanced switch-mode power supply and highly efficient cooling system, ideal for rack-mounting and reliable operation. Media Core 600, when connected to the Internet, also provides easy access to thousands of Internet radio stations, plus streamed music services from Rhapsody. It connects easily to a range of control options, including wirelessly to the free Sooloos App for iPad, iPhone, and iPod Touch. Connect Media Core 600 to any set of Meridian DSP loudspeakers to create an all-Meridian, fully digital, distributed audio system.
Price: $12,000. meridian-audio.com

Esoteric K-01 Player
The K-01 is the flagship model of Esoteric's all-in-one advanced D/A converter and Super Audio CD/CD player series. Derived from the award winning P-01/D-01 series separates, the top-loading K-01 is equipped with Esoteric's new VRDS-NEO "VMK-3.5-20S" mechanism, which has advanced substantially beyond existing VRDS models. The K-01 follows the philosophy of the D-01, and has two 32-bit monaural D/A converters with eight parallel/differential DAC circuits used per channel, as well as separate L/R power supplies. The on-board, high-precision clocking circuit has also been improved. The power supply for each circuit block has been optimized by using four on-board power transformers. Three sets of 192kHz/24-bit digital inputs, supporting the most advanced high-sampling-rate digital sources are also built-in. USB driver software supports 24-bit/192kHz asynchronous transmission for connecting with a PC.
Price: $22,500. teac.com/esoteric

Jeff Rowland Aeris DAC
The svelte Jeff Rowland Aeris DAC uses a cross-platform USB interface that is fully plug 'n' play with all operating systems, including PC, Mac, or Linux. It also sports its own output-level control for direct connection to power amplifiers. Making use of an asynchronous buffer, voltage-controlled crystal oscillators, and a 24-bit D/A converter the Aeris DAC delivers bit-perfect conversion and reduces total jitter to less than 10 picoseconds RMS from any input. Internal circuitry, both audio and digital, is fully isolated within individual aluminum pockets of the Aeris' precision-machined chassis, which itself is milled from a solid block of aircraft grade 6061-T6 aluminum—a feature that provides exceptional thermal stability, RFI/EMI shielding, and resonance control. The main power supply is isolated inside an external, machined-aluminum chassis. Eleven precision high-speed regulators provide low impedance and low noise "point of load" DC current to all analog and digital circuits. Includes remote control.
Price: $9800. jeffrowland.com

Why a Music Server?

Robert Harley

Imagine sitting in your listening chair and deciding that you want to hear a particular piece of music. You get up, walk over to the racks upon racks of CDs, turn your head sideways to read the labels, and scan the CD spines for the desired disc. You then open the jewel case, take out the CD, put it in a transport mechanism, and start the disc playing.

Now imagine instead that there's a touchscreen display next to your listening seat which shows you your entire music collection as album covers. The titles can be presented alphabetically, by genre, by composer, conductor, orchestra, and, in many cases, by what musician plays on the album. You touch the album cover and the music begins playing instantly.

Welcome to the world of music servers.

That's just the tip of the iceberg of how a music server can transform your relationship to your music library. Say, for example, that in that piece of music you just selected you found yourself greatly enjoying the drumming of Peter Erskine. On what other records in your collection does Erskine play? A couple of finger-taps on the touchscreen display shows you all the titles featuring this virtuoso drummer, ready for instant listening. Or consider this scenario we've all found ourselves in: We know what *kind* of music we're in the mood to hear, but not the specific title. Choose a title on the touchscreen that's in the genre you're interested in and the music server rearranges the album art on the screen to show you titles you *might* want to hear. Or simply choose a genre and allow the server to surprise you by playing music from that genre for as long a time as you specify (a feature that's great for parties).

> **A music server not only allows you to explore your music collection in new and rewarding ways, it also has the potential of delivering better sound than is available from CD.**

The server is not just a storehouse for your music collection, but also a portal for accessing new music. Say you're driving home from work one evening and hear a piece of new music on the radio that grabs you. When you get home simply turn on the server and download the album to make it part of your library. In fact, the distinction between the music on your server and music available for download is blurred; you have instant access to a vast music library from the listening seat, all organized in a way that lets you find exactly what you want. Speaking from personal experience, having a 24-hour, seven-day-a-week music store next to the listening seat can get costly very quickly. But it offers unprecedented opportunities to make musical discoveries. Anyone who has lived with a music server quickly finds it indispensable. In fact, the concept of music being stored on individual pieces of plastic that must be inserted into a machine will one day seem like a quaint anachronism—and already does to many listeners.

MUSIC SERVER SOUND QUALITY: BETTER THAN DISC

A music server not only allows you to explore your music collection in new and rewarding ways, it also has the potential of delivering better sound than is available from CD. Music read from a hard-disc drive sounds better than it does when read from an optical disc. That's correct—music ripped from a CD to a music server can offer greater fidelity on playback from the server than from the original disc itself when decoded through the same digital-to-analog converter.

How can a copy sound better than the original? Consider what goes on during playback of an optical disc: a laser reads the data off a disc spinning at 200–500 RPM and performs error correction on that data on the fly. The optical system has one pass to get it right. Moreover, the crucial clock that controls the digital-to-analog conversion process is tied to the clock that controls the spinning disc.

Now consider what happens when you rip that data to a hard-disc drive on a server. The disc-ripping software can read areas of the disc repeatedly if errors are detected, moving on only when the data are perfectly accurate. Those data are stored on a hard drive and treated like any other file by the computer. Reading those data off a hard drive is a cake-walk compared with the challenge of keeping a laser tracking a spinning, wobbling, eccentric optical disc, with only one chance at getting all the information off that disc correctly. Moreover, there's evidence that the quality of the disc itself (specifically in the pit and land structures that encode the audio data) affects sound quality—a factor obviated by hard-disc storage.

I've compared the sound of state-of-the-art CD transports to music servers playing the same music through the same digital-to-analog converter and digital cable. I first ripped the CD to the server, put the CD in the transport, and compared the two. The sound from the server was smoother in the treble with a more delicate rendering of treble detail. The soundstage was also improved when music was sourced from the server—more spacious, better focused in imaging, increased in depth—providing an overall more convincing impression of instruments existing in space within the recorded acoustic. There was also a greater sense of ease and musical involvement. In short, transferring your CDs to a server will improve the sound of your CD library—provided that the server is configured and set up correctly. **tas**

Remember! You can go to another section by clicking here.

Music Server File Formats Explained

Robert Harley

All music-management software for DIY music servers, as well as many turnkey systems, offers you the choice of storing your music in one of several file formats. Your choice of format is important on a number of levels: It affects sound quality, the amount of disc-storage space required for a given amount of music, and future compatibility and flexibility.

File formats can be divided into three categories: uncompressed, lossless compression, and lossy compression. The first category consists of two commonly used file formats called WAV and AIFF. A WAV file is an unadulterated bitstream that is identical to the source data. That is, a WAV file created from ripping a CD contains the identical PCM data that was recorded on the CD. Consequently, WAV files consume the most disc space for a given playing time—10.5MB per stereo minute for CD-quality audio (44.1kHz/16-bit). The alternative uncompressed file format is AIFF, or Audio Interchange File Format, first developed by Apple Computer. The advantage to WAV and AIFF files is that they are "universal currency," in that they can be converted to another file format in the future with no loss of quality.

You can cut nearly in half the amount of disc capacity consumed by a music file without sacrificing sound quality simply by using a lossless compression format such as Free Lossless Audio Codec (FLAC). FLAC produces perfect bit-for-bit accuracy to the original data and is sonically transparent. That is, a file converted to FLAC consumes about 50% to 60% of the space of a WAV file, but when "uncompressed" results in a bitstream that is indistinguishable from the original data. Other lossless compression formats are Apple Lossless Audio Codec (ALAC) and Windows Media Audio (WMA) Lossless. FLAC is more likely to be found on Windows-based servers, and ALAC (more commonly called "Apple Lossless") on Apple-based servers. You can rip CDs to a Macintosh in FLAC and play them back, but the process requires additional third-party software.

The third category of file formats, the lossy compression schemes, is anathema to high-end values and should be avoided. These systems reduce the number of bits in the datastream by throwing away musical information. Unlike the uncompressed and lossless compression formats that provide a perfect bit-for-bit reconstruction of the original datastream, lossy compression formats permanently remove information that cannot be recovered. Examples of lossy compression format are MP3, WMA, and the Dolby Digital format used to encode film soundtracks. These formats can operate at a variety of bit rates, with the higher bit rates delivering better sound quality than the lower bit rates. Nonetheless, with the cost of storage so low, there's no reason to use a lossy compression format. Note that iTunes delivers music using a lossy compression format. Full CD-resolution and true high-resolution downloads are available from audiophile-oriented sites. **tas**
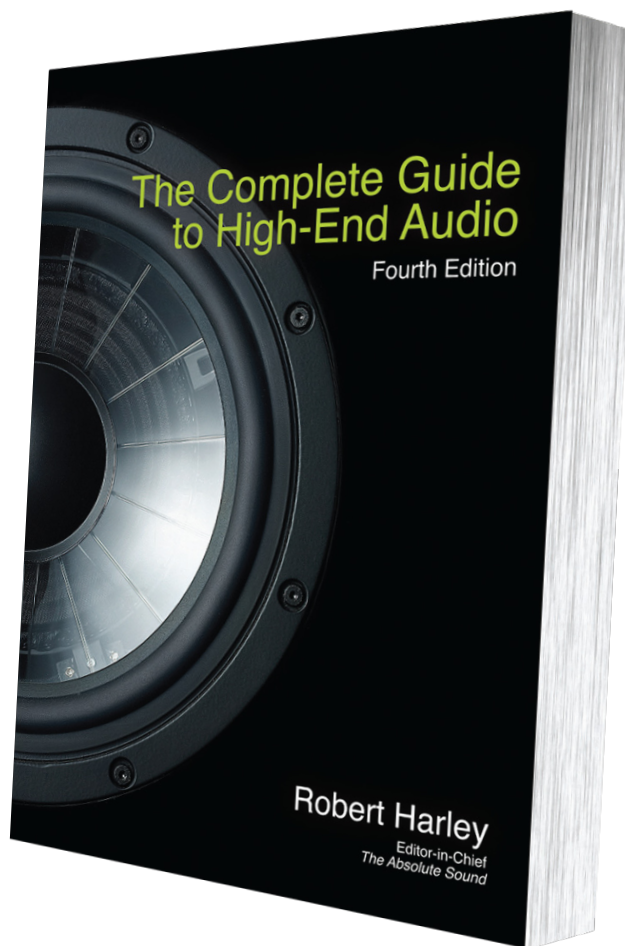
> **You can cut nearly in half the amount of disc capacity consumed by a music file without sacrificing sound quality by using a lossless compression format such as Free Lossless Audio Codec (FLAC).**

EQUIPMENT REVIEWS

Disc Players

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OPPO

Oppo BDP-95 Blu-ray/ Universal Player

A High-End Icon(oclast)

Chris Martens

Over the past several years, Oppo has steadily been carving out a healthy chunk of the disc-player market and for all the right reasons. From day one, the firm has followed a singular vision that involves building versatile players that combine often shockingly high levels of performance and value in equal measure. Exactly how high those levels of performance really are in an absolute sense may be a matter of some debate, but absolutely no one (either in the high-end universe or anywhere else) would deny that Oppo has emerged the "big dog" in terms of offering sonic value for money. As a result, Oppo's disc players have been winning friends, influencing people, and merrily re-writing the rulebooks that define exactly how much sound (and picture) quality one can reasonably expect from universal-disc player for a given sum of money.

But one other aspect of Oppo that's also deeply admirable is their practice of continuous product improvement. Let's call this the "never-miss-an-opportunity-to-make-a-good-thing-better" impulse that, in my view, defines Oppo as a true high-end manufacturer (albeit one whose products are, by design, affordable). From year to year, the Oppo folks just keep on pressing forward—never resting on their laurels (of which they've garnered quite a bunch), so that each new Oppo model really *is* better than the last—and in genuinely meaningful ways.

Those of you who have followed Oppo since its inception know that the firm's practice has been to produce really good, full-featured standard models, but then to offer somewhat more costly hot-rod models targeted specifically toward sound quality-conscious music lovers. This basic practice continues with the firm's new second-generation Blu-ray/universal players, where Oppo's BDP-93 ($499) serves as the standard model, while the just-released BDP-95 ($999) stands as the flagship, audiophile-grade model.

In the past, Oppo's flagship players were often "tuned to the nines" versions of its standard models, but that's no longer the case. While the BDP-95 shares some features in common with the BDP-93, it is in fact an entirely different player in ways that run more than "skin deep." Let me begin this review, then, by listing some of the BDP-95's distinguishing features and characteristics.

AUDIOPHILE HIGHLIGHTS

Apart from externally obvious differences, the BDP-95 differs from the BDP-93 (and from most other competing Blu-ray/universal players on the market), by providing a distinctive array of audio features, as highlighted below.

- **Very high quality 32-bit DACs**: The BDP-95 uses two 8-channel, 32-bit ESS SABRE[32] ES9018 Reference Audio DACs, which ESS claims "is the world's best performing 32-bit audio DAC solution targeted for high-end consumer applications and professional studio equipment." The ES9018 is similar to the

DACs some high-end manufacturers use in multi-thousand-dollar, two-channel CD/SACD players.

- **Dedicated stereo analog output**: The BDP-95 offers a dedicated stereo analog output with "specially optimized ES9018 DAC and output driving stages," where "each output is driven by four DAC channels stacking together to achieve even higher performance." The stereo output offers two sets of output connectors: one set with RCA single-ended connectors and the other with XLR balanced connectors. Oppo states that, "the balanced output features a true differential signal path all the way from the DAC to the 3-pin XLR connector."

- **Multichannel analog output**: The BDP-95 also offers a set of 7.1-channel analog outputs, which are driven by the second of the player's two ES9018 DACs.

- **Substantial, low-noise power supply**: The BDP-95 incorporates a toroidal power

EQUIPMENT REVIEW - Oppo BDP-95 Blu-ray/Universal Player

transformer "custom designed and built by Rotel," which is said to offer "superior power efficiency and much lower exterior magnetic field over traditional laminated steel core transformers."

• **Coaxial and optical digital outputs**: The BDP-95 provides both coaxial and optical digital outputs so that it can be used as a digital transport to feed outboard DACs.

• **Expanded bass-management options**: Relative to earlier BDP-83-series players, the BDP-95 gives users an expanded range of subwoofer crossover options, including settings for: 40Hz, 60Hz, 80Hz, 90Hz, 100Hz, 110Hz, 120Hz, 150Hz, 200Hz, and 250Hz.

• **Powerful connectivity options**: The BDP-95 is equipped with a hard-wired RJ-45-type Ethernet port, a wireless-n Wi-Fi adapter, an eSATA port, and two USB 2.0 ports. As shipped, the player can play music, video, or photo files from USB or eSATA drives, and can also stream certain types of content from the Internet. It cannot, however, function as a USB DAC, although it unofficially provides DNLA Server-like "Experimental Functions" that potentially allow the BDP-95 to play digital audio files stored on PCs attached to your home network.

• **Rich disc and media format support**: The BDP-95 is a true universal player that supports: Blu-ray Disc, Blu-ray 3D, DVD-Audio/Video, SACD, HDCD, CD, Kodak Picture CD, AVCHD, MP4, DivX, MKV, FLAC and WAV from recorded discs or, where feasible, from USB or eSATA drives.

As you look back over this list of highlights, ask yourself this: have you ever seen a $999 universal player that offered an audiophile-oriented features set like this? I certainly have not, which further underscores the value on offer here.

For those of you who are home-theater/movie enthusiasts, let me simply mention in passing that the BDP-95's video features are every bit as versatile and performance-minded as its audio features. To learn more about the video aspects of the player, visit AVguide.com where you can read the review I recently published for our home theater-oriented sister magazine *The Perfect Vision.* But for now, let's focus on the BDP-95's sound.

Have you ever seen a $999 universal player that offered an audiophile-oriented features set like this?

The BDP-95, like many high quality players, benefits from a generous amount of run-in time and needs plenty of warm-up time before giving of its best. It also tends, as do many other high-end players, to benefit from extra care taken in selecting associated power cords and

signal cables, so that some experimentation is the order of the day. Odd though this may sound, the BDP-95 deserves—and in a sense almost demands—to be used with very high quality cables that could potentially cost as much (or more) than the player does. Once these preliminary issues are resolved, however, the sound of the BDP-95 becomes very special indeed—especially if you choose to listen through its balanced XLR outputs.

As I perceive things, the BDP-95 has three signature qualities that set it apart from the competition. First, the player has a highly detailed sound, yet a sound where the different aspects of "detail" are so thoroughly and beautifully integrated that you tend not, at first blush, to notice just how much sonic information the player is able to retrieve. When I use the term "detailed" to describe the BDP-95, then, I am thinking of a complete package that includes subtle timbres, textures, harmonics, transient sounds, reverberations and echoes, and especially spatial cues. Put all of these elements together and listen to a familiar piece of music and you may conclude that, with the Oppo in play, there's suddenly more "there" there.

Second, the BDP-95 is remarkably smooth sounding, which is all the more impressive when you consider that it also highly detailed. In my experience, the qualities of detail and smoothness don't necessarily travel well together, so that it's a rarity to find them working synergistically as they do in the Oppo. There's real sonic "magic" in this combination of virtues—sort of the audiophile equivalent of gain without pain. In the case of the BDP-95, detail may be what draws you in at first, but the player's relaxed, effortless smoothness is what keeps you listening, hour after hour.

Finally, the BDP-95 is capable of exceptionally three-dimensional presentations, assuming your chosen recordings are up to the task. Three-dimensionality, more than any other quality, gives the Oppo an overarching and holistic quality of sophisticated musicality, making it sound like a higher-priced component than it actually is. During my listening tests, for example, I compared the sound of a series of discs being decoded by the digital front end of the excellent Anthem Statement D2v A/V controller ($8499) vs. the sound of the same discs played through the BDP-95. I also did similar comparisons between the BDP-95 and two comparatively expensive reference players I had on hand: a Musical Fidelity kW SACD player and a Rega Isis CD player. The results were eye opening.

First, I found the BDP-95 consistently outperformed the Anthem controller's digital front end—sounding smoother, more detailed, and *considerably* more three-dimensional. I also

EQUIPMENT REVIEW - Oppo BDP-95 Blu-ray/Universal Player

found the Oppo was thoroughly competitive with (though perhaps not fully the equal of) my two high-end reference players. While I ultimately preferred my reference players to the Oppo (no great surprise there, given the 8X-to-9X price differentials involved), the Oppo was in no way outclassed by them. On the contrary, the Oppo sounded, in a qualitative sense, much more *like* the premium-priced reference players than unlike them. In short, the Oppo proved to my satisfaction that it could "play with the big boys."

The BDP-95 is remarkably smooth sounding, which is all the more impressive when you consider that it also highly detailed.

Let me provide a handful of illustrations to show how these qualities play out with real-world recordings. As a starting point, listen to Musica Nuda's cover of the classic Beatle's tune "Come Together" from *Live à Fip* [Bonsai Music]. Musica Nuda represents a category-defying collaboration between vocalist Petra Magoni and acoustic bassist Ferruccio Spinetti, where the duo's core sound has roots in acoustic jazz, but with classical, pop/rock, and even techno/punk influences also in evidence. Thus, the track opens the Spinetti vigorously playing the familiar opening theme of "Come Together" on his acoustic bass while we he what turns out to be a processed form of Magoni's voice supplying what would normally be percussion and other instrumental accents. As the song evolves, it becomes clear that Spinetti's bass will not only be used to supply the traditional "bass part" of the tune, but will also—through Spinetti's sheer

dexterity and inventiveness—provide other instrumental parts, as well (so that, in essence, one hears acoustic bass playing the role of an ensemble). The BDP-95 makes it easy to hear Spinetti apply various playing techniques as he draws multiple distinct voices from his chosen instrument.

At the same time, Magoni applies her expressive and malleable voice in a highly creative way as well, sometimes supplying traditional vocals, but at other moments shifting radically in pitch and texture to produce what almost seem like sounds produced by otherworldly instruments or even a synthesizer (an illusion reinforced by selective use of presumably soundboard-controlled reverb and echo effects). Again, the Oppo is so detailed that it invites you to explore and savor not only the core sound of Magoni's voice, but also the many unexpected twists and turns it takes along the way. Though there are basically just two people performing, the illusion is that of hearing a large and eclectic ensemble at play.

But one subtle yet very important aspect of the Oppo's rendering of the song is the deft manner in which it provides low-level sonic cues that let you know the song is not—despite its occasional use of electronic effects—a traditional studio recording, but rather a live performance. In an almost subliminal way, you become aware that—behind the occasional technical pyrotechnics—the performance is being

captured in a real recording venue before a live audience (you can almost *feel* the energy of the crowd). A delicious three-dimensional moment occurs about a minute into the song as Magoni belts out the song's signature chorus line "Come together…" from center stage, only to have a voice positioned at the far right rear corner of the soundstage complete the line, softly singing, "…over me." As the song powered to an abrupt close, the pent up energy of the crowd erupts into enthusiastic applause that, through the Oppo, sound remarkably believable and realistic—not like "canned" white noise.

For another fine example of many of the Oppo's musical strengths at play, try listening to the third ("The Alcotts") movement of Charles Ives *A Concord Symphony*—a piece originally written as a piano sonata and later orchestrated by Henry Brant, as captured in a lovely live performance featuring the San Francisco Symphony conducted by Michael Tilson Thomas [SFS Media, Multichannel SACD]. Though only a bit more than six minutes long, this movement spans quite a range of orchestral moods, introducing everything from gentle, contemplative woodwind themes, to vigorous

and at times quite angular string passages, on through to powerful and sometimes deliberately dissonant brass and percussion outbursts.

Throughout the movement, the BDP-95 impressed me favorably in several ways. First, it caught the distinctive timbres of each orchestral section in a rich and vibrant, yet never overstated way. Instead, you hear what Brant once described as the "athletic surefootedness" of the orchestration conveying Ives' musical ideas in "clear, vivid, and intense sonorities." This player is all about getting the tonal colors and textures of instruments right.

Next, the sheer smoothness of the BDP-
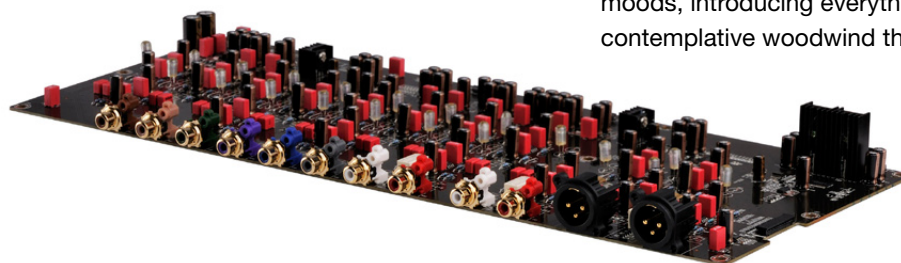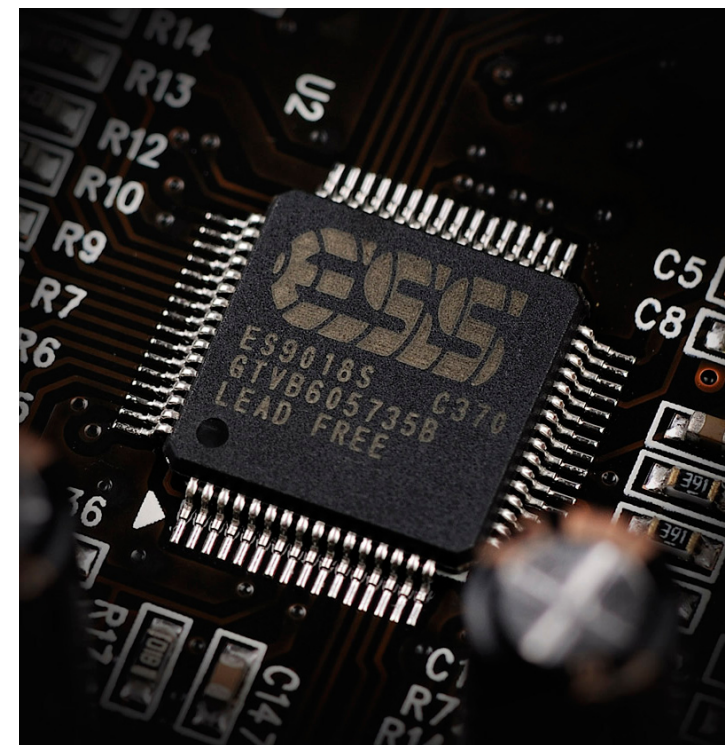
EQUIPMENT REVIEW - Oppo BDP-95 Blu-ray/Universal Player

95 allowed it to navigate the more angular and dissonant aspects of Ives' themes in a way that revealed their intentional (and sometimes startling) idiosyncrasies, while at the same time allowing their richness and underlying beauty to shine through. If you know Ives' music, then you may have found, as I have, that some disc players tend to turn it into a strident, jagged-sounded mess, but not so the Oppo. String tones, for example, sounded rich and buttery smooth, while brass section sounds had appropriate energy and bite, burnished with golden overtones. While the Oppo never hid or glossed over the at times quirky aspects of Ives' themes, its inherent smoothness and tonal richness invited listeners to embrace the broader sweep and flow of the composition.

Finally, the BDP-95's three-dimensionality lets me hear that this was a live recording (captured in Davies Symphony Hall, San Francisco), so that through myriad small sonic cues I got a sense of the orchestra as a living, breathing entity interacting with hall and the audience. Specifically, the Oppo's ability to retrieve low-level details gave me a sense for the acoustics of the recording venue and, on louder passages, to hear how the orchestra energized the entire hall, filling it with sound. It is important to bear in mind, though, that the Oppo isn't one of those players that pursues detail for detail's sake; instead, the player's many small sonic details coalesce to form an integral, organic whole that simply "sounds right." If you stop to think about it, that's one of the highest complements any disc player could earn.

Are there caveats, here? There are a few, though it seems almost churlish to mention them given the many things the BDP-95 does well, and for so little money. From an operational standpoint, and with the needs of a coming generation of computer audiophiles in mind, I would love to see Oppo configure this player so that it can serve as a USB DAC, and I would also love to see the player's DLNA Server-like "Experimental Functions" become fully supported. Sonically, the Oppo already delivers very good-to-excellent performance in most areas, so that the player could perhaps best be improved by simply expanding upon its current core strengths: detail, smoothness, and three-dimensionality. About the only area where the BDP-95 might immediately benefit from a bit of extra work would be in the area of low bass performance, where some competing players do offer a somewhat deeper and more powerful presentation with a bit better pitch definition in an absolute sense. But with that said, let me also be clear on one fundamental point: absolutely nothing I've heard at or even vaguely close to the BDP-95's price can compete with its sound quality.

The bottom line is this: the BDP-95 is by far the finest Blu-ray/universal disc player Oppo has yet produced. If you can afford one, then put the Oppo right at the top of your short list. If you can afford something more expensive, strongly consider buying the Oppo anyway. It's that good. 𝖙𝖆𝖘

SPECS & PRICING

Oppo BDP-95 Blu-ray/Universal Disc Player

Disc/file formats supported: Blu-ray Disc, Blu-ray 3D, DVD-Audio/Video, SACD, HDCD, CD, Kodak Picture CD, AVCHD, MP4, DivX, MKV, FLAC and WAV from recorded discs or, where feasible, from USB or eSATA drives.

HDMI audio bitstream support: Dolby TrueHD, Dolby Digital Plus, Dolby Digital; DTS-HD Master Audio, DTS-HD High Resolution Audio, DTS-Digital Surround; SACD via DSD bitstream or LPCM conversion, LPCM 7.1-channel, 5.1-channel and 2-channel.

Onboard decoder support: As above. Outputs 1080p at: 24Hz, 50Hz, 60Hz

Video outputs: Two HDMI, one component video, one composite video.

Digital audio outputs: Two HDMI, two digital (one coaxial, one optical).

Analog audio outputs: One 7.1-channel output, two stereo outputs (on single-ended via RCA jacks, one balanced via XLR connectors).

Frequency response: 20Hz–20kHz (-0.3dB), 20Hz–96kHz (-2.5dB)

Signal-to-noise ratio: >130 dB (A-weighted, mute), > 115 dB (A-weighted, unmute)

Other connections: Ethernet and Wireless-n (for firmware updates, BD-Live content, Netflix and Blockbuster-On-Demand content streaming, and for "Experimental Functions"), two USB ports (for content playback from USB drives), eSATA port, IR in/out, RS-232 (optional).

Dimensions: 4" x 16.875" x 12.25"

Weight: 15.4 lbs.

Warranty: One year, parts and labor

U.S.

Price: $999

Oppo Digital, Inc.
(650) 961-1118
oppodigital.com

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Marantz SA8004 CD/SACD Player

Nothing To Criticize, A Lot To Rave About

Paul Seydor

Not long ago my colleague Robert Greene posed himself a question, "Why are people reluctant to buy expensive turntables?" Then he shared his thoughts in an e-mail with Robert Harley, Jonathan Valin, and me. Recalling a visit to my place to hear a brand new Acoustic Research XA turntable—an almost fifty-year-old design that sold for the princely sum of $87 in 1968—I had found on eBay, he told RH and JV:

"We sat down to listen and what came out? Glorious music in every sense, including the strictly sonic. The AR arm and turntable were in no way embarrassed by being played through such a superlative set of electronics and speakers [McIntosh C46/MC402, Quad ESL 2805] and in such a nice room acoustically. The result was truly wonderful—pure, warm, low distortion, naturally balanced sound indeed, music essentially as it really is . . . . [If] one had listened to this system without knowing what the turntable/arm used was, one would have found rather little to complain about. I think no one would have said, wonderful speakers and electronics, but what on earth is wrong with the turntable. This would just not have happened. Of course, in direct comparison to other setups, one would have noticed differences and on occasion perhaps improvements, but on its own the playback was very satisfying indeed."

REG's answer to his own question? Because "vinyl is a technology that works so well by nature that it sounds wonderful even in simple forms . . . the inexpensive stuff is just too good!" (Is a cliché here reversed—the very good now the enemy of the excellent?)

I thought about REG's e-mail as I began this review of Marantz's new SA8004, a superb-sounding two-channel SACD and CD player priced at $999. It seems to me that after more than a quarter century compact disc technology has long since reached the point that vinyl itself did a few decades ago. I am not referring to the long-standing debate about whether digital is

"as good as vinyl"—that's an argument I don't engage in any more (I just look for another bar). What I mean, rather, is that digital technology has advanced so that, like vinyl, it just *works*, reliably, dependably, predictably. Some of the strongest evidence for this is the proliferation of really good moderate- and lower-priced compact disc players TAS has reviewed the past several years. Surely even vinyl zealots would have to admit that it has been a very long time since even an inexpensive player from a reputable manufacturer has exhibited anything approaching the sonic nasties of early digital. And if the presence of excellent-sounding modestly priced vinyl gear makes the extremely expensive stuff a cautious proposition in these economically parlous times, how much more is this true of digital, where last year's state of the art is almost certain to be bested by next year's moderately priced gear?

I knew there was something special about this new Marantz when I unboxed it. Many inexpensive components feel so light and cheaply made that if differently shaped they could easily serve as Frisbees. Not so the 8004. While no back-breaker,

it weighs in at a substantial seventeen pounds and looks impressive, to boot. It's obvious someone was paying attention to controlling chassis vibration and resonance. (I know that weight as such and a quality "feel" are at best imperfect indicators as to how good a component will sound; but I've never cared for equipment that looks or feels cheap, even if it sounds great, and there's enough good-sounding budget-priced gear around to prove that inexpensive doesn't necessarily have to equate to chintzy.)

Setup's a matter of plug (a generic pair of interconnects is provided, though I used Kimber Select), play, and start listening, which I did, following a 24-hour break-in with Reference Recordings's *Test & Burn In CD*.

First up was a new Christmas release from the Anonymous Four, *The Cherry Tree Carol* [HMUSA, SACD]. A signature characteristic of recordings by the producer Robina Young and her engineer Brad Michel is how ideally they always manage to mediate focus and ambience. The four women are arrayed across the front, spread not too far apart and somewhat

EQUIPMENT REVIEW - Marantz SA8004 CD/SACD Player

closely miked. Yet this is not a dry-sounding recording; indeed, a nice ambience is captured to create the illusion of singers performing in a recognizable space. There is nothing sonically "demanding" here except that any edginess or lack of midrange purity and naturalness will be instantly revealed as roughness or peakiness, while top-end constriction will exact a penalty in reduced atmosphere. This is also an exceedingly smooth recording in every respect, the four voices blending, separating, blending again like gossamer threads. But it should not sound warm as such, instead a bit cool, even austere, though the reverberation has a satisfying decay. While any resonant acoustic will obscure articulation of the lyrics somewhat, the microphones are proximate enough that the words remain very intelligible, especially during solo passages (e.g., the lyrics of the title carol can be totally understood without recourse to the booklet).

Not that the Marantz lacks warmth (more on this anon), as a drop-dead-gorgeous new recording of Bach's cello suites demonstrates. The soloist is Luigi Piovano, first cellist of Rome's National Orchestra of the Academy of Santa Cecilia: Piovano's tone is golden and he has been recorded in a suitably accommodating acoustic in sonics that are almost voluptuously refulgent. (Lovers of the cello or the Bach should search this one out!)

Just because I started with small material—that is, numerically small—doesn't mean this player can't do the big stuff. Indeed, one of its strengths is the ease, grip, and aplomb with which it handles Mahler symphonies or Wagner operas, throwing wide and deep soundstages. Take the extremely demanding "Forging Song" from *Siegfried* in the

Solti recording, the Marantz didn't miss a trick, the soundstage reproduced exactly as the producer John Culshaw described it, with the title character dead center toward the rear while the screaming Mime runs helter-skelter. And if it's testicular slam you need to be reassured about, chase up a copy of Hindemith's juggernaut Symphony in E flat in Leonard Bernstein's pile-driving recording—slam that'll knock you into the next county.

> ## Its presentation of all the tactile aspects of music—textures, tonal colorings and shadings, and that elusive sense of body—is unusually organic and integrated.

Something different, something blue—*Kind of*, to be precise, first in its two-channel SACD iteration, which for my tastes competes with the fancy vinyl reissues for listenability and vitality. The same virtues extend to the Legacy version, albeit in very slightly diminished measure. Using the same recording, in a direct comparison to the Benchmark DAC1 (fed from the 8004's digital out), the Benchmark brings a bit more detail and resolution and an impression of fractionally greater neutrality. Does the 8004 suggest a hint of warmth? Perhaps, but we are talking subtleties here.

A few years ago a friend of mine was buying a new CD player and had passed up a very expensive and highly reviewed model in his price range. When I asked him why, he replied, "Because-it-sounds-like-this," imitating the excessive articulation (i.e., "tunefulness") for which it was known, but which he felt was merely mechanical sounding. The 8004 will brook none

of that. I'm not sure if the company's designers specifically voice its components, but the ones I've both heard myself and read about from reviewers whose judgments I trust suggest that musicality defined in terms of roundedness and freedom from any sort of fatigue and harshness are priorities. Yet the Marantz is still capable of a very high order of resolution, indeed.

And something more. There's an aspect to the reproduction of very inexpensive digital components that is difficult to define but that I often hear. It has to do with a subtle papery quality, a certain "thinness." I employ the quotation marks because I don't mean it in the sense of tonal balance, but as regards solidity, body, substantiality in the reproduction of voices and instruments. I hear none of these shortcomings from the 8004. Indeed, its presentation of all the tactile aspects of music—textures, tonal colorings and shadings, tastiness, that elusive sense of body—strikes me as unusually organic and integrated, while being very transparent and grain-free.

Somewhere around this point in the review of a budget-priced component one is supposed to start voicing caveats and reservations. Not to be deficient in this regard, let me say that I can imagine some listeners finding the 8004 perhaps a bit too smooth, given current tastes for components with rising top ends that result in an excessively crisp and, to my ears, a too bright presentation. So, yes, let me grant a slight touch of warmth, though that might also be a function of its satisfying sense of body. Whatever, no complaints here, because it sounds perfectly right and natural to me—I'm buying the review sample. The reality is that this player leaves me

with nothing I feel like criticizing and a lot to rave about.

Which is to say that we have come full to the modest AR XA turntable with which I began this review. There are obviously "better" players out there for a great deal more money, but plug the Marantz SA8004 into the most demanding setup you can find, put on a well recorded program, and sit back and listen. Trust me, you will not think there is anything wrong with the reproduction because what you'll hear is truly wonderful—pure, warm, low in distortion, music essentially as it really is. **tas**

SPECS & PRICING

Marantz SA8004 SACD Player

Playback formats: Disc—SACD, CD, CD R/RW (CD DA/MP3/WMA); USB—MP3/WMA/WAV/AAC, HDAM, and HDAM SA2

DAC: Cirrus Logic CS4398

Frequency response: SACD, 2Hz–50kHz (-3dB); CD, 2Hz–20kHz

Dimensions: 17 3/4" x 4 5/16" x 13 9/16"

Weight: 17.2 lbs

U.S.

Price: $999

MARANTZ AMERICA, INC.
100 Corporate Drive
Mahwah, N.J. 07430-2041
(201) 762-6500
usmarantz.com

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Exotica for the Masses

Simaudio Moon i-1 Integrated Amplifier and Moon CD-1 Compact Disc Player

Wayne Garcia

I f you're a regular reader of this magazine, you know that flipping through these pages you'll soon find yourself in a section we call "The Cutting Edge." This is the place where my colleagues, and I, on occasion, review audio's priciest, sexiest, and, as the name says, most exotic cutting-edge components. Like the pages found in another type of magazine, "The Cutting Edge" has the power to induce lust from afar. But no matter how sexy its centerfolds, "The Cutting Edge" isn't merely a place to display gear porn. It's a forum for in-depth explorations of the finest technology and ideas this industry has to offer—no matter how out-of-reach the products may be to most of us.

But for anyone whose been exposed to such superbly musical and pricey items, "Exotica" can also prove to be hazardous. After all, give most people a taste of luxury—be it an audio component, automobile, wristwatch, or bottle of wine—and it's hard to go back. In other words, how do you drink Two-Buck Chuck once you've tasted Gevrey-Chambertin?

Thankfully, there's also a wide middle ground—that zone, for example, where highly satisfying and beautifully made wines are to be found for less than the price of a tank of gas. The same basic analogy, if not the price at the pump, holds true for audio gear. While there's a lot of good stuff out there that doesn't cost an arm and a foot, the real joy comes in discovering affordable components that provide outstanding sound and lasting musical satisfaction.

Which brings us to Canada's Simaudio.

Like many audio manufacturers, Simaudio builds a wide range of products. Naturally, many of the lessons Simaudio learned developing its upper-level Evolution Series are to be found in the company's more affordable gear.

Two shining examples are available in the form of Simaudio's latest entry-level components, the $1500 (each) CD-1 Compact Disc player and i-1 integrated amp. What I like so much about these designs can be stated simply: While they lack the final degrees of air, detail, frequency extension, and transparency found in those "Cutting Edge"-bound products, there's an organic nature, a "whole greater than the sum of the parts" quality, that makes this combination surprisingly wonderful to listen to.

Lower noise floors—meaning a noticeable lack of electronic haze, hash, and barely heard aural cobwebs—are one of the areas in which today's gear clearly whomps the best of the past. Curiously, because we're not talking about obvious noises such as a hum or a buzz, this is the type of virtue that you don't really notice until you realize that something you usually hear isn't there anymore. The sensation manifests itself not only in a greater sense of dynamic and rhythmic freedom, but also in transparency—the feeling that you're peeling back the curtain on an event that took place in another time and place, but that still feels like it's occurring right now, in your room.

This silence, or perhaps more accurately stated, this ability to let each recording's own energy and ambience announce itself from the get-go, is one of the triumphs of this moderately priced stack. To hear what I mean, play the L.A. Philharmonic and Esa-Pekka Salonen's recording of Lutoslawski's Symphony No. 4 [Sony Classics]. The piece unfolds slowly, starting with a quietly scored passage of harps and skittering strings. The Simaudio duo makes you feel like you've placed your head through the looking glass into another world. A soundstage of impressive breadth and depth opens up, defying both speaker and room boundaries. Strings overlap laterally in beautiful impressionistic layers, yet a solo violin passage is in perfect focus. The symphony builds to a shimmering climax, but with no sense that the sound is emerging from two point sources.

Though rated at a solid but fleet 50Wpc (100Wpc into 4 ohms), the i-1 proved to have good dynamic headroom. No compression or audible stress occurred on this or subsequent

EQUIPMENT REVIEW - Simaudio Moon i-1 Integrated Amplifier and Moon CD-1

Remember! You can go to another section by clicking here.

torture tests, one of which was Jeff Beck's "Brush With the Blues" from the CD *Who Else!* [Epic]. The tune begins as a slow, snaky blues workout before Beck goes psycho on his Strat. To get the track's full sonic and emotional impact and to truly appreciate Beck's dazzling virtuosity—at one point he hits a sustained high note that will rattle your fillings—as well as the throbbing bass and hammer-blow drums that root it all, the cut begs to be played loudly.

The i-1 was again impressive, though keep in mind that my listening room is small, and both sets of the speakers I used in the evaluation are rated at 4 ohms.

A singer, such as Nina Simone, benefits from this Simaudio pair's musicality, too. Her hushed, haunting interpretation of Billie Holiday's "Don't Explain" from the *Four Women* box set [Verve] also emerged from a deep, beautifully described background. A flute flutters behind her at stage left, while her piano and an acoustic bass lay a rich foundation for her smoky vocal growls and intimate purrs.

As magical as these listening experiences were, I also heard a slight darkness to the overall sound, most notably, perhaps, in the upper treble, as well as a thickness in the midbass region heard with cellos, basses, and percussion. I noticed this in the Beck disc as a slight fattening of the bass guitar, in the Nina Simone song as a reduction of air in the flute, in the Lutoslawski symphony as a slight diminution of the detail and tension it usually has. But these things are only noticeable in comparison with much pricier gear, and they are ultimately, for me, of little concern. Because in addition to that behind-the-curtain quality mentioned above, these Simaudio components present music with an organic wholeness I'm not quite sure I've experienced at this price point before.

By using the phrase "organic wholeness," I'm not trying to co-opt buzzwords from the food industry. Rather, I'm trying to convey the sensation of musicians playing with and off each other the way we hear them in live performances and the feeling that the entire sonic spectrum, even if not as extended as way up and way down as the priciest stuff, is built from the same tonal, dynamic, and expressive materials.

In short, with its latest and most affordable CD player and integrated amplifier, Simaudio, like a neighboring wine from a less exalted vineyard in Gevrey-Chambertin, offers a mighty fine taste of the good stuff, at a price most of us can appreciate. **tas**

WHAT MAKES THEM TICK

Available in black and silver finishes, both the i-1 and CD-1 are cosmetically minimalist and quite solidly built. The i-1's feeling of power-in-reserve comes from the 320VA toroidal transformer and 20,000mF worth of capacitance that occupies most of the interior left side. The preamp section utilizes JFET inputs, while the amplifier uses but a single pair of bipolar output transistors per channel. The back panel offers five line-level inputs, while the faceplate provides an eighth-inch mini-jack for personal media players and a quarter-inch headphone jack. The all-aluminum chassis was designed to be rigid and vibration free, and the i-1 was designed to be powered up at all times, to ensure consistently optimal sound, but also to operate at low temperatures for a lengthy operational life.

Built on what appears to be the identical all-aluminum chassis, the CD-1 boasts a proprietary CD transport mechanism, as well as Simaudio's in-house software and hardware technologies. Burr-Brown PCM1793 DACs provide 24-bit/192kHz decoding and 8x oversampling, while an RS232 port provides communication links in custom-install situations and for any firmware updates. Like the i-1, the CD-1 was designed to be powered on at all times. And while the red LED front-panel display is nice and large, I wish Simaudio allowed for the thing to be dimmed or shut off. It's very bright in a darkened room at night, so when I was not using it, I put the unit in the "standby" mode. **WG**

SPECS & PRICING

Moon CD-1
Outputs: RCA analog, S/PDIF digital, IR, RS232, SimLink
Dimensions: 16.9" x 3" x 12.75"
Weight: 14 lbs.

Moon i-1
Power output: 50Wpc into 8 ohms, 100Wpc into 4 ohms
Inputs: Five pair line-level, IR, RS232, SimLink
Outputs: One pair preamp
Dimensions: 16.9" x 3" x 12.75"
Weight: 22 lbs.

ASSOCIATED EQUIPMENT
TW Acustic Raven One turntable, Tri-Planar Ultimate VII tonearm, Transfiguration Orpheus cartridge; Artemis Labs LA-1 linestage and PL-1 phonostage; Kharma MP-150 mono amplifiers; Simaudio Moon LP 5.3 phonostage; Totem Acoustic The One and Kharma Mini Exquisite loudspeakers; TARA Labs Zero interconnects, digital cable, Omega speaker cables, The One power cords, and AD-10B Power Screen; Finite Elemente Spider equipment racks

U.S.
Moon CD-1
Price: $1500
Moon i-1
Price: $1500

SIMAUDIO LTD.
95 Chemin du Tremblay, Unit 3
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J4B 7K4
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simaudio.com

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Cambridge Audio Azur 840C CD Player

The best CD playback under $5k for…$1799

Robert Harley

The new Cambridge Audio 840C CD player left me shaking my head in wonderment at how Cambridge can sell this much CD player for $1799. The 840C is packed with advanced features, sophisticated technologies, and high-quality parts that one finds in digital products costing upward of $10k. Here's a sample of the 840C's technology: custom transport mechanism; custom upsampling digital filter running on a 32-bit DSP chip; differential digital-to-analog converters; digital inputs; digital upsampled outputs; and balanced analog outputs. (See Features and Technology sidebar for details.)

Such an impressive feature and technology list, however, tells you nothing about how the player sounds. To know that, you must listen. Dropping the 840C into my reference system, I was stunned by its overall sound quality. I knew immediately that the 840C wasn't a player to be measured against similarly priced products, but was worthy of comparison with reference-grade digital front ends.

For starters, the 840C doesn't sound anything like an $1800 CD player. It had a resolution, refinement, ease, grace, and musicality that were instantly recognizable as being different from every other product in the category. In fact, it's hard to know where to begin praising the 840C. We could start with any part of the sonic fabric, but I'll choose the treble reproduction, an area where CD players often reveal their shortcomings. The 840C had a delicacy, refinement, and sophistication in the top octaves that must be heard to be believed. Most digital near this price—indeed, most digital at any price—tends to make cymbals sound like undifferentiated bursts of white noise, with no inner character or clue as to the mechanism by which the sound was created. By contrast, the 840C had a completely natural top end that was smooth and gentle, yet bursting with fine inner detail which gave high-frequency-rich instruments a remarkable timbral realism. Listen, for example, to Jack DeJohnette's delicate understated cymbal work on Michael Brecker's fabulous new (and, unfortunately, last) CD *Pilgrimage*. I could hear every nuance of his exquisite playing, and the cymbals were infused with a wealth of finely filigreed detail in their shimmer and decay. Many $5k players don't approach the 840C's beautiful rendering of the top octaves.

The midrange was equally well served by this combination of resolution and ease. Instrumental tone colors were vivid and alive, as though the 840C had access to a wider color gamut than other CD players. Most CD players anywhere near this price tend to homogenize timbres by overlaying them with a common synthetic character; the 840C portrayed timbres with a stunning naturalness. The natural rendering of tone color, coupled with the overall ease, made the 840C musically vivid without being sonically vivid.

The 840 also had a soundstage dimensionality that I haven't heard before in a sub-$5k digital front end. The 840's spatial presentation reminded me of the first time I heard the Theta DSPro Generation III, with its spectacular sense of sculpted instrumental outlines "hanging" in the soundstage and separated by near-tangible air. The 840C presented tightly focused images, with sharp outlines that were surrounded by a sense of palpable bloom. The result of the 840C's ability to present instruments as distinct objects in three-dimensional space was a heightened ability to hear what each musician was playing. The subjective consequences of this objective change in the presentation cannot be overstated; rather than hearing a congealed and synthetic mass of sound, the 840C brought the music to life by conveying a convincing impression of individual musicians. In addition, the 840C resolved reverberation tails down to a very low level, which further added to the illusion of hearing instruments in a large acoustic space.

Music through the 840C had an organic "rightness" and fundamental musicality that's hard to describe. I heard a sonic coherence that translated to an enhanced ability to hear into the music and understand it more deeply. This was partly the result of the 840C's remarkable ability to keep individual instrumental lines distinct, and partly because of the player's tremendous sense of ease, smoothness, and liquidity. This player is amazingly free from midrange glare (often manifested on the leading edges of piano notes) and metallic hardness in the treble. Despite the

FEATURES & TECHNOLOGY

The 840C offers a host of inputs and outputs beyond those traditionally found on CD players. These connectivity options take advantage of the 840C's custom digital filter, and include two digital inputs for decoding external sources (each input offers coax or TosLink jacks) and digital outputs with selectable sampling frequency, word length, and dither on/off. These features allow you to use the 840C as a DAC, or to upsample digital signals (either from the internal CD drive or for processing external digital sources) for output to an outboard DAC. (See Specs & Pricing for a full list of supported sampling rates and word lengths.) You can even name the two digital inputs, with the name appearing on the 840C's front-panel display when that input is selected.

The custom filter is built on an Analog Devices "Blackfin" 32-bit DSP chip running upsampling software provided by a company called Anagram Technologies. Cambridge calls the filter algorithm Adaptive Time Filtering (ATF), presumably because the filter is optimized for time-domain response (conventional digital filters are optimized for frequency-domain response). The filter upsamples the 44.1kHz/16-bit audio from the CD (or external source) to 384kHz/24-bit.

Dual Analog Devices 24-bit DACs convert the digital data to analog signals differentially. That is, the left and right channels are each split in the digital domain to create balanced signals, and converted to analog with two DACs per

channel (one for each phase of the balanced signal). This technique reduces DAC-induced distortion (artifacts common to both halves of the balanced signal cancel due to common-mode rejection) and lowers the noise floor. It requires, however, double the number of DACs and analog output stages compared with conventional conversion. Another benefit of differential DACs is that a balanced signal is created without the penalty of a phase splitter in the analog domain. This is the right way to create a balanced output signal from a digital source, rarely used because of the expense.

The 840C gives you the option of adding dither, a small amount of noise that increases resolution at the expense of a slightly higher noise floor. A selection in the menu allows you to turn the dither on and off.

To top it off, the 840C employs a custom transport mechanism that's considerably beefier than standard-issue transports. A custom clock drives the DACs, and reportedly has less than 130 picoseconds of correlated jitter.

Finally, the power supply is massive, with a huge toroidal transformer, lots of filter caps, and rows of power-supply voltage regulators. I counted a whopping nineteen TO-3 regulators, about triple the number usually found in CD players of this price. Independent regulators for each subsystem increase the isolation between circuits, which often translates to better sound. **RH**

840C's utter grace and ease, the player is tremendously good at resolving fine detail. This rare combination of ease and resolution is an important factor in musical involvement and long-term listening satisfaction.

Dynamically, the 840C was good, but not out-of-the-ballpark great as it is in every other sonic criteria. Microdynamics were rendered with good resolution, but the 840C isn't the last word in slam, impact, and "jump factor." Through the balanced outputs, however, the sound is considerably punchier, with tauter and more muscular bass. If you have balanced inputs on your preamp, you should use them.

CONCLUSION

The Cambridge 840C CD player delivers the best CD playback I've heard from any player under $5k—and it costs $1799. Not only is the 840C easily the greatest value in digital sources in my experience, it must be considered one of the greatest bargains in all of high-end audio. Even if your budget for a CD player is considerably more than $1799, I encourage you to audition the Cambridge 840C. I could live with the 840C at the front end of my $100k reference system—it's that good. **tas**

SPECS & PRICING

Cambridge Audio Azur 840C CD Player
Analog outputs: Balanced on XLR jacks, unbalanced on RCA jacks
Digital outputs: Coaxial on RCA, optical on TosLink
Digital inputs: Coaxial on RCA, optical on TosLink (two each)
Digital input word lengths supported: 16–24 bits
Digital input sampling frequencies supported: 32kHz, 44.1kHz, 48kHz, 88.2kHz, 96kHz, 176.4kHz, 192kHz
Digital output sampling frequencies supported: 32kHz–192kHz (pass-through); 48kHz, 96kHz, 192kHz upsampled
Analog output upsampling: 384kHz/24-bit

Digital-to-analog converters: Dual Analog Devices AD1955 24-bit
Digital filter: Analog Devices Blackfin DASP-BF532 32-bit DSP, running ATF software, upsampling to 384kHz/24-bit
Dimensions: 16.9" x 4.5" x 14.7"
Weight: 18.7 lbs.

ASSOCIATED EQUIPMENT
Wilson MAXX 2 loudspeakers; Mark Levinson No.326S preamp; Mark Levinson No.433 power amplifier; MIT Oracle MA loudspeaker cables, MIT Magnum MA interconnects; Shunyata Hydra-8 power conditioner and Shunyata Anaconda power cords; custom-built room treated by Acoustic Room Systems

U.S.
Price: $1799

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- 2x, 4x and DSD up-sampling in the 32 bit domain
- 32 bit digital volume attenuator
- 3 high-resolution digital inputs (USB, SPDIF, optical)
- FIR and short-delay apodizing filters
- VOSP SACD/CD playback mechanism
- Digital re-clocker (2nd PLL).

D-07
- 32 bit dual mono DAC / 32 Bit Digital Attenuator
- PCM to DSD conversion
- 2x, 4x and DSD up-sampling in the 32 bit domain
- Word Sync and 2nd PLL
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Bryston BCD-1 CD Player

One for the Ages

Alan Taffel

BCD-1 CD Player

There has been a lot of talk lately about "buying your last CD player," and understandably so. The format has come under assault from above and below. At the low end, we find Millennials downloading compressed MP3s rather than purchasing CDs. And the high-enders, largely unsatisfied with CDs from the get-go, can barely wait to do their own downloading of files whose resolution trounces that of the silver disc, and whose playback from magnetic drives is, by all empirical evidence, sonically superior to CD's optical mechanism. CD's plight only worsens when surveying the portable music scene. The Discman is long dead, replaced by iPods and their ilk, and automakers can't incorporate MP3-compatibility fast enough. No doubt, then, the future for music reproduction—at any resolution, whether for the home or portable— will be computer-based (with maybe a little Blu-ray thrown in), leaving CD an orphan.

And yet, having had no real alternative for the majority of releases over the past few decades, we audiophiles have accumulated a substantial catalog of CDs. So the timing seems auspicious to buy that last CD player—one built for the ages, one that will do full musical justice to our libraries, even as new releases inevitably slow to a trickle.

But what makes a CD player an ideal "last" player? Upon the arrival of the equivalent historical moment for LPs, the industry and consumers agreed that "last turntable" really meant "expensive turntable." This widely accepted premise was not mere marketing hype; it was grounded in the reality of turntable production. Turntables (and tonearms, for that matter) are, first and foremost, exercises in mechanical engineering. In this field, principles do not change over time; mass and materials are paramount considerations; and R&D is costly because the industry at large applies so little intellectual capital (relatively speaking) to the field. Meanwhile, manufacturing output is low, so production costs are inevitably high. For all these reasons, building a truly great turntable has always been a pricey proposition. And buying the most expensive one you could afford as your last made sense.

However, CD players are entirely different. True, these components, like turntables, must employ solid mechanical engineering. But once the bits are off the disc, the bulk of what CD players do is digital. Clocking, jitter reduction, D-to-A conversion, and filtering are the biggies on the block diagram. And those functions, like all silicon-based processors, are subject to Moore's Law, which basically states that you can count on an exponential growth in power even as costs plummet. The internal components of CD players also benefit from a massive intellectual capital investment and from economies-of-scale. So the best, last CD player may not be the most expensive one at all—it's more likely to be the most recent.

Enter the new Bryston BCD-1, the venerable Canadian firm's first CD player. Why introduce such a unit now, when its competitors have had comparable models in their stables for years? Simply put, the company decided to wait out the SACD/DVD-A format war. The fact that there was essentially no winner allowed Bryston to do what it really wanted to do all along: eschew DVD-based drive mechanisms—which are compulsory on universal players but which compromise CD sound because the clock speed is not an even multiple of CD's 44.1kHz sampling rate—in favor of a purpose-built, CD-only drive.

To this it has added a DAC that employs an advanced, hybrid multibit/Delta-Sigma, 24-bit/192kHz, 128x over-sampling Crystal chip, and rigidly synchronized both the DAC and the transport to a master clock. This arrangement essentially abolishes the jitter inherent in S/PDIF connections. The final block in the diagram is the analog output stage. Here, rather than pulling a chip-based op-amp off the shelf, as most CD-player manufacturers do, Bryston pressed into service its own highly refined, fully discrete, true Class A circuit.

All these premium parts nestle within a handsome, rigid chassis that exemplifies Bryston's legendary reputation for build-quality. Ditto the solid aluminum remote, which offers satisfying heft, positive tactile feedback, and backlighting. The chassis also houses both balanced and unbalanced outputs, S/PDIF, AES/EBU, and TosLink digital outputs (though I can't imagine why you'd want to bypass the internal DAC), a 12-volt trigger, and an RS232 jack for software upgrades. And thanks to Moore's Law and the other aforementioned fortuitous trends,

EQUIPMENT REVIEW - Bryston BCD-1 CD Player

a BCD-1 can be yours for the staggeringly reasonable price of $2695.

Still, features, internal goodies, and brawny construction, while necessary, are insufficient to qualify a CD player for the ages. Performance, too, must be extraordinary. And here is where the full scope of Bryston's achievement becomes apparent. CD sound simply doesn't get much better than this. Is the BCD-1, even at its relatively modest price, of reference caliber? Unquestionably. It certainly matches the considerable virtues of my own reference, a superb Goldmund transport and DAC combo that has, overall, shamed everything else I've previously heard. But the Bryston gives no ground in musically critical areas such as dynamics, where it delivers the full measure of dramas large and small, and detail resolution.

The reader might now have the impression that, aside from these minor distinctions, the Bryston and Goldmund setups sound remarkably similar. Well, they don't.

The latter capability is closely related to—but not the same as—the ability to separate and allow the listener to follow multiple musical lines. This is where the BCD-1 really rises above competitively priced units. The Resolution Audio Opus One CD, for example, is only slightly more expensive than the Bryston, and boasts many of its own charms. But the BCD-1 is simply in a different league, with the greatest disparity being what Linnies would call the ability to "follow the music." A fine example is the addictive Radiohead track, "Everything in its Right Place" from *Kid A* [Capitol]. The song

sounds great through the Resolution, but the Bryston allows me to hear everything going on within the deeply textured mix. Of course, this feat requires excellent resolution, but both players have that. Only the BCD-1 brings out these hidden musical layers, and effortlessly integrates them within the musical whole. This is something reference-level players do, while lesser units do not.

Admittedly, the Bryston and my own reference are not always equal—in some areas, the Bryston is better. Its timing, for example, is quicker; not only are attacks sharper, but rhythms in general are tighter. And while both the reference and the Bryston deliver wonderfully realistic timbres, the BCD-1 is ever-so-slightly better at capturing an instrument's unique essence. Listen, for instance, to the opening of Prokofiev's *Romeo and Juliet* as captured on the excellent Mercury CD. The composer allows each section of the orchestra its moment in the spotlight, and the Bryston shines a light on each with brighter illumination and, thereby, reproduces them with more recognizable sonic truth.

The reader might now have the impression that, aside from these minor distinctions, the Bryston and Goldmund setups sound remarkably similar. Well, they don't. With the Prokofiev the Goldmund has a richer, more voluptuous sound; the Bryston is lighter and cleaner. Note my deliberate use of the word "light" rather than "lean." I do not care for components that are tonally lean, that is, those that rob music of warmth and natural opulence. Never once did the BCD-1 strike me as lean or analytical (leanness' kissing cousin). So what is going on here? The answer is that the Bryston exposes the Goldmund as being

darker than neutral, with a false richness caused by a mild boost in the midbass. The effect is not unpleasant (and is, in fact, rampant in high-end components), but can cause trouble with certain source material. One example is the ultra-pure Nils Lofgren *Acoustic Live* CD [Vision]. Through the Goldmund, Nils' guitar sounds mildly but inescapably bloated. The Bryston, with its greater neutrality, depicts the guitar far more naturally. Further, the BCD-1's superior timing renders the entire album much more lively and buoyant. Light, but not lean.

To fully experience what the Bryston is capable of requires a few set-up considerations. First, do not skimp on interconnects; this player deserves good ones. Second—and this almost goes without saying—replace the stock power cord with an audiophile-grade unit. Finally, although Bryston did put some effort into vibration isolation, the job is not complete until the BCD-1 is perched atop a good set of cones. I used those made by Goldmund, which are well worth their price if you can find them, and their effect was not subtle. Have you read elsewhere that the BCD-1 can sound "dry," or that its soundstage is a mite squished? Those potential detriments are real, but they fall by the wayside if you use good cones and ancillary equipment. The biggest effect of the cones is in the bass, which becomes much clearer. But the cones also reduced glare and increased spatiality. Obviously, these are benefits you wouldn't want to miss.

In sum, I am as surprised as anyone to discover that the best "last" CD player might well be an unprepossessing, modestly priced machine. But the Bryston BCD-1 has all the chops to qualify for the honor, from state-of-the-art digital

components, to an audiophile-grade analog output section, to the feature flexibility and durability that will allow it to serve its role over the long run. Most importantly, it gets the music just right. This player is like a friend who takes a book from the shelf and opens it to a particularly enticing passage for your reading pleasure. That is what the BCD-1 will do for your CD library: Open it, and present it to you in all its glory. tas

SPECS & PRICING

Bryston BCD-1 CD Player
Analog outputs: RCA single-ended analog, XLR balanced
Digital outputs: S/PDIF, AES/EBU, TosLink
Dimensions: 19" x 3.1" x 11.2"
Weight: 18 lbs.

U.S.
Price: $2695

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Esoteric SA-50 CD/SACD Player

The Swiss Army Knife of the Digital Age

Robert Harley

Esoteric's new SA-50 CD player offers a host of features and capabilities unimaginable to the original designers of the CD format. Not just a CD player, the SA-50 will spin SACDs, decode other digital sources (including a music-server's output via USB), switch between digital sources, and drive a power amplifier directly. If you have no analog sources in your system, the SA-50 obviates the need for a preamplifier. This architecture not only saves money, but provides better sound quality. After all, the best preamplifier is no preamplifier.

Moreover, the SA-50 is packed with sophisticated technology. The unit offers a wide range of user-selectable upsampling options, a two-stage PLL reclocking circuit to reduce jitter from external sources, a dual-mono implementation of a new 32-bit DAC, a clock input, and a new minimum-phase digital filter (selectable) that doesn't introduce pre-ringing (a type of time-domain distortion).

Although well built and attractive, the SA-50 looks more business-like than Esoteric's more costly offerings. Priced at $5800, the SA-50 is clearly intended to be an extremely high-value product that provides a multifaceted answer to the needs of today's complex digital sources. In fact, it struck me that the SA-50 is nearly functionally equivalent to the $23,000 dCS Puccini/U-Clock combination I reviewed in Issue 200. The Esoteric lacks an asynchronous USB interface and external clock, but will do much of what the dCS does at a fraction of the price.

How can Esoteric offer so much for so little? The only concessions to cost that I could see were the flat front faceplate (more expensive Esoteric models have sculpted front panels), a plastic rather than a metal drawer, and…that's it. The unit is built like a tank, weighing in at nearly 40 pounds and featuring Esoteric's three-point vibration-isolation feet. The transport mechanism is a more cost-effective implementation of Esoteric's vaunted VRDS transport, known in the SA-50 as VOSP (Vertically aligned Optical Stability Platform). Although scaled down in implementation, the transport nonetheless maintains the key element of clamping the entire disc surface rather than just a small area at the disc's center. The custom toroidal power transformer is huge, accounting for a good percentage of the unit's heft.

Let's take a closer look at the SA-50's rich feature-set. The Esoteric offers balanced and unbalanced outputs, along with either fixed or variable operation. In the fixed-output mode, the SA-50 functions as a conventional CD player, with the volume controlled by your preamplifier. In the variable mode, the SA-50 drives a preamplifier directly with volume controlled by the SA-50's remote control. The amount of attenuation is shown in the front-panel display. Note that you must go into the set-up menu to activate the variable-output mode (the default is fixed output). This could cause a problem the first time you connect the SA-50 directly to a power amplifier and play a disc, expecting to be able to adjust the volume via the remote control—you'll get a surprise in the form of a full-level signal driving your power amplifier.

You can specify in the set-up menu whether pin 2 or pin 3 of the XLR analog-output jack is "hot." This allows you to determine if the SA-50 is polarity inverting or not. If used with a "pin-2 hot" power amplifier, choosing "pin-2 hot" in the set-up menu means the SA-50 is not polarity inverting.

Three digital inputs (USB, coaxial, TosLink) are provided along with front-panel source-switching between these inputs. The combination of a volume control and multiple inputs allows the

EQUIPMENT REVIEW - Esoteric SA-50 CD/SACD Player and DAC

SA-50 to serve as your system's control center, selecting sources and adjusting the volume. Note, however, that the USB input is limited to 48kHz/16-bit data. This is surprising given that most listeners' music servers will be loaded with high-resolution files, and that 96kHz USB is common. The coaxial and TosLink inputs are, however, specified at 192kHz/24bit.

The SA-50 offers four upsampling options. The most basic is called "ORG" (for "original") which simply puts the signal through an 8x oversampling digital filter. In the "2FS" mode, the unit upsamples 44.1kHz (either from an external source or from the internal disc drive) to 88.2kHz (or 48kHz to 96kHz). The "4FS" mode converts 44.1kHz data to 176.4kHz (and external 48kHz data to 192kHz). Finally, selecting "DSD" converts the PCM data to the 2.8224MHz single-bit Direct Stream Digital format for conversion to analog. This latter feature, in my experience with this and other players with the same capability, offers better sound than when the PCM signal is converted to analog. In addition, when playing an SACD, the DSD bitstream is kept in its native format and converted to analog with a single-bit DAC. This is the only way to hear the full glory of DSD.

You also have the option of two digital-filter algorithms, "FIR" (finite impulse response) or "S_DLY" (short delay). The FIR filter is a conventional type that's been used since the first days of the CD format (although, here, in a more powerful and sophisticated implementation). The S_DLY filter is a minimum-phase type that adds no pre-ringing to the signal. The recent trend has been away from FIR filters, which some believe are responsible for CD's hardness and flat soundstaging, toward

minimum-phase filters. Note that when playing SACDs, or converting PCM to DSD, neither of these filters is used. (The single-bit DSD format doesn't require a reconstruction filter, which is perhaps one reason DSD sounds superior to standard-resolution PCM.) The filter is the new AK4399 from the Japanese semiconductor manufacturer AKM.

A rear-panel BNC jack accepts a word-clock signal from an external clock. If you don't use an external clock, the SA-50 uses a dual-PLL for receiving external digital signals. A dual-PLL reduces jitter, although it slightly increases the time the SA-50 takes to lock to a digital input. (The PLL isn't used when playing a disc in the SA-50's transport.) Finally, a digital-output jack is provided for driving another digital device.

All these adjustments and configurations are accessed through the set-up menu and front-panel display. The menu system is straightforward and simple to use. I must also comment on the SA-50's outstanding remote control. It feels great in the hand, the volume buttons are large and fall naturally beneath the thumb, and the buttons are laid-out well.

Inside, the SA-50 uses dual-differential DACs along with a fully balanced analog-output stage. That is, a balanced signal is created in the digital domain, converted to analog differentially by two DACs per channel (one for each phase of the balanced signal), and then amplified/buffered by two analog stages per channel. This is the right way to create balanced outputs. The DACs are a new chip from AKM, the AK4392, a dual 32-bit device with an internal filter and volume adjustment, although the internal filter isn't used in this application.

LISTENING

I first evaluated the myriad filter and upsampling options to determine which was best in my system. In comparing the FIR filter to the S_DLY filter, I thought that the FIR filter was more immediate and vivid sounding, with slightly better grip in the bass. The S_DLY filter was smoother in the treble, a little softer overall, less incisive, and more relaxed. The S_DLY filter setting also produced a more spacious soundstage. I preferred the S_DLY setting, although you should listen to both. I can imagine some listeners preferring the FIR filter's more up-front presentation. This filter choice was, however, rendered moot by my preference for the PCM-to-DSD conversion mode, which doesn't invoke either filter. The conversion to DSD was decidedly superior to any of the PCM upsampling modes, which all had a trace of glassiness overlaying the treble compared with the DSD conversion. Converting PCM to DSD resulted in a smoother and more relaxed treble, along with a greater sense of ease overall.

With these settings, and with the SA-50 driving the Pass XA100.5 amplifiers directly via its balanced outputs, the Esoteric player exhibited some outstanding qualities that were surprising given the machine's extensive capabilities and reasonable price. The SA-50's most salient characteristic was a palpability and immediacy in the midrange that fostered a sense of "you are there" realism. The presentation was incisive, upfront, and vivid, with crystal-clear delineation between images and the space around them. In its ability to present the music as individual instruments existing in space, the SA-50 was exceptional. The soundstage was the antithesis of thick, confused, or opaque. In addition, the

SA-50 presented instruments against a jet-black background, along with a halo of air around the images.

The SA-50's immediacy was partly the result of a slight tendency toward forwardness in the upper midrange that emphasized certain instruments. In addition to giving the presentation greater presence, this slight "spotlighting" also tended to somewhat dilute the saturation of tone colors. For some systems, the SA-50's somewhat forward rendering may be a liability. For my system and taste, the SA-50 was right at the threshold of being too assertive. Nonetheless, I greatly enjoyed the sense of palpability and the feeling that nothing was coming between me and the instruments. For example, Buddy Guy's great fusion of modern and traditional blues, *Sweet Tea*, starts with a track that features just him and an acoustic guitar. The SA-50 beautifully conveyed the track's eerily realistic presence and intimacy (both sonically and in the expression of the lyrics), projecting the images just a little farther in front of the loudspeakers than I've heard from my reference digital playback.

Going back to the SA-50's soundstaging, the Esoteric's excellent separation of musical lines greatly contributed to the player's ability to convey a recording's musical values. The clarity with which the SA-50 resolved the timbral and spatial qualities of instruments was no doubt a factor in the consistently powerful sense of music-making it produced. That is, the presentation never sounded like a collection of sounds. Rather, I heard a strong sense of musical coherence, meaning, and expression that not all digital front ends get right. It's an intangible quality that has nothing to do with treble balance, timbral purity,

EQUIPMENT REVIEW - Esoteric SA-50 CD/SACD Player and DAC

or most other sonic criteria, but one that in many ways defines the listening experience.

The SA-50's bottom end was full, tuneful, and articulate, with good dynamic impact on kick drum. I also enjoyed the Esoteric's powerful rhythmic drive and expression. The extreme bass was slightly rolled off compared to that of other digital front ends I had on hand for comparison. This had only a very slight effect on the impact of bass drum, for example, but was audible with very low frequencies such as organ pedal point. Again, the bass roll off didn't affect the timbre of instruments, only the sense of weight and room pressurization from organ pedal tones such as those on Track 7 of Rutter's Requiem [Reference Recordings].

Esoteric has always been at the fore front of delivering great sound from the SACD format and the SA-50 continues that tradition. When playing SACD, the gap between the SA-50 and my reference narrowed considerably. SACDs through the SA-50 had tremendous resolution, ease, spaciousness, and delineation of instruments.

However, when I used the SA-50 as a DAC for my music server (connected via the USB input), I found that the sound noticeably dropped in quality. Switching to the USB input made the sound hard, grainy, and bright, lacking air and transparency. But even comparing CD playback from the SA-50's disc drive

against the same music that had been ripped to the server, the disc playback was significantly better sounding in every way. This is at odds with my previous experience, which suggests that music read from a hard drive sounds better than the same data read from a CD (and decoded by the same DAC). The culprit is not the SA-50's digital-input circuitry or DACs, but rather the USB interface itself. I discovered this by running USB from the server into a dCS U-Clock, which receives the data via USB and converts it to S/PDIF, and then feeding this S/PDIF to the SA-50. The U-Clock's Asynchronous USB interface eliminated the detrimental sonic effects heard when driving the SA-50 directly from the server. If you plan to connect a music server to the SA-50 via the USB interface, I suggest adding a USB-to-S/PDIF converter such as the $495 Bel Canto USB Link 24/96. The Bel Canto lacks Asynchronous USB, but nonetheless should produce better sound than going straight into the SA-50 with the server's USB output. Moreover, I expect to see stand-along Asynchronous USB-to-SPDIF converters on the market later this year.

Out of curiosity, I also used the U-Clock's clock output to drive the SA-50, resulting in another improvement in sound quality. One would probably never use a $5000 clock with a $5800 player, but the comparison was interesting, nonetheless. It's fascinating to listen to

the effects of timing precision on digital-audio reproduction, and I jump at any opportunity to expand my knowledge and experience.

CONCLUSION

The Esoteric SA-50 is a remarkable piece of equipment, performing many of the different functions required of today's evolving digital front ends. The SA-50 is the ideal solution for those with no analog sources who would like to forego using a traditional preamplifier. Even without these features, the SA-50 would stand on its own at the price for a CD/SACD player. The Esoteric also delivers musically, with a tremendous sense of palpability, clarity, transparency, and resolution. The presentation tends toward the incisive and vivid side, a characteristic that will suit some systems and listeners more than others.

My only reservation about the SA-50 is that the sound through the USB input doesn't begin to suggest the full quality of which the SA-50's outstanding DACs are capable. This is a fundamental limitation of an "adaptive mode" USB interface, but one that can be overcome with an outboard USB-to-SPDIF converter.

With that minor caveat, I can enthusiastically recommend the Esoteric SA-50—the Swiss Army Knife of the digital age. tas

SPECS & PRICING

CD/SACD player and DAC

Disc formats: CD, SACD, CD-R/CD-RW

Analog outputs: Unbalanced on RCA jacks, balanced on XLR jacks

Digital inputs: USB ("B" type connector, input up to 48kHz/16-bit), coaxial (input up to 192kHz/24 bit), TosLink (inputs up to 192kHz/24 bit)

Digital outputs: Coaxial, TosLink

Word clock input: TTL levels, 75 ohm, BNC jack (can lock to 44.1kHz, 48kHz, 88.2kHz, 176.4kHz, 100kHz)

Dimensions: 17 3/8" x 6" x 13 7/8"

Weight: 39.6 lbs.

ASSOCIATED COMPONENTS

Wilson Audio Sasha loudspeakers, Basis 2800 Signature turntable with Basis Vector 4 tonearm, Air Tight PC-1 Supreme cartridge, Aesthetix Rhea Signature phonostage; PC-based music server (built by Goodwin's High-End), Classé Audio CDP-502 CD/DVD-A player, Berkeley Audio Design Alpha DAC, dCS Puccini/U-Clock CD/SACD player and DAC; Pass Labs X20 preamplifier; Pass Labs XA100.5 power amplifiers; MIT Oracle MA interconnects; MIT Oracle MA speaker cable; Running Springs Audio Dmitri, Shunyata Hydra-8, Hydra-2, and V-Ray AC conditioners, Shunyata Anaconda, Python, and King Cobra CX AC cables; Shunyata Dark Field cable elevators; room custom designed and built, acoustic design and computer modeling by Norm Varney of AV RoomService, acoustic treatment and installation by Acoustic Room Systems (now part of CinemaTech)

U.S.

Price: $5800

TEAC AMERICA, INC.
Esoteric Division
7733 Telegraph Road
Montebello, CA 90640
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Cary Audio Classic CD 303T SACD Professional Version SACD Player

Versatility to the Max

Wayne Garcia

One thing you have to love about our audiophile community is the way we passionately embrace and hold on to technologies that the rest of the planet either views as outmoded or, in some cases, was never even aware of to begin with. Conversely, we are also more than happy to accept new technologies—should they prove to be sonically superior to those that came before. And frequently, these sometimes very different approaches to music playback—tube versus solid-state and analog versus digital are the two most obvious examples—simply coexist for as long as there are enough people willing to support them.

When it comes to the current state of digital audio reproduction, it's understandable that many of us are feeling a bit of anxiety on just where things are headed. Standard Red Book CD, whose reproduction quality has become markedly better during recent years, is teetering on the verge of… uh, I'm not really sure what. Although more and more people are downloading music, CDs don't (yet) appear to be disappearing. SACD, which is essentially a non-issue to the mainstream listener, appears to be thriving in the audiophile

market, where many vinyl reissue specialists are releasing the same titles on both LP and SACD, and, I'm told, with equal commercial success. At the same time, music servers, and we won't even get into the ubiquitous iPod, are increasingly popular for at least a few obvious reasons: less physical clutter—who, I ask, actually likes CD jewel cases?—higher resolution, as well as easy access to an entire music library. Still, some of us like owning, handling, and interacting with physical media—something that no digital

replacement can duplicate, and, hence, one of the LP's undeniable attractions and advantages. Just as e-readers are becoming another way to read books, newspapers, and magazines—for many of us, though not yet this writer—music servers are becoming another avenue to the enjoyment of music.

> **Cary's latest digital disc player is a remarkably complete digital-audio playback device, with selectable upsampling frequencies of 96, 192, 384, 512, and 768kHz.**

Given that we're standing in the middle of this digital audio crossroads, it seems logical for anyone contemplating the purchase of a new digital disc player to pause and ask a simple question: How, for at least the foreseeable future, do I envision using this thing? If you

want maximum flexibility and at least a smidge of a feeling that your new component won't be obsolete before you open the box, then you'll want something like Cary Audio's Classic CD 303T SACD Professional Version SACD player.

Although that cumbersome nomenclature includes the SACD tag, don't be misled—Cary's latest digital disc player is a remarkably complete digital-audio playback device. Its bag of tricks includes Red Book CD playback with selectable upsampling frequencies of 96, 192, 384, 512, and 768kHz. That's an unusually large range—big enough to keep the tweakers among us busy for eons. But the advantages of upsampling may have its limits, as we'll get into shortly. The player also decodes HDCD-encoded CDs, which, though hardly a new technology, remains one of the most satisfying ways to achieve higher performance from the compact disc. SACD playback is, of course, a given here, both in two-

EQUIPMENT REVIEW - Cary Audio Classic CD 303T SACD Professional Version SACD Player

channel and multichannel modes. And unless told to do otherwise, the player automatically selects the SACD portion of multi-layer discs. Moreover, the 303T's internal clock processes DSD at 22.5792MHz, which is double the norm.

But high-quality CD and SACD playback is only part of what the 303T has to offer. It can also be employed as a standalone DAC complete with RCA, TosLink, and 24-bit/192kHz USB inputs. The latter is compatible with Windows (but not with the MAC OS), and Cary includes a CD-ROM to get you set up. Digital outs are available via AES/EBU, TosLink, and coax connections, while analog outputs offer single-ended RCA as well as balanced XLR.

> **Selecting the tube output section didn't simply add warmth and a touch of romance to the sound; it also added rewarding layers of complexity, air, and texture.**

Speaking of analog choices, the 303T's analog output can run in either solid-state or tube mode, and the latter's suite of four 12AU7s can be easily accessed, rolled, or replaced by means of a separately removable tray inserted into the unit's top plate.

As I said, the 303T offers maximum flexibility—as well as ease of use. As you might imagine, all of these playback options require lots of switching. In that regard, both the 303T's front panel and its remote-control wand are busier than those found on most disc players. But the Cary design team has somehow managed to arrange all these buttons in a fashion that, with just a bit of use, quickly becomes familiar. In my opinion this ease

of use is crucial to the end user of a device that is, after all, supposed to bring us pleasure, not another dose of aspirin.

At 34 pounds the 303T is solidly built and very nicely finished. The chassis sits on four adjustable conical feet designed to minimize vibrations from whatever rack or shelf the unit resides on.

Finally, the 303T carries a price tag of $6500. Not exactly pocket change, but also not crazy money in our high-end world, especially given what this machine not only offers in flexibility but also in sonic excellence.

The 303T's basic sonic signature is on the warm side, and that's true in the solid-state mode, too, before ever engaging the tube output stage. Yet this player doesn't sound sluggish, just warm in a way that highlights musical expressiveness (as opposed to more angular, detail-oriented designs that may arguably deliver more impressive hi-fi effects). For instance, at the start of Gershwin's "Oh, Lady, Be Good," from Analogue Productions' terrific-sounding edition of *Count Basie and the Kansas City 7*, we hear Basie's nimble piano work, the trench-digging bass of Ed Jones, and Sonny Payne's silky brushwork. Soon the rest of the ensemble joins in—two tenor saxes, trumpet, and guitar. The 303T swings right along with the music's ebbs and flows, presenting the tune with a matter-of-fact ease, clarity, and balance.

Michael Tilson-Thomas and the San Francisco Symphony's beautiful DSD recording of Mahler's Symphony No. 2 [SFS Media] is a great challenge to individual components as well as to the systems we evaluate them in. It's got air and dynamic range galore, as richly varied a palette of tone colors and textures as you'll find, orchestral scoring with the complexity to drive any system batty, as well,

of course, as two female vocal soloists, a large mixed chorus, and a crushing final statement for pipe organ. Again, the 303T showed its inherent warmth, impressive composure, and sweeping musicality. It also delivered a convincingly massive soundstage of well-layered three-dimensionality, with plenty of ambience. On the other hand, if you want every last inner detail, exclamation mark, or tympani stroke to "wow" you, well, that's not what Cary's 303T is about. What it will "wow" you with is a continuity of musical expression that, as noted above, sweeps you away with the musical whole.

After enjoying a large range of SACDs and CDs over the 303T, I chose an old favorite of average sound quality but undeniably great music, *Thelonious Himself* [OJC/Riverside], to describe what you might expect to hear from this amazing array of playback options. In the standard 44.1kHz setting, Monk's piano was big, clean, and rich, with a healthy percussive strike and good, if somewhat "clangy" harmonic structure. At 96kHz the sound became airier, more focused, transparent, and less brittle up top. The 192kHz setting was richer, and less strident still, but perhaps not as immediate sounding. Moving to 384kHz introduced a noticeably rolled-off top end as well as a veiling effect. And, for these ears, both the 512kHz and 768kHz modes were not only too soft and veiled, but also started to lose the music's structure and continuity of line.

Selecting the tube output setting didn't simply add warmth, or richness, and a touch of romance to the sound; it also added consistently rewarding layers of harmonic complexity, air, and instrumental texture. After sliding back and forth for comparisons sake, I must tell you that I ended

up leaving the 303T's tube stage engaged for the remainder of the evaluation period.

You, naturally, may disagree, which is part of what makes our hobby so much fun. But one remarkable thing about Cary's 303T CD/SACD/DAC is that, no matter what digital audio media one may want to play through it, the owner of this design has access to an unusually complete range of media and playback options. And though no device is absolutely "future-proof," the folks at Cary have come up with a terrific-sounding design that at least offers a glimpse of that possibility. **tas**

SPECS & PRICING

Cary 303T SACD player
Compatible discs: SACD, CD, CD-R, CD-RW
Outputs: Balanced XLR, single-ended RCA (analog,),
AES/EBU, coax, TosLink digital)
Tube complement: Four 12AU7
Dimensions: 5" x 18" x 15"
Weight: 34 lbs.

U.S.
Price: $6500

CARY AUDIO DESIGN
1020 Goodworth Drive
Apex, North Carolina
(919) 355-0010
caryaudio.com

Esoteric K-03 CD/SACD Player/DAC

Puzzles and Prizes

Alan Taffel

The Esoteric K-03 is not your usual CD/SACD player. Its designers have innovatively combined a highly refined CD/SACD transport and a full-function DAC within the same luscious chassis. Although on the surface that doesn't seem so unusual, consider that very few digital players even bother to include an input for external devices like PCs, music servers, and cable boxes. Many is the time I wished I could plug one or more of those sources into a CD or universal player that I *knew* had a great internal DAC. Recently, more designers have incorporated the digital input feature into their players. But even those models generally offer only one such input, and that interface is virtually never USB.

In contrast, the K-03 is a real DAC. There are three digital inputs—coax, TosLink, and USB—as well as filter and oversampling options, just like an outboard unit. There is even an input for an external word clock. You won't see *that* on many other digital players. Nor is the USB input an afterthought; it offers asynchronous clocking via one of multiple drivers, and supports the highest bit rates. The benefit of Esoteric's approach is compelling: Owners get the inherent sonic advantages of having the transport and DAC in one box, along with the versatility and multi-source support normally available only from stand-alone DACs.

This benefit also carries over to the K-01, the K-03's big brother. Although I begged for the $22,500 flagship, with its quad power supplies, magnesium disc clamp, and sixteen(!) DACs, a sample could not be made available in time for this issue. I was forced to "settle" instead for the penultimate $13,000 K-03. This model makes do with just two power supplies, a Duraluminum clamp, and a paltry eight DACs. Specifically, the K-03 employs eight 32-bit AKM DACs (four per channel). The two units do not share identical transports, either, but they do have the same new clock and discrete, fully balanced analog modules.

Besides being a hybrid player/DAC, the K-03 is also unusual in that it requires a great deal of configuration before it can—or should—be used. Most digital players are plug-and-play, but if you do that with this one you will not experience anything like its ultimate sound. Unfortunately, setup is no small task. First, one must endure a break-in period that amounts to over a month of continuous play. The second complication is due to those aforementioned upsampling, filter, and driver options. There are *four* upconversion choices, *three* USB drivers, and *five* filter settings. If that isn't enough, the settings interact with each other, thus requiring a listen to every combination in order to find the best—a rather daunting proposition.

And it needn't be, because it turns out quite a few of these options are clearly inferior and could have been omitted without sacrifice. Why they were included is a puzzle. If you are curious about my adventures in Optionland, they are recounted in the sidebar, "A Surfeit of Settings." If not, I can still save you a lot of trial and error by telling you that the "S_DLY1" "apodising" filter is the best, the choice between 2x and 4x upsampling is a matter of personal preference (although either one stomps the other options), and the asynchronous "HS_2" USB driver is the only one worth considering. I should also note that none of these is among the unit's default settings, which are invariably the worst in their respective category. Another puzzle.

Thankfully, configuration needs to be done only once. The K-03 may not be plug-and-play, but at least it's set-and-forget. And once set, any lingering consternation begins melting away. From a functional standpoint, the Esoteric has the silky-smooth operation, the weighty remote, and the rock-solid reliability one would expect in this price range. There are thoughtful touches, too, like the way the menu takes you to the parameter you last changed.

The K-03 has quite a few operating modes, and most—but not all—of them deliver reference-

EQUIPMENT REVIEW - Esoteric K-03 CD/SACD Player/DAC

quality sound. Surprisingly, the least impressive (which is not to say *un*impressive) mode is CD playback. I'll explain why this is so surprising later on, but for now let me describe the sound. First the good news. With CDs, the K-03 sounds beautiful, delivers exceptional detail and spatial depth, and is tonally ravishing. On the other hand, it is missing the ultimate resolution, openness, and freedom from digital edginess that makes today's reference gear so relaxing and engaging.

For an illustration, listen to "Bydlo" from the Colin Davis rendition of *Pictures at an Exhibition* [Philips]. Although this is not a stellar recording overall, the track is a good test of timing and resolution. With respect to the former, the low strings that open the track should maintain a lumbering pace but should never plod. Through the K-03, plod they do. As for resolution, listen to the snare drum that enters at about 1:30 into the track. Through my reference player—a combination Goldmund Mimesis 36 transport and the dCS Debussy DAC—the snare drum's "papery" quality is quite distinct. On the Esoteric, that quality is missing, as are other subtleties such as the hall acoustics that surround and firmly place the solo horn's position at the rear of the stage.

Of course, my reference rig is significantly more expensive than the K-03. Heck, the dCS all by itself costs nearly as much as the Esoteric player/DAC combo. My disappointment with the latter's CD sound is not so much because it does not equal the reference gear, but because it does not equal its own performance in other modes. So now that I have CD out of the way, let me turn to those.

First and definitely foremost is SACD playback. Simply stated, the K-03 is the best SACD player I have heard—not by a mile, by a marathon. When playing my standard SACD torture test, Stravinsky's Suite from *L'Histoire du Soldat* [Pentatone], the Esoteric imbues both strings and horns with a heretofore unheard burnished quality that amps up the disc's already astonishing realism. The K-03 brings out—but never exaggerates—every detail. Nor is there a trace of edginess, and music has tremendous drive. As good as this hybrid disc's CD layer sounds through my reference system, the K-03's SACD rendition simply annihilates it.

The K-03 is one sweet DAC. Rhythms are unflagging, details emerge clearly and naturally, and listener fatigue is non-existent. Dyanmics are suberb, as well.

This difference is not simply due to the higher inherent resolution and analog-like nature of the SACD format. I played, for example, the "Out of the Woods" track from Nickel Creek's eponymous first album through the estimable Marantz UD9004 universal player. The Marantz ably captures the liquidity that differentiates SACD from CD, but the UD9004 does not come close to matching the scale and sonic clarity of the K-03.

I was in for another treat when I connected my CD transport to the Esoteric's coax input. Oh my, the K-03 is one sweet DAC. As with SACD playback, rhythms are unflagging, details emerge clearly and naturally, and listener fatigue is non-existent. Dynamics are superb, as well. All of these factors raise the emotional quotient

mightily. Indeed, as a pure DAC, the Esoteric is very much in dCS territory. It can't quite match the latter's sublime effortlessness and bass definition, but the Esoteric feels more open and light on its feet.

By now it should be apparent that playing CDs through the K-03's DAC via an external transport yields results that are distinctly superior to the all-in-one mode. Take "Bydlo," for example. Using an external transport, those previously missing hall acoustics reappear. Further, the external transport wrings out all the music's drama, while the internal transport is emotionally circumspect. On a raft of material the K-03's DAC always proved more detailed and neutral—and therefore more enjoyable—when driven externally.

This should not be the case. Self-contained players at this level almost always sound better than an outboard transport, no matter how good, driving their DAC. The benefits of eliminating an S/PDIF connection and having a single master clock are nearly insurmountable. That the Esoteric does not follow this pattern is particularly puzzling. The only explanation I can see is that the K-03's internal transport, for all its pedigree, is holding back the unit's CD sound. Obviously, though, this comment does not apply to that same transport playing SACDs.

Back to the K-03 as a DAC, there is still USB to discuss. Here, again, Esoteric offers options. Three drivers are available. The first will load automatically when you connect the K-03 to a computer, while the other two must be downloaded from Esoteric's Web site. As already mentioned, I experienced the best results with "HS_2," the only driver that supports asynchronous clocking. However, all three drivers sounded significantly

better when supplemented with ASIO, which bypasses any and all OS detritus, so I highly recommend you download the free ASIO4ALL for your media player of choice (sorry, Mac users). Using ASIO has the added benefit of enabling the K-03 to automatically adapt to sample and bit rates, relieving you from manually setting those in the OS to match each source file you play—a major pain.

Once everything is set, the magic begins. I listened to the new 96/24 version of Tom Petty's "Refugee," downloadable from HDtracks, and the sound was stunningly open, clear, and revealing. From the same source I downloaded a high-res

SPECS & PRICING

Esoteric K-03 CD/SACD Player/DAC

Outputs: Stereo balanced analog, stereo balanced single-ended

Other connections: Coax, TosLink, and USB digital inputs; word clock input

Dimensions: 17 1/4" x 6 3/8" x 13 1/4"

Weight: 61.75 lbs.

U.S.

Price: $13,000

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7733 Telegraph Road
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EQUIPMENT REVIEW - Esoteric K-03 CD/SACD Player/DAC

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file of Keith Jarrett's *Köln Concert* and found it to be equally engaging. The real achievement of the K-03's USB is that it seems to have banished entirely the artificiality that has plagued this interface. USB is a fast-evolving format, which is fortunate since it started out sounding so bad. Now, even though I proclaimed the dCS Debussy the best USB I had ever heard only a few months ago, I am bound to say that the Esoteric is even better.

Lastly, the K-03 can serve as a linestage. Like many recent digital players and DACs, the K-03 has a built-in volume control and can directly drive a power amplifier, obviating the need for a separate linestage so long as all sources are digital. In this mode, the K-03's performance is largely dependent on the choice of music. Digital volume controls always cause a loss of resolution when they are turned down, and so pop/rock recordings, with their high mastering levels prompting a volume reduction, fare poorly. But classical music, which does not have hyped levels and so can be played with higher volume settings, sounded nearly identical to my reference linestage—an extraordinary accomplishment.

In sum, the Esoteric K-03 is a brilliant concept, and delivers in nearly every category. Though its builders went overboard on configuration options, the rest of the design is unassailable, as is its sonic performance in all but one area. CD playback is good, and disappoints primarily compared to the K-03's SACD, DAC, and USB performance, all of which are of reference caliber. I would urge you to give the Esoteric a listen, to hear just how good these can be. tas

A SURFEIT OF SETTINGS

Of the five filter settings, "None" is the inexplicable default. In this mode, the K-03 sounds airless and sickly, and dynamics are voting in absentia. "FIR1," a fairly standard filter with a variable frequency cutoff, is much better, with a breath of air and a modicum of timbral resolution. Unfortunately, FIR1's dynamics are so exaggerated that on accented notes I felt like I was being jabbed in the eyes. Moving along to "FIR2," which is the same as FIR1 but with a fixed 80Hz cutoff, the sound takes on softer leading edges and dynamics are more manageable, though accents remain off-putting.

With "S_DLY1" we move into "apodising" filters and the difference is stark. The entire presentation is less in-your-face, and there is more air. Dynamics finally settle down into a natural stance, and a reduction in midrange bloat makes it easier to hear things like individual violin strings. Meanwhile, though "S_DLY2" is also an "apodising" filter, it seems to ring plenty. Moreover, this option doesn't do much (though it does a little) to alleviate the issues I found with the FIR filters. FIR2 and S_DLY2 sound better or worse depending on the upsampling setting, but at their best neither compares with S_DLY1, the only option that really sounds "right."

As with filters, "None" is the default upsampling setting. And as with filters that is a shame, for this setting is characterized by sluggish rhythms, coarse dynamics, and uninformative timbres. Switching to "2X Upsampling" reveals what a poor decision it would be to leave the K-03's

upsampling disabled. Here, rhythms snap into place and for the first time are unconfused. Also for the first time, instruments and dynamics can, when the music calls for it, display a degree of delicacy. "4X Upsampling" adds freer (but not exaggerated) dynamics and greater rhythmic swing, though this setting is not quite as clean-sounding as 2X.

"Upsampling to DSD" is an option for which I had high hopes. Selecting it, I was struck by how completely different it sounded compared to every other setting. On orchestral material, I almost felt as if I were listening to a different string section—one with a markedly darker sound. That's not necessarily wrong (who knows how the real strings sounded), but the DSD setting was also noisier than—and not nearly as well-sorted dynamically as—the 2X or 4X options. As a result, in this mode certain pieces of music, like the second movement of the Vaughan Williams Sea Symphony, lose much of their mystery and subtlety. I therefore recommend choosing between 2X and 4X upsampling, either of which is a valid choice that will be determined by personal preference.

The standard USB driver, dubbed "Normal," is not bad, but it suppresses depth and transparency. Adding ASIO reaps a marked improvement in both areas. Drums and vocals emerge as if from behind a scrim, dynamics flourish, and there is much more detail with no downside. Nonetheless, the "HS_1" driver is worlds better, and better still with ASIO. HS_1,

though, betrays the synthetic string sound I hear from so much USB. "HS_2" is clearly the best of the lot, being more spacious and having better bass definition. Here, for the first time, ASIO does not make a night and day difference, though it does supply better-behaved rhythms and is generally airier.

As discussed in the main piece, ASIO would be recommended even if it made no sonic difference, because it gives the K-03 the ability to dynamically adapt to the incoming source's sample and bit rate. This is a feature no computer audiophile should be without. Its absence means either manually changing sample rates all the time, or leaving them set to the highest the OS will support, and suffering the harmful sonic effects of asynchronous upsampling.

dCS Puccini CD/SACD Player and Puccini U-Clock USB Converter/Clock

Virtuoso!

Robert Harley

I've lived with quite a few of the most ambitious digital-playback products in my twenty-one years as a full-time re-viewer, but somehow never managed to audition a unit from England's Data Conversion Systems (dCS) until now. That's a shame, because the dCS Puccini CD/SACD player and U-Clock combination has turned out to be one of the world's great digital front ends.

The company dCS has a long history of technical accomplishments in both professional and consumer audio. The firm pioneered many cutting-edge advancements, including the proprietary "Ring" DAC found in all its digital-to-analog converters (see sidebar). Mike Story, dCS's founder, was also at the forefront of high-resolution digital audio long before it was a commercial reality. I attended a paper he presented at an Audio Engineering Society convention in the early 1990s in which he correctly posited that the sonic improvement

rendered by high sampling rates was the result of improved time-domain performance due to the relaxed filter requirements. That's accepted wisdom today, but it was revolutionary nearly twenty years ago. Over the decades dCS has addressed such topics as upsampling, PCM-to-DSD conversion, jitter, noise-shaping, the time-domain performance of digital filters, and other issues long before they became part of the high-end mainstream.

dCS is again taking the technology lead with the U-Clock, a device that vaults the sound of the company's Puccini CD/SACD player into new sonic territory while simultaneously expanding its functionality to incorporate state-of-the-art decoding of high-resolution digital audio from a PC-based music server.

The $17,999 Puccini player is the same model Jonathan Valin commented on in his review of the Scarlatti, dCS's $67,000 three-box statement product (Issue 183). Jonathan concluded that the Scarlatti was the best digital he'd heard, an opinion apparently shared by quite a few high-

end manufacturers judging from the number who have purchased the Scarlatti for their own development work or trade-show demonstration. Jonathan also thought that the less-than-third-the-price Puccini was very nearly as good as the reference-quality Scarlatti.

The $4999 U-Clock improves the Puccini's sound quality by delivering an ultra-precise clock to the player, reducing jitter. As has become abundantly apparent, great-sounding digital audio requires extraordinarily precise timing in the conversion of digital data to an analog waveform. My review of the $16,000 Esoteric G-0Rb rubidium clock (Issue 180) created skepticism among certain readers that human ears can detect timing variations that are measured in picoseconds (see, for example, the letter from Dave Martson in Issue 198). The objections to expensive outboard clocks are not based on these readers' own listening

experience, but purely on theoretical grounds—conventional clocks should be good enough, in their view. But there's a simple way to determine for yourself if jitter is a factor in digital audio reproduction—listen to a Puccini with and without the U-Clock engaged. As we'll see in the report on my listening impressions below, the difference is not subtle.

The U-Clock's second important function is to allow the Puccini CD/SACD player to operate as a digital-to-analog converter for PC-based music servers that have a USB output. The U-Clock takes in digital audio data from a PC on the USB interface and converts it to S/PDIF for presentation to the Puccini. That might not sound like a big deal—one can buy a box for $250 that

EQUIPMENT REVIEW - dCS Puccini CD/SACD Player and Puccini U-Clock USB Converter/Clock

UNDER THE HOOD

The Puccini features an Esoteric transport mechanism (with a custom drawer) under dCS's custom-software control. In fact, all the software inside the unit is written by dCS. This software can be updated by downloading new code distributed by dCS on a CD. The custom digital filter is implemented in two DSP chips and two field-programmable gate arrays. Four filter types are available, selectable from the front-panel menu system. Filter 1 has the widest bandwidth and is the recommended setting. Filters 2 and 3 roll off at progressively lower frequencies. Filter 4 is the "measurement" filter, and isn't intended for listening. The filter choice affects the amount of out-of-band noise allowed through the system. All the filters are FIR linear-phase types.

When you select PCM-to-PCM upconversion on the front panel, the digital filter feeds a modulator that converts the PCM data to the 5-bit format required by the Ring DAC. If you select PCM-to-DSD upconversion, the filter's output goes through an additional step of converting PCM to DSD before the modulator that creates the 5-bit Ring DAC code.

The Puccini features exactly the same Ring DAC found in the $67,000 Scarlatti. This DAC, developed by dCS in 1992 and under refinement since, completely eliminates a source of distortion in conventional off-the-shelf DAC chips. It is implemented with 20 discrete devices per channel. Its fundamental nature lends itself to converting DSD signals, which is one reason why the Puccini sounds the best in this upconverting mode. (For a cogent explanation of how the Ring DAC works, see Jonathan Valin's sidebar on page 109 of Issue 183.) The Ring DAC's balanced output feeds a fully discrete Class A output amplifier. This is the signal that appears on the XLR jacks. The single-ended signal is buffered by an op-amp-based circuit so that output levels are consistent between the balanced and single-ended outputs (a balanced circuit inherently is 6dB higher in level). The power supply is a hybrid of switching and linear supplies that was newly developed from scratch for this latest generation of products.

The Puccini is a very advanced product, both in its design and capabilities for the user. Note, however, that the Puccini requires greater owner involvement than other CD players with its selectable upsampling, selectable filters, upgradable software, and extensive menu system. **RH**

does the same thing—but dCS has engineered a state-of-the-art USB interface that introduces absolutely no sonic compromises. Rather than considering USB a limiting factor in PC-based audio sound quality, dCS believes USB is the optimum interface if engineered correctly.

In most digital interfaces, including S/PDIF and AES/EBU (a variant of S/PDIF), the source component (the CD transport or PC music server, for examples) is the master clock to which the receiving device must lock. Virtually all USB DACs operate in this way, which is known as "Adaptive Mode." Asking the receiving device to lock to the source's clock is problematic for several reasons. Although the USB interface was never designed for transmitting high-quality audio, it inherently has the ability to allow the receiving device to control the data rate from the source device—a feature not possible with S/PDIF, AES/EBU, or even FireWire.

dCS has developed its own technology for exploiting USB's built-in "feedback" system which allows its own high-precision clock to serve as the master, forcing the source (the PC-based music server) to slave to that clock. This technique, called "Asynchronous Mode," transforms the USB interface into a high-quality interface. Rather than the computer establishing the clock precision (not a good idea for many reasons), the entire audio system is clocked by a high-precision crystal inside the U-Clock. Note that an asynchronous USB interface doesn't automatically confer low-jitter and better sound; it still must be implemented with a high-quality circuit.

Moreover, locating this asynchronous USB interface in a separate chassis (the U-Clock) rather than in the DAC itself has many benefits. First,

SPECS & PRICING

Puccini CD/SACD player/DAC
Conversion: dCS Ring DAC
Sampling frequencies: Up to 96kHz/24-bit
Inputs: S/PDIF (x2) on RCA, clock on BNC
Outputs: S/PDIF (x2), balanced analog on XLR, unbalanced analog on RCA
Dimensions: 18.1" x 4.4" x 15.8"
Weight: 26.6 lbs.

U-Clock
Outputs: Clock signal on BNC (x4), S/PDIF on RCA (x2)
Inputs: USB
Dimensions: 18.1" x 2.3" x 16.1"
Weight: 16.7 lbs.

U.S.

Puccini CD/SACD player/DAC
Price: $17,999
U-Clock
Price: $4999

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EQUIPMENT REVIEW - dCS Puccini CD/SACD Player and Puccini U-Clock USB Converter/Clock

noise in the PC is isolated from the DAC by the U-Clock. Second, the DAC needn't incorporate another clock running at a frequency unrelated to the audio-based clocks. Multiple clocks running at different frequencies within the same chassis can introduce cross-contamination.

The U-Clock is an apparently simple, yet brilliant, solution to adapting a CD player (the Puccini) to the needs of music-server owners. It solves sonic compromises of the USB interface with state-of-the-art design and implementation in a separate chassis, as well as allowing music-server users to decode files through the Puccini's outstanding DACs.

Although Jonathan covered the Puccini as a CD player in his review, let's recap the machine's highlights. The unit is simply stunning visually, with gracious curves and an unusual surface pattern etched into the shiny aluminum front panel. My only complaint is that the front-panel button markings are small and hard to read, a problem that diminished with familiarity. The drawer mechanism of the Esoteric-sourced transport is all-metal and operates silently and smoothly. A front-panel display allows the user access to a wide range of controls through an extensive menu system. One of these controls allows the user to select whether and how the signal is upsampled. One option is to convert any resolution PCM (from CD or files from a music server) to DSD before decoding (the other option is PCM-to-PCM upsampling). I found that the PCM-to-DSD conversion sounded the best, and this was the option I used for nearly all my auditioning. The display shows the clocking status via a clever icon of two gears meshing. The Puccini will decode 44.1kHz, 48kHz, 88.2kHz, or

96kHz, all with up to 24-bit word length. Note that it will not decode 176.4kHz (such as Reference Recordings HRx files) or 192kHz.

The rear panel offers both balanced and unbalanced outputs, along with digital inputs and outputs (two each on RCA jacks). A BNC connector accepts the clock signal from the U-Clock. The Puccini has a variable output, enabling it to drive a power amplifier directly. You can select a maximum output level of 2V or 6V; I recommend the 2V setting if you are driving a preamplifier.

The U-Clock matches the Puccini visually, and the two look stunning together. The front panel has just two pushbuttons and three LEDs. The leftmost button and accompanying LED is intriguing, to say the least. Marked "Dither," it modulates the clock edges in a controlled way in an effort to improve sound quality. It's counterintuitive that changing the timing of the clock edges could make the Puccini sound better, but dCS found that this small variation "exercises" the PLL in the Puccini and results in better sound. The modulation is easily filtered by the PLL. You can judge for yourself simply by turning dither on and off. The second button selects the clock frequency, either 44.1kHz (used for 44.1kHz sources and multiples of 44.1kHz, including SACD) and 48kHz (for 48kHz and 96kHz sources).

LISTENING

I started by listening to the Puccini as a CD and SACD player without benefit of the U-Clock. It was immediately apparent that this was one serious contender for the best digital I'd heard. The sound was immensely appealing, particularly the gorgeous, liquid, and glare-free midrange.

The presentation was a bit set-back rather than forward, with tremendous depth, clarity, and transparency. There was also an intangible sense of sonic coherence that manifested itself as a kind of "musical rightness." Whatever the Puccini was doing, it was different from other great digital I've heard.

After getting a general impression of the Puccini itself, I engaged the U-Clock. One little front-panel button-push vaulted what was already a spectacular sound into entirely new territory. The U-Clock snapped images into sharp(er) focus, increasing the sense of clarity, precision, and definition I had enjoyed from the Puccini alone. The heightened focus had a profound effect on the sense of instruments existing within an acoustic. Without the U-Clock, reverberation tended to be connected to the image itself, as though the image and the hall were merely variations of the same sonic cloth. With the U-Clock, the instrumental image was presented as a clearly defined object existing within an acoustic space rather than simply fused to it. The instrument and the surrounding acoustic were presented in a closer facsimile to what we hear it in life.

That was just the beginning of the U-Clock's magic. The Puccini's reproduction of timbre, which already had a bell-like clarity, was taken to a new level by the U-Clock. Timbres had greater palpability and realism, partly the result of less grain and edge (which were already very low) and partly because of greater resolution of textural detail. Similarly, the U-Clock made the Puccini's reproduction of transient information even more lifelike. The leading edges of piano attacks, for example, had a trace of edge that vanished with the U-Clock engaged. Listen, for example, to the

wonderful new recording of Vassily Primakov performing Chopin mazurkas on Bridge Records. The U-Clock made the piano more lifelike in transient attack, in richness of tone color, and particularly, in the sense of space surrounding the instrument. I pulled out this CD as a diagnostic tool to listen for specific sonic attributes of the U-Clock but immediately forgot about the sound and listened to the entire disc, completely captivated by the compositions and Primakov's expressive performance. Such an experience is always the sign of a great component.

In short, if you own a Puccini the $4999 U-Clock is an essential upgrade.

The Puccini/U-Clock combination was "plug 'n' play" with regard to the USB interface. I connected a generic USB cable from my fan-less, drive-less PC server to the U-Clock, selected the appropriate input on the Puccini, and the system played back my music files at a variety of sampling rates. I listened to files at 44.1kHz, 88.2kHz, and 96kHz from the server, as well as CDs and SACDs played in the Puccini's transport.

Getting back to the sound of the Puccini/U-Clock combination, I found myself consistently and deeply engaged with the music. The dCS pair had a different presentation than I've heard before from digital that is difficult to describe. The Puccini/U-Clock was distinguished by a pristine clarity of timbre along with a crystalline-like transparency of soundstage. It simply lacked the artifacts we associate with digital, such as a synthetic gray pall overlaying tone colors, grain and glare embedded in timbres, and a sense of haze or opacity between you and the music. Instruments and voices were vivid and alive, yet the presentation was never forward. In fact, the

EQUIPMENT REVIEW - dCS Puccini CD/SACD Player and Puccini U-Clock USB Converter/Clock

sound was relaxed and engaging despite the sense of immediacy. Background vocals were revelatory in that I could clearly hear the timbres of individual voices and how they blended into each other. I was also struck by the sheer realism of Neil Young's guitar on some 96kHz/24-bit tracks from *Harvest* sourced from the music server; it had more "guitarness" and less of a mechanical sound than I've heard from this track before. I got the impression of greater density of information, but not in an analytical way. I've heard a number of digital products that sound very clean, precise, and transparent, but those qualities are often accompanied by a mechanical character, a coldness or a stark sterility that doesn't foster musically intimacy. The Puccini/U-Clock's central triumph was the ability to sound super-pristine and precise, yet simultaneously warm and involving.

An analogy that came to mind to describe the Puccini/U-Clock's density of tone color and liquidity of timbre is of two identically colored bed sheets, one made from 600-thread-count cotton and the second made from 400-thread-count material. Put the 400-count sheet through the wash a few times and leave it in the sun for a day. Now compare the two sheets. The 600-thread-

count sheet is finer in texture, smoother, and more continuous. It's also more richly hued and vibrant. The Puccini's rendering of instrumental timbre is like that of the 600-thread-count sheet, while most other digital is analogous to the 400-thread-count sheet.

In addition to this remarkably naturalistic rendering of timbre, the Puccini threw a stunning sense of space and depth, revealing the size of the hall and the spatial relationships between instruments. In addition, the background was jet-black which further highlighted the sense of image tangibility. The pair's exceptional low-level resolution contributed to expansive sound as fine spatial cues in the back of the soundstage were rendered with great clarity. Reverberation decay was stunning in the way it maintained resolution down to the lowest levels, the smoothness of the decay, and the way it seemed to hang in space. This is one area where state-of-the-art modern digital is vastly better than earlier efforts, which truncated reverberation decay and sounded coarser and coarser at lower and lower levels.

I found the Puccini/U-Clock highly involving rhythmically. The bass was extremely punchy and dynamic, with a very tight and controlled quality. I heard a dynamic coherence from top-to-bottom, as though the music "gelled," heightening the feeling of musicians locking into a groove.

There's one area in which the Puccini/U-Clock significantly distances itself from all competition, and that is in the reproduction of very fine high-frequency transient detail. I was floored by the Puccini's resolution of micro-detail—think brushes on cymbals, shakers, the zils on a tambourine, gently struck triangles, and güiro. The lower the level and the more transient the nature of the

signal, the greater the extent to which the Puccini outshone other digital I've heard. Information that was simply blurred by other digital was resolved with pristine and vivid clarity by the Puccini. For example, the triangle on Rachmaninoff's Symphonic Dances had a delicacy that vividly conveyed the mechanism by which the sound was made. It wasn't just a high-frequency transient, but a pitch accompanied by a strong sense of attack, ringing, and decay. But the track that most dramatically illustrated the Puccini's unmatched performance in this area is the beginning of "Valentino" by Victor Feldman on the JVC XRCD title Audiophile (a compilation of two records made in the 1980s, engineered by the great Alan Sides). The track starts with a rain stick behind Hubert Laws' gentle flute passage. I've listened to this track countless times over the years, but have never heard the individual beads moving through the rain stick with such startling clarity. I point this out not because I enjoyed this quality for its own sake, but rather to illustrate how the Puccini accurately conveyed very fine transient detail, and how this fidelity fostered a sense of hearing the instrument itself rather than a reproduction of it.

It occurred to me that one reason the Puccini/U-Clock rendered timbres with such realism could be this fabulous resolution of low-level detail, particularly low-level transients. Musical waveforms contain a richness of micro-dynamic structure (a reed moving back and forth, for example); accurately conveying that structure makes instrumental textures and tone colors more lifelike. Although we're not consciously aware that the timbral realism is derived from this micro-transient information, it's simply one less cue to the brain that we're hearing a reproduction

rather than the instrument itself.

Although I don't have nearly as much experience with cutting-edge SACD playback as I have with CD, I thought the Puccini/U-Clock's rendering of SACD was the best I've heard. Interestingly, however, the Puccini/U-Clock's reproduction of CD was so good that it narrowed the gap I usually hear between CD and SACD.

Finally, you're probably wondering how the Puccini/U-Clock compares with the other great digital I've heard lately, including the Meridian 808.2 and Spectral SDR-4000 Pro CD players, as well as the Berkeley Alpha DAC. Starting with the Alpha DAC, the Berkeley unit was a bit more forward in spatial presentation, presenting the front of the soundstage a little closer to the listener. The Puccini's bass was leaner and tighter, with the Alpha DAC sounding "bigger" in the bottom end but somewhat less controlled. The Alpha DAC excelled at macro-dynamics with greater impact on timpani strokes, and also with a warmer and fuller rendering of bass guitar. As great as the Alpha DAC is, the Puccini/U-Clock combination bested it overall with a smoother rendering of midrange textures, a heightened sense of space, and, particularly, the resolution of transient detail. The Alpha DAC was at a disadvantage in the comparisons in that it was fed from the same music server as the Puccini, but through an AES/EBU interface rather than through the U-Clock that locked the computer to its timing reference. Also, keep in mind that these are two very different products; the Alpha DAC will decode up to 192kHz sources and has no USB input, disc drive, or SACD capability, but costs less than one-quarter the Puccini/U-Clock's price.

The other contenders for the state-of-the-art

EQUIPMENT REVIEW - dCS Puccini CD/SACD Player and Puccini U-Clock USB Converter/Clock

in digital playback (at least in my experience), the Meridian 808.2 and Spectral SDR-4000 Pro, make an interesting contrast with the Puccini/U-Clock. The Spectral and dCS better the Meridian in resolution of low-level detail, transient fidelity, and bass definition. But the Meridian excels, uniquely, in its portrayal of dimensionality—the impression of three-dimensional instruments in three-dimensional space. The 808.2 is also remarkable in its reduction of hardness and glare, particularly in poor-sounding CDs. The Spectral's strengths are in its portrayal of soundstage depth and resolution of fine spatial and timbral detail. I thought the Puccini/U-Clock rendered midrange textures with greater warmth and palpability. All four products have their own virtues, and all are contenders for the state of the art.

Finally, you really need to hear the Puccini/U-Clock driving a power amplifier directly to fully appreciate its clarity and resolving power. Even the best preamplifiers shave off some detail and diminish the sense of immediacy and transparency that are the Puccini's hallmarks.

CONCLUSION

The dCS Puccini/U-Clock pair is an extremely sophisticated piece of engineering. Rather than working within the limitations of off-the-shelf technology, dCS has developed a number of innovative and advanced technologies to extract the maximum performance from digital media. That effort has paid off in the listening room—the Puccini/U-Clock delivers an enormously appealing and involving musical presentation that is in many ways competitive with the state of the art, and in some aspects establishes a reference-quality level of performance.

The dCS' sound was different from other top contenders I've heard, and I struggled to put that difference, and its effect on musical involvement, into words. But if I had to boil it down to a single idea, it would be that the Puccini/U-Clock simply presents more musical information to the listener without calling attention to the fact that it's presenting more information.

I can't overstate how much I enjoyed music through the Puccini/U-Clock; it was absolutely enthralling on CD, SACD, and high-resolution sources. This is a digital front-end I could live with for the rest of my life. tas

Jonathan Valin Comments

Well, I was supposed to write a sidebar comment to this review, but what can I say that Robert hasn't already said better in this brilliantly worded and precisely accurate assessment?

I was very curious to see how my best friend and colleague in this industry would react to the Puccini, since he has so much more experience with the finest digital front ends than I do (and than virtually anyone else in this business does). Don't take it as vanity on my part if I say I am delighted that he heard the Puccini as I do. It's not ego, believe me; it's relief. When I reviewed the Scarlatti/Puccini several moons ago, I thought both were "the best digital" I'd heard, but I thought this for a very specific reason and, let me add again, I thought this without having the vast comparative experience that Robert has with digital sources. My reason for loving the Scarlatti and Puccini was that both sounded like analog sources without sacrificing digital virtues. By sounding like analog sources I don't mean they made CDs sound like LPs, exactly. I mean that they shared with record and tape players a more "holistic" presentation than digital typically provides.

To my ear, digital has always sounded–to greater or lesser extents–flat in aspect and piecemeal in presentation. CD and SACD present the trees, all right, right down to the veins in the leaves, but they invariably seem to lose sight of the scope, spaciousness, and sheer volume of the forest. It's not that CDs and SACDs aren't often sonically impressive–and lifelike. LPs do not typically have the extension and dynamic impact of CD/SACDs, particularly in the bottom octaves; nor do they typically have the sheer crystalline clarity of digital sources. But…digital sources do not have what analog (at its best) has: a realistic warmth of timbre and richness of texture inextricably coupled with a lifelike bloom and body that make instruments and vocalists seem three-dimensionally "there"–perhaps a bit less "look-at-me" detailed than digital but more rooted, more present,

more complete, more real. CDs and SACDs make musicians sound the way highly detailed photographs look; LPs and tapes make musicians sound the way statues in a statue garden look.

With the dCS Puccini and the Scarlatti par excellence, this changed. The details they were adding didn't just amount to hearing, oh, three more second violins more distinctly in the string section (although you could hear three more second violins more distinctly); rather, I was hearing the whole string section (and each violin in it) with a new-found fullness of timbre and texture and an unparalleled (in the digital realm) three-dimensionality and ambient clarity. The dCS's details didn't stand out as individual parts; instead, the parts it was adding were making more complete wholes. Robert put this better than I did when he said that the Puccini "simply presents more musical information to the listener without calling attention to the fact that it's presenting more information." This is precisely correct.

With the addition of the U-Clock, the gap between the Puccini and the Scarlatti (which has its own clock) has narrowed. Robert once described the way timing errors (jitter) affect digital sound by analogizing an unclocked or poorly clocked CD player to hand-held binoculars–with their inevitable image blur caused by the shaking of your hands–and a precisely clocked CD player to image-stablized binoculars–which "freeze" what you're looking at, as if you've taken a jitter-free picture of it. Once again, I can't improve on this. This is exactly the difference that I heard with the U-Clock. What was a bit fuzzy–almost literally "jittery"– snapped into focus.

To an extent the U-Clock gives you the best of both the digital and analog worlds: increased (because better focused/timed) detail, and increased (because better focused/timed) wholes. The U-Clock is a no-brainer recommendation, as is the Puccini itself.

EQUIPMENT REVIEWS

DACs

An Affordable Hotrod DAC for Apple Portable Devices

High Resolution Technologies iStreamer DAC

Chris Martens

High Resolution Technologies (HRT) has built a series of affordable, high-performance USB DACs collectively called MusicStreamers, which have won critical acclaim from many high-end audio publications, including *The Absolute Sound* and *Hi-Fi+*. But with the release of its latest product, the $199 iStreamer, HRT has embarked on a new path by offering—you probably guessed this from the product's name—a dedicated DAC created specifically for use with iPods, iPhones, and iPads.

PRODUCT CONCEPT

Many of us embrace the idea that iPods, iPhones, and iPads are wonderfully convenient portable devices, but ones whose performance is—as a matter of practical necessity—limited by Apple's inherently low-cost onboard DACs and analog audio sections. In short, music lovers who prize sound quality have long felt that a better sounding approach would be to pull digital audio data directly from the iPod/iPad/iPhone, and then to feed it to a higher quality outboard DAC, which is precisely where the iStreamer comes in.

The concept of extracting digital audio data from Apple's portable devices for external processing is not a new idea. Several years back, Wadia's Model 170 iTransport was the first digital audio dock for iPods, though the 170 required a separate, external DAC of the user's choosing. Later, Peachtree upped the ante with its innovative iDecco integrated amp/DAC/dock, which not only extracts digital data from iPods, but also provides an excellent onboard DAC and integrated amp section. Other manufacturers, including Marantz, have built disc players with digital connections for Apple products.

But several things make the iStreamer unique. First, and perhaps most obvious, is the price. At $199, the iStreamer is the only Apple-compatible DAC that actually costs less than a modern-day Apple iPod Classic. Second, the iStreamer is noteworthy in that it doesn't try to be all things to all people; it's an Apple-specific DAC that handles all Apple-supported data formats, but that does not attempt to handle other formats not supported by Apple portable devices, which helps to hold costs down. Third, the iStreamer provides a host mode interface, meaning that the iStreamer takes complete control of the data exchange between the Apple device and the DAC for what HRT describes as "jitter-free" performance (in other words, the data transaction is locked to the iStreamer's—not the Apple device's—clock).

Is the iStreamer targeted toward Audiophiles with a capitol "A?" Yes and no. The iStreamer sounds very good for its price, but does not claim to be the best DAC available (HRT offers higher-end models that go after that part of the spectrum). Instead, the iStreamer is for music lovers who love their Apple devices but want "something better" in the sound-quality department.

> **At $199, the iStreamer is the only Apple-compatible DAC that actually costs less than a modern-day Apple iPod Classic.**

EASE OF USE

The iStreamer is extremely simple to use. You simply hook up the wall-wart power supply, connect a left/right pair of RCA cables to your amp/receiver, and then hook up a traditional Apple-type docking/USB connector. Once you flip on your Apple device, one of the iStreamer's three data rate lights (32k, 44.1k, or 48k) will start flashing—denoting that the iStreamer is synced to your Apple—and you're good to go.

There are a several small caveats to note: The iStreamer is not portable (it's tethered to its wall wart-type power supply). The iStreamer does not provide digital audio outputs, so that there's no way to use it as a digital dock to feed data to an outboard DAC. The iStreamer can be sensitive to the condition of the Apple dock/USB cable, so that you'll want to make sure the connectors are in good shape and correctly inserted on the Apple end (the iStreamer may not be able to achieve or maintain proper sync if the connector is seated askew in the Apple). In a break with common practice, the iStreamer's data "sync" lights flash on and off continuously when sync is achieved (for most other DAC makers, a flashing light indicates that the DAC is trying to achieve sync, while a solid light indicates that data sync has successfully been achieved). Finally, while the iStreamer supports many Apple devices (as listed below), it does not support some of the earlier generation iPods (I believe because they may not support the iStreamer's required host mode interface).

None of these caveats constitutes a "showstopper" by any means, but they are certainly points worth bearing in mind for prospective iStreamer buyers.

EQUIPMENT REVIEW - High Resolution Technologies iStreamer DAC

SONIC CHARACTER

In general, I found the iStreamer offered smooth, neutral tonal balance with solid but not exaggerated bass, a good measure of midrange openness, and very smooth highs that seemed to be rolled off just a hair. For many listeners, these characteristics point toward a DAC that is eminently musical—never cold, brash, brittle-sounding, or analytical. The benefit, of course, is that the iStreamer lets users completely bypass the slightly raw, coarse-sounding sonics of Apple's own DAC and analog audio electronics to step up to a whole different (and better) class of performance.

But the iStreamer's signature characteristic is an uncanny quality of rock-solid stability, which manifests itself as a sonic sense of purity of timbre and "locked-to-a-clock" rhythmic precision. Another benefit is a certain desirable quality of ease in the iStreamer's presentation. I can't say for sure, but I believe these positive qualities may very well be attributable to the HRT's "jitter-free host mode interface."

LISTENING TESTS

I put the iStreamer through several sets of comparisons tests, with outcomes as noted below.

iPod Classic with and without iStreamer

For this comparison I conducted three sets of tests, always using the same set of lossless, CD-resolution files stored on an iPod Classic. All listening was done through Ultimate Ears In-Ear Reference Monitor (IERM) headphones.

I began by listening to the iPod Classic connected directly to my pair of Ultimate Ears In-Ear Reference Monitors. Next, I used Moon Audio Silver Dragon line out dock (LOD) cable to connect the iPod Classic to an ALO Audio Rx Mk II portable headphone amplifier. Finally, I inserted the iStreamer in the signal path, connecting the iPod Classic to the iStreamer and using the iStreamer's analog outputs to drive the ALO headphone amp.

> **The iStreamer lets users completely bypass the slightly raw, coarse-sounding sonics of Apple's own DAC to step up to a whole different class of performance.**

The iPod did a decent job with the Ultimate Ears 'phones, which are reasonably easy to drive. Even so, I noted a few shortcomings. First, bass was not as deeply extended or well-controlled as I would have liked, mids were somewhat forward-sounding and a bit splashy, while fast-rising, high-energy treble transients had a subtly aggressive, "pingy" quality that inexperienced listeners might mistake for "detail," but that wasn't really right.

The iPod Classic gave significantly better performance when used with a high-quality LOD cable and portable amp, exhibiting improved bass performance and smoother, better balanced, and more open-sounding mids and highs. But, even with the help of the cable and amp, the iPod still showed slightly less extended bass than would be optimal, residual hints of midrange forwardness, and highs that—though greatly improved—still showed traces of splashiness and a subtle lack of focus.

With the iPod Classic, the iStreamer DAC, a portable headphone amp, and high-quality Headphones, the playback system took another step up in performance, showing even better tonal balance, improved bass pitch definition and solidity, mids that sounded at once smoother yet also more transparent, and an overall presentation that was noticeably more coherent and focused.

Playing the iPod Classic through a good LOD cable and headphone amp improves its sound quality significantly and offers a good portable solution. For desktop environments, however, adding the iStreamer elevates sound quality further still, giving a significant jump in performance that makes the modest iPod sound more like a high-end source component in the process.

iStreamer vs. Computer-based Desktop System

For this comparison I conducted two listening tests. I began by listening to lossless CD-resolution digital audio files from a Windows PC (running iTunes) connected via USB inputs to a NuForce Icon HDP combination USB DAC/headphone amplifier. Next, I changed out the digital front end of the system, playing

SPECS & PRICING

High Resolution Technologies iStreamer DAC

Accessories: Power supply, power supply cable, Apple-compatible dock/USB cable, RCA interconnect cables

DAC: 16-bit, with support for 32kHz, 44.1kHz, and 48kHz data rates

Digital input: USB

Transfer protocol: Host Mode (master)

Analog output: Stereo analog audio (RCA jacks)

Dimensions (H x W x D): .875″ x 2.3″ x 4.6″

Weight: Not specified

U.S.
Price: $199

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EQUIPMENT REVIEW - High Resolution Technologies iStreamer DAC

the same lossless digital files through an iPod Classic that was connected to the iStreamer, with the iStreamer's analog outputs driving the analog audio inputs of the NuForce Icon HDP amp. All listening was again done through Ultimate Ears In-Ear Reference Monitors.

The PC/NuForce-based system sounded very clear, but offered somewhat lighter tonal balance than would have been ideal. Relative to typical non-PC-based systems, the PC/NuForce-driven system offer better midrange focus, exceptional transient detailing, extended highs, and overall stability and coherence comparable to the results achieved with the iStreamer in the system.

> **HRT's iStreamer provides an easy-to-use and extremely good-sounding means of connecting Apple devices to high-end systems.**

The iPod/iStreamer/ NuForce-based system offered slightly warmer and more full-bodied tonal balance than the PC-based system did, with superior bass weight, midrange focus that was good (though perhaps not quite up to the Icon's level of performance), good transient detailing and smooth highs, and with just a trace of treble rolloff.

Both the PC-based and iPod/iStreamer-based desktop systems offered better sound quality than could be achieved using just the analog outputs of an iPod as the "front end" of the system. Which digital source one prefers—the PC + NuForce DAC vs. the iPod Classic + iStreamer DAC—largely will be a matter of listening tastes. The PC + NuForce DAC combo will appeal to

those who prize clarity, transient speed, and treble extension, though its sound might be too lightly balanced for some tastes. The iPod Classic + iStreamer DAC combo, on the other hand, will appeal to those who favor a smoother, more relaxed, and more organic sound, while giving up only a subtle degree of apparent transient speed and definition. The iPod + iStreamer combo's superior bass performance and warmer tonal balance will, I think, strike many listeners as offering the more forgiving and "musical" sound overall.

SUMMING UP

HRT's iStreamer provides an easy-to-use and extremely good-sounding means of connecting Apple's modern portable audio devices to high-end desktop, home stereo, or home theater systems. The iStreamer provides significantly better sound quality than any analog iPod dock can hope to match, while its host mode interface gives the iStreamer a coherent, well-focused sound that enables it to play with the big boys—delivering credible high-end sound from Apple's popular portable audio devices. **tas**

UNEQUALED AUDIO FIDELITY

From the inventors of HDCD and designers of the Pacific Microsonics Model One and Two

Alpha DAC

"The Alpha DAC is a spectacular bargain. It is my outboard converter of choice for both CD and high-resolution sources—regardless of price."
—Robert Harley, *The Absolute Sound*, Issue 189

Alpha USB

"The Berkeley Audio Design Alpha USB is a breakthrough product that not only overcomes the limitations of the USB interface, but provides a state-of-the-art method of getting audio out of a computer."
—Robert Harley, *The Absolute Sound*, Issue 214

When reproducing the finest high-resolution digital audio recordings, the **Alpha DAC** combined with the new **Alpha USB** delivers a level of audio fidelity unequaled by any other type of audio recording, including first generation analog master tapes and direct-to-disc vinyl records. To hear for yourself, visit your Berkeley Audio Design dealer.

www.berkeleyaudiodesign.com
510-277-0512

BerkeleyAudioDesign®

High-Resolution Technologies Music Streamer Pro USB DAC

A Simple Upgrade for Computer-Audio Systems

Robert Harley

The new Music Streamer Pro DAC from High-Resolution Technologies is the most minimalist product I've ever reviewed. The unit is simply a block with a USB input on one end and analog-audio outputs on the other. The chassis is too small to accommodate XLR jacks, requiring adapters to take advantage of the Music Streamer Pro's balanced outputs. The adapters convert the "TinyQ" balanced jacks to standard XLRs. The MS Pro gets its power via the USB cable when connected to a computer; there is no other way to power and connect this DAC.

The MS Pro is the latest in a series of inexpensive, bare-bones USB DACs from High-Resolution Technologies. The company made quite a splash two years ago when it introduced the original Music Streamer at a retail price of just $99. Since then, the company has steadily moved up-market, culminating in the $499 Music Streamer Pro reviewed here. The Pro features asynchronous USB connection, which means that the clock controlling the Music Streamer's DAC runs independently from the computer's clock, resulting in lower jitter. In fact, any USB interface that isn't asynchronous is a non-starter in my view.

Connection and operation couldn't be simpler. Connect the MS Pro to your Windows-based PC or Macintosh and the computer's operating system recognizes the Music Streamer and you're ready to listen. I auditioned the Music Streamer Pro with my fanless, driveless PC-based server playing a variety of CD-quality and high-resolution files. Connection was via AudioQuest's Carbon USB cable.

Sonically, the Music Streamer Pro was good for the money, but not a giant-killer. On the plus side, it had a generous and warm bass that gave the presentation a fully fleshed-out sound rather than the thin "skeletal" sound of many inexpensive DACs. Midrange timbres were relatively clean, and the soundstage had a nice transparency. I was also impressed by the sense of depth, overall soundstage size, and air between instrumental images. The Music Streamer Pro managed to avoid the "cardboard cutout" syndrome that plagues the spatial presentation of many inexpensive digital products. Dynamics were surprising for the price.

Although full, the bottom end was somewhat loose and ill-defined. Bass guitar lines seemed to lag slightly behind the beat, fostering the impression of slower tempi along with the feeling of musicians not quite as locked into the groove. The treble had a bit of hardness, but that's to be expected at this price.

Overall, the Music Streamer Pro is a bargain considering that it offers an asynchronous USB interface and fully balanced operation. It's also easy to use and small enough to share shelf space with a full-sized component. The unit has some laudable sonic attributes, including a large soundstage, wide dynamics, and a warmish bass balance This warmth, however, comes at the expense of bass definition and rhythmic drive, both of which were somewhat lacking. The Music Streamer Pro won't find a home in an ambitious high-end system, but it offers a simple and good-sounding upgrade path for computer-based audio systems. **tas**

SPECS & PRICING

High-Resolution Technologies Music Streamer Pro USB DAC
Type: USB DAC
Interface: Asynchronous USB
Analog outputs: Balanced on TiniQ jacks
Output level: 4.5V RMS at 0dBFS
Supported resolution: 96kHz/24-bit maximum
USB type: 1.1
Dimensions: 2.1" x 1.2" x 5.6"

U.S.
Price: $499

HIGH RESOLUTION TECHNOLOGIES LLC
Los Angeles, CA
highrestech.com
(323) 967-7447

Cambridge Audio DACMagic

Magic Act!

Alan Sircom

Cambridge Audio is not a name you see often in the pages of *Hi-Fi+*. It's a UK-designed, Chinese-built range of audio electronics that's designed to fit snugly into the low to medium end of the separates hi-fi market. Nevertheless, looking past the price tags, the company has some exceptionally high-tech and distinctly hi-fi-friendly products in its line, of which the $429 DACMagic is merely the latest. Except that it's also one of the most important products to hit the stores this year.

There's a deep-seated sense of no-BS about Cambridge Audio products that could be considered an inversion of some aspects of high-end sensibilities. Although well-made, there's no brushed-gold, inch-thick casing, no heatsink carved to spell out the name of the designer's cat, no Nuclear Football remote control. These products are functional on the outside, built smart on the in. And the DACMagic is no exception; it's a black or silver paperback (in size and weight), with a separate plug-top power supply. The front panel has buttons to control power, source

selection (USB and two sets of TosLink or S/PDIF coaxial inputs) and choice of filter slope, and five LEDs to determine the incoming sampling rate. It upsamples to 24-bit/192kHz precision thanks to Adaptive Time Filtering, developed in partnership with Anagram Technologies in Switzerland. The chipset features a pair of Wolfson's 8740 DACs coupled to a 32-bit Texas Instruments DSP, and it has balanced XLR and single-ended RCA outputs, as well as a TosLink and S/PDIF coaxial out. OK, dedicated DAC twitchers will crow that the Wolfson 8741 is the chip du jour, but re-read this paragraph again, cross-referencing it with the following words–"four-hundred-twenty-nine dollars"–and try not to be impressed.

We've seen DACs before. We've even seen cheap DACs before. What makes this one so important? Simply that this isn't just a digital-to-analog converter; it's a human-to-audiophile converter. You see, insert this little box in between someone's PC or PS3 and an amp, and the regular muggle turns into a hi-fi buff by the end of the first bar or the at the sound of the first gun report. And that's the point of this DAC, and in particular this review.

Viewed from the perspective of *Hi-Fi+*, the

DACMagic is the kind of money one might spend on a power cord. On the other hand, the idea of a DAC itself, let alone one costing as much as $429, is absurd to real people. That is, until they hear one. Then they change… they nod their heads and give you that knowing "now I get it" look.

So, there are two options for you, to spread our particular concept. Buy one and loan it to your friends or buy lots and give them to friends. The result is the same thing–if you loan it out, you won't get it back.

We did just that. We pitched up to what is arguably the least audiophile bunch around–a handful of twentysomething gamers huddled round a Sony PlayStation 3 and a copy of Guitar Hero. Turning up with an amp, speakers, and a DAC was dangerously "dad stuff" for them, but less than a minute into the experience, the guy wielding the plastic Gibson did precisely what we expected…up went the volume, a broad

smile emerged, and he became more and more animated. A quick rendition of "Back in Black" and "Paradise City" (some things never change) and they started asking prices and asking questions about iPods and laptop outputs. Where not more than 20 minutes earlier, there was a look of outright disgust at all this "old man" hi-fi gear, now they were asking where they could buy the stuff (which one subsequently did within a week). Getting the product back to finish off the review was tough; I could out-manoeuver them easily, but their superior thumb speed was a bit of a problem.

The DACMagic has been dipped in a lot of purple prose in the media. There are those touting it as the best converter that's on the market, possibly the best that's ever been heard. Let's be sensible; it's a remarkable $429 DAC, but it's still a $429 DAC. It will help make a cheap CD player sound like something between a Cambridge

EQUIPMENT REVIEW - Cambridge Audio DACMagic

Audio Azur 740C and an 840C (which is still saying a lot… these are a fine pair of CD players, and take the resampling up to dizzy 384kHz heights). It won't intrinsically improve a good, current player costing more than about $1500, or a decade old player that cost $2200 or more when new. It does, however, have an uncanny habit of latching on to some frankly shonky datastreams, the sort of back-of-the-attic digital relics that are only dragged out to see how good a lock some DACs have. The DACMagic is very, very good at reading less-than-frisky digits from tired old players, then.

Sonically, what you get is a fundamentally clean and honest presentation, one that stays just the right side of brightness. It's a paragon of neutrality (that'll be the Swiss connection) making the most of even old acoustic recordings of bluesman Blind Blake come to life. Stereo is wide, not especially deep or high, but extremely solid. Even when handling large-scale orchestral works, instruments are locked down in their own distinct spaces within the mix. But the big thing is detail; there's loads of it, precise from top to bottom, and every Hertz an accurate one. No rolled-off highs, suppressed mids, or flattened bass lines. Far from it; in fact, the bass manages to balance perfectly between energy and depth. Bass lines–even those found in dub reggae–are easy to define and full of drama.

Like any good DAC, it makes its presence felt more by the lack of sonic signature than its intrusiveness. In fairness, this doesn't make for a big step up for most CD players, but the USB option is a very different story. Suddenly, iTunes becomes a legitimate audiophile source option (the DACMagic is easy to hook to PC and Mac alike) and the freedom it imparts to the sound of Apple Lossless files is little short of revelatory.

The filter options are interesting. They represent mild shaping of the sound, instead of fundamental changes from filter to filter. Both "linear" and "minimal" phase seemed to work best, with CD mildly better through the cleaner, deeper-bass presentation of the "linear phase" setting and Apple Lossless files more suited to the more cuddly and warm "minimal phase." The Steep filter didn't have a big place in my listening

tests, but neither was it out of place compared to the others; it gets lost with orchestral music but was wonderful for making sense of those really early Louis Armstrong Hot Fives recordings, through both CD and computer audio sources. It's a mild preference, only determined by lengthy listening sessions; when it's right, the sound just snaps into some kind of temporal focus, making transients more precise.

There are two schools of thought here, depending on your anal-retentive score. Some will fiddle with the filters, find what they like best, and leave well alone. Others will determine what filter shape goes best with every single recording on each source and adjust accordingly. Fortunately, the DACMagic isn't bothered either way.

What limits does the $429 price tag impose? Well, the case itself is thin, and the lettering on the case is printed and will probably rub off in time. We'd also love to hear what it would sound like with something a bit more potent than the plug-top power supply it comes with. Oh, and if we were being really picky, ST optical and AES/EBU would be nice. Notice though… none of these price tag limitations affect either the performance or the build of the DACMagic, but are really just idle musings.

Cleverly, Cambridge Audio's DACMagic does everything right. It fights impressively above its weight, but not to the point where the likes of PS Audio should be worried. More importantly, the combination of keen price and easy connectivity to computer audio and gaming products adds a new dimension, turning gamers and iTunes users into proto-audiophiles, seemingly in a matter of seconds. Say 'hello' to the hi-fi wow factor, 21st Century style. **tas**

SPECS & PRICING

Cambridge Audio DACMagic

D/A Converters: Dual Wolfson WM8740 24-bit DACs

Digital filter: Texas Instruments TMS 320VC5501 DSP upsampling to 24-bit/192kHz

Analogue filter: 2-Pole Dual Differential Bessel Double Virtual Earth Balanced

Frequency Response: 20Hz to 20kHz (+/-0.1dB) - steep filter disabled

Signal to Noise Ratio: -112dB

Digital input word widths supported: 16-24bit (16-bit for USB)

Digital input sampling frequencies supported: 32kHz, 44.1kHz, 48kHz, 88.2kHz, 96kHz

Audio output up-sampling: Fixed 24-bit/192kHz

Weight: 1.2kg/2.6lbs

U.S.

Price: $429

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Musical Fidelity M1 DAC and V-Link USB Adapter

Raising the Bar in Affordable DACs

Steven Stone

I know the cynics in the audience may have a hard time believing this, but sometimes when I receive a component for review I don't know its price. In the case of the Musical Fidelity M1 DAC and V-Link USB adapter, time had erased their price-point from my memory banks. When I finally discovered how little they cost—$699 and $169, respectively—I was more than pleasantly surprised. Practically anyone who can afford a Mac Mini can probably also spring for a M1 DAC and V-Link, creating a playback system that will enthrall all who give it a listen.

According to Musical Fidelity, "the M1 and V-series offers state-of-the-art performance for a low price. It is done by solid commercial principles and state-of-the-art circuit/PCB design. The V-series has no extraneous anything. It is state-of-the-art circuitry laid out to perfection with no trimmings, packaged in a simple low-cost housing made in large numbers." How well does Musical Fidelity succeed at bringing the state of digital art to the masses? Well enough to elicit a gentle tugging at purse strings.

FIDELITY AIN'T CHEAP

Musical Fidelity has a blue-chip reputation that comes from making A-grade audio components for over two decades. Headed by designer Anthony Michaelson, Musical Fidelity specializes in electronics, and was among the first companies to make a high-end digital-to-analog converter (DAC). The V Series components were a big leap for Musical Fidelity, from the rarefied heights of "If you have to ask what it costs you can't afford it" to "I'll take one for each room." Historically, few audio companies have succeeded in covering such a wide price range without shortchanging some parts of their lines. But Musical Fidelity has managed to consolidate its position at the über-high end with products such as its Titan Class A power amplifier, while simultaneously creating the new M1 and V Series budget lines.

Technically, what differentiates the M1 DAC from its competition are its extremely low-distortion circuits (with overall distortion of less than 0.005% across the entire frequency band). The M1 DAC also uses a Class A analog output circuit that generates 2.25V RMS via its RCA single-ended outs and 4.5V RMS from its balanced XLR outputs. In addition, the M1 employs a special choke-filtration system that acts like a power conditioner for its entire power supply.

Physically, the M1 DAC is an exercise in simplicity. The front panel is all black except for a tiny inset silver nameplate. The rest of the faceplate is empty save for the small white printing and tiny LED condition lights. There are only two control buttons on the front panel—one for on/off and the other for selecting the active input source.

Small LED confidence lights show the incoming sample rate, input source, and whether the M1 is properly upsampling the signal to 192/24. The M1 DAC doesn't have a remote or an adjustable output level. It supports every sample rate from 32kHz to 192kHz and has inputs for coaxial S/PDIF, TosLink, AES/EBU, and USB.

The just-released V-Link USB-to-S/PDIF/TosLink converter box takes a USB 1.1/2.0 input and outputs a S/PDIF signal via either TosLink or S/PDIF RCA connectors. The V-Link supports up to 96kHz/24-bit sample-rates. Its digital heart is the USB receiver TI TAS 1021chipset from Texas

EQUIPMENT REVIEW - Musical Fidelity M1 DAC and V-Link USB Adapter

Instruments. Additionally the V-LINK uses a discrete crystal-oscillator for its clock reference and a discrete output chipset. The TAS1021 has been used by both dCS and Wavelength in their highly regarded USB DACs. Both wrote their own proprietary software for the TI TAS 1021 that enables asynchronous operation. Musical Fidelity also developed its own asynchronous USB-protocol interface software for the TI TAS 1021.

Why did Musical Fidelity create the V-Link if its M1 DAC already had a USB input? The simple answer is that product cycles are much shorter with computer gear, and in the 1½ years since the M1's USB input was developed, Musical Fidelity has had time to develop a much better USB application that uses the latest asynchronous-data-acquisition methodology. The engineering and sonic differences between the M1 DAC's USB and the V-Link's USB input circuit implementation are not subtle.

"MUSICAL" AND "FIDELITY" ARE LIKE "COUNTRY" AND "WESTERN"—THEY GO TOGETHER

For most of my review the V-Link and M1 DAC were tethered to my desktop system (see the review equipment list), and I connected the M1 DAC's balanced outputs directly to a pair of PSI Audio A 14M powered/active monitor speakers. I adjusted levels via the speakers' individual volume controls. This allowed me to hear the M1 DAC without the sonic limitations of a preamp and an extra run of cables. I also used the M1 DAC via its single-ended RCA outputs connected to my Accuphase P-300, which has its own volume controls so I could go preamp-less. Using either signal chain I could easily replace the M1 DAC with one of my other reference DAC/preamps for

A/B comparisons.

Before comparisons let me talk about the sound of the M1 DAC by itself. First, forget about the USB input. It's an ancient (by computer-time standards) USB implementation that sounds quite flat, both dimensionally and dynamically, when compared to the M-1's other inputs. Pretend it doesn't exist. After all, you've still got three other inputs—RCA S/PDIF, TosLink, and AES/EBU. If you require a USB input add the V-Link and you'll be on a level playing field vis-à-vis the sonic capabilities of the other inputs.

While on the subject of USB and the V-Link, I can almost without reservations recommend it to anyone with a high-end DAC that lacks USB capabilities. It renders USB almost as well as the Empirical Audio Labs Off-Ramp 3 USB converter box. To compare the V-Link to Empirical I used the V-link's coaxial output and the Empirical's AES/EBU output connected to the M1. This allowed for virtually instantaneous A/B switching via the M1's front panel and the Audio Midi Setup when I used iTunes. Using Pure Music and Amarra was slightly more complicated. I needed to change the output device in Sound Preferences as well as in the program's individual preference pane and then reboot the playback program. Still, with a bit of practice I got to the point where I could switch from A to B in less than 15 seconds.

In matched-level A/B tests the V-Link falls short of the Empirical Audio Off-Ramp 3, primarily because it's slightly less three-dimensional than the Off-Ramp. However, in resolution and dynamics the V-Link was indistinguishable from the Off-Ramp. I spent some time listening to that old audiophile chestnut, from Rickie Lee Jones' *Pop Pop*, "My One and Only Love," and can't

say I've ever heard freer or more independent dynamics from each of the instruments. Rickie Lee's voice, the acoustic guitar, acoustic bass, and accordion all displayed a dynamic autonomy I rarely hear in a recording but often experience at a live performance. These continuous micro-dynamic adjustments that musicians make throughout a concert came through transparently and effortlessly.

With computer audio there's usually more than one way to get a musical file from storage to transducer. The more computer audio gear I review, the more ways I've found to hook that gear up. I started the review with the V-Link attached to the M1 running into the PSI A 14M speakers. But since the M1 has only fixed-level outputs, I couldn't control the subwoofer volume level except by adjusting the subwoofer's input-level controls separately. This is easy if your subwoofer is within arm's reach, as mine is, but for most folks it isn't practical.

After a month I inserted the Bel Canto DAC 3.5 into the system. This opened up quite a few new routing options for comparison, since the DAC 3.5 has a volume control, two analog outputs, and an analog bypass. This way I could compare the Musical Fidelity M1 DAC's analog output to the Bel Canto DAC. Also the Bel Canto 3.5 VB allowed me to hook up the V-Link to both the M1 and DAC 3.5, and to use the Empirical Audio Off-Ramp 3 with both DACs. This permitted quick A/B switching between DACs, USB converters, and various DAC inputs. Later I also substituted the Wyred4Sound DAC-2 and the Weiss DAC202 for the Bel Canto 3.5 VB.

As I mentioned earlier, the V-Link is a wonderful way to get a 96/24 USB signal into any DAC

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EQUIPMENT REVIEW - Musical Fidelity M1 DAC and V-Link USB Adapter

that lacks USB inputs such as the Weiss DAC 202. Using the V-Link into the Weiss' S/PDIF input produced results that were very close to the Weiss' FireWire input. The FireWire input was still superior—it had better depth rendition and more of a three-dimensional presentation overall. But in other sonic respects the V-Link held its own. I was especially impressed with the V-Link's ability to retain all the dynamic subtlety I experienced through the FireWire input. I also re-compared the V-Link to the Empirical Audio Off-Ramp converter. Through the Weiss, differences between the V-Link and the Empirical Audio Off-Ramp 3 were magnified. Once more the Off-Ramp revealed more dimensional information, but it also had greater dimensional solidity and more precise imaging, especially in defining the edges of each instrument in space.

Getting back to the Musical Fidelity M1 DAC. When I used it directly attached to an active speaker or an amplifier with an adjustable input-level control, it was noticeably more revealing and delivered a higher level of fidelity than when I inserted a preamp into the signal chain. Even when I used the most expensive RCA cables I had available—a pair of $1840 Cardas Clear 1-meter interconnects—the overall sound was still compromised compared to a direct-to-the-amp connection. By the end of the review period I fervently wished the M1 was available with adjustable level controls so that it could be used without additional devices and cabling in the signal path. That way everyone could hear how good this DAC is in "stand-alone" mode.

When compared to a far more expensive DAC, such as the Weiss DAC 202, the M1 proved to be remarkably close in performance when it was

connected directly into the PSI A 14M speakers. Sure, the Weiss was still superior, but overall the similarities between the two DACs were striking. Both produced equally large soundstages, solid palpable images, fast transient attacks, and wonderfully compelling listening experiences. Yes, if I could only chose one I'd go for the Weiss, but after a few minutes with the Musical Fidelity M1 it was difficult to focus on its deficiencies, such as they were, because its overall performance was so solid and musical.

RAISING THE BAR

Robert Harley recently sent all TAS reviewers a letter [published in TAS issue 213] from reader Bob Anselmo. Anselmo wrote, "It's time to raise the bar on lower-cost equipment. We should expect lower-priced stuff to sound better than it did in the past." I completely agree.

The Musical Fidelity V-Link and M1 DAC certainly changed my opinion of what level of performance can be expected from a budget DAC and converter box. I won't claim they equaled the performance of far more expensive electronics such as the Weiss DAC 202, but unless you plan to conduct tightly controlled A/B tests, you will be hard-pressed to hear the Musical Fidelity M1's shortcomings. Unlike budget components of yore, where pleasant "grayish" sound was the best you could reasonably expect, the M1 is musical, arresting, and involving in the way a good audio component should be, regardless of price. Even if you can afford to pay more, maybe even a lot more, I strongly advise you to listen to and live with the Musical Fidelity M1 DAC and V-Link converter for a week in your system. Consider the bar raised. tas

Peachtree Audio iDac

Much More Than A DAC

Steven Stone

As you might surmise by its name, the Peachtree Audio iDac is a DAC (digital-to-analog converter) that has an iPod dock. And while the dock is part of its feature set, it's far from the whole story. The iDac also has provisions for two S/PDIF, two TosLink, and one USB input. That's a whole bunch of digital inputs. But the iDac needs this plethora of inputs to fulfill its true function—a one-box hub for all the digital music sources on your computer desktop or in your living room. The iDac does a mighty fine job being the digital ringmaster for your audio circus.

Regular readers should be quite familiar with Signal Path's Peachtree Audio line of electronics. Its Nova, iNova, Decco, and iDecco and have garnered lots of glowing reviews, here and elsewhere. And for good reason. Peachtree has a knack for producing high-value products aimed at both entry-level and computer audiophile markets. The iDac has the potential to be its most successful product yet.

THE FEATURE SET

As you would expect from a current generation D/A, the iDac supports 96/24 via its USB input and 192/24 via S/PDIF. Using the latest ESS 32-bit Sabre32 DAC chipset as part of a multi-layer DAC board that has over 450 individual components, Peachtree has developed a DAC that can cope with both high- and low-jitter sources. The iDac also uses a linear instead of a switching power supply as well as eleven regulated sub-supplies to lower internal noise and interference.

According to Peachtree Audio, "the ESS Sabre32 DAC's patented jitter-reduction circuit re-clocks the digital signal before passing it through a high-resolution 24bit/192kHz bit-perfect processor. Re-clocking is needed for computers and most streaming or hard-drive-based music servers because most have high levels of jitter and/or noise. Transformer-coupling each digital input eliminates noise, typically associated with ground problems and switching power supplies. The USB connection is galvanically isolated, eliminating noise generated by your computer's switching power supply and greatly improving sound quality."

The Sabre DAC doesn't re-clock in the traditional way. Instead it uses something called a "hyper stream modulator" inside the chip to reduce jitter at the master clock. "A patented technique is used to re-create the audio data in a crystal-controlled low-phase-noise clock domain completely isolated from the clock domain of the transport medium."

On the back of the iDac, on either side of the two RCA analog-output connectors, are pushbuttons. On the left resides a "digital filter switch" which gives you the choice of either an NOS (no oversampling) or NAL (non-aliasing) digital filter, the second of which will upsample the signal to 192/24. On the right you'll find a "hi-bit/lo-bit" switch, which changes the way the digital processor uses its DAC chip. In lo-bit mode, the DAC uses dual processing, while in hi-bit mode the iDac switches to quad processing. As you might guess by the location of these two switches, they are not for everyday toggling back and forth. Instead I'd advise new owners to run their own listening tests, decide which settings they prefer and then leave 'em alone.

Given its name, it would be weird if the iDac didn't have a built-in iPod dock. But what makes the iDac's dock special is that it is a pure digital connection that takes the digital signal from the iPod without going through the iPod's internal

D/A. The iDac joins the Wadia iTransport 170i as one of the few (but expanding) list of products that offers a direct digital connection to an iPod.

Which iPods does the iDac support via its digital dock? The iPod Touch (first through current generation), iPod Classic (80GB, 120GB, 160GB only—no earlier versions supported), and iPod Nano (second, third, fourth and fifth generation) all will deliver a direct digital connection when tethered to the iDac. In theory the iDac's internal DAC can even support higher bit-rates than the 48/16 limit imposed by iTunes, but since no iPods support any higher bit rates, 48/16 is the current limit.

I mentioned earlier that the iDac has a plethora of inputs. But its audio output set is limited. The iDac offers only one pair of line-level analog audio outputs and these outputs are at a fixed level. This means that you will need to have something else in your system to attenuate and control volume levels. In a conventional system a

EQUIPMENT REVIEW - Peachtree Audio iDac

preamp, AV/controller, or the volume controls on your integrated amp or receiver will accomplish this task. But it's too bad that you must use some kind of additional device to control volume since any additional device in the signal chain will have some negative impact on transparency and detail, which are two areas where the iDac excels. I'll discuss this with more specifics in the sound section.

The iDac has a strong family resemblance to other Peachtree Audio products. Housed in a piano-black case (it comes in a soft cloth bag to keep the finish pristine), the front panel is simple and uncluttered with six source-selector buttons and one slightly larger on/off button. A nice ergonomic touch: When you select a source that is inactive or nonexistent the blue circle around the buttons glows intermittently, while with a "good" source the circle has a steady glow.

The iDac's remote control leans toward the minimalist side with only source selection, on/off, and controls for navigating your docked iPod. I would have liked to have seen a mute button, but if you use the iDac with a receiver or preamp, their mute controls will suffice. Perhaps the next version could also move the NOS/NAL and hi/lo bit-rate toggle switches onto the remote; there's plenty of empty real estate available to accommodate them.

THE IDAC SOUND

Since the iDac has a fixed-level analog output I coupled it with a Reference Line Preeminence One passive preamp for most of the review. But I also did some listening sessions where I connected the iDac directly to the inputs of my Accuphase P-300 power amplifier and used the amp's built-

in level controls to adjust system volume. The P-300 has two sets of input connectors and the ability to switch between them via a front panel toggle. This feature makes it possible to do real-time A/B listening tests between different DACs at my desktop without having to move my head during the changeover (since the amp is within arm's reach underneath my desk). The P-300 is also great on winter days when the waves of heat coming from under my desk serve as an effective space heater. This summer, the P-300 goes back in the closet.

So what's up with the two stealth buttons on the backside of the iDac? The instruction manual doesn't give much info as to why or which setting is "better." To hear the sonic differences between hi-hit/lo-bit and NOS/ NAL settings, I hooked up the iDac directly to the P-300 power amplifier. With only a one-meter pair of AudioQuest Colorado cables between the iDac and the P-300 I could clearly hear differences between the settings. Most listeners will prefer, as I did, NOS setting. The NAL sounded too tight and matter-of-fact with less bloom and a smaller overall soundstage. I also preferred lo-bit to high-bit, low-bit was more relaxed and to my ears, more natural. In some systems, especially those that have their own internal or room-enhanced bloom, the NAL settings might prove to be a more synergistic match-up. I advise you to try all the combinations.

My real-time A/B comparisons between DACs turned out to be especially interesting. First I put the Musical Fidelity M1 DAC up against the iDac, using the internal USB inputs. The iDac won this face-off easily with superior performance in every meaningful sonic category. But the tables turned

when I inserted the V-Link USB adapter (which converts USB to S/PDIF) into the M1's signal chain. The M1/V-link combo was on the same sonic level as the iDac. I preferred the Musical Fidelity combo's solidity and dimensionality to the iDac, but the iDac still excelled in lateral image focus and slam. The iDac also produced a larger overall image size.

Next I compared the iDac's USB input with the Musical Fidelity V-Link feeding the iDac's S/PDIF input. Much to my chagrin I heard the same sonic differences between the iDac USB and the V-link inputs as I had heard between the iDac and the Musical Fidelity M1 fed by the V-link. Through the V-Link the image was more three-dimensional and each instrument seemed to be better defined and more substantive. Perhaps USB implementations are becoming more critical to the overall sound than the DACs themselves? [See my review of the Berkeley Alpha USB converter this issue. —RH]

Further tests were certainly in order, so I replaced the V-Link with Empirical Audio's newest version of the Off-Ramp 4 USB-to-S/PDIF adapter box. Differences between the iDac's stock USB circuitry and the Empirical Off-Ramp were startling. When I used Empirical Audio's Off-Ramp 4 for USB translation duties I finally heard the iDac's capabilities when fed a low-noise low-jitter signal. Wow! The soundstage grew noticeably in size from what the iDac was able to produce with either the V-Link or its own internal USB converter. Along with the expanded soundstage size, dimensionality and image solidity also improved. Spaces between various instruments and the edges of individual instruments were better defined by the Off-Ramp than by other USB solutions. Whatever the reason,

the Empirical Off-Ramp/iDac combination turned out to be the best sound I got from the iDac. As economically unlikely as it may be to couple a $1000 DAC with a $1500 USB converter, the combination proved to be competitive with the best digital front ends I've reviewed, regardless of price or connection type.

And what about that iPod dock? That, too, turned out to be a pleasant surprise. When I conducted level-matched A/B tests between a docked iPod Classic 160 and the USB feed from my Mac Pro computer I could not hear any difference between the two. None, nada, zip. I tried AIFF, Apple Lossless, and even 320 BPS

SPECS & PRICING

Peachtree Audio iDac

Type: USB DAC with iPod dock

Frequency response: 5Hz-100kHz

S/N: 118dB "A" weighted

Output voltage: 2V RMS

Output impedance: 10 ohms

Dimensions: 3.5" x 9" x 10"

Weight: 8 lbs.

U.S.

Price: $999

PEACHTREE AUDIO

(704) 391-9337

signalpathint.com

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EQUIPMENT REVIEW - Peachtree Audio iDac

MP3 files and the results were the same—no discernable differences.

Given how different the iDac can sound depending on the quality of the digital source, does it have its own intrinsic sound? Yes, it does. To me, the sign of a good D/A is whether it can translate and retain the micro-dynamics of each instrument. On "Holiday for Sweet Louise" from one of my longtime reference CDs, 3d Matinee's Meanwhile, the iDac kept everything in this rock-and-roll mix from getting messy. On this 3d Matinee cut, the surge of the keyboards works in oddball rhythmic counterpoint to the electric guitar line, each pulses or surges at different moments. This subtle interplay comes through clearly with the iDac. The core iDac sonic character is musical and non-fatiguing while still being precise and detailed.

How does the iDac stack up against the Wyred4Sound Dac2 I reviewed recently? I'd have to call it almost a sonic dead heat, but the Dac2 wins hands-down when it comes to flexibility and overall value. On USB inputs the iDac sounded less dynamic than the Dac2. The iDac also lacked some of the Dac2's imaging precision and dimensionality. When both the iDac and Wyred4Sound were fed the same signal from an Empirical Audio Off-Ramp 4, sonic differences vanished. In matched-level A/B tests I could not reliably tell one from the other. But when you consider the Dac2's better USB implementation, excellent built-in volume control, and ability to drive a pair of balanced XLR outputs as well as a pair of single ended RCA outputs, you see why I think it's a better overall value. But if you already have a high-quality preamp, the iDac would be a more cost-effective choice.

With a high-definition digital signal the iDac can create an almost holographic soundstage. I played a bunch of high-def music through the iDac—both my own recordings and those from Reference and MA. In imaging palpability and dimensional definition the iDac pretty much disappeared. I wasn't listening to the iDac as much as listening through it.

WHY IDAC?

In the automobile business there's an old and somewhat crude saying, "There's an ass for every seat." Once you remove the intrinsic condescension of the phrase, it concisely conveys the concept that every car is made with a particular customer in mind, and a salesman's job is to match the customer with the best product for his needs. So if we look at the iDac this way, what would be the best place for an iDac?

I see the iDac as an ideal step-up/catch-up product. It's perfect for the audio newcomer who wants to upgrade his computer audio system or an experienced audiophile who wants a way to bring all those digital audio and video sources like X-Boxes, Apple TVs, Blu-ray players, computers, and even iPods into his two-channel analog system.

But there is a third customer for the iDac. Although it's priced under $1000 I suspect that many 3+-year-old DACs that lack high-def capabilities, even those with substantially higher original price tags, could be replaced by an iDac. The sound that an iDac can pull from even a 320bps MP3 file on an iPod is pretty amazing. But with non-lossy 16/44, 96/24, and 192/24 music files coming in through its S/PDIF inputs, the iDac demonstrates its full sonic potential. This is one sonically serious DAC whose performance transcends its name. tas

Benchmark Media Systems DAC1 HDR DAC/Preamplifier

Giant-Killer?

Paul Seydor

When Benchmark Media's DAC1 digital-to-analog converter was introduced a few years ago, it not only garnered rave reviews, but those reviewers who performed rigorous laboratory tests only wound up confirming Benchmark's claims that it was a virtually distortionless device.1 At the time Benchmark essentially serviced the professional audio market, where its products are widely used—it's probably no exaggeration to say that at some point in the production of better than half of all digitally based music releases a Benchmark product is used for monitoring. But as happens every now and then, word gets out about really exceptional professional products—the original Rogers LS3/5a is a classic case in point—and soon audiophiles start buying them.

Although the DAC1 is designed principally as a DAC, it is outfitted with a headphone amplifier and a high-quality volume pot, the latter enabling it to control headphone level and output level to an amplifier, but only for the digital source it is converting. Many audiophiles who purchased these units soon discovered that bypassing their own preamplifier made for even more transparent reproduction. This led Benchmark to introduce the DAC1 PRE, which adds two more coaxial inputs to the existing coaxial and TosLink inputs of the DAC1, plus a USB input and a stereo analog input, thus making it possible for audiophiles to hook up, say, their turntables (via phono preamplifiers) *and* their digital sources,

including computers, all of which could be switched and level-controlled by the PRE. It's hard to imagine Benchmark was here not looking beyond industry professionals to consumers. Many audiophiles, however, wanted all the DAC1 PRE had to offer plus the convenience of remote control operation, which brings me to the subject of this review. The new DAC1 HDR consists of a DAC1, with the expanded input selection of the DAC1 PRE, together with a remote control handset that operates volume, input selection, on/off, and mute functions. The "HDR" stands for "HDR VC," a propriety motor-driven Alps potentiometer that, quoting the manual, "avoids the dynamic range limitations of digital volume

controls and the distortion and noise introduced by IC-based analog volume controls."

Inasmuch as the DAC1, which constitutes the digital circuitry of the DAC1 HDR, is a known commodity, I shall not expend much print describing either its workings or its sound. I refer you instead to Robert Greene's thorough review in Issue 183. A man hardly given to hype or overstatement, REG judged the sonic performance so neutral, transparent, noise-and-distortion-free, and source-accurate that he pronounced digital's initial promise effectively realized: perfect sound, albeit *within* the Red Book CD standard.2 This correlates with my own experience of the DAC1, which I've used for some three years now with a variety of stand-alone or integrated transports. According to Benchmark, the DAC1 circuitry inside the DAC1 HDR is identical to what is in the original DAC1, and my listening tests confirm this. Given a bit-accurate transport,3 the DAC1 HDR reproduces your CDs

with something approaching peerless accuracy. This doesn't necessarily mean you'll like what you hear, but that isn't Benchmark's concern—truth in reporting is. Inasmuch as there are many ways for the digital nasties to enter the chain from recording through processing through manufacturing, it's perfectly understandable why accuracy as such may not be every audiophile's highest priority. But if it is yours, you will have to spend a great deal more money to buy greater accuracy than what this Benchmark is capable of, and even then the improvements, such as they are, will be incremental rather than dramatic. (It remains a fact of life—a happy fact for audiophiles—that as regards electronic components, especially digital ones, advancements in technology constantly lower the price point for ever-higher performance.)

"Super-clean, super-clear, super-quiet, super-transparent." These are the first notes I scribbled down several moments after I had recovered from

EQUIPMENT REVIEW - Benchmark Media Systems DAC1 HDR DAC/Preamplifier

the initial burst of Ingrid Fliter's Steinway from her scintillatingly essayed program of the complete Chopin waltzes on EMI. She and her instrument are nicely captured with a good balance of focus and ambience such that when played back at moderate levels ("moderate" here means too loud for conversation), this recording affords a good row D or E perspective. As I was in a piano mood, next up was Martha Argerich's justly acclaimed performance of Ravel's *Gaspard de la Nuit* (DG). Here the perspective is close up and very personal, with a wealth of detail on display, both musical and extra-musical, from the delicacies of Argerich's touch and phrasings to the pages of the scores being turned (at one point, just after the opening of the "Scarbo" movement, you can hear either the pianist's or her page turner's lips part). None of this extra-musical detail, incidentally, is due to any brightness or edginess that inheres in the Benchmark circuitry, nor does it detract from the you-are-there experience of Argerich's thrilling performance.

As noted, inasmuch as I am already familiar with the DAC circuitry of the DCA1 HDR through a variety of front ends, I was most interested in evaluating the unit as a front end in and of itself, so I concentrated on a variety of vinyl and SACD sources. Once I report that it seemed to me to afford a fractionally more transparent window back to the sources than just about anything I've used so far, I don't really have a whole lot more to say. The *Christmas Revels* is an LP I've used

You will have to spend a very, very great deal more money to find a better DAC or linestage than are in this unassuming but nearly perfect unit.

hundreds of times in evaluating equipment: When the Revelers wander in talking among themselves I heard at once greater inner detail and clarity allied to a more convincing presentation of the group, the soundstage, and the space than from any other control unit I've used, with the possible exception of Musical Surroundings's SuperNova (which dispenses with a high-level gain stage altogether), and the SuperNova is scarcely better,

if it is better at all (review Issue 200). When Room summons the group to attention by banging on a pan, the walls of the space are in clear and present evidence as never before. This motley group of players, including a children's chorus and all manner of "olde" instruments, is also one of the toughest tests of tone color I know. Again, the DAC1 HDR splashed their hues all over the walls in a riot of holiday festivity.

Suffice it to say that the tonal neutrality and resolution so widely observed in the DAC1 has been carried over into the HDR's linestage. Components this electronically transparent can be very frustrating to write about it because they don't give you a whole lot to describe. The DAC1 HDR is the same size as all the previous products in this series, which is to say so elegantly diminutive as to invite the word "cute," strange as it may seem when applied to a piece of professional gear. Of course its small size must mean that it lacks truly wide dynamic range, the H(igh) D(ynamic) R(ange) nomenclature surely mere wishful thinking. Think again: I put on the most dynamically challenging source material I could find, including the classic direct-to-disc *For Duke*, the *Sheffield Drum Test Record*, Jim Boyk's SACD *Tonalities of Emotion*, any number of Telarc's symphonic productions that feature their Big Bass Drums (all in SACD)—never did I hear the slightest indication that anything was being withheld, damped down, suppressed, compressed, distorted, or unrevealed, regardless of level. And bass response is by any measure simply firm, ample, and strong—try the thundering coda in Richard Goode's *Waldstein* (Nonesuch). All this from a DAC/preamp that costs $1895, is light enough to hold in one hand, has a footprint

smaller than an iPad and is only about three times as thick (i.e., high), and that has the sheer gall to house its power supply in the same tiny chassis as the digital and audio circuitry. Metaphors fail me—but think, Toto, this isn't Kansas any more or Alice, this is surely Wonderland indeed.

Criticisms? Virtually none related to sonic performance. I wish the excessively bright blue LEDs could be dimmed and some kind of volume

SPECS & PRICING

Benchmark Media Systems DAC1 HDR DAC/ Preamplifier
Analog input: One pair, unbalanced
Digital inputs: Five (one USB, one optical, three coaxial)
Outputs: One unbalanced pair, one balanced pair
Headphone outputs: Two
Dimensions: 9.5" x 1.725" x 8.5"
Weight: 3.5 lbs.
Warranty: 5 years

U.S.

Price: $1895

BENCHMARK MEDIA SYSTEMS, INC.
203 East Hampton Place, Ste. 2
Syracuse, NY 13206
(315) 437-6300
benchmarkmedia.com

EQUIPMENT REVIEW - Benchmark Media Systems
DAC1 HDR DAC/Preamplifier

indicator were visible from across the room. I do wonder about the decision to include only one pair of analog inputs, which forces users like me and I suspect many, if not most, readers of this magazine, who enjoy both vinyl and SACD, to endure the inconvenience of disconnecting the one in order to connect another. There is no room on the back of the chassis for another set of analog inputs unless Benchmark dispenses with two of the three coaxial digital inputs (which, I think, could easily be done: How many of us need more than one, especially with a TosLink and a USB input on hand?). Or perhaps the company could make an accessory outboard switching box with two or three additional analog inputs. Although the channel-to-channel tracking of the HDR's volume control is as precise as any I've ever used and far, far more so than most, I'd still like a balance control, but, again, there's no room for one here. Neither is there a tape monitor or EPL loop or stereo/mono switching.

All of which only underscores the obvious—the DAC1 HDR is no more or less than what it purports to be: a high performance digital-to-analog converter with an analog stereo input and essential but minimalist linestage functions, including volume control and input selection from a remote handset. Still, much as I have enjoyed using this superb product, I can't help but wonder if Benchmark doesn't have more products up its very inventive sleeves. I for one would stand in line to try out a true full-function, non-minimalist preamplifier from this company, one that, in addition to everything found on the DAC1 HDR, also provides a few more analog inputs, both RCA and XLR, a stereo/mono switch,

and a balance control. Of course, it would have to be larger, probably twice the size of this unit, but that still leaves a pretty compact package. In the meantime, if your control and input needs are covered by the DAC1 HDR, my final word is that you will have to spend a very, very great deal of money to find a better DAC and linestage than are in this unassuming but nearly perfect unit. If I may be indulged one last allusion to the world of children's stories: This little Jack may not slay every giant out there, but I'll wager he'll hold his own against any of them. **tas**

[1] Benchmark publishes the most thorough test data of any manufacturer I've ever encountered: 19 of the instruction manual's 52 pages are taken up with performance graphs, measurements, and specifications.

[2] That is an important qualification: good as Red Book digital has gotten these last several years, the improvements wrought from higher resolution digital formats (SACD, DVD-Audio, Blu-ray, etc.) are not to be minimized.

[3] Because the DAC1's Ultralock circuit eliminates all jitter except any that might inhere in the source itself, many consumers infer that the quality of the transport is irrelevant. This would be true only if all CD transports were bit-accurate. Some aren't, and to that extent will yield unpredictable results with the DAC1 and also every other DAC. Alas, as with so many audio products, there is no necessary correlation between accuracy and price: Many cheap CD players and transports are bit-accurate, while some expensive ones aren't. Owing to the high intrinsic performance of Benchmark DACs and DAC/preamps, if you're getting weird results, I'd suspect the transport. [*For another view on the subject of bit accuracy in CD transports, see my blog "Transport Bit Accuracy" on theabsolutesound. com.* —RH]

◀ PREVIOUS PAGE

Bryston BDA-1 and Audio Research DAC7

Two Revolutionary DACs

Alan Taffel

From the founding of this magazine, Harry Pearson did our industry a great service by establishing a reference: the sound of unamplified live music in a natural space. As a result, designers began to pay attention to such crucial sonic phenomena as soundstages, timbres, dynamics, and the localization of musicians. However, an unintended consequence of HP's radical reference was that the industry became so wrapped up in recreating the sound of live music, it often paid less attention to the experience of hearing live music.

Listening to live music differs from listening to a recording, and not just in sonic terms. I find, for instance, that when I am at a live concert I can comfortably absorb and grasp new material on a first hearing. Yet when listening to a recording of new music, I usually need several playings to reach the same level of appreciation. Also, at a live performance an audience witnesses not only the physical but also the musical interplay between performers. In contrast, even very good-sounding audio systems often fail to convey the interplay that made a particular performance unique. And though stereo setups strive mightily to present a facsimile of the recording venue within our listening space, that is very different from the feeling of having a live venue—and its attendant acoustics—surround you. To cite one final example, consider how refreshed one feels after a live performance, as opposed to the fatigue that can result from even a short home listening session.

This dichotomy between live and recorded listening experiences began to occupy my noggin after I spent time with two new DACs from Bryston and Audio Research, the $1995 BDA-1 and $3495 DAC7, respectively. These components are, in my view, at the vanguard of a new digital era. Like the best analog systems, they deliver not only state-of-the-art sonics, but propel the listener right into the heart of performances. This is exactly what happens in the live listening experience.

What these two DACs do is a bit difficult to convey because our industry has not spent decades establishing a vocabulary to describe the nature—as opposed to the sound—of live music. However, here

EQUIPMENT REVIEW - Bryston BDA-1 and Audio Research DAC7

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goes. HP often used photographic terms to help readers, through visual analogies, understand his sonic descriptions. I will enlist another of the senses—taste. For those who have ever eaten truly sublime food, you know what happens: You put a bite in your mouth; at first, individual flavors assert themselves, some more urgently than others; then those tastes begin to intertwine, playing off each other; finally, they form an integrated whole that is in perfect balance and makes complete sense. The wondrous part about this experience is that it requires absolutely no effort on the part of the diner; the process washes over you, explaining itself as it goes, leaving you with nothing to do but enjoy and appreciate the artistry that made it.

This is also what happens during a live concert—just substitute instruments, notes, and musical lines for flavors. I submit that it is what should happen when we listen to an audio system. Unfortunately, most audio systems deliver the sonic equivalent of a heavy-handed stew. Flavors are inextricably mashed together, and sorting them out takes real work. This phenomenon is so pervasive that I believe most audiophiles do not even know they are hard at work. The system tells them "what" is happening musically, but the listener is left to fill in the "how" and the "why." This is tiring!

The Bryston and ARC DACs reveal, Linn-style, the individual strands that make up the music's fabric. But, as with tasting miraculous food, that is only the first step. They go further, allowing the listener to then hear how each instrumental line relates to the others,

how they trade off, and why the composer wrote the music as he did. The same goes for the musicians themselves. Each has decided to play his line a particular way, yet each is simultaneously listening to his fellow musicians and making adjustments so his part fits and enhances the whole. Believe it or not, these DACs make all this plain as day.

Listen, for example, to the first in the lovely collection of Dvořák's *Serenades from Bohemia* [Praga]. Through either of these DACs, one can clearly hear the purposefully steady tempo set out by the violin; the trading of the thematic line from strings, to piano, to woodwinds; the way each instrument uses dynamics to momentarily take the spotlight or step out of it; and the perfect synchrony of the ritard as the piece concludes. Sure, the recording also sounds good, with sweet timbres and a finely rendered soundstage, but this is so much more.

Reading the above, without hearing the units themselves, one might too easily plop the BDA-1 and DAC7 into the category of "analytical" components. In a way, they are, in that they reveal a great deal about what is going on below the surface. But we typically associate the term "analytical" with cold-sounding products that lack cohesion and soul, and require the listener to synthesize all the information into an integrated whole. That is manifestly not the case here. Instead, both these DACs are warm as the sun and, like that great bite of food, not only reveal the ingredients, but explain and combine them for you. This makes these DACs not only the most informative I have ever heard, but also

the easiest and most relaxing to listen to. This is no mean feat. Indeed, it is revolutionary.

Although these abilities go a long way toward recreating the live experience, that experience is still not quite complete. One thing neither of these DACs can do is transport listeners to the original performance venue, a limitation inherent in their ability to play only two channels. Stereo will never enfold listeners in an acoustic embrace the way good multichannel can. However, for a host of practical reasons, most of us must simply set that particular dream aside.

In terms of sheer sonics, the Bryston and ARC share many qualities—but also differ in presentation and operation. By now this may go without saying, but both DACs deliver richly detailed timbres, have astoundingly high resolution, offer dynamics both nuanced and bold, and can paint wide deep soundstages. Both can be exquisitely delicate, as on the opening of the Beach Boys' "Wouldn't It Be Nice," and both can rock out, as they do on the Stones' classic "Gimme Shelter." To my chagrin, both the BDA-1 and DAC7 handily put my reference unit to shame.

The ARC is the brighter of the two, but not in any negative sense of the term. The DAC7 just shines a brighter floodlight on the proceedings, and places the listener closer to the musicians than does the Bryston. The ARC also boasts better bass—with terrific definition and character—which gives it a warmer overall tonality. For its part, the Bryston offers more precise timing, slightly purer timbres, and sharper leading edges. This last characteristic renders the Bryston a bit more revealing, but

SPECS & PRICING

BRYSTON BDA-1
Inputs: Two (each) digital USB, RCA, TosLink, AES/EBU, BNC
Outputs: Analog balanced XLR and single-ended RCA
Resolution: 192/24 (S/PDIF), 48/16 (USB)
Dimensions: 17" x 1.75" x 11.25"
Weight: 18 lbs.

AUDIO RESEARCH DAC7
Inputs: Digital USB, XLR, RCA, BNC, TosLink
Outputs: Analog balanced XLR and single-ended RCA
Resolution: 192/24 (S/PDIF), 48/16 (USB)
Dimensions: 19" x 5.25" x 10"
Weight: 11.5 lbs.

U.S.

BRYSTON BDA-1
Price: $1995

BRYSTON LIMITED
677 Neal Drive
Peterborough, Ontario, Canada K9J 7Y4
(705) 742-5325
bryston.ca

AUDIO RESEARCH DAC7
Price: $3495

AUDIO RESEARCH CORPORATION
3900 Annapolis Lane N.
Plymouth, Minnesota USA 55447-5447
(763) 577-9700
audioresearch.com

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EQUIPMENT REVIEW - Bryston BDA-1 and Audio Research DAC7

also makes the ARC more forgiving on inferior recordings.

Differences between these DACs also relate to musical presentation. Although both DACs deliver the "heart of the performance" experience, they do not go about doing so in quite the same way. The Bryston is the more probing of the two, relaying the greater wealth of information about the composition and the performance. To extend the food analogy, the Bryston is more like deconstructionist cuisine. The ARC, on the other hand, is more organic. One is less aware of the whys and wherefores of the music, and more aware of how it fits together. These comments should not be taken as mutually exclusive; the Bryston also pulls everything together, and the ARC also presents gobs of musically relevant information. The difference is a matter of subtle emphasis.

The biggest sonic difference between these two DACs is heard only when comparing their USB inputs. Neither of them surmounts USB's inherent limitations, and neither supports high sampling rates or deep bit-depths. However, the ARC's USB input is clearly superior to the Bryston's, which sounds dull and cloaked. With the help of a bright-leaning USB cable to compensate, such as the Synergistics Tricon, the Bryston achieves a satisfactory result—which is about as good as USB gets. But the ARC needs no such assistance, since its USB input is neutral from the get-go. Indeed, using the Synergistics cable with the DAC7 places much more emphasis on USB's high-frequency foibles than anyone should be forced to endure.

From an operational perspective, each DAC has its own advantages. The BDA-1's front panel features an incredibly useful LED arrangement

that displays both the incoming sample rate and the upconverted rate. Two things to note here: First, upconversion is switchable on the Bryston, allowing purists to bypass it (though it really does improve the sound); and, second, the Bryston always upconverts to an integer multiple of the incoming rate rather than to some fixed maximum. The LEDs confirm, at a glance, the incoming rate (all the way up to 192/24 for S/PDIF), whether upconversion has been selected, and, if so, the new rate. The ARC has no comparable display.

Both the Bryston and the ARC offer a bounty of digital source options—though the BDA-1 has more of them—including a BNC connection. Here again, the Bryston is more user-friendly, allowing direct selection of the desired source from the front panel. The ARC's front panel requires the user to scroll through inputs round-robin style, though its remote does permit direct selection. The DAC7's remote also enables users to control the "transport" (play, stop, pause, skip) functions of their PC's music server software, a slick and highly convenient feature. The BDA-1 does not ship with a remote, but the company separately offers the BR2 ($350), which also boasts the PC transport feature.

Clearly, though, the big news is not their feature sets but the level of live-performance musicality both of these DACs achieve. Choosing between them comes down to matters of preference and priorities. There is a lot of buzz right now about various digital products, some of which have astronomical prices and some of which have exotic designs. The fact that these two DACs have neither only reinforces their revolutionary nature. Believe me, this is where the buzz belongs. tas

Wyred 4 Sound DAC-2 Digital-to-Analog Converter

High Value

Steven Stone

Just as the Tea Party is a reaction to what its constituents see as a runaway tax-and-spend government, so the resurgence of do-it-yourself and "high-value" products reflects a rejection of price-no-object components by audiophiles of modest means. I empathize with this viewpoint. Over the years I've become less and less enamored of products above my own completely personal price-points. I won't dispute that new stratospherically priced components often deliver a technological edge. But chances are good that the new methodology will be licensed to other manufacturers and incorporated into less expensive products in a matter of months. And then there is this: No matter how expensive and beautifully made a five-year-old DAC may be, its performance will be challenged by many far-less-expensive current-production DACs. Sorry, but that's the way technology works. For me, it makes more sense to spend big money on components that will not be eclipsed in six months or a year, such as a power amplifier or speakers, rather than a DAC.

Then along comes the $1499 Wyred 4 Sound DAC-2. It's certainly priced attractively. Yes, it's way more than a HRT Music Streamer II USB DAC ($149) or even a Music Streamer II+ ($350), but way less than a Weiss DAC 202 ($6670). Psychologically, $1500 is a figure that divides "might buy's" from "hell no's." And when you factor in that the Wyred 4 Sound DAC-2 includes a very-well-thought-out analog preamplifier that can replace a separate preamp in an all-digital system, the DAC-2's obvious value and reasonable price make it even more enticing.

The DAC-2 can serve as the control center for a high-end near-field desktop, a two-channel room system, or even in a multichannel system, which means that most, if not all TAS readers could easily find a place for a DAC-2 in their systems.

TECH TOUR—PARTS AIN'T DESIGN

As I enumerate the stuff inside the Wyred 4 Sound DAC, please consider that just because the DAC-2 uses an ESS 9018 Sabre32 32-bit DAC doesn't mean it will sound identical to other units that use this same part. DAC parts only have a "sound" within the context of the hardware and software in which they are employed.

Wyred 4 Sound's designer, EJ Sarmento, began with that Sabre chip and then formulated a fully balanced design around it. The Sabre is an eight-channel device that can be configured as a quad-differential circuit with four differential DACs per phase per channel, which delivers 132dB of dynamic range. Because jitter is inevitable in an S/PDIF digital stream, the DAC-2's Sabre ESS 9018 deals with jitter in a clever way—by disregarding the clock signals coming from the source. Instead the Sabre re-clocks by instituting a discrete digital delay that can affect either the positive or negative edge of each duty cycle by up to 50%. The processor accesses the width of each digital pulse, compares it to past pulses, and assigns the pulse a particular quantified width. Then the device processes each pulse in turn with no attempt to re-time the clock, it merely time-stamps the information and passes it downstream. According to Wyred 4 Sound, this methodology makes it possible for the DAC-2 to accept up to 50ns of random and 200ns of sinusoidal jitter with no audible effects. Technically, this is an asynchronous system, since the data flow is controlled by the DAC, not the computer. But this is not the same asynchronous methodology used by Wavelength, which focuses on the interface between the computer and the DAC.

Other technical features of the DAC-2 include automatic 386x oversampling, an oversized toroidal power transformer, 35-amp bridge-rectified power supply with 88,000uF filtering, proprietary low-ESR "super caps," Schottky bridge rectifiers, a fully discrete output stage using dual-differential input amplifier stages and Dale RN55d resistors, and a 32-bit digital volume control. All the circuit boards—digital, analog output, and USB input—are designed so they can be upgraded to allow for some degree of future-proofing.

Physically, the DAC-2 is compact, taking up only a half-rack width of 8 ½ inches. The front panel features a matte-finished face available in either black or silver with three buttons (up, down, and power), and a vacuum fluorescent (VFD) display. The remote control is an inexpensive plastic job with volume, balance, power on/off, input selector, phase, mute, and HT bypass. This last button lets you route a two-channel analog

EQUIPMENT REVIEW - Wyred 4 Sound DAC-2 Digital-to-Analog Converter

signal through the DAC-2 so it won't alter the volume of that input.

On the back of the DAC-2 you will find an on/off switch, two RCA coaxial S/PDIF inputs, two TosLink inputs, one AES/EBU input, one I2S2 input (via HDMI), and one USB input. The DAC-2 also has one pair of balanced XLR outputs, one pair of unbalanced RCA outputs, and one pair of "Bypass" analog inputs. The DAC-2 is capable of accepting up to a 192kHz 24-bit signal. It accomplishes this via a proprietary asynchronous USB driver. If you're a Windows user, you're already familiar with drivers, as it seems that virtually every hardware device requires one be installed prior to operation. Mac users may be less at ease with drivers, as most come pre-installed in the Mac OS. Being primarily a Mac user, when I first set up the DAC-2 I didn't install the driver, the result being dead silence. After glancing through the owners manual I discovered the driver CD, installed the driver, and then all was well.

SET UP AND DAILY USE

Most of the time the DAC-2 remained in my near-field desktop computer audio system (see Associated Equipment for specific list of gear), but it also spent some time in my large room system. In my computer system I set up the DAC-2 so that it received a USB input from the computer, an AES/EBU input from the output of an Empirical Audio Off-Ramp 3 being fed USB, a TosLink S/PDIF input from the computer, and an RCA coaxial S/PDIF input from my Oppo DBP-80 universal player

During the 60+ days I had the DAC-2 in my systems it never malfunctioned in any ergonomic or performance parameter. My only quibble is

that the acceptance angle for the remote control is rather narrow, especially in the vertical plane. Unless I lowered the remote so it was nearly parallel with the faceplate, its commands were not acted upon. A domed rather than inset IR receiver might solve this little problem.

Front-panel design always comes down to a battle between visual simplicity (fewer buttons and knobs) and the complexity of commands needed to make a three-button system work with the fewest sub-menus. The DAC-2 has only three buttons, so you need to do a double-button push to get into the settings menu. To switch from volume-control mode to input-control mode you must push the "power" button, which in this case doesn't power down the DAC-2, but switches it between these two modes. Simple? Well, sort of. Just a note—if you push the power button fast, it will change between volume or input mode. If you push the button and hold it down, it will power down the unit. My problem was that it was far too easy to be in the wrong mode, so, instead of adjusting the volume, I'd be changing inputs. My advice—stick with the remote control.

Nestled in the set-up menu is something called "IIR bandwidth." No, it's not for adjusting the frequency of your remote control. Instead it means "infinite impulse response," and it adjusts the filter's bandwidth. You may choose 50k, 60k, or 70k. You also have a choice of two roll-off slopes (fast and slow), brightness level for the front-panel display, and the option for each individual input to be either a fixed or variable output source.

THE SOUND

Sonically the DAC-2 delivers on its promises. The overall sound has a solidity and weight that are

both arresting and involving. Much of this sonic goodness stems from the DAC-2's lack of low-level noise and digital artifacts.

A good part of the DAC-2's apparent clarity comes from its ability to portray both lateral and dimensional information unambiguously. I never found myself wondering exactly where an instrument or sound was within the soundstage. One of my reference cuts for imaging precision is "Punchbowl" off Punch Brothers' *Punch* album (reviewed Issue 208). Since the sessions were recorded live with five musicians clustered around one main stereo pair of microphones (similar to how you would record a string quartet), it is a good test of how well a system can preserve and uncover dimensional and locational cues. The DAC-2 captures the interplay between the mandolin and fiddle as they play identical lines and how their decays trail off differently based on their different physical locations, reverberating off the rear and sidewalls of the recording space.

While the DAC-2's presentation is certainly fast and incisive, it never leads with an electronic edge. I wouldn't go so far as to say the DAC-2 is tube-like, since it adds little, if any, harmonic warmth or additional depth to the soundstage. The DAC-2's analog section presents music with a clarity and precision that will keep your left-brain fully involved. On The Band of Heathen's tune "Let Your Heart Not Be Troubled" from their *One Foot in the Ether* album, the DAC-2 had no difficulty unraveling the complex multitracked parts for my listening pleasure.

As with the other USB and FireWire DACs I've reviewed, I spent lots of time comparing the iTunes with Amarra, Pure Music, and AyreWave players through the DAC-2. In every case it was

easy to hear the superiority of these software solutions over iTunes through the DAC-2. The three D's—depth, definition, and dynamics—all improved. I was especially aware of this step up in quality on orchestra recordings, such as my own high-definition recordings of the Boulder Philharmonic (down-sampled from DSD to 96/24). Although the overall soundstage and image size didn't change appreciably, the spaces between

SPECS & PRICING

Wyred 4 Sound DAC-2 Digital-to-Analog Converter
Type: USB and S/PDIF DAC with built-in volume control
DAC: ESS Reference audio (ES9018) 32-bit DAC
Inputs: Two coax inputs, two TosLink inputs, one AES/EBU input, HT bypass inputs (via DC trigger,) 24-bit/192kHz asynchronous USB input
Output impedance: 100 ohms
Driver: Proprietary for 32/64-bit Windows XP, Vista, and Mac OS above 10.4
Dimensions: 8.5" x 4.125" x 13.5"
Weight: 16 lbs.

U.S.

Price: $1499

WYRED 4 SOUND
2323 Tuley Rd Unit A-C
Paso Robles, CA, 93466
(805) 237-2113
wyred4sound.com

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EQUIPMENT REVIEW - Wyred 4 Sound DAC-2 Digital-to-Analog Converter

instruments were more pronounced and each instrument seemed more palpable and three-dimensional. Also all three programs preserved more of the delicacy and air in the string sections and woodwinds than iTunes could muster.

I installed the DAC-2 in my large-room system principally to see how it compared with my longtime reference Meridian 568.2 controller on two-channel digital sources. Since the Meridian 598 transport has RCA coaxial as well as Meridian's proprietary MHR (Meridian High Resolution) connectors, I was able to do closely matched A/B comparisons. The DAC-2 and Meridian 568.2 sounded much more similar than I expected. The 568.2 was slightly darker harmonically with less upper-frequency air and shimmer. The Meridian also had less upper-midrange dynamic contrast and speed, and the faintest haziness around individual instrument outlines. Soundstage size and depth through the two units were identical, yet the DAC-2 was more incisive in terms of locational cues and subtle dimensional details.

Because I'm a glutton for punishment I also placed the DAC-2 in my small-room system, so I could compare it to the Lexicon MC-12 HD on two-channel sources. I ran the TosLink output from an Oppo DBP-83SE to the DAC-2 and its RCA coaxial output to the Lexicon. In comparison to the DAC-2 the Lexicon sounds somewhat "grayish," with less sense of dynamic life and contrast. Although their soundstage sizes were very similar, it was easier to locate parts and listen into the mix through the DAC-2. The Lexicon wasn't murky, but it didn't have quite the clarity or ease of the DAC-2. When I switched inputs so the DAC-2 got the RCA coaxial and the Lexicon got

the TosLink, nothing changed; the DAC-2 was still audibly superior.

Some of the sonic differences between the Wyred 4 Sound and the Lexicon could be attributed to the fact that the Lexicon's output was from its single-ended rather than balanced main outputs, because the DAC-2's bypass inputs are only single-ended. Years ago, when I first installed the Lexicon in my system, I compared its single-ended RCA outputs to its balanced XLR outputs and found that balanced was clearly superior in terms of depth, dynamics, and overall musicality. So some of the "grayness" I was hearing from the Lexicon was because of the single-ended connection.

Moving the DAC-2 back to my desktop computer system I compared the DAC-2's USB input with the performance of the Empirical Audio Off-Ramp 3 USB converter. Using Amarra 2.0 in its new "stand-alone" mode, where it operates without iTunes, allowed me to do relatively fast, but not instantaneous, A/B comparisons. I still had to shut down Amarra, change inputs on the DAC-2, change output devices in OS X's sound preference panel, and then re-open and restart Amarra—a process that took me about 30 seconds when I got all the moves down pat. Given that this was not an ideal A/B setup, I still heard some subtle differences between the two front ends. The Off-Ramp had a slightly more distant perspective, with greater sense of ambience, but not quite as much immediacy. It was as if I had been moved back three or four rows in the concert hall. On the 176.4/24 high-resolution recording from MA Recordings, La Segunda, (which was played back at 88.2 by the Off-Ramp due to its 96k upper limit and at 176.4 through the DAC-2),

the DAC-2 had better depth recreation and a more solid feel. Rarely have I heard a recording and playback that sounded more like a live microphone feed.

Finally it was time to compare the DAC-2 to the Weiss DAC 202, the "big dog" in my DAC stable. Since I had to switch the two DACs in and out of my system, I couldn't do any matched-level A/B comparisons—all my notes are from long-term listening sessions.

Where the Empirical Audio Off-Ramp seemed to render a more distant perspective than the DAC-2, the Weiss DAC 202 seemed to provide a more close-up view of the event, as if I'd moved up a couple of rows. The DAC 202 also managed to retain more spatial information, with an even greater sense of dimensionality and depth. The level of difference was subtle, and more on the order of the differences you'd hear between two comparably-priced premium cables than between a $1500 DAC and one that costs $6500.

After extensive time with both units it was clear that the DAC 202 bests the DAC-2 in all audible performance parameters, but the DAC-2 is so close to the performance to the DAC 202 that most people would be shocked to learn that there was a $5000 difference in their prices. Even more importantly, I never felt that burning desire to return to the DAC 202 while I was listening to the DAC-2, which gives you an idea of how good the DAC-2 is. For $1499 this DAC-2 is nearly a giant-killer.

FINAL THOUGHTS

Since The Absolute Sound is a print publication, there ain't no way (except on rare occasions) we're gonna be the first to "publish" a review of a component as au courant and in-demand as the Wyred 4 Sound DAC-2. The question on my mind, when I read other extremely positive reviews, was, "Have these folks ever heard a state-of-the-art DAC to compare against the DAC-2?" Because I know full well what happens when you're confronted by the best you've heard—you get a wee bit overenthused.

After my time with the DAC-2 I can't help but be impressed with it. Wyred 4 Sound has combined a rich feature set with remarkable performance at a price that makes it hard to beat. While I haven't heard every DAC (who has?), I have yet to hear any USB DACs under $1500 that I like better, and I doubt that you will either, at least for the time being. Factor in the DAC-2's current 192kHz high-resolution capabilities and 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. Will the DAC-2 get you back in touch with your first seminal high-fidelity experience? There's a high probability that it will. tas

Accuracy *and* Musicality

Berkeley Audio Design Alpha DAC

Robert Harley

Most audio products designed for professional use are sonically inferior compared to the best high-end "consumer" components. That's because, in the pro world, features, functionality, and reliability under harsh conditions take precedence over sound quality. And odd as it might sound, most professionals are more price-sensitive than audiophiles.

A notable exception is the new Alpha DAC digital-to-analog converter from Berkeley Audio Design. Although outfitted with professional features, the Alpha DAC is designed to bring first-rate digital-to-analog conversion to both pros and audiophiles. That's not such a stretch considering that the Alpha DAC's design team also created High Definition Compatible Digital (HDCD), as well as the Pacific Microsonics Model Two, widely considered among both pros and knowledgeable consumers as the finest analog-to-digital and digital-to-analog converters extant. The Model Two is the professional high-resolution A/D converter and HDCD encoder found in the world's best mastering rooms. Most of the Pacific Microsonics design team reformed

as Berkeley Audio Design, turning its attention to making a D/A converter capable of decoding high-resolution sources.

The Alpha DAC's professional orientation is apparent from the unit's form factor, features, and front-panel layout. The small chassis is sturdy and nicely finished, but utilitarian by high-end standards. Similarly, the front panel is all business and no fluff; the panel is loaded with essential controls and indicators. A large alphanumeric display shows the input sampling rate, from 32kHz to 192kHz. Small LEDs indicate the selected input, signal lock, HDCD decoding, polarity inversion, and the selected digital filter. A pair of buttons next to the display adjusts the output level; the Alpha DAC is designed to

drive a power amplifier directly. The display automatically switches to show the output level when the output level is being adjusted. Although you can, of course, treat the Alpha DAC like any other source component and run its output through your preamplifier, the Alpha DAC only reveals its full potential with no preamplifier in the signal path—particularly when decoding high-resolution sources. A remote control allows you to adjust level, channel balance, polarity, muting, input selection, and display dimming.

Another of the Alpha DAC features that reveals its professional roots is the selectable digital filter. This feature isn't provided so that you can "tune" the DAC to your system or personal preference. Rather, there's one filter that is

absolutely optimum (Filter 1), while the other filters allow mastering engineers to hear how their work will sound on the digital filters typically found in consumer CD players. One of the filters is identical to the Pacific Microsonics PMD-200 found in many high-end CD players. Incidentally, the availability of the PMD-100 and PMD-200 in the mid-1990s greatly improved the sound of CD players at the time. The Pacific Microsonics 8-X oversampling filter (with HDCD decoding) replaced the ubiquitous NPC filter chip, rendering a wholesale improvement in CD sound.

A rear-panel RJ-45 jack marked "BADA" (Berkeley Audio Design Alpha) is designed to accept encrypted high-resolution from an outboard decoder of DRM-protected music. The

EQUIPMENT REVIEW - Berkeley Audio Design Alpha DAC

idea is that the outboard box would strip music of DRM and then encrypt the data for transmission over the BADA interface. This arrangement meets the licensing laws while providing listeners with uncompromised playback of DRM-protected files. Although such forward-thinking is welcome, record labels are quickly abandoning DRM. Another forward-thinking touch is the Alpha DAC's ability to receive upgraded firmware simply by playing a CD with the upgrade encoded on it and feeding the CD transport's output into one of the Alpha DAC's digital inputs.

The Alpha DAC's design and build-quality are somewhat paradoxical. On one hand, the unit employs unusual and extremely sophisticated circuits. On the other, it doesn't look like a conventional high-end product, lacking the massive power transformers, banks of power-supply regulators, large discrete output stages, and other staples of high-end DACs. The unit features Analog Devices DACs followed by op-amp output stages (the op-amps' markings have been removed). Berkeley says the op-amps were chosen not to save cost or board space, but because they best realized the technical and sonic goals of the Alpha DAC. The unit employs some proprietary technologies and is meticulously hand-tuned: Each unit is aligned, burned-in under load for seven days, aligned again by hand, and listened to before being boxed for shipment.

The Alpha DAC is based largely on the body of knowledge developed by the Pacific Microsonics design team in creating High Definition Compatible Digital (HDCD) and the professional Model Two HDCD encoder/decoder. The Alpha DAC's primary designer is Michael "Pflash" Pflaumer, a Pacific Microsonics co-founder who, among other achievements, wrote the DSP code that realized HDCD encoding and decoding (including the PMD-100 and PDM-200 digital filters). For the Alpha DAC, he wrote a new and advanced digital filter that runs on an Analog Devices SHARC processor. Although the Alpha DAC's components and circuit topology are considerably different from that of the Model Two, this new product reflects the fundamental insights gained during the multi-million-dollar design effort on the Model Two.

LISTENING

Evaluating the Alpha DAC was more complicated than auditioning a conventional D/A converter. Rather than simply connect a CD transport to one of the Alpha DAC's digital inputs, I had the ability to play high-resolution and standard-resolution files courtesy of a PC-based music server described elsewhere in this issue in the article on Reference Recordings' HRx format. My description in that article of the sound of the 176.4kHz/24-bit HRx files is in large part a description of the sound of the Alpha DAC.

Starting with the Alpha DAC decoding standard-resolution (44.1kHz/16-bit) sources from the music server, the Alpha DAC delivered some of the best-sounding CD playback I've heard. First, the Alpha DAC has that rare (and musically important) quality of resolving lots of information without sounding analytical, hyped, or "hi-fi-like." The Alpha DAC presents to the listener a tremendous amount of low-level detail such as delicate spatial cues, the finely filigreed harmonic structure that defines instrumental timbres, and the gossamer-like quality of the very end of reverberation tails. Most digital products truncate this information, or present it as coarse and grainy rather than with a silk-like delicacy. A visual analogy is a pixilated image on a digital TV transmission with poor reception. The lower the signal level, the greater this effect.

The Alpha DAC is highly resolving at all signal levels, but it's this ability to dig down into the lowermost levels that elevates its performance into the top level of digital playback. Just as important, the Alpha DAC doesn't call attention to its resolution; rather, it is suave, understated, and refined. It's the kind of resolution that conjures a vivid impression of the mechanism by which an instrument created a sound, the palpability of tone color, and the precise spatial relationships between instruments within a recorded acoustic. All this information is delivered in a completely natural and unforced way, fostering a tremendous sense of ease, relaxation, and musical involvement.

The Alpha DAC is also capable of huge dynamic contrasts, along with a lightning-fast portrayal of transient information. The

SPECS & PRICING

Berkeley Audio Design Alpha DAC
Input sampling rates: 32kHz–192kHz
Input word length: Up to 24-bit
Digital inputs: AES/EBU on XLR jack, S/PDIF on BNC, optical on TosLink, BADA-encrypted RJ-45
Analog outputs: Balanced on XLR jacks, unbalanced on RCA jacks
Digital filtering: Multiple options
Analog output level: Variable in 0.1dB steps; Channel balance adjustment in 0.05dB steps
Dimensions: 16.5" x 1.75" x 10.4" (19" rack-mount option)
Warranty: Three years parts and labor
Weight: 9 lbs.

ASSOCIATED EQUIPMENT
Basis 2800 Signature turntable with Basis Vector 4 tonearm, Dynavector XV-1S cartridge, Aesthetix Rhea phonostage; PC-based music server (built by Goodwin's High-End), Spectral SDR-4000 Pro CD player, Classé CDP-501 CD/DVD-A player, Sony SCD-9000ES SACD player; Spectral DMC-30SS preamplifier; Spectral DMA-360 and Pass Labs XA100.5 power amplifiers; MIT Oracle MA interconnects; MIT Oracle MA loudspeaker cables; Shunyata Hydra-8, Hydra-2, and V-Ray AC conditioners, Shunyata Anaconda and Python AC cables; Shunyata Dark Field cable elevators; room custom designed and built, acoustic design and computer modeling by Norm Varney of AV Room Service, acoustic treatment and installation by Acoustic Room Systems (now part of CinemaTech)

U.S.

Price: $4995

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EQUIPMENT REVIEW - Berkeley Audio Design Alpha DAC

music swings effortlessly from quiet to full-scale with tremendous speed, but with no sense of etch on the transient leading edges. The rest of my system is particularly adept dynamically (Spectral DMA-360 amplifiers, MIT Oracle MA interconnects and cable, and Wilson X-2 Series 2 loudspeakers) which allowed me to hear the full measure of the Alpha DAC's extraordinary lifelike reproduction of transients, lack of smearing, and ability to present music's dynamics intact.

Perhaps because of this tremendous agility I was more aware of the musicians' rhythmic and dynamic inflections. There was a distinctly greater sense of the music sounding upbeat and lively, of the band "locking into" the groove, and of a heightened physical involvement in the music. The music had a greater coherence, energy, and life that was akin to the difference between a tight band playing on a good night vs. on a great night. This is a DAC that involves your whole body in the music in a visceral and emotional way, not just in an intellectual abstraction of the sound's component parts.

The Alpha DAC's sense of transparency was startling. The presentation had a pristine clarity and vividness that seemed to strip away a fine scrim between the soundstage and me. Although the sound had a sense of precision, definition, and alacrity, it was never cold, sterile, or analytical. I've heard digital products that cover up the digital nasties with a syrupy romantic sound, as well as others that one might admire intellectually for their precision but not enjoy musically. Never before have I heard an outboard DAC that combines musical vividness with such ease, grace, and involvement as the Alpha DAC. Similarly, timbres were simultaneously immediate and palpable, yet

gentle and relaxed. There's long been a conflict in the high-end about "musicality vs. accuracy." I believe this is a false dichotomy; accuracy is musicality when the sound is truly accurate and not merely a hi-fi-like representation of reality.

Finally, the Alpha DAC is simply sensational in its soundstaging. Instruments and voices are localized with scalpel-like precision within a huge and transparent acoustic. In addition to getting right the macro-aspects of depth and width, the Alpha DAC was supremely adept at resolving very fine spatial details that define an instrument's relation to the surrounding acoustic. I heard a wonderful sense of bloom and what Jonathan Valin calls "action"—the impression of air expanding dynamically from an instrument.

The Alpha DAC was so good it invited comparison with the best CD playback I've heard, the Spectral SDR-4000 Pro CD player. This comparison wasn't definitive; playing the same music in the Spectral via its transport mechanism, or through the Alpha DAC sourced from the music server, introduced variables. The Spectral had the advantage of no digital interface in the signal path and thus lower potential jitter. The Alpha DAC, on the other hand, was fed the output of a solid-state memory which, in my experience, produces a better sound than reading data from an optical disc on the fly. In addition, the Alpha DAC and the Spectral both fed a preamplifier via unbalanced outputs, obviating the Alpha DAC's advantageous ability to bypass a preamplifier and drive a power amplifier directly. It was also impossible to compare the two devices on a level playing field when sourced from the same transport mechanism; the Spectral's digital output is coaxial on an RCA jack; the Alpha

DAC's coaxial digital input is on a BNC connector. Although a BNC transmission line is technically superior to one terminated with RCA connectors, introducing an RCA-to-BNC converter would not have provided a fair comparison.

Nonetheless, the juxtaposition was interesting; the two products sounded remarkably similar. The Alpha DAC was a bit more forward and lively through the midband and lower treble, with timbres sounding more "illuminated from within." The SDR-4000 Pro had a slightly more distant spatial perspective and greater midrange liquidity. This character of the Alpha DAC was, however, somewhat dependent on the AES/EBU cable feeding it. The Alpha DAC and Spectral had similar degrees of resolution, although the Spectral had a bit more air and space around instrumental images. The SDR-4000 was also a bit warmer in the midbass, with greater texture.

I describe the Alpha DAC's sound relative to the Spectral not because they compete with each other; they are very different products that serve different needs. Rather, I compare them to illustrate just how close the Alpha DAC comes to the state-of-the-art (at least in my experience) in CD playback, and indeed, to note how similar they sound. Interestingly, they both employ a custom digital filter that incorporates HDCD decoding running on an Analog Devices SHARC DSP.

But listening to the Alpha DAC with only 44.1kHz/16-bit sources is like driving a Ferrari without leaving first gear; it's thrilling, but only hints at the possibilities. When decoding 176.4kHz/24-bit files from the music server, the Alpha DAC is jaw-dropping. The resolution of fine detail I heard from CD was taken to an entirely new level. In fact, I concluded that the Alpha DAC sounds so

detailed with 44.1kHz/16-bit sources because it was designed fundamentally to resolve the extraordinarily low-level detail of high-res sources. When fed Reference Recordings' 176.4kHz/24-bit files from the music server, the Alpha DAC driving the Spectral DMA-360 amplifiers through MIT Oracle MA and Wilson Alexandria X-2 Series 2 loudspeakers produced the most thrilling audio experience of my life. See my accompanying article on Reference Recordings' HRx format for more on the Alpha DAC's sound with high-resolution sources.

CONCLUSION

The Berkeley Audio Design Alpha DAC is the best-sounding outboard digital-to-analog converter I've heard. The fact that it decodes high-resolution sources of any sampling rate and word length is icing on the cake. Although the Alpha DAC is spectacularly great on CD, this converter really shows its prodigious resolution, dynamic agility, and soundstaging capabilities when fed 176.4kHz/24-bit digital audio. Moreover, the Alpha DAC's feature set, operation (instant locking to changing sampling rates, for example), and ability to drive a power amplifier directly expands the product's utility and makes it ideal for the next generation of high-resolution music servers that is just around the corner.

The real story, however, is that this performance and functionality is possible at $4995. Although not inexpensive, the Alpha DAC nonetheless competes with, and outperforms, much more expensive converters. At $5k, the Alpha DAC is a spectacular bargain. It is my outboard converter of choice for both CD and high-resolution sources—regardless of price. tas

Weiss DAC 202 192kHz/24-bit FireWire DAC

The Ultimate Music-Server DAC?

Steven Stone

As high-end consumer audio systems become more like pro studios and higher bit-rate digital files on hard drives become more common, it should come as no surprise that the differences between pro audio gear and consumer audio gear are shrinking. Some companies such as Bryston, EMM, and Manley have been creating components for both markets for years. Another name on this pro/con manufacturers list is Switzerland's Weiss Engineering.

Daniel Weiss founded Weiss Engineering in 1985. Based on his experience working on digital products for Studer Revox, Weiss decided to concentrate on the design and manufacture of equipment for mastering studios. Weiss' first product, the 102 system, is still current with 96/24 capabilities. In the early 90s Weiss Engineering brought out its Gambit Series, which included a stand-alone equalizer, de-noiser, A/D unit, D/A unit, and sampling-frequency converter. Sony, BMG, EMI, Warner, Hit Factory, Abbey Road, Teldec, Telarc, Gateway Mastering (Bob Ludwig), Bernie Grundman Mastering, Masterdisk, and Sterling Sound, all use Weiss units in their mastering suites. In 2001 Weiss introduced its first consumer D/A, the Medea, followed in 2004 by the Jason transport. To support this level of innovation, Weiss Engineering Ltd. employs five full-time employees in its engineering department.

The DAC 202 is the second consumer FireWire DAC from Weiss. The first was the Minerva DAC. The Minerva was a glorious flop—beautifully made, wonderful sounding, it sold like ice water in northern Alaska. The reason for its lack of sales was simple: The Minerva was a Weiss DAC 2 with a different front panel. Everything inside the Minerva was identical to the DAC 2 and the DAC 2 was $2000 less. Anyone who wanted a Minerva bought a DAC 2 and naturally Minerva sales suffered as a result. The DAC 2 is still part of Weiss' professional line-up, while the Minerva is not.

How does the DAC 202 differ from the DAC 2/Minerva? According to Daniel Weiss, "The DAC 202 has a completely different analog section. In addition the DAC 202 has a remote control, headphone output, and a digital word clock input/output. The DAC 2 lacks all of that. But

the DAC 2 and 202 do have some of the same digital parts, including the 32-bit DAC chip." The 202's analog circuit uses an ESS 32-bit 9018 DAC chip configured for two converter channels per one analog audio channel. It's coupled to a minimalistic signal path that uses only a current-to-voltage converter and a balanced driver as active devices.

> **The DAC 202 is the most transparent DAC I've ever heard, and it's not just transparent on high-resolution sources. Even MP3s benefit from its ministrations.**

SETUP

Setting up the Weiss DAC 202 was simple. After downloading and installing Weiss' latest FireWire driver, I connected the Weiss to my Mac Pro via a standard FireWire cable and then connected the DAC 202 to my amplifiers via analog XLR connections. Next I selected the Weiss as my

audio device via Apple's Sound preferences in the system preferences file. Finally I opened iTunes plus either Amarra or Pure Music and began playing any and all of the digital music files in my library. The setup for Windows is a bit more involved, but not demoralizingly so. If you insist on using a Windows machine, the DAC 202 can oblige. If you don't have a FireWire-enabled computer you can still use the DAC 202 via its S/PDIF RCA or AES/EBU digital inputs.

My ergonomic impressions of the DAC 202 were overwhelmingly positive. During its time in casa Stone the 202 has had zero connectivity issues. This was a pleasant change from the Minerva, which I used prior to the DAC 202's arrival. Frequently the Minerva would mysteriously become unselected as an output device and silence would ensue. Correcting the problem required resetting the computer's audio device preferences and disconnecting and reconnecting the FireWire cables. But the DAC 202 exhibited none of this squirreliness. Amarra, Pure Music, and, most importantly, Mac's audio MIDI set-

EQUIPMENT REVIEW - Weiss DAC 202 192kHz/24-bit FireWire DAC

up control panel immediately recognized and supported the 202. Unlike many so-called plug-and-play devices, the DAC 202 really does work correctly the first time and every subsequent time you use it.

Controls on the front panel are as minimalist as you can get. There's only one knob, an LCD display, and a headphone jack. If you push the knob in it switches from user mode, where it serves as a volume control, to menu mode, which gives you access to the 202's settings and adjustments. The 202 also sports an IR remote control, which is far easier to use than the front panel knob. More than occasionally when I tried to adjust the DAC 202's volume with its front panel volume control nothing happened. That was because the 202 had slipped into menu mode instead of volume mode.

Employing the remote solves this user-error problem. The remote includes controls for volume, input source, and mute. It also gives you the ability to change absolute polarity and switch from digital filter A to B. Depending on the source, switching absolute phase can have an audible affect. Some audiophiles claim that over 10% of all commercial recordings sound better when their absolute polarity is inverted. The digital filter is an upsampling filter. According to Weiss, "Filter A has a steeper frequency response than B. Future DAC 202 software will offer even more filter choices." The audibility of your filter selection will depend on the source as well as the other components in your reproduction chain.

The DAC 202 is chock full of proprietary digital technology, which you can learn about through several white papers on the Weiss Web site. Rather than parrot Weiss' excellent technical description of its Jitter Elimination Technologies (JET) Phase Lock Loop (PLL) jitter-reduction system, I'll direct you to its site, which also has the DAC 202 instruction book in PDF form, the latest FireWire drivers, firmware, and test files for determining if your computer is producing bit-perfect output.

Is the DAC 202 musical? My answer is a resounding yes! It never got fatiguing. Even after days of marathon listening, the rudest takes were still listenable.

On the analog side Weiss has come up with a clever way to optimize its digital volume control. The DAC 202 has four "coarse" volume levels that are set via relays. Fine level adjustments are done with a rotary encoder in the digital domain. This design reduces the operating range of digital volume control so it never has to do very much attenuation, and therefore remains within its optimum operational parameters for lowest distortion and resolution loss.

THE SOUND OF THE RECORDING STUDIO

For you readers who've skipped the first half of the review to jump to "the good stuff," I will give you this incentive to read further—the DAC 202 is the most transparent DAC I've ever reviewed. And it's not merely transparent on high-resolution sources. Even MP3s benefit from its ministrations. Here's an example: I tuned into the Internet radio station Folk Alley and it was playing Mary Chapin Carpenter's "Twilight" at 128k bits per second. I have this song in my iTunes library in 320kbps MP3. When I went from the Folk Alley stream

to my own file it was immediately obvious how harmonically threadbare the 128bps stream was compared to a 320bps MP3. Also the 320kbps MP3 version had substantially more inner detail and low-level information. But even the 128kbps stream was more than merely listenable through the DAC 202, although its limited resolution and harmonic shortcomings were obvious.

Although the DAC 202 has that "pro sound" (i.e. low distortion and coloration), it is not dry or sterile. In the past many audiophiles avoided pro gear because they felt that it lacked the bloom and musicality of top-shelf consumer gear. I didn't notice any of these sonic phenomena through the DAC 202. Instead, it was a nearly perfect window opening onto the "house sound" of whatever I was listening to. Here's a good example of the DAC 202's transparency and fidelity to original sources; on Nickel Creek's fine album *Why Should the Fire Die*, the tune "Somebody More Like You" uses a combination of well-recorded acoustic instruments and studio effects. The separation between the acoustic instruments and the studio effects, such as the out-of-phase synthesizer wash, was immediately obvious through the DAC 202. Also the differences in soundstage size between the acoustic instruments' environment and the larger artificial space where electric instruments reside was more apparent through the DAC 202 than through any other DAC I've auditioned.

But is the DAC 202 musical? My answer is a resounding yes! One sign of the DAC 202's musicality is that it never got fatiguing, even after days of marathon listening. I've been recording my bluegrass band's live gigs lately using a Zoom Q-3 in 96/24 mode and editing with Audacity

(which had no problems recognizing and interfacing with the DAC 202 at higher bit–rates), and some of these recordings aren't exactly pretty. The DAC 202 made me aware of just how unpretty they were, but I never felt like my ears were getting shredded. Even the rudest takes were still listenable.

Whenever I receive a USB or FireWire DAC for review I invariably run "the Amarra test" on it. The test is simple: Turn on the iTunes software, plug-in Amarra, and see if the sound improves due to

SPECS & PRICING

Weiss DAC 202 192kHz/24-bit FireWire DAC
Digital inputs: XLR, RCA, TosLink (optical), two FireWire.
Supported sampling frequencies: 44.1kHz, 48kHz, 88.2kHz, 96kHz, 176.4kHz or 192kHz on any of the inputs, except TosLink
Digital outputs: XLR, RCA, two FireWire
Analog outputs: Balanced on XLR, unbalanced on RCA
Dimensions: 8" x 3" x 12"
Weight: 7 lbs.

U.S.
Price: $6670

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EQUIPMENT REVIEW - Weiss DAC 202 192kHz/24-bit FireWire DAC

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Amarra's bypassing of iTunes audio processing. I also performed the "Pure Music test" using the Pure Music software package which implements a similar bypass of iTunes and Apple MIDI control audio functions. The DAC 202 revealed subtle but pervasive sonic improvements with both of these software packages. Dimensional solidity, inner detail, and overall musicality all moved up a notch. For all my critical iTunes listening through the DAC 202 I used one of these iTunes augmentations.

Every summer I try to attend the Rockygrass Academy in Lyons, Colorado. For the last couple of years I've been permitted to make recordings from the mixing board during the morning mini-concerts. I used my Korg MR-1000 to record in DSD format and then use Audiogate software to make 96/24 WAV files. Last year I had an opportunity to record Darrell Scott's morning set. The results were outstanding. His cover of Joni Mitchell's "Urge For Going" was a sonic knockout, even if it was a mono mix. But even in mono the three-dimensionality of each instrument and the separation between instruments were astounding. Sure, they were all piled up in the middle of the soundstage, but even with this center-stage monophonic traffic jam each instrument occupied its own unique spot in the mix, both spatially and harmonically. Bryn Bright's acoustic bass came rumbling up from the back of the soundstage, like a caboose at the back of a big ol' musical train.

Because of the lack of extraneous mechanical and electronic noise, even the spaces between instruments seemed better defined through the DAC 202 than with other DACs I've used. This lack of noise and grain also heightens dynamic contrasts, especially in the micro-dynamic domain. Subtle dynamic differences that were obscured in DACs with noisier analog circuits came through with far greater clarity because of the DAC 202.

Macro-dynamics were also outstanding through the DAC 202. My Darrell Scott recording was done with no limiters or compressors, and the dynamic contrasts are scary. When the entire band leans into the final chorus the DAC 202 still managed to preserve the not-so-subtle dynamic variations from forte to triple forte without getting harder sounding or losing any of its finesse.

If you want to personally hear what the creators of some of your favorite albums hear, I can think of no component that will get you closer to a mastering suite.

And how did the DAC 202 compare with its predecessor, the short-lived Minerva? The DAC 202 had less electronic grain so it sounded less mechanical. The DAC 202 was also less fatiguing after all-day nearfield listening sessions. Both DACs dredged up equal amounts of information and detail and as sonic tools they were both superb. As I mentioned earlier in this article the DAC 202 has a completely different analog section, and the sonic differences I heard between the DAC 202 and the Minerva were a result of the DAC 202's analog circuitry—surprise, surprise.

Since I live to do comparisons (and you love to read them), I listened to the Empirical Audio Off-Ramp 3 USB DAC connected via its S/PDIF coaxial output to the DAC 202. On 44.1 material played through iTunes without using Amarra or Pure Music, I could not hear any noticeable differences between the Off Ramp 3 using USB and the DAC 202 using FireWire. I wanted to try this test using either Amarra or Pure Music, since they both offer a higher level audio quality than iTunes alone, but they "capture" the audio stream and prevent you from changing the sound device in Apple's Sound Preferences control panel while they are running. This makes A/B testing far less do-able. Moving on to higher bit-rate 96/24 files played through Audacity, I couldn't discern enough differences between the Off-Ramp and the DAC 202 to have a preference. On some material I felt the Off-Ramp's presentation was slightly softer and more euphonic, but on most commercial recordings I really couldn't reliably tell which unit was playing in A/B tests.

I would be remiss if I didn't mention the DAC 202's excellent headphone output. The DAC 202 has its own dedicated amplifier, separate from the 202's main outputs. It sounds just like the DAC 202's line-level outputs—clean, powerful, solid, and still highly musical. I used it with both Grado RS-1 and Sennheiser HD-580 headphones. In both cases the sonic personalities of the headphones, not the headphone amplifier, were the most prominent components in the signal chain. With both headphones the DAC 202's headphone amp supplied ample power, dynamic drive, and control. The bass was well articulated through both headphones. And while a headphone can never deliver the visceral qualities of an in-room transducer, especially on lower frequencies, the DAC 202s headphone amplifier lets you know the bass is very much alive and kickin'.

MAYBE IT'S TIME TO TURN PRO

It's difficult to remain complacent when you review a component that outperforms anything you've reviewed in the past. The natural tendency is to go into rave mode and turn the review into fan-boy screed. I have gone to considerable lengths to avoid this, even though the DAC 202 is good enough to cause a meltdown of any reviewer's critical facilities. As to whether the DAC 202 performs at a level equal to Weiss' $19,000 Medea or other manufacturers' mega-buck DACs, I don't know. I suspect at worst the DAC 202 will come very close, especially via its FireWire inputs.

But if you want to personally experience what the creators of some of your favorite albums hear, I can think of no component that will get you closer to a mastering suite than the Weiss DAC 202. If your computer has FireWire capabilities and you want a reliable, ergonomically elegant, and superb-sounding DAC, the Weiss DAC 202 is here, waiting for you. tas

dCS Debussy DAC

A Budget dCS?

Alan Taffel

I n 2003, Bentley Motors shocked the automotive world by introducing the Continental GT. Until then, acquiring one of Bentley's bespoke motor-cars would set the purchaser back several hundred thousand dollars. But the Continental GT cut in half the price of entry into those rarified ranks. Initial skepticism was short-lived; one look at the car's sumptuous interior and prospec-tive buyers knew they were in the presence of a true Bentley. The GT became an instant success, neatly illustrating that value has appeal even at extravagant price points.

Now comes the Debussy DAC from dCS, the digital specialist whose products are every bit as top-shelf—and pricey—as those of its fellow Brits at Bentley. Prior to the Debussy's arrival, owning an "entry-level" dCS DAC was an $18,000 to $24,000 proposition. The new model, which carries an $11,000 price tag, aims to be dCS' Continental GT.

Of course, while eleven grand may break new ground for dCS, it's still a lot of dough for a DAC. Still, no realistic consumer would expect dCS gear (or Bentleys) ever to be downright cheap. The real question is whether the Debussy measures up to dCS standards and, in so doing, confers true value to its buyers. To find out, we need to pop the hood.

There is a reason dCS products are expensive.

While a Bentley is characterized by luxurious, hand-crafted materials, dCS gear is crammed full of costly DSP chips, hybrid power supplies, discrete clocks, and gate arrays. Custom software harnesses this powerful coterie to execute a complex digital dance that comprises multiple stages of synchronous upsampling, conversion to dCS' proprietary 5-bit format, and finally processing by the firm's lauded Ring DAC, with a choice of in-house programmed filters on the side.

One might think a "budget" dCS would employ fewer custom parts and more off-the-shelf hardware and software. But that approach would not result in a true dCS, any more than badge-engineering a VW would yield a Bentley. Indeed, the more one examines the Debussy's innards, the

more one comes away wondering how it can be sold at its price. Consider: The Debussy includes the *exact same* control board—responsible for everything from power delivery, I/O, upsampling, clocking, and format conversion—as the universally acclaimed Paganini and Scarlatti models. Its Ring DAC and fully balanced Class A output stage, both discrete modules, are *identical* to those in the Debussy's expensive siblings, as are the software-based digital filters.

Nor did dCS skimp on features. Again, one might reasonably expect a loss of connection flexibility, but the Debussy will handle an RCA, BNC, and a pair of AES sources. (As usual, the BNC input was the best sounding.) Further, there are both single-ended and balanced outputs. Other welcome ingredients include a front-panel sample-rate display, the ability to accommodate an external word clock, and the world's heaviest remote control.

All this would be sufficient to justify excitement about the Debussy's debut, but dCS ups the value equation considerably with the inclusion of a USB port. An easy, inexpensive add-on, you say? Not in this case. Elsewhere in the dCS line, adding a USB interface requires an entirely separate (and dearly priced) box (the $4999 U-Clock). The Debussy is the company's first and only product to incorporate both the Ring DAC and USB in one box. Furthermore, this is no ordinary USB interface. It's good up to 96/24, and is of the asynchronous variety, which allows the DAC rather than the PC to control timing, resulting in far less jitter. This port is the icing on the Debussy cake.

So where, exactly, does the dCS newcomer cut corners? Mostly in areas that do not affect the sound. Much was saved, I am told, through the exclusion of an alphanumeric front-panel display. In its place are a series of LEDs that indicate input and filter selection, sampling rate, and volume when the Debussy is used as a linestage (more on this later). Personally, I found the LED array perfectly serviceable. Another limitation:

EQUIPMENT REVIEW - dCS Debussy DAC

The Debussy has only two filter options, whereas higher models offer more. This is another non-issue as far as I'm concerned; I was perfectly content with the recently released apodising filter, which I found significantly more natural than the standard version. In any case, future filters that dCS deems superior will be downloadable. One final omission was upsampling to DSD—a feature that impressed me mightily at the 2010 CES. This particular exclusion did disappoint me, and I'm still lobbying dCS to find a way to include it in the Debussy. Call me greedy.

Ultimately, of course, a car is judged by a test drive and an audio component is judged by listening (unless you're Julian Hirsch). So, does the Debussy's advanced and costly componentry deliver the sonic goods? I won't mince words: The Debussy is flat out fabulous. Let me count the ways.

This is not the first DAC I have reviewed that employs an elaborate processing scheme. For the most part, I have been unhappy with such units. The behind-the-scenes frenzy of digital calculating seems to find its way into the listening experience. That is, such DACs *sound* like they're working hard, which prevents the listener from relaxing into the music. DCS knows of and is careful to avoid this phenomenon. The company uses more processors (oops, there goes the cost curve) so that each is taxed less. This explains, for example, why upsampling is performed in stages rather than all at once. Perhaps it also accounts for the fact that, despite all the binary manipulations taking place, the Debussy sounds unfailingly natural. Music winds out of this DAC like thread from a spool.

> **The Debussy not only unearths every musical line; it also makes following those lines concurrently an effortless task. It simply delivers more musical details.**

In virtually every way, the Debussy sounds terrific. Dynamics, depth, and detail are present in copious quantities. If the music so beckons, this DAC's tone is as ravishing as long, lustrous hair. Indeed, early in my time with the Debussy I identified a certain pervasive smoothness that was fine for some source material, but inconsistent with the more ragged elements of my CD collection (e.g. the MFSL remaster of the Pixies' wonderfully raw *Doolittle*). However, inserting some good cones underneath the chassis completely extinguished this minor coloration. So I settled down to some serious listening.

"On the Beach at Night Alone," from Ralph Vaughn Williams' choral *Sea Symphony* (Telarc) is an enthralling piece that whisks the listener on a journey from the solitude of one man's thoughts to the "vast similitude" of the cosmos. I listened to this piece through the Debussy feeling that it could hardly be more engrossing. The DAC's sheer quantity of spatial, dynamic, and musical information added up to an all-encompassing, emotionally shattering experience.

By comparison, my Bryston BDA-1 reference DAC (a Golden Ear recipient, mind you) seemed restrained. The chorus lacked high-end extension, sounding almost muted. Dynamics were still excellent, yet the Bryston could not achieve the cataclysmic climax delivered by the dCS. Bass was nowhere near as thunderous. Finally, the Bryston buried some instruments in the mix, whereas the Debussy would never subject any player to such ignomiy. In this respect, having a Debussy is like buying a new analog front end; in both cases you get to discover previously buried treasure within familiar tracks.

On smaller scale works, like the Stravinsky Suite from *l'Histoire du Soldat* (Pentatone), the gap between the dCS and the Bryston narrowed. Neither held an advantage with respect to pacing, and instrumental timbres were virtually indistinguishable. Still, the Debussy delivers more detail—you can hear the hall reverb far more clearly right from the first note—and a deeper stage. And the Debussy not only unearths every musical line; it makes following all those lines concurrently an effortless task. Ivor Tiefenbrun would be well pleased with this DAC.

SPECS & PRICING

dCS Debussy DAC
Inputs: Digital USB, RCA, AES/EBU (2), BNC, WORDCLOCK
Outputs: Analog balanced XLR and single-ended RCA, 2V or 6V user-selectable
Maximum resolution: 192/24 (S/PDIF), 96/24 (USB)
Dimensions: 17.6" x 2.6" x 15.5"
Weight: 1.4 lbs.

U.S.
Price: $10,999

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EQUIPMENT REVIEW - dCS Debussy DAC

Getting the Debussy's USB Right

No matter what anyone—or any manual—tells you, USB is not plug-and-play. Not if you want to get the best sound from this interface. Overcoming your PC operating system's inherent limitations is the first challenge. If you are running Windows XP and follow the Debussy manual's instructions, for instance, you will likely end up (knowingly or not) invoking Windows' Direct Sound, which means bits will pass through the dreaded kernel mixer. Following the manual to the letter, I achieved what I have come to view as typical USB sound: smeared rhythms, closed extension, and screechy strings. In short, yuck.

Regardless of your Windows OS, what you want to do is bypass all its junk by using the far superior, professional standard ASIO driver set. (Another option is the recent WASAPI, but I did not have time to experiment with it.) Most music-playing software packages, like Media Monkey, support ASIO. The problem is that the Debussy doesn't. However, a nifty, freely downloadable package called ASIO4ALL solves the problem. Not only are these drivers bit perfect, they dynamically adapt to the

source material's sample rate. This is an important provision, because standard PC (and Mac) drivers asynchronously upsample data to the highest supported sample rate—a sonically injurious process. ASIO4ALL will work with virtually any playback software except iTunes. Once you've heard the way ASIO restores USB's air and dynamics, you will never go back.

After sorting out software, there is still the matter of cables. As I have stressed before, *USB cables make a demonstrable difference*. For my tests with the Debussy, I experimented with five of them, ranging from the Brand X variety that comes with printers to audiophile models. The winner this round—just as in the last time I conducted a USB cable survey—was the unpretentious Belkin Gold Series. The difference this cable makes is not remotely subtle. Depending on what you are comparing it to, it can be the difference between music and wallpaper. Here, blessedly, is one area of the high end that does not require spending a fortune; the Belkin costs under five dollars.

I should reiterate here that the Bryston to which I compared the Debussy is an excellent DAC, and at $1999 represents a different sort of value. The above contrasts are not intended to denigrate the BDA-1, but to convey just how good the Debussy is compared to "run of the mill" reference gear. This point hit home even harder when I played the SACD layer of the same Stravinsky disc through the estimable Marantz UD9004 universal player. In the past, even the best CD rigs have paled beside the Marantz's way with an SACD. Not this time. Here, it was the SACD, higher resolution notwithstanding, that sounded literally pale by comparison to the Debussy's Red Book presentation. That's how good the Debussy sounds—it can upend the normal CD/SACD pecking order.

One area where I find the Debussy ever-so-slightly wanting is rhythm. It's not that the DAC lacks a sense of pace—far from it. However, I have heard other digital pieces, such as the Burmester 089 CD player I'm reviewing (good gosh, it's even more expensive!) whose rhythms are more concrete. I doubt this deficit would even be noticeable without a direct comparison to the rare product that is superior in this category. As such, rhythm is not so much a Debussy weakness; rather, it simply is not among its lengthy list of strengths. I suspect the addition of the external Puccini clock—a nice down-the-road upgrade for Debussy owners—might improve things in this area.

That leaves me with one last strength and one real weakness to report. Let's get the bad news out of the way first. Although the Debussy is fully equipped to directly drive an amplifier, I found it quite unimpressive in this capacity. As a linestage, the dCS sounds rather boring, with compressed dynamics and undifferentiated timbres. Only its vocal purity, quiet background, and solid imaging elevate it above mid-fi. I tried the Debussy in this mode with two different amps, with the same

results. So, although being able to use the Debussy as a linestage would up the value quotient even further, I must instead recommend using this DAC as a DAC, sending its analog outputs through a good dedicated linestage. In my experience, that is the only way to hear what the Debussy can really do.

And the last thing to report that this DAC does is deliver the best USB sound I have ever heard. Setup must be done with care (see sidebar), but the results can be extraordinary. This is the first USB I have listened to—and longtime readers know that I've heard and dismissed quite a few—that is truly in the same sonic territory as (if not quite the equal of) the best S/PDIF. With my trusty Dvorak *Serenades* (Praga), the Debussy's BNC input was just slightly less grainy and more dynamic than USB. Mary Guathier's "Falling Out of Love" from *Mercy Now* was again awfully close. In USB mode, Mary's voice moved forward and exhibited some sibilance plus, once more, a hint of grain.

With higher-resolution source material, the gap widened—in S/PDIF's favor. The Classic Records 96/24 recording of Ravel's *Pavanne pour une infant defunte* was clearly less timbrally pure when played via USB. Too, the Debussy's USB does not sidestep the lax rhythms I have noted in every other USB DAC. On the other hand, strings, which are usually USB's bête noir, here had nary a trace of the shrillness that would normally afflict them. Overall, the Debussy's BNC input is its most convincing; however, the USB port is very nearly its musical and sonic equal—and that's saying a lot.

I have no idea if the dCS Debussy will meet with the same success Bentley's Continental GT has enjoyed. What I do know is that it delivers everything its builders intended in terms of performance, usability, and yes, value. Is it a "true" dCS? You don't even need to look under the hood to find out. Just listen. tas

EQUIPMENT REVIEWS

Music Servers & Accessories

Berkeley Audio Design Alpha USB Interface

The USB Problem Solved At Last

Robert Harley

The high-end audio industry has a remarkable track record of making fundamentally limited technologies sound good. Most of these technologies were created for mass-market consumption where low price is the overriding design mandate. But because these technologies have become the standard, the high-end industry has had no choice but to attempt to create silk purses out of sows' ears.

Here are just a few examples: the CD format (its creators would be astounded at the advances in just, say, digital filtering); the RCA plug and jack (compare the connections on 1970s equipment to today's Cardas or WBT RCAs), and the Compact Cassette (Nakamichi 1000, anyone?).

A more recent, and perfect, example is the Universal Serial Bus, or USB interface. Designed to connect computer peripherals, USB was never intended to be a high-resolution digital-audio interface. But the rapid growth of computer-based music systems has, once again, foisted upon the high end a standard that requires exceptional re-engineering to meet the demands of high-quality music reproduction. Because of the contributions of high-end designers, today's best USB interfaces are light years beyond the basic implementations.

In this overall drive toward a good-sounding USB interface, one company stands out for pushing the envelope—Berkeley Audio Design. The company that brought us the amazing Alpha DAC has turned its considerable engineering chops and uncompromising work ethic toward solving the USB interface problem. The Alpha USB reviewed here—only the second product from Berkeley—has been nearly two years in the making, largely, I surmise, because Berkeley is run by engineering-driven perfectionists who kept discovering during the design process better and better techniques and implementations. Berkeley is the kind of company that would repeatedly delay a product launch until it had wrung out every last bit of performance.

WHY DO WE NEED A USB CONVERTER?

Before looking at the Alpha USB in detail and considering its sound quality, let's review the options for getting music out of a computer-based server and into a digital-to-analog converter (DAC). For now, we'll ignore the turnkey systems such as Sooloos and the Olive O6HD (reviewed in this issue) to focus on do-it-yourself servers based on a personal computer.

The first option is to install a soundcard that has built-in DACs in your PC. You simply connect the soundcard's analog outputs to your preamplifier and you're in business. The compromises of this approach are fairly obvious—the inside of a computer is not the best place to perform digital-to-analog conversion. The second option is to use a soundcard's S/PDIF or AES/EBU digital output for connection to an external DAC. In our previous issue (May/June) Karl Schuster surveyed and reported on eight such sound cards. This approach requires opening your PC to install the card, and configuring software. Moreover, building a PC server that is "bit transparent" (one that doesn't change the ones and zero representing the music) is easier said than done.

The third method, which is by far the most popular, is to simply run a USB cable from the computer to a DAC equipped with a USB input. Although simple in practice, the USB interface audibly degrades the signal passing through it, even in the better implementations. As noted, USB was never designed for audio; it is a "packetized data" format in which data are split up into discrete chunks, wrapped up with information about those chunks, transmitted, and then put back together at the receiving end. This is in sharp contrast with the continuous bitstream of digital audio formats such as S/PDIF. Moreover, until recently USB has been limited to a maximum sampling frequency of 96kHz. And let's not forget that many of us have older DACs that still sound good but lack a USB input. It is for these reasons that my own server, which I use exclusively to play high-resolution music, is a PC fitted with a Lynx AES16 card that outputs its digital signal as AES/EBU on an XLR plug. (I also use a Meridian Sooloos to access my CD music library.)

The solution to these myriad problems is an outboard box that takes USB from the computer and outputs S/PDIF—if this can be accomplished

EQUIPMENT REVIEW - Berkeley Audio Design Alpha USB Interface

without compromising sound quality. Although USB converters are widely available, none could be considered an all-out assault on the state-of-the-art. Rather, they are largely utilitarian in purpose.

Which is where the Berkeley Alpha USB comes in. Berkeley's goal with the Alpha USB was not just to create the best-sounding USB interface, but to completely eliminate the problems of USB and build a state-of-the-art solution for getting music out of a computer and into a DAC. Concomitantly, Berkeley wanted to create a product that allowed anyone, not just those with technical expertise, to realize state-of-the-art computer-based audio performance. When Berkeley's Michael Ritter told me about the Alpha USB, he invited me to compare its sound with that of the AES/EBU output of the Lynx AES16 card in my fan-less, drive-less PC server, a setup that many considered the state-of-the-art in computer audio (see my review of this system in Issue 189). The PC with the Lynx card starts off with the considerable advantage of never converting the audio data to the USB format.

If the Alpha USB did sound better than my PC (with both driving the same DAC), it would not only represent a breakthrough in sound quality, but make it much easier for non-geeky music lovers to enjoy the benefits of computer-sourced audio.

FUNCTIONAL AND TECHNICAL DESCRIPTION

The Alpha USB is housed in an unusual chassis that's significantly larger than that of most USB converters, but smaller than full-sized components. As with the Alpha DAC, the Alpha USB features a no-cosmetic-frills chassis. The front panel's only indicator is a single LED that lights up amber when powered on, and then switches to green when connected to an active USB source (the computer). The rear panel is also minimalist, with AES/EBU and S/PDIF outputs (the latter on a BNC jack) selectable via a small toggle switch. The Alpha USB lacks an RCA output because the RCA's characteristic impedance is 50 ohms, not the 75-ohm standard for S/PDIF. A BNC interface, which has a characteristic impedance of 75

ohms, is vastly superior to RCA for carrying digital audio. AC power is via an IEC jack. There is no power switch; the unit, which draws 4.5W in standby mode, is designed to be left continuously powered. Macintosh computers running Snow Leopard or later will automatically talk to the Alpha USB. Windows users will need to install driver software, which is included on a CD.

The Alpha USB is the epitome of "form follows function." The unusual chassis size and shape were chosen for sonic performance. In broad terms, the Alpha USB isolates the "dirty" USB circuitry from the "clean" digital-audio output, and provides a high-precision clock for that audio output. The USB input jack is mounted on a plastic insert rather than directly in the chassis to prevent capacitive coupling of noise between the USB input and the digital-audio output. This isolation between the two "halves" of the Alpha USB is improved by powering the circuitry associated with the USB input from the computer via the USB bus, and powering the clocks and digital-audio output drivers from a clean, linear power supply built into the Alpha USB. Berkeley extends this isolation concept by recommending that the computer be powered from one AC outlet and the Alpha USB, DAC and analog components from another, and that the USB cable between the computer and Alpha USB be routed away from the Alpha USB's chassis and any analog cables or components. It goes without saying that the Alpha USB's USB interface is "asynchronous," meaning that the Alpha USB's output clock is not locked to the computer's clock. That is, the timing precision of the Alpha's S/PDIF output

is not affected by the computer. A conventional "adaptive" USB interface, in which the computer serves as the master clock, is simply a non-starter for critical applications.

The digital audio signal is clocked with one of two precision oscillators, one for the 44.1kHz family of sampling frequencies (44.1kHz, 88.2kHz, and 176.4kHz)

SPECS & PRICING

Berkeley Audio Design Alpha USB Interface
Type: USB-to-S/PDIF converter
Input: High-speed USB 2.0, Type B jack
Output: Switch selectable–coaxial SPDIF on BNC, balanced AES on XLR
Supported sampling rates: 44.1kHz, 48kHz, 88.2kHz, 96kHz, 176.4kHz,
192kHz
Supported word lengths: Up to 24 bit
Supported operating systems: Apple Macintosh and Microsoft Windows
Mains power: 100 or 120 or 240VAC, 50/60Hz, IEC power input connector
Power consumption: 3 Watts line, 1.5 Watts USB, designed for continuous operation
Dimensions: 10.5" x 2.5" x 5" (including feet)
Weight: 2.5 lbs.

U.S.

Price: $1695

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

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EQUIPMENT REVIEW - Berkeley Audio Design Alpha USB Interface

and the other for the 48kHz family (48kHz, 96kHz, 192kHz). To give you an idea of how meticulously designed and executed the Alpha USB is, consider that each oscillator is measured on a $90,000 instrument that creates a spectragraph of the oscillator's phase noise. Only the best measuring parts go into the Alpha USB; the rest are rejected. This measurement process costs many times what the actual part costs, but Berkeley discovered that this performance parameter was crucial to sound quality.

The Alpha USB's designer, Michael "Pflash" Pflaumer, comes from a multidisciplinary background that includes writing the DSP code that made HDCD possible, analog design, and RF design. When the first microprocessor became available, Pflash wrote his own operating system for it and built a computer around it. It's rare for an engineer who can write DSP code to also have an intimate understanding of how electromagnetic fields behave. It's even rarer when that engineer is also a musically sensitive listener who uses his ears to guide product development.

SYSTEM CONTEXT AND SET-UP NOTES

Berkeley sent to me a bare-bones iMac ($1199) loaded with music, including most of the Reference Recordings HRx 176.4kHz/24-bit titles (which I also have on my PC server). They also shipped a Straightwire Info-Link AES/EBU cable and a Straightwire USB-Link USB cable (both 1.5 meters). The iMac ran iTunes along with the latest version of Pure Music, a $129 piece of software that improves sound quality (see Steven Stone's review of Pure Music in Issue 209). Berkeley recommends Pure Music, and includes with the Alpha USB a $25 discount coupon for

the software. The complete digital front end of an iMac, Alpha USB, Alpha DAC, Pure Music, Straightwire USB and Straightwire AES/EBU costs about $8300.

> **The Berkeley Alpha is a breakthrough that overcomes the limitations of USB and also provides a way of getting state-of-the-art sound from a computer.**

The iMac was unbelievably easy to set up and use; my Windows PC was a different story. To use the Alpha USB with a Windows machine, you must install the supplied driver. This will let the PC and Alpha USB talk to each other but will not allow you to realize the system's full sonic potential. You must also install an ASIO driver to avoid data corruption. For Windows XP users, the available ASIO drivers will degrade the sound. Machines running Windows 7 can use a more sophisticated WASAPI driver that reportedly delivers performance as good as that possible from a Macintosh. Installing the ASIO driver on my XP machine was a hassle. Moreover, switching between the Lynx card and the USB output required going into a couple of menu layers in MediaMonkey to manually change a setting.

Considering my experience with both an iMac and a Windows PC, I can unequivocally say that the Macintosh is vastly superior (and I've used PCs for all other computing since the late 1990s). The Mac is far more elegant, easier to set up, better sounding, and doesn't require that you jump through hoops to realize its optimum performance. Even if you own a PC that you are thinking of using as a server, I encourage you to

spend the $1199 for an iMac—you'll be glad you did.

LISTENING

I began by comparing the sound of the iMac/Alpha USB to my PC-based server equipped with the Lynx AES16 card. The bitstreams from each computer driving the Berkeley Alpha DAC were identical. How do I know this? Every HDCD recording carries a flag in the least significant bit that identifies the recording as HDCD-encoded. An LED on the Berkeley Alpha DAC's front-panel illuminates when this flag is detected. If the data were corrupted, that LED would not illuminate. In all my tests, the HDCD LED remained lit when playing any Reference Recordings title (CD or high-res file). It follows that the bitstreams were identical for non-HDCD recordings, as well.

Listening to 176.4kHz/24-bit Reference Recordings HRx files through the Constellation Audio electronics ($65k Altair preamplifier and $140k Hercules power amplifiers) driving Sonus faber's $200,000 flagship loudspeaker is as critical a situation as one could devise. These exquisite recordings contain so much fine information, dense spatial cues, micro-transient detail, and rich timbral colors that any degradation is instantly identifiable. The playback system is truly of reference quality.

Comparing the PC-based server with the Lynx card to the iMac and Alpha USB (by changing which AES/EBU cable was connected to the Alpha DAC) revealed that the PC setup I'd thought was the state-of-the-art was actually a step down from what was possible. Simply put, the Alpha USB took the system to another level of resolution and musicality.

Although the PC server sounds phenomenally great, the iMac/Alpha USB was better in virtually every sonic criterion and inferior in none. First was the sense of space and the naturalness of the staging. The iMac/Alpha USB had a slightly less forward perspective along with much greater soundstage depth. Although the front of the stage was a little set back, instruments in the back of the hall sounded much more distant. The sense of bloom around individual instrumental outlines was more realistic and palpable. The Alpha USB made the PC/Lynx sound slightly congested and homogenized by comparison.

Within this sense of expansive space, I could more easily hear fine recorded detail, particularly instruments toward the back of the hall. Switching over to the Alpha USB system was like sharpening the focus on a camera; very-low-level detail that had been just a bit indistinct or smeared snapped into vivid clarity. I thought I had heard the HRx recordings in their full glory with my PC server, but I was astonished to discover another level of resolution and clarity.

The treble through the Alpha USB was smoother and, paradoxically, slightly more prominent. The presentation wasn't brighter, just more alive and vivid. The treble had greater texture and increased density of information, yet was more finely filigreed and delicate. The top octaves also had greater smoothness and ease, particularly on high-level, high-frequency transients such as the upper octaves of fff piano passages.

The sense of hearing more information was partially the result of the Alpha USB's superior rendering of transient detail. Listen to a Latin percussion instrument such as the güiro; the Alpha USB better resolved the instrument's

EQUIPMENT REVIEW - Berkeley Audio Design Alpha USB Interface

dynamic envelope to create a greater impression of hearing the instrument itself rather than a re-creation of it. Percussion seemed to "pop" from the soundstage with greater life. Micro-transients were also noticeably superior; listen to brushes on cymbals, to triangles, and to tambourine. By better resolving this low-level transient detail, the Alpha USB made the presentation more lifelike and musically vivid. I could better hear the mechanisms by which the sounds were created, which is always a good sign.

Finally, the Alpha USB had a greater sense of ease on high-level peaks, particularly during dense and complex passages. The music got louder more gracefully, with smoother textures and less homogenization of images. The presentation remained more coherent during the loudest orchestral passages, contributing to the overall sense of ease and involvement I experienced.

Because the Alpha DAC was receiving the same bitstream from both music servers, the only difference was in the timing precision—jitter.

Some of these differences could be attributable to the different computer platforms, so I compared the sound of the Lynx card in my PC to the PC's USB output through the Alpha USB. The Alpha USB sounded better overall, although the disparity was not as great as when comparing the PC/Lynx to the iMac/Alpha USB. The specific sonic characteristics were the same, but the magnitude of the difference was reduced, suggesting that the Macintosh platform has a sonic advantage over the PC. That difference might be erased by a PC running Windows 7 and a better WASAPI driver, but I was unable to

hear that configuration. Even if a Windows 7 PC can sound as good as the Macintosh, the Apple platform is much more pleasant to use.

CONCLUSION

The Berkeley Audio Design Alpha USB is a breakthrough product that not only overcomes the limitations of the USB interface, but provides a state-of-the-art method of getting audio out of a computer. Moreover, the Alpha USB makes this reference-quality performance available to non-technical music lovers who have a Macintosh and a DAC.

Though the Alpha USB's $1695 price is considerably more than that of other USB converters, the Alpha is a bargain when you consider that it provides a simple, foolproof path for creating a state-of-the-art music server. Moreover, the entire digital front end of the iMac, Alpha USB, Alpha DAC, Pure Music software, and Straightwire digital interconnects costs about $8500. That's not chump change by any measure, but it's eminently reasonable for a music server and a DAC that deliver this level of performance. I listened to this digital front end as a source for electronics and loudspeakers that together cost more than $400k, yet never felt that the digital source was the weak link in the chain. In fact, I had the opposite reaction: This source allowed me to hear these ultra-exotic electronics and loudspeakers at their best.

One day computer-based music systems will be simple to set up, foolproof, ubiquitous, and uncompromised in sound quality. The Berkeley Alpha USB represents a giant leap forward in realizing this goal. tas

The Absolute Sound Card Survey

## An Epic Adventure, Four Years in the Making!

Karl Schuster

I beg your indulgence for a moment, as I step into the Way-Back Machine, and emerge on a quiet afternoon in the spring of 1986. I am working at a high-end audio store, and on this particular day I find myself perusing the most recent issue of *The Absolute Sound*. Therein, I find a rave review of the Audio Research D-115 Mk II power amplifier by veteran contributor Andrew G. Benjamin. Our store is a top-ten Audio Research dealer, so I am naturally pleased that AGB's effusive enthusiasm and insightful delineation of the D-115's merits will fuel further interest in our premier line of tube electronics. However, I feel some consternation at the magnitude of AGB's praise, especially his unequivocal proclamation that "perhaps if I heard a better device, I would catch some faults here or there." I think to myself, "Well, I guess he hasn't heard the Audio Research M-100s." I know from personal experience that, while the D-115 is indeed a sensational $3000 stereo amplifier, the $5000-per-pair Audio Research M-100 mono amplifiers are better still.

This little anecdote kept popping into my head as I contemplated how to present the results of four years' worth of comparative listening tests, covering eight different sound cards. While I haven't heard every sound card out there, I sure have spent a good deal of "quality time" testing many of the prime contenders.

In the December 2007 issue, I outlined the basic architecture of a computer-based music server, and detailed what I had heard using the analog outputs of the Echo Mia MIDI ($199)

sound card and Echo Gina 3G ($439) external PCI audio interface. These preliminary tests instilled a deep appreciation for the uniquely compelling sonic benefits of computer-sourced music playback, most notably "an unprecedented solidity, providing a welcome foundation to the music" and "a certain ineffable *rightness* to the music server's sound, which somehow makes the process of listening to digital music easier, less demanding of subconscious mental effort."

I can happily report that long-term evaluation
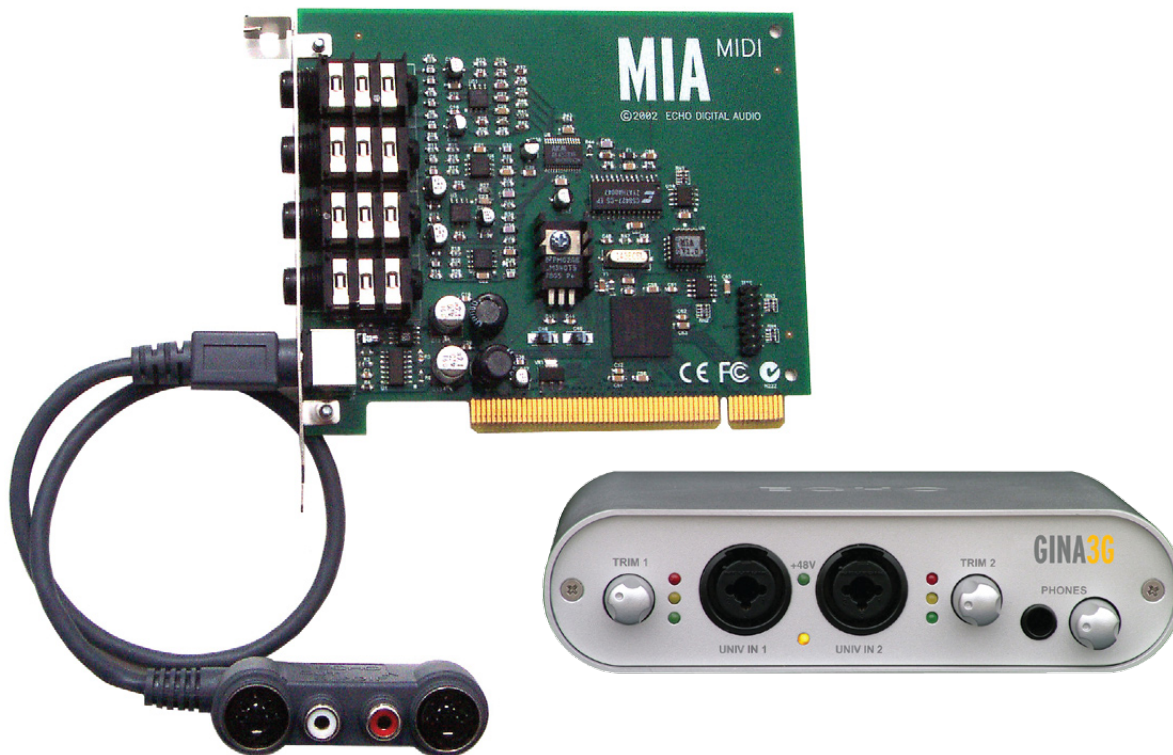
of another half-dozen sound cards has only confirmed the validity of those first impressions. While the performance of USB, FireWire, and network-streaming devices continues to evolve and improve, the design of internal PCI audio interfaces is already a mature science, based on established engineering standards and robust, proven technologies. The best sound cards set a benchmark against which alternative technologies must ultimately be judged.

As most readers of *The Absolute Sound* have discovered, every new listening experience adds to our accumulated knowledge base. Over the course of this survey, the process of auditioning each successive product added further

perspective and context, ultimately culminating in a surprising and exciting conclusion.

After my initial tests of the analog outputs of the Echo Mia MIDI and Gina 3G, I turned to their digital outputs, driving my Bryston BDA-1 digital-to-analog converter. Despite my preconceptions favoring the digital connectivity-architecture of the Gina 3G, I was surprised to find that the digital output of the Mia MIDI actually sounded better, with a more incisive, open, and rhythmically agile presentation. While the analog outputs of both Echo cards sounded fine at their respective price points, the Bryston DAC is in an entirely different league, as has been described in these pages in Issues 193 and 194.

# EQUIPMENT REVIEW - The Absolute Sound Card Survey

Since both Echo cards are limited to a maximum sample rate of 96kHz, I moved on to products that support all the common sample rates up to 192kHz. First up was the M-Audio Audiophile 192 ($200), which features RCA digital S/PDIF jacks on its mounting bracket, and balanced analog outputs on quarter-inch TRS phono jacks at the end of a short DSUB breakout cable. I auditioned the Audiophile 192's analog outputs only briefly, finding trade-offs versus the Echo Gina 3G. The Audiophile 192 exhibited a slightly washed-out character, rendering music with a desaturated tonal palette compared to the Gina 3G's more substantial and colorful presentation.

In contrast, the digital output of the M-Audio Audiophile 192 handily outperformed those of both Echo cards. I won't bother listing all the areas in which sound quality improved, since the bottom line was simply this: You name it, and it got better. The net effect was a wholesale increase in both sonic realism and musical communicativeness. However, as I continued to use the Audiophile 192 for many months, I ran into a few problems. Its driver was more temperamental than that of the Echo cards, occasionally freezing up and requiring a computer reboot. It was also

subject to occasional dropouts, which I rarely encountered with the other cards tested. Finally, the Audiophile 192 was not capable of transmitting a 176.4kHz signal via its digital output; rather, it downconverted this sample rate to 88.2kHz.

The next contender proved to be quite a delight, with some unique, appealing features. The $139 ESI Maya44's RCA digital S/PDIF output jack is also optimally located on the card's mounting bracket, accompanied by an optical TosLink S/PDIF jack. Two pairs of single-ended analog inputs and outputs are efficiently configured on stereo quarter-inch TRS phono jacks. The output stage of the second analog output pair (Channels 3 and 4) features a very low source impedance that can drive headphones directly. In use, the Maya44 operated with rock-solid stability, but unfortunately also downconverted 176.4kHz signals to 88.2kHz via its digital output.

The Maya44's coaxial S/PDIF digital output delivered a light, airy sound via the Bryston DAC. Its character was rather impressionistic, lacking a bit of body, foundation, and tonal richness. Direct comparison with the M-Audio Audiophile 192 clearly favored the Maya44, which delineated instrumental lines with greater ease, and allowed music to flow and breathe with a compellingly engaging naturalness. The Audiophile 192
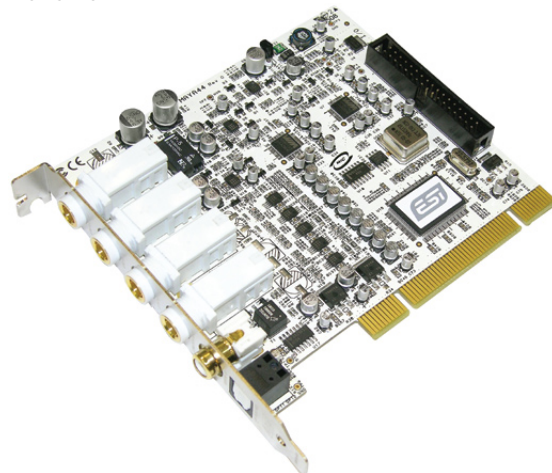
sounded more forward, with a weightier bass range, but the Maya44 rendered the sounds of actual instruments playing in the lower octaves with superior pitch definition, rhythmic subtlety, and dynamic inflection.

The intrinsic sonic character of the Maya44's analog outputs paralleled that of its digital output, a theme that was consistently repeated across the span of this survey. The Maya44's second analog output pair was the better of the two, with punchier, tighter bass and superior focus. Sonic artifacts such as slight grain, a bit of edginess on transient peaks, and diminished soundfield scale were clearly evident when the Maya44 was pitted against the far more capable Bryston DAC, but these colorations were so astutely balanced that they never interfered with the Maya44's ability to play music with a lively sense of involvement. Setting the Maya44's analog-domain output-level control to -2dB reduced the sharpness on high-level peaks to an appreciable degree. All told, the Maya44 once again raised the bar another notch closer to the elusive goal of sonic realism.

Moving on to the feature-rich RME Hammerfall HDSP-9632 ($549), I was finally rewarded with a sound card that could transmit all sample rates including 176.4kHz via both coaxial S/PDIF and balanced AES/EBU digital outputs. Alas, I immediately ran into a snag due to the manner in which Foobar 2000 interacts with RME's ASIO Hammerfall DSP driver, used to assign ASIO channels to the card's physical outputs. This problem threatened to require manually switching between three different ASIO configurations depending on the sample rate of the file being played—clearly an untenable situation.

Fortunately, the freeware utility ASIO4ALL

came to the rescue, since it can limit the number of ASIO channels visible to the music playback program. With ASIO4ALL, a single Foobar ASIO channel-assignment configuration enabled the HDSP-9632 to play files of any sample rate without incident. Both the RME ASIO driver and the universal ASIO4ALL driver bypass the

SPECS & PRICING

ECHO DIGITAL AUDIO
echoaudio.com
Prices: Echo Mia MIDI, $200; Echo Gina 3G, $439

ESI AUDIOTECHNIK GMBH
esi-audio.com
Prices: ESI Maya44, $139; ESI Juli@, $199

LYNX STUDIO TECHNOLOGY, INC.
lynxstudio.com
Prices: Lynx L-22, $749; Lynx AES16, $695

M-AUDIO
m-audio.com
Price: M-Audio Audiophile 192, $200

RME
rme-audio.de
Price: RME Hammerfall HDSP-9632, $549

FOOBAR 2000 (FREE)
foobar2000.org

ASIO UNIVERSAL ASIO DRIVER (FREE)
asio4all.com

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EQUIPMENT REVIEW - The Absolute Sound Card Survey

Windows audio subsystem to deliver a bit-accurate signal to the sound-card hardware, and I heard no audible difference between the two options.

I also tested another music playback program, Media Monkey, which assigns ASIO channels in an entirely different manner. Media Monkey was able to use RME's native ASIO Hammerfall DSP driver to play files of all sample rates via either the HDSP-9632's stereo analog or digital outputs without any problem whatsoever.

When I began assessing the RME card's S/PDIF output, my reactions were mixed. The RME sounded clean and full, in a forthright and controlled manner, but also exhibited a closed-in, muted quality that lacked the beguiling ebullience of the Maya44. However, as I revisited familiar source material, I found myself able to hear deeper into the music via the RME, discerning musical strands that the Maya44 left buried in the mix.

Then I happened to play one selection—Ravel's *Pavane pour une Infante défunte*, performed by the Minnesota Orchestra under Stanislaw Skrowaczewski (extracted from Classic Records 24/96 DAD 1025)—that abruptly ended the need for further comparison, so unexpected and definitive were the differences between the two cards. When I played the *Pavane* through the RME card, after a few seconds I just laughed out loud, blurting "Oh! Now I get it!" From the opening notes, the music immediately unfolded with a stately, measured, restrained, "walking" rhythm. Searching for words in my head, I first thought "funeral," but discarded that descriptor as too somber; this was lighter, but still reverential and formal. So I decided that it sounded more like a "procession."

I did a quick Wikipedia search, and found *pavane* defined as "a slow processional dance that enjoyed great popularity in the courts of Europe during the sixteenth and seventeenth centuries." While this piece sounded lovely via the Maya44, the RME card presented it in an entirely different, revelatory (or perhaps Ravelatory) light, clarifying the meaning and structure of the music with obvious, manifestly superior lucidity. Even more significantly, it made clear that Skrowaczewski's interpretation of the work masterfully conveyed the composition's essence, and that both the original recording and the subsequent digital transfer successfully preserved the expressive dynamic shadings and timing cues of that performance.

The RME card's AES/EBU digital output didn't fare as well, sounding brighter and more mechanical, with an electronic haze that diminished the magical rhythmic dexterity so evident via coaxial S/PDIF. My initial impression of a mellow, midrange-centric, overly restrained character was confirmed and amplified when I auditioned the RME card's analog outputs, which reminded me of a vintage Linn Sondek turntable in both virtues and liabilities: musically expressive, yet colored by excess warmth; curiously curtailed reverberant decays; and an intimate, close-up perspective that flattered recordings of small ensembles, but never opened up enough to convey the power of a full orchestra.

After my positive experience with ESI's Maya44, I approached the oddly named ESI Juli@ card ($199) with eager anticipation. The Juli@ card features a clever analog input/output configuration, with a reversible plug-in board that is fitted with RCA jacks on one side, and balanced

TRS jacks on the other. Alas, I was disappointed to find the coaxial S/PDIF RCA jacks relegated to thin breakout cables tenuously connected to the card by a loose-fitting multi-pin DIN plug.

Despite my concern about this less-than-ideal digital output configuration, the Juli@ immediately distinguished itself as something quite special driving the Bryston BDA-1 DAC. Rather than the Maya44's euphonic, water-color presentation, the Juli@ illuminated the music with unprecedented clarity, focus, and resolution. Rather than the RME's restrained, closed-down character, the Juli@ breathed with life, air, and energy. The Juli@ card rendered complex timbres effortlessly, with an unsurpassed ability to convey note shape—the dynamic and harmonic growth and decay of each note over time.

Spatial cues were presented with captivating verisimilitude. Finally, here was a card that preserved the full scale of well-recorded orchestral performances, with three-dimensional instrumental body and natural reverberant decay into the ambient air of the recording venue. Of particular note was the Juli@'s imperturbable dynamic stability. Even during energetic crescendos, the Juli@ card maintained a consistent perspective; instruments in the rear of the soundfield *stayed* in the rear of the soundfield. In contrast, every other sound card in this survey exhibited sonic artifacts during high-energy peaks, either coarsening textures, adding glare, or projecting the sound forward.

This sense of unflustered, well-behaved equanimity categorically differentiated the ESI Juli@ from every other sound card tested. Whether it was delicate chamber music, free-wheeling improvisational jazz, or raucous rock 'n'
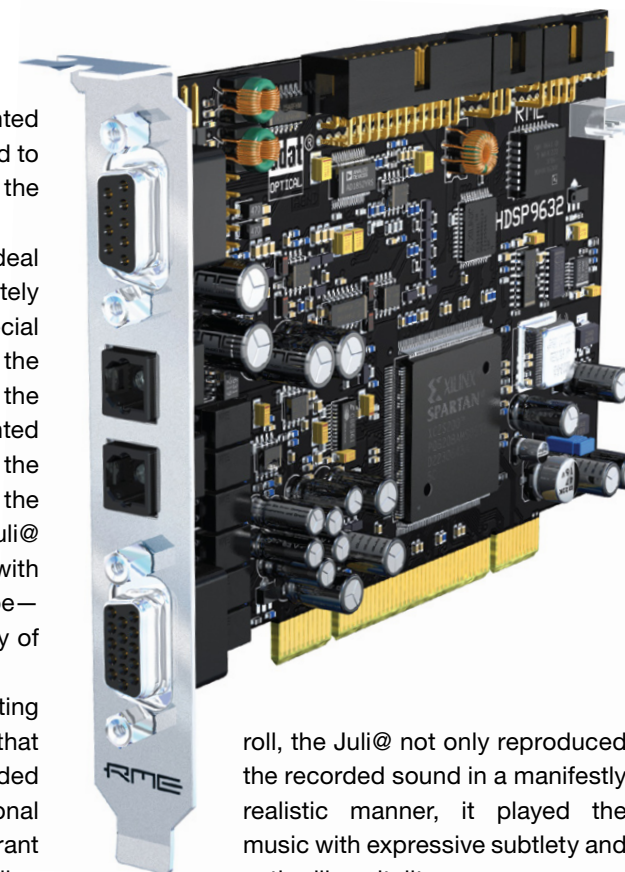
roll, the Juli@ not only reproduced the recorded sound in a manifestly realistic manner, it played the music with expressive subtlety and enthralling vitality.

After hearing the standard-setting performance of the Juli@'s digital output, I crossed my fingers as I connected its RCA analog outputs to my preamp. My hopes were rewarded, as the sound of the analog outputs mirrored the considerable merits of the card's digital output. No, the Juli@ card's analog outputs did not deliver the soundstage width and depth, unfettered dynamic peaks, rich tonal complexity, and rhythmic precision of the Bryston DAC, but the *essence* of the presentation was uncannily preserved, only on a smaller, less well-resolved scale. The Juli@'s cardinal virtues remained obvious via the RCA analog outputs, diminished in degree, but not in character. Even

EQUIPMENT REVIEW - The Absolute Sound Card Survey

NUTS 'N' BOLTS

The computer used for these tests is quite ordinary, a nine-year-old PC with a 1.47GHz AMD processor, 1GB of RAM, and no geeky tweaking of arcane operational parameters. The machine is utterly representative of a stock generic PC, other than having been fitted with a quieter power supply and fan-speed regulators to reduce mechanical noise. Thus, the conclusions drawn from these tests can confidently be expected to represent what anyone can achieve with minimal effort.

For consistency over the course of this project, I have continued to use the Windows XP operating system. I encountered no problems installing any of the cards or their software drivers, even with two or three cards installed simultaneously. All of the cards can utilize the professional standard ASIO protocol that bypasses the Windows audio subsystem, thereby insuring bit-perfect signal transfer and playback of music files at their native sample rates. I set ASIO latency to 512 samples in each card's on-screen control panel, which allowed playback of all supported sample rates without drop-outs or audible glitches. I used the program Foobar 2000 for playback and library management. Mac Pro users are advised to choose a music player that circumvents the limitations of Apple's CoreAudio architecture, such as Decibel, Pure Music, or Amarra.

Due to limited physical space, most sound cards use non-standard connectors for analog and digital inputs and outputs, such as multi-pin DSUB or DIN jacks. In such cases, the cards are usually supplied with either short breakout cables fitted with RCA or XLR jacks, or cables of modest-length (typically 6 feet) that can be attached directly to nearby audio components. Unfortunately, these stock cables are often of the "included with a cheap VCR" caliber. Substitution of higher-quality after-market cables offers the potential of further improving performance, as with any critical analog or digital source component. I auditioned each sound card with the cables provided by the manufacturer, and also tested custom upgraded cables wherever possible.

after periods of extended listening, I found myself shaking my head in disbelief, admitting to myself that "I could live with this," so well-balanced were the sonic compromises of the Juli@'s RCA analog outputs. However, I was utterly chagrined to find that the Juli@'s balanced TRS analog outputs sounded comparatively shrill, unrefined, and fatiguing. I sighed with relief after reversing the Juli@'s analog output board again, returning to the sweet, even-tempered music flowing forth from its single-ended RCA analog outputs.

The final two sound cards arrived courtesy of well-respected Lynx Studio Technology. The L-22 ($749) is a stereo card capable of internal decoding of all sample rates up to 192kHz for playback via its analog outputs, but limited to a maximum of 96kHz via its AES/EBU digital output. The AES-16 ($695) is a versatile multichannel digital-only card with comprehensive onboard mixing capabilities. All inputs and outputs on both cards are provided on DSUB connectors.

The L-22's balanced analog outputs delivered a visceral, full-bodied presentation, rich with tactile immediacy. However, under dynamic conditions, it failed to measure up to the ESI Juli@'s poise and composure. At low to moderate signal levels, the L-22 exhibited a smooth liquidity, but high-level peaks sounded edgy and forward. It offered a vivid, up-front perspective, but with reduced depth, flattened dimensionality, and

truncated reverberant decays. The L-22 seemed to emphasize whatever was most prominent in the mix, spotlighting the loudest element of the music with explicit directness and presence, but at the expense of overshadowing quieter instruments playing simultaneously. The L-22's AES/EBU digital outputs sounded remarkably consistent with its analog outputs. Considered on its own, the L-22 certainly sounded nice enough, but repeated comparisons only emphasized its colorations when compared against the more coherent, neutral, better-behaved ESI Juli@.

The Lynx AES-16's AES/EBU digital outputs exhibited an obvious family resemblance to those of the L-22, but with superior performance across the board. Compared to the L-22, the AES-16 sounded cleaner, with improved solidity, tighter focus, reduced temporal smearing, and a more engaging sense of drive. Substituting upgraded AES/EBU cables improved performance even further. Still, direct comparison with the ESI Juli@ laid bare the AES-16's limitations: lackluster rhythmic alacrity, homogenized tonal colors, and a subtle opacity that obscured note decay, hall ambience, and low-level dynamic inflections in a manner analogous to the absorption of sound in humid air. Perhaps I might have had a more enthusiastic reaction to the Lynx cards had I auditioned them earlier, but after I heard the extraordinary ESI Juli@, it became abundantly clear just how
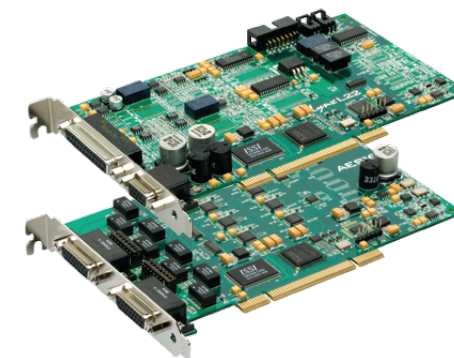
much of the music the Lynx cards were missing.

Over the course of this survey, I often found myself frustrated with the trade-offs among the various contenders. Fortunately, the remarkable performance of the inexpensive ESI Juli@ card completely changed the game. Not only did its single-ended RCA analog outputs trump those of the other products reviewed, but its digital output established a new reference standard. Thus, the Juli@ merits an unqualified recommendation, in both relative value and absolute performance. Apparently, I'm not the only one who appreciates the ESI Juli@'s virtues; several high-end manufactures are already using modified Juli@ cards inside their computer music products.

It only seems fitting to close this survey by shamelessly lifting the concluding lines from AGB's amplifier review that I mentioned at the outset: For this reviewer, the ESI Juli@ is the first computer music source component "to succeed in forsaking sonic artifacts. In fact it does more. For a mere electronic device, it luminescently renders the truth of the music." tas

High-End Sound From an iPod

Wadia 170 iTransport iPod Dock

Robert Harley

As great a product as the iPod is—and it is truly spectacular—it has an Achilles' heel for discriminating listeners: its digital-to-analog converter and analog output stage. The iPod's D/A converter and output amplifier are by necessity sonically compromised, restricting the iPod's usefulness. No serious listener would use an iPod at the front end of a high-end system. That's a shame, because the iPod is a brilliant device in its functionality, execution, and user interface. It can also store hundreds of hours of music with perfect bit-for-bit accuracy to the source.

Leave it to Wadia Digital to create a product that capitalizes on the iPod's strengths while completely eliminating the sonic shortcomings that have relegated it to ancillary listening environments. That product is the 170 iTransport, the first Apple-sanctioned dock to tap into the iPod's digital bitstream and present that bitstream to an outboard digital-to-analog converter of your choice. The iTransport allows you, for the first time, to bring the iPod's functionality to a high-end system with no excuses—for just $379.

The 170 iTransport looks like a traditional Wadia product in miniature, all the way down to its pointed feet. The flat top surface holds the docking connector, which accepts all iPod models courtesy of a supplied variety of dock inserts. The rear panel presents the iPod's digital output in S/PDIF format on an RCA jack. You simply connect this output to any outboard D/A converter and the iPod's sound quality is now determined by the quality of that D/A converter. For those of you without an external D/A converter, the iTransport offers analog outputs. Note that the iTransport doesn't have an internal DAC; rather, the iTransport simply routes the iPod's analog outputs to the iTransport's rear-panel jacks. For those with video iPods, the iTransport offers S-video and component-video outputs. An external power supply plugs into a rear-panel jack.

Controlling the iPod via its click-wheel is made easier by the open iPod-mounting design (iPod docking stations in which the iPod is flush-mounted make operating the click-wheel difficult). With certain iPod models (Nano G1, iPod Video), the click-wheel interface is disabled when inserted into the iTransport, and a small supplied remote control provides basic functions, such as track forward/backward and pause/play.

The iTransport was extremely simple to set up and use. I unpacked it, popped in my iPod Classic, and was listening to music within two minutes of opening the box.

As expected, the iTransport sounded like the DAC to which it was connected. I store music on my iPod using Apple Lossless, which provides perfect bit-for-bit accuracy to the original with about a 40% reduction in storage requirements compared with uncompressed WAV files. In listening comparisons between the iTransport and the CDs from which the music was ripped, I thought the iTransport had a slight advantage. The iTransport had just a bit more space, bloom, and ambience than the CD. The recorded acoustic was slightly bigger, the spatial perspective was a bit more distant, and the sense of air surrounding instrumental images was somewhat more tangible and defined. The differences were slight, but noticeable. This impression is consistent with what I've heard when comparing music on CD with the same music read from a hard-disk drive.

The iTransport's slightly-better-than-CD sound quality is a bonus; the real reason to buy the iTransport is that it turns your iPod (which you probably already own) into a music server worthy of feeding a high-end system. Anyone who's used the iPod knows how much easier it is to access music using the click-wheel than finding the CD and inserting it in a player. It equates to more time listening and less time looking through racks of jewel boxes. **tas**

Bel Canto USB Link 24/96 and Focusrite Saffire Format Converters

Upgrading the Sound from Your PC

Alan Taffel

Suppose your mind is captivated by the possibility of using your PC as a music server, but your system is saddled with an ancient—that is, more than a year old—DAC that does not support USB. Short of buying a whole new DAC, are there other options? Don't worry; Bel Canto's got your back. The company recently released the USB Link 24/96, a cleverly conceived USB-to-S/PDIF converter that is less expensive, more flexible, and better-sounding than the alternative of a USB DAC.

Even those who never made it through *PCs for Dummies* will have no trouble installing the Link. The size and shape of an aluminum cigarette pack, the device has no controls and sprouts but one USB input and one BNC S/PDIF output. Bel Canto generously packages the Link with a short BNC-to-BNC S/PDIF cable as well as a BNC-to-RCA adapter. Simply connect that cable between the Link and your DAC, snake a USB cable from your PC to the Link, and you are pretty much ready to roll. No need to load any drivers, or even to plug the Link into a wall since, like many USB devices, it draws its juice from the host PC. You must, however, point your operating system's audio output to the new device—a simple process. Once in operation, the Bel Canto invisibly goes about its business.

That business, of course, is not just reconciling formerly incompatible PCs and DACs, but doing so in a musically satisfying way. In this, the Link has some long odds to overcome since its raw material is the sonically challenged USB interface. Fortunately, the Bel Canto does a fine job of moving the sound significantly closer to good S/PDIF.

Indeed, in every system I tried, having this converter in the loop actually sounded better than directly connecting a PC to a USB DAC. How, you ask, can that be? Isn't it always best to have fewer boxes and a pure signal path? Normally, yes. But the Link in effect replaces the cheap, off-the-shelf USB input chip found in most DACs with a far more sophisticated and costly module. Then it delivers its payload to the DAC's S/PDIF input, which is invariably the best-sounding. These advantages apparently outweigh any losses caused by the Link's conversion process.

To illustrate, consider the sound of two USB DACs with and without the Link. Straight-through USB to the Bryston BCD-1, for example, yields a pallid, airless, "mid-fi" sound. The Audio Research DAC7's USB input is notably more extended, tuneful, and tonally rich than the Bryston's, but the ARC in straight-through mode still exhibits flashes of USB's tell-tale traits: sloppy rhythms, flat dynamics, and a vacuum-sealed soundstage.

Placing the Link in front of either DAC—especially the needier Bryston—substantially elevates the sound. Listen, for instance, to the highly illuminating Dvořák *Serenades from Bohemia* [Praga] that I used throughout these tests. The Link produces tighter timing, a whiff of fresh air around instruments, tonality captured in pastels rather than black-and-white, and dynamics with hints of actual bloom. Plus, the violins aren't nearly as irritating. To reap these benefits, be sure to use a good USB cable, such as the Belkin Gold Series ($49.99). And although the supplied Stereovox VC2 BNC cable is quite good, I was able to eke out a tad more air, greater bass authority, and a slightly quieter, more relaxed presentation by swapping in my reference Empirical Design 118 ($105).

Yet for all its superiority to straight-through USB, parity with straight-through S/PDIF eludes

EQUIPMENT REVIEW - Bel Canto USB Link 24/96 and Focusrite Saffire Format Converters

the Link's grasp. The Bel Canto minimizes but cannot completely eliminate USB's unfortunate attributes. This is not the Link's fault; it is USB's fault. Still, the fact remains that where USB through the Link provides whiffs of air, S/PDIF offers pillows. Where the Link elevates USB's colors to pastel, S/PDIF delivers enamels. And even modest S/PDIF is perfectly capable of steady timing, ripe dynamics, and strings that do not shriek. Partial USB, the Link seems to tell us, is better than total USB; but the best USB is no USB at all.

Unfortunately, if this conclusion holds true for other USB converters, as I suspect it does, it creates a quandary for would-be PC music server users. Once again the question arises: Are there other options? Well, some PCs do have an S/PDIF output, but that is usually TosLink—hardly a step up from USB. At the same time, quite a few PCs include a FireWire interface. In the pro- and home-recording communities, FireWire is the standard for PC audio. These facts led me to wonder how FireWire might compare to USB.

Finding out was easy. Because of its preference for FireWire, the pro recording industry offers a plethora of boxes that can, among other things, convert FireWire-to-S/PDIF. Even more conveniently, I happen to own one, the Focusrite Saffire, which I use in my home studio. The Saffire's primary mission is to bring microphones and line-level musical signals into a PC to be recorded. However, along the way it does perform the desired conversion function, coincidentally at the same price point as the Link. (A less elaborate version, the Saffire LE, offers identical conversion capability for a hundred dollars less.)

Like the Link, the Saffire handles files with resolution up to 96/24. Otherwise, though, the two are quite different. For instance, the Saffire requires loading multiple sets of drivers, only one of which (ASIO) permits on-the-fly sample rate adaptation. And whereas the Link has and needs no user interface, the Saffire's operation is governed by a non-intuitive PC-based control panel. Finally, the Link offers a high-performance BNC output whereas the Saffire must make do with RCA.

But the biggest difference between these two converters—and where the Saffire comes out on top—is in their sonics. Frankly, the Saffire's sound bowled me over. From it emerges a gorgeously rich, relaxed, airy, rhythmically cohesive, flesh-and-blood presentation that is the antithesis of USB. No, this converter cannot match reference-level S/PDIF in inner detail, instrumental body, or bass definition. Perhaps someday these, too, will arrive courtesy of a FireWire converter built to high-end standards. In the meantime, the Saffire's blissful freedom from USB's foibles constitutes a genuine breakthrough in extracting audiophile-grade sound from a PC.

The contrast between the Saffire and the Link was evident no matter the source material. I heard it on the intricate, aforementioned *Serenades*, but also on extremely simple tracks like "That Dress Looks Nice on You" from Sufjan Stevens' *Seven Swans*. This song has very little going on instrumentally; Stevens' vocal is backed by a plaintive acoustic guitar figure, supplemented occasionally by a banjo.

There are no big dynamic swings, no depth-charge bass notes, not even many expressive nuances. In other words, there are not a lot of bits (of information) to work with. And yet, with FireWire as the PC output and with the Saffire managing the digital hand-off, the song comes into its own. Timing snaps into place, the banjo strings acquire realistic bite, and the vocal and guitar sound both more lifelike and more coordinated with each other.

On high-resolution material, the difference between interfaces and converters is even more striking. Listen, for example, to Rebecca Pigeon's "The Raven," which can be downloaded at 96/24 from HDtracks. The high-res version of this song is awesome in its purity, regardless of interface. But what a relief it is to revel in the benefits of that resolution—the effortlessly open highs, the wealth of timbral information—without any accompanying USB crud. Through the Link, this HD music file sounds better than the CD, but only marginally so. The Saffire/FireWire version utterly stomps the CD.

For those with a suitable PC and the willingness to tackle a greater operational challenge, the Focusrite Saffire is the best way I have found to derive high-end audio from a PC. But FireWire's surprise showing in this test should take nothing away from Bel Canto's achievement. The USB Link 24/96 is a smart and timely idea, a piece of cake to install, and invisible in operation. For those who opt for USB's simplicity and ubiquity, it is easy to recommend. tas

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 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

EQUIPMENT REVIEW - Channel D Pure Music Software

Remember! You can go to another section by clicking here.

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.

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,

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-res with ease.

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

SPECS & PRICING

Channel D Pure Music Software
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

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Logitech Squeezebox Touch

High-Res Wireless For $299

Steven Stone

The debate over whether wireless audio or computer-based audio are "good enough" to be part of a high-end system is still raging among some audiophiles. For others the debate is already over. As AHC discovered when he reviewed the Meridian Sooloos music server in issue 204, if you can afford a server that costs five figures you can have great sound from a hard-drive-based system. But what about those of us who can only pony up three figures? Can you achieve CD-competitive (or better) sound from an "entry-level" product such as the new $299 Logitech Squeezebox Touch? I sure hope so.

I see strong signs that the price/performance differential between expensive and budget-priced front-ends is shrinking. In the rapidly approaching new music-library-based audio world, consumer electronics, computer manufacturers, and pro-audio firms will all have devices suitable for use in high-end audio systems. Logitech is a prime example of a company that's involved in this new audio revolution. Its latest wireless audio device, the Squeezebox Touch, builds on two prior generations of Squeezebox wireless devices. Offering 96/24 capabilities, a responsive touchscreen display, and a more robust and open-ended interface, the Touch looks like the ideal "first step" for audiophiles who have yet to make the jump to a high-performance wireless music device.

THE BASICS

The Logitech Touch's main function is to play music files. The files can come from many sources, including your iTunes library or the primary music library on your computer or music libraries on USB and NAS drives, as well as Rhapsody, Internet radio, Pandora, Last FM, and other on-line music services. It is called Touch because of its 4.3" (actually 2 ¼" by 3 ¾" display area) touchscreen. All functions can be controlled and accessed from this touchscreen. The Touch also comes with a remote control to operate it from your listening chair, if your listening chair is close enough to the Touch to read its screen (for me the decipherability limit is five feet.) If not, you can use the Touch with the Squeezebox Duet remote, which has a full color screen. You can

also control the Touch via your iPod or iPad via a free app available through Apple's App Store.

The Squeezebox Touch supports most formats including WAV, AIFF, Apple Lossless, FLAC, WMA, WMA lossless, AAC, and MP3. It can also transcode formats through its Squeezebox Server software (more on this later). The Touch does not require a computer running Squeezebox Server to play Internet radio or on-line music services. A server is required as one of the ways to play tracks from your primary home music library — the other way is to play from a USB memory stick or SD card inserted into the Squeezebox Touch. This USB drive connection gives you access to any local music library on a USB drive without forcing you to turn your computer on. Some users have

even set up the Touch so it runs exclusively off a USB drive with no local or Internet connectivity whatsoever (except for initial setup, which will require Internet access).

I used the Touch with a variety of different manufacturers' USB sticks to conclude that this method is both reliable and easy. But I never managed to get the Touch to recognize my Newer Technology V3 drive due to its OS X file system. The Touch only supports USB drives with FAT 16, FAT 32. NTFS, and ext2/ext3 files systems. Also it's vitally important to use a drive that has its own external power supply that does not depend on the USB connection for its power. The Touch's power supply can't fully support the power requirements of a USB drive through its

www.theabsolutesound.com

EQUIPMENT REVIEW - Logitech Squeezebox Touch

USB connection. If you are planning to use the Touch primarily with a USB drive, consult a list of supported devices here on Slim Devices' Web site.

The Touch's most exciting new feature is its ability to play 96/24 high-def music files. It's the first under-$300 wireless server that supports these higher-resolution files. During the review period I played 96/24 files from my USB sub-libraries and main library via both Wi-Fi and hard-wired Ethernet sources successfully. Only once, during months of play, did the Touch stop in the middle of a song. Merely pushing the pause and play buttons instantly solved that problem.

Initial setup for the Touch was simpler for me than for a new user since I've been using a Squeezebox Duet and have already set up a Squeezebox Server and Squeezebox system. The Squeezebox Server software and my Duet remote immediately recognized the new Touch and I was listening to music from my main music library via Wi-Fi in under fifteen minutes. If you do have set-up problems Squeezebox user forums on Slim Devices' Web site will be immeasurably helpful. Several Squeezebox experts, such as John Swenson, frequent the forums regularly.

TWO WAYS TO GET HOOKED UP

The Touch has both WiFi and Ethernet connectability, but if I were a betting man I'd wager that over 80% of Touches will be connected via WiFi. I tried it both ways, and like the general public most of my listening was done via WiFi connected to 170GBs

of music in my iTunes music library. Some users on the Squeezebox forums claim that Ethernet offers better fidelity than WiFi, but I didn't hear any repeatable, recognizable fidelity differences between these two connection methods. Obviously if you experience dropouts via WiFi then switching to a hard-wired Ethernet connection will reduce incidents of dropouts due to the connection. Will it sound better this way? That depends on your WiFi connection.

In both my systems I used the Touch primarily with its digital outputs. I understand that some readers intend to use the Touch's internal DAC instead of its digital outputs, so I spent some time listening to its analog output through my Stax Nova headphones. Compared to the Weiss DAC 202 via its RCA coaxial connections, the Weiss exhibited a more extended top end with a greater sense of air. I also found the Weiss to be more dimensional. But the Touch's own internal DAC isn't bad. It's musical and smooth, and has excellent dynamic contrasts. Its sins were primarily of omission, with a slightly darker and more forgiving nature than the Weiss.

MULTIPLE LIBRARIES

I mentioned earlier that many people are using the Touch as a stand-alone unit with a music library connected via the USB inputs. While this is an excellent solution for those who don't or won't install a wired or wireless network to access digital music files, it does limit your options to only one music library. Part of the beauty of the Squeezebox system

is its ability to handle and access multiple libraries. Each household member can have his or her own library, which the Touch can access via only a few selections from its screen.

My preferred method for using the Touch's USB connection is with what I call "micro libraries," which are USB memory sticks between 2 and 8GB with specific types or genres of music. The Touch's Squeezebox Lite software has memory limitations that make it happier with smaller libraries as opposed to fully populated big ol' 1TB USB drives. Large drives cause the SBL software operations to slow down upon installation, when the Squeezebox software is initially indexing the drive. These slow-downs can affect response time to commands, especially those involving skip and search functions. Once fully indexed, larger drives will respond with the same speed as smaller USB cards, but any time you switch USB drives the indexing process will affect the Touch's response time until SBL finishes its indexing duties. With small USB cards instead of larger mechanical hard drives, the Touch's SBL software can quickly index the music and be fully responsive moments after the drive is connected.

At the 2010 CES a friend of mine loaned me a USB memory stick with the complete Beatles In Stereo boxset on it. This was one of a limited run of 30,000 copies. When I got home I could hardly wait to hear the 24-bit FLAC file versions of my fave Beatles songs on the USB stick, but when I tried to play them on my Mac I couldn't. Even the latest

SPECS & PRICING

Logitech Squeezebox Touch

System requirements: For access to music on your computer use Squeezebox software with the minimum system requirements: 256MB RAM and 100MB of available hard disk space. With any of the following operating systems—Macintosh: OS X 10.3 or later; Windows: 733MHz Pentium running Windows 2000, Windows XP, Windows 7, or Windows Vista; Linux/BSD/Solaris/Other; Perl 5.8.3 or later. Broadband Internet connection required for Internet radio & music services.
Audio formats: MP3, FLAC, WAV, AIFF, WMA, Ogg Vorbis, HE-AACv2, HD-AAC, Apple Lossless, WMA Lossless, APE, MPC and WavPack supported through transcoding
Internet radio: Support for MP3, Ogg Vorbis, HE-AACv2 and WMA formatted Internet Radio streams
Wireless interface: True 802.11g wireless networking; support for 802.11b and 802.11g routers and access points; throughput up to 54Mbps, high speed PCI interface to radio module
Ethernet interface: Shielded CAT5 RJ-45 connector, connects to any 100Mbps or 10Mbps network (with Auto MDX)
General: USB host connector for accessing music and photos via USB drive or USB key; SD card slot for music and photos; supports sampling rates up to 24 bit / 96 kHz; stereo analog (RCA), digital optical, and digital coax output; 4.3" (11cm) 24-bit color LCD with capacitive touchscreen

U.S.

Price: $299

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6505 Kaiser Dr.
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EQUIPMENT REVIEW - Logitech Squeezebox Touch

Apple OS 10.6.4 operating system wouldn't natively decode FLAC files. Sure, I could load them individually into Audacity for playback, but if I wanted to play the entire stick's contents from the 44.1/24-bit files I was out of luck. Using the Squeezebox Touch all I had to do was copy the 44.1/24-bit FLAC files onto another USB stick, load the stick into the Touch's USB slot, and a minute later I was listening to the entire Beatles library in 24-bit splendor! That was good day sunshine indeed.

DAY-TO-DAY OPERATION

During the past two months that I've had the Touch up and running in my system I've found it to be substantially more stable and glitch-free than the Logitech Duet. Dropouts caused by connection errors have been virtually non-existent with the Touch, while in the past the Duet has occasionally given me problems. I should mention that Logitech's latest version of the Squeezebox Server Software might also be part of the reason that the Touch has been so stable and responsive. Other users on the Squeezebox Forums also have reported improved performance with the Duets and version 7.5.1.

When I reviewed the Sonos system and compared it to the Duet using the previous version of Squeezebox Server software, the Sonos proved to be ergonomically more elegant. But the Touch with SC 7.5.1 and SBL brings the Squeezebox system much closer to the ease of operation of the Sonos. Now, regardless of whether my computer is on or not, I can access and enjoy my music files through the Touch. That alone is a massive improvement over previous Squeezebox systems.

MID-FI, HI-FI, OR ULTRA-FI?

Once the Touch was set up in my room system, the fun really began. Since both the TosLink and coaxial digital outputs are active, it was easy to try different digital connection schemes and compare the results. At first I set up the Touch so it ran through my Meridian 518 digital-processing device before the signal went into my Meridian 568.2 controller. With this hook-up method, the sonic differences between the Squeezebox Touch and Duet were nil. Going back and forth using the same 44.1/16 digital file source I couldn't reliably tell one from the other. But once I hooked both units directly into the Meridian 568.2 via their RCA coaxial outputs, the differences between the units were more pronounced. I noticed the Touch had a more three-dimensional soundstage with a better sense of depth. Also dimensional cues and subtle low-level details were more apparent through the Touch. When I tried streaming Internet radio sources, I couldn't discern any sonic differences between the Touch and the Duet. MP3 files through the two units were also essentially identical. My conclusion was that to hear the Touch's sonic improvements over the Duet you have to use at least a lossless 44.1/16 file for your listening tests.

Next I compared the Touch to a Meridian 598 DP DVD/CD transport with the original CD in the Meridian and its matching digital file on the Touch. Overall, I'd call this comparison a sonic dead heat. The 598 was a bit more harmonically lush, but this lushness came at the expense of dynamic contrast and inner detail. The Touch was more matter-of-fact with greater dynamic ease. Both displayed riveting levels of inner detail and musical texture, but the Meridian emphasized the source's musicality while the Touch brought the music's dynamism to the forefront.

The best sonic results from the Touch came when it was playing higher-resolution files. I listened to a slew of my own 96/24 recordings through the Touch and it never failed to produce outstanding results. I was especially impressed by the Touch's ability to reproduce spatial information. I have a recording I made of a live concert from the bluegrass band Long Way Home. The recording was done in a small one-room wooden schoolhouse west of Boulder, CO. The band was recorded with one stereo pair of Schoeps Collette microphones connected to a Grace Lunatec V-3 microphone preamp in M/S mode. Through the Touch it was easy to place each instrument in the soundstage and hear the wall and floor reflections. Even very subtle dimensional cues were obvious, such as the way the acoustic bass's notes bloomed and expanded as they interacted with the room.

GOIN' FOR THE TOUCH

Economically speaking, a $300 device is not high end. But high-end audio isn't only about economics. Performance matters. Judged strictly by that yardstick the Logitech Squeezebox Touch qualifies as a legitimate high-end component. Alone, the Touch produces musical and detailed sonics and can deliver 96/24 music files to your eager ears. Coupled with a top-echelon DAC the Touch can take you well past the first scrum in high-end sonics into the center of the playing field.

If you haven't dipped your toes into the ocean of wireless and computer audio, the Logitech Squeezebox Touch would be an excellent craft for your maiden voyage. For less than the cost of a pair of top-quality one-meter interconnects, you can enjoy your digital music files, even 96/24 files, anywhere in your home. Let's face it: Early adopters aren't usually thought of as thrifty types, but considering its price and capabilities, purchasing a Squeezebox Touch may be the most parsimonious audiophile purchase you'll ever make. Recommended? Oh, yes! tas

Linn Products Majik DSI Digital Stream Music Server

You Say Magic, Linn Says Majik DSI

Steven Stone

Everyone has heard of Linn, right? Well, I suppose there might be a few still-wet-behind-the-ears audiophiles for whom Linn is new, but for the rest of us it's a familiar name. Most audiophiles fall into one of two camps when discussing Linn—pro or con. Few are neutral. Why? Perhaps because Linn has always done things its own way, with a whole-systems approach to audio. Almost no one buys just one Linn component. More often than not people have entire Linn systems, right down to the cables.

Linn and Linn dealers have long been in the forefront of the subjectivist school of audio. Linn's founder, Ivor Tiefenbrun, first brought the audio terms "tunefulness" and "pace" into the audiophile lexicon.

For many older audiophiles Linn's most technologically important product will always be its Sondek LP-12 turntable. Still in production after more than 30 years, the LP-12 is, without doubt, a true classic. But for a company to survive today it must embrace the brave new world of computer audio and digital music. Linn has attacked the problems of computer audio with gusto and come up with a unique approach—the Linn Majik DSI integrated digital stream player. It combines all the features and ergonomics of an old-school analog integrated amplifier with those of a digital music renderer. The DSI can play any music file, analog or digital, from a phonostage, tape recorder, tuner, CD player, hard drive, ROM disc, flash card, USB thumb drive, or iTunes music library. The DSI combines all the functions of a stream player, A/D, D/A, analog preamplifier, and class A/B power amplifier in one compact chassis.

Linn has long employed a "systems" approach to audio, so it should come as no surprise that Linn applied similar thinking to its computer audio products, including the DSI. Since some aspects of a computer audio system are out of Linn's control (such as USB 2.0 and USB 3.0 specifications and drivers), Linn decided to avoid problems by only supporting setups and applications where it could guarantee an acceptable level of performance. You won't find a USB input on the DSI because Linn feels that bringing computer audio files into the DSI via an Ethernet connection is better. By employing Ethernet the DSI avoids all the USB issues—asynchronous vs. synchronous file transfer, excessive jitter, throughput limits, and sample and bit-depth limitations. The DSI also relegates burning and storage duties to other components in your system. Instead, the DSI supports and recognizes any and all NAS (Network Attached Storage) drives on your network. These external drives are file sources while your computer acts as the server to access and send these files via Ethernet to the playback part of the system, which is called the renderer. The DSI makes music from the raw music files delivered to it from the server.

One major ergonomic difference between the DSI and most music servers is that Linn does not support or recommend a wireless WiFi connection between the DSI and the rest of your computer audio system. Instead Linn strongly suggests a wired Ethernet connection or Home-Plug. Home-Plug is a system that uses your home's AC wiring to move data. With a sender plugged into your AC on one end and a receiver on the other, data can move at speeds up to 85Mbps, nearly as fast as Ethernet's 100Mbps, and almost 50% faster than WiFi's 55Mbps data throughput rate. For maximum transfer rates Home-Plug devices need to be on the same AC circuit. If the Home-Plug's signal goes through a circuit-breaker (as it will if the sender and receiver are on different circuits), its data rate will be substantially reduced. Even in this situation Linn suggests Home-Plug instead of WiFi because other devices such as microwaves and wireless phones will not affect the Home-Plug system's transfer rate as they do with WiFi.

For over a year I've been using wireless audio

EQUIPMENT REVIEW - Linn Products Majik DSI Digital Stream Music Server

systems from Logitech and Sonos that use WiFi connections. During that time I've had no issues with dropouts, but neither system supports files with data rates faster than 96/24. With the Apple TV, which also uses WiFi, I've occasionally had issues with stuttering on high-def material (usually corrected by letting the Apple TV's buffer fill more completely before I watch). During the review period the DSI performed flawlessly with nary a dropout or connection problem via Ethernet or S/PDIF.

An important part of the Linn experience is purchasing Linn equipment from an authorized Linn Dealer. This isn't merely a way to make you spend additional money. Linn dealers are trained by Linn on how to install and set up all its products. Just like the LP-12 turntable, a network-aware music server such as the DSI requires proper initial setup to perform optimally. A Linn dealer will come to an owner's home to set up and install its DSI. If changes need to be made in the DS1's setup, Linn supplies its Linn Konfig software for re-configuring the system.

Since a Linn dealer will install this control software, I won't go into great detail about its ergonomics. I watched Linn's U.S. distributor Darrin Kavanagh do the initial setup. It took him about 45 minutes, not including a drive to a local computer superstore to buy an Ethernet switchbox so we could hard-wire the DSI into my home network.

When it comes to signal paths simple is always better. That's a basic audiophile truism. The DSI offers two ways to handle an analog signal. For maximum quality the DSI can maintain the analog signal in the analog domain throughout the entire signal path. This is ideal for sources such as external phono preamps, tuners, and

SACD players. Or you can use the DS1's internal A/D to digitize the incoming analog signal.

Why would you want to put an analog signal through an additional A/D and D/A cycle? So it can be made available to any other music-render on your home network. Using the current Linn set-up and configuration program, you can designate whether an analog input remains pure analog or is converted to digital. Linn has plans for future software upgrades for the DSI that will add server functions (storing music files for delivery to other music renderers) in addition to its current function as a renderer.

Linn has attacked the problems of computer audio with gusto and come up with a unique approach: the Majik digital-stream player.

Overall the DSI has a classical British appearance, understated and businesslike with a noticeable absence of bling. An elegant front panel with six buttons and a large blue LED screen is all that greets the eye. Other Linn products, including the Akurate and Majik DS, share this refreshingly minimalist design. It is especially well suited to the DSI, however. The left side of the panel has the mute and volume-up/volume-down buttons next to a headphone jack. The right side of the DSI has an on/off button and a pair of up/down source buttons. You'll also find a 1/4" stereo headphone output and a stereo mini-plug line-level input on the front. The display can be programmed to rename any input, and when the DSI is streaming music files from your network the display can show basic metadata such as the artist and song title.

The rear panel of the DSI is densely populated by three RCA S/PDIF and three TosLink digital inputs, one Ethernet input, four RCA single-ended analog inputs, one TosLink digital output, one RCA digital output, one fixed-level Zone Two analog output, one variable-level analog output, a pair of five-way speaker binding posts, an AC connector, and an on/off switch. The variable-level outputs can be connected to an amplifier for bi-amping a system or to a subwoofer, which is how I used it.

During most of my review I had the DSI situated under my desk, so I could reach (but not see) its front-panel controls. Most of the time I adjusted the volume and chose the input source from the front panel rather than from a remote. After a day or two I got very used to being able to feel where all the buttons were located and could make adjustments without looking. It's nice to use a product that's not only attractive but also so ergonomically well thought out.

The DSI comes with a fairly ordinary baton-shaped remote control. It doesn't even light up. Judging by the dual-function buttons, which have titles such as "Lip Synch" and "Video Adjust," this remote is used by several Linn devices besides the DS1. Fortunately you don't have to use the remote to operate the DSI.

Early in the design process Linn engineers decided to concentrate on what they knew. Instead of reinventing the wheel, Linn uses other folks' tires. An example of this thinking is reflected in Linn's eschewing USB in favor of a hard-wired Ethernet connection. By eliminating USB, Linn avoided all USB's problems and the cost of providing support to solve them.

Because the DSI is an open-architecture UPnP

(Universal Plug and Play) device, it can see and be seen by other devices that are Ethernet-aware such as an iPhone or iPod Touch. The DSI can also be accessed and controlled by software written by third parties. Bookshelf Apps developed just such an app, called Songbook (for which go to http://bookshelfapps.com/about.php).

SPECS & PRICING

Linn Products Majik DSI Digital Stream Music Server

Supported file types: FLAC, ALAC, WAV, AIFF, AAC, MP3

Audio sample rates: 7.35k, 8k, 11.025k, 12k, 14.7k, 16k, 22.05k, 24k, 29.4k, 32k, 44.1k, 48k, 88.2k, 96k, 176.4k, 192k

Power output: 90W RMS per channel into 4 ohms

Signal to noise ratio: > 115dB

Frequency response: 6Hz–25kHz (–3dB into 4 ohms)

Dimensions: 80 mm x 381 mm x 355 mm

Weight: 10.78 lbs.

Price: $4200

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EQUIPMENT REVIEW - Linn Products Majik DSI Digital Stream Music Server

Songbook is in its second incarnation. Apple took so long to approve the first version that the developer had time to write a complete update. Songbook's interface, both on the Mac desktop and on an iPhone, is very intuitive and Mac-like. Besides playing any music file, including everything in my iTunes playlists, Songbook can build playlists, shuffle songs, rip CDs into a music library, and find cover art. While I didn't test every function, most worked as advertised. Unfortunately my version 2.1.0 wouldn't successfully save playlists. I'm hoping later versions will fix this.

On an iPod Touch, Songbook works especially well. You can control one or more DSIs from anywhere you have a WiFi signal with Internet access. If you're an experienced iPod Touch or iPhone person, you'll feel right at home with the software's layout. Although I own not one but two 160GB iPod Classics, I hadn't used an iPod Touch before. After I got my special iPod Touch finger-moves down, I found it easy to use. At times Songbook was a bit slow executing commands. Whenever this occurred the problem was inevitably a result of a poor WiFi connection. Walking closer to my Apple Airport Extreme solved the problem.

I prefer using iTunes over Songbook, but my bias may be a result of my longtime use and greater familiarity with iTunes rather than any intrinsic or fundamental shortcomings in Songbook's ergonomics.

So, how does this technological marvel sound? Would it be a surprise you if I told you the DSI sounds addictingly good? My one sentence sonic description of the DS-1: "It has the musicality of the Meridian 561/518/Edge AV-6 system coupled with the resolving powers of the Weiss Minerva DAC." In short, this is the kind of component that may well encourage you to get off the audiophile-equipment-churning merry-go-round for a long breather.

Linn recommends some break-in time for the DSI's full sonics to bloom. I tried to withhold judgments for the first 100 hours of use, but heck, I'm human, and I can't listen without forming opinions. Who can? Yet right away I was drawn in by the DSI's combination of superb low-level resolution and euphony. On my own live recordings of the Boulder Philharmonic, the DSI displayed excellent depth and dimensionality combined with benchmark low-level detail.

Regardless of what speakers I listened with, the DSI did a superb job of placing instruments precisely in a three-dimensional space.

The DSI's dynamic capabilities varied depending upon what speakers it was tethered to. Dynamic contrast was best with the Paradigm S-1 speakers, followed closely by the ATC SCM7 monitors. The Aerial Acoustics 5B displayed less dynamic snap, but I suspect this was a result of its lower efficiency and more difficult impedance curve. Linn's specifications rate the DSI at 90 watts RMS into 4 ohms. That's only 45 watts into 8 ohms, so don't expect it to be able to be able to drive low-sensitivity speakers to deafening levels in a large room.

Most of the time I listened to the DSI with a subwoofer connected. And despite its relatively low power output, the DSI did a fine job in upper-bass punch. Even with the lower-efficiency Aerial 5Bs I had no problem getting a smooth blend between speakers and subwoofer. While I did hear some compression at higher volume levels with this combo, the DSI amplifier section still delivered a satisfying level of slam at lower levels.

Regardless of what speakers I listened with, the DSI did a superb job of placing instruments precisely in three-dimensional space. In comparison to the April Music Stello Ai-500, the DSI seemed slightly more dimensional. Also the DSI achieved a higher level of overall musicality. Especially on less than stellar recordings the DSI focused me on the music rather than on the amusical aspects of the recording. But the DSI was not able to keep up with the Stello in micro- and macro-dynamics. This was most likely due to the Stello's much beefier amplifier section.

I spent a good deal of time comparing the DSI 's Ethernet connection to its RCA coaxial digital input fed by the Empirical Audio Off-Ramp 3 with Ultraclock. Frankly, the differences were very minor. The Empirical produced slightly better depth, but in dynamics, detail, harmonic balance, and overall musicality the two were eye-crossingly close. Even on 96/24 material the Ethernet connection kept pace with the Empirical. My conclusion, after a couple of months of listening, is that Linn's Ethernet solution to the USB/Firewire conundrum is both elegant and effective. I suspect that in the future many other companies will adopt this UPnP Ethernet connection for their music rendering systems.

Another trial that the DSI passed easily was the Amarra test. When I turned the Amarra software program on and off I could easily tell a difference through the DSI/Empirical Off-Ramp combo. Using a direct TosLink connection from my Mac Pro to the DSI, the differences between Amarra on and off vanished. Since Amarra works only with iTunes as a plug-in, I couldn't use it for the DSI's Ethernet connection, which requires the Songbook software program.

Horses for courses is a phrase I first read in a Dick Francis novel, but it certainly applies to audio as well as horse racing. The DSI was designed to be a one-box integrated solution for music lovers who want a component that can play virtually any music file, analog or digital, in any room of their home. The DSI delivers the convenience of a wireless music server, but it does require an Ethernet or Home-Plug connection.

While the DSI could be used in any room, it is ideally suited to a smaller room or for nearfield listening. That's because its power amplifier is only a modestly sized 90 watts RMS into 4 ohms. I used it primarily in my computer desktop system. And while I could hear some dynamic compression at higher volume levels with the least sensitive speakers in my arsenal, with decently sensitive speakers the DSI shouldn't be in danger of running out of power.

Overall the DSI is a winner. It does all the things you've come to expect sonically from a Linn component. It's tuneful, with excellent pace and musical delivery. On the practical side, it's painless to set up (thanks to your local Linn dealer) and simple enough to use that even your mother can operate it successfully. If you want to join the digital-music-file revolution, the DSI will let you surf on a leading technological wave without sacrificing any musical enjoyment in the process. tas

Micromega WM-10 WiFi Music Streamer

A Music Library At Your Fingertips

Tom Martin

We recently received and hooked up the Micromega WM-10, an unassuming black box that seems to be exactly what a lot of people are looking for, even if they don't know it yet. You see, the WM-10, launched at CES 2010, has the ability to play high-quality audio streams remotely using the WiFi (802.11n) wireless system.

WHAT EXACTLY DOES IT DO?

Some people find the parts and pieces of computer audio a bit confusing. So, before we get into the WM-10 specifically, let's review the general idea on offer here.

Products like the WM-10 assume that the customer has (or is willing to buy) a music server. A music server is a device for:

• Downloading music files via an Internet connection
• Ripping music files from CDs
• Storing music files (on a hard disk or other mass-storage device)
• Organizing the library of music files that you've stored
• Streaming the music files (creating a bitstream usable by a D/A converter)

Since a music server can be configured using an existing PC or Mac, almost everyone is in the situation of having a music server. Audiophiles may wish to have a dedicated music server, of course,

to avoid competition for resources between computing activities and music activities.

A music server could be hard-wired to a D/A converter and from there plugged into an amplifier and speakers or headphones. As an alternative, one could purchase a music client (also known as a networked music player or music streamer). Here we use the term music client because it fits the server-client metaphor borrowed from computing (where the server is the base for massive central data storage and organization and the clients are remote devices for viewing and lightly manipulating data). The music client receives music data streams from the music server over some kind of network connection and then performs D/A conversion for input to an existing audio system.

So, in summary, the WM-10 is a music client. It is designed to work with a music server (e.g. a Mac with iTunes and WiFi). The WM-10 receives WiFi music streams from the server and does D/A

conversion. You plug the WM-10 into your preamp or receiver via a stereo analog connection. You control the songs that are being streamed from the server using an iPod Touch or iPhone.

WM-10 DESIGN

With the above in mind, the concepts behind the WM-10 are pretty simple and easy to grasp. The first idea is that audiophiles probably want their PC and their audio systems in different rooms. This isn't some whole-house audio-lifestyle BS, it is desirable because PC components tend to be noisy and the easiest way to deal with this is to relegate PCs to another room. It can also be a practical matter. Audio equipment is often located in a more relaxed living space in the home than computer gear, which is assigned to a functional workspace. The WM-10 is therefore conceived as a remote client.

The second idea behind the WM-10 is that such remote clients should be wireless. Many homes are not fully wired with Ethernet connections in all the right places, so wireless is simply easier. One could use AC power-line networking, but some designers are concerned about how well this works.

The third notion behind the WM-10 is that audiophiles will want a high-quality music client, if they want one at all. While the WM-10 starts life as a humble Apple Airport Express, Micromega has redesigned it in two critical areas. It has installed a higher-quality power supply and redesigned the clock circuitry for better D/A performance. Finally, instead of a TosLink digital output, they've used a coaxial S/PDIF output for easier and better interfacing to external DACs (should you not want to use the internal DAC of the WM-10).

While on the subject of quality, I would add that Micromega chose the Airport Express because it uses an excellent chipset, which is capable of decoding 24-bit/176kHz and 24-bit/192kHz high-resolution files. As a practical matter, this is more a future-proofing capability than something you can use right now. The Flamjaset protocol embedded in WiFi doesn't yet allow these high-resolution data rates.

To install the WM-10, you simply plug it into your preamp/amp/receiver via RCA stereo outputs on the back of the WM-10. Next, you set your music server's wireless interface to connect to "Airstream." The WM-10 then is directly

EQUIPMENT REVIEW - Micromega WM-10 WiFi Music Streamer

connected to your music server; it doesn't go through your network, with the result that, if your music server is also your PC, you can't use the Internet over WiFi while streaming songs. Using an Apple iPod Touch or iPhone, you download a free application called "Remote" from the App Store. The iPod then sees your iTunes library on your PC and allows you to select albums, artists, and songs just as you do with the locally stored songs on the iPod itself. The iPod relays your commands via WiFi to the server and the server obeys. Volume can be controlled via the iPod or via your preamp.

HOW WELL DOES IT WORK?

I was impressed with the ease of installation. Computer setup often involves the strategic application of profanity and a search of forums for the secret to making two components designed to a "standard" talk to each other. But in this case, setup took about 15 minutes and everything worked the first time.

I only encountered two problems. The first involves "sleep mode." When you start a listening session, it simply takes a while for everything to wake up and connect with the WiFi systems (direct from the WM-10 for music streams and via your WiFi network for remote commands). Related to this, you have to remember to set the iPod Touch to "Auto-Lock: Never," so that it won't go into sleep mode. If it does go into sleep mode, it take a while when it wakes up to find the WiFi signal, which makes it a very slow remote. Neither of these are WM-10 issues; they are byproducts of using ancillary systems that have a sleep mode.

The second problem is more obscure, but some

of you will want to know about it. My server has the Amarra player installed and integrated with iTunes. For whatever reason, the volume control on iTunes and the volume control on Amarra fight with each other when you change songs using the iPod Touch. There may be a way around this, but I haven't found it.

HOW DOES IT SOUND?

I compared the sound of the WM-10 to that of my reference EMM Labs CDSA player, as usual evaluating differences in the context of the absolute sound. This might seem unfair, given that the EMM Labs player cost $10,000 in its (very recent) day. My goal, I can assure you, was not to set an impossible task for the WM-10. Rather, I wanted to know, as readers will, how closely the WM-10 came to DACs that are roughly state-of-the-art.

The good news is that the WM-10 offers a smooth and relaxed sound that avoids some of the cheap and nasty distortions that can mar digital playback. Too many DACs in my experience have some additive distortion in the treble range that distracts from the sense of the music being real. The WM-10 isn't like that; the player mostly gets out of the way and doesn't impose its flavor aggressively.

I would also characterize the WM-10's flavor as being slightly warm, not because it emphasizes bass, but because the mid-treble range comes across in a somewhat reticent fashion. If you have a system that leans toward an edgy or cold sound, the WM-10 might balance things out nicely.

Bass response and definition also are quite good. For example, on the Alison Krauss disc

Forget About It [Rounder], the title track has a kick drum that is very well defined on the WM-10. However, I wouldn't swear that this is due to something special in the WM-10's reproduction of low frequencies; it may simply be that the more laid-back treble presentation makes it easier to focus on the bass quality.

The WM-10's treble sonics also deliver a good sense of depth. I came to qualify this observation after listening to many discs. I think the WM-10 should be described as offering a more distant perspective. With the WM-10, you simply feel as if you are sitting farther from the band or as if the band is pushed farther behind the plane of the speakers. I find this perspective helps create a sense of virtual reality, because the sound is less locked onto the speakers.

When we get to the actual presentation of micro-dynamic detail we find the place where I think the WM-10 can be bettered by very good DACs directly connected to your preamp. For example, on Eva Cassidy's Live At Blues Alley [Blix Street Records], the opening track has the announcer introducing Eva. With the EMM Labs DAC you really get the sense of the club venue as you hear the decay of the PA reverberating. The WM-10 delivers less of this ambient information. Eva snaps her fingers on Irving Berlin's "Cheek to Cheek," and the EMM makes the echo of the snaps more realistic sounding.

The differences I'm talking about are subtle. Experienced listeners would note that the Micromega doesn't dig into the music quite as far as some of the very best DACs, but this isn't a hit-you-over-the-head kind of difference; it is more of the latent sense of a very fine veil over the performance. At $1595, looking for an EMM

level of transparency probably isn't a realistic expectation.

Mostly, if you heard the WM-10 without comparison, I think you'd say it sounds really good. I don't think you'd immediately comment on its transparency, but you'd probably comment on its smoothness and sense of low distortion. Given the convenience of playing music from a PC or Mac, with your complete music library at your fingertips, this sound will be more than adequate to make you forget about silver discs in trays. tas

SPECS & PRICING

Micromega WM-10 WiFi Music Streamer
Inputs: WiFi (802.11n)
Outputs: One single-ended stereo analog output (RCA connectors), one coaxial S/PDIF digital output
Formats: 16 bit/44.1kHz (AAC, AIFF, Apple Lossless, MP3, WMA, WAV)
Dimensions: 16.9" x 2.75" x 9.8"
Weight: 8.8 lbs.

U.S.
Price: $1595

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Olive O6HD Music Server

A Truly Outstanding Server in a Highly Competitive World

Anthony H. Cordesman

Let me begin with the punch line. The Olive O6HD offers very high sound quality, a simple way of storing large amounts of music, and ergonomics which may take a day or so of getting used to, but are straightforward and functional, and can be expected to improve over time. If you want a very-high-quality unit that does not involve fighting your way through a computer and is backed by first-rate service in loading your collection, this is an excellent choice.

BEFORE YOU BUY: THE UNCERTAIN VALUE OF A STAND-ALONE MUSIC STORAGE SYSTEM

Now, let me start to get complicated. I live with computers. I design and run complex databases covering major areas of national security, use computers to write books, and use them to communicate on a global basis. At the same time, I do not allow any purchase of developmental or error-prone software, or any system I have to fight to use. Anyone who pulls a "geek factor" around me is going to be fired, and the computer music storage and audio systems I have seen to date all fall into the far-too-difficult-to-justify-my-time category, and are so optimized around popular music that their handling of metadata are nightmares for those storing and using a large classical-music collection.

This brings me to the context in which the Olive O6HD, Sooloos, and similar systems must be judged. CD and SACD players without a digital input are already dinosaurs. In fact, I would not pay more than 1/15th of list price for any such unit—new or used. I would also question the competence of any manufacturer or dealer that keeps selling them. They do not compete in the present and they have no future. Unlike analog, no one is going to want to preserve CD-player sound quality.

There is broad professional and audiophile consensus that CDs sound better stored and streamed. High-resolution HD downloads clearly sound better than CDs and equal or surpass SACD and DVD-A. With servers, you can store the equivalent of thousands of CDs in a small box, and get better sound quality and easier and far more flexible access to your collection. Moreover, you can start shifting away from an obsolete 44.1kHz/16-bit standard to the much-better-sounding higher sampling rates that are now available from sources like HD Tracks.

The question, however, is whether you need a dedicated device like the Olive O6HD or Sooloos.

I put this challenge to Olive, and it made the following case:

"When we originally designed our product it was based on the idea that there is a certain target group that does not want to have a PC involved when listening to music (as opposed to someone who prefers to use a streaming client with a PC as the server). We have done extensive research with our customers trying to identify why they have bought our product over other solutions.

"The number-one quoted reason why our customers chose Olive was that they were not satisfied with the sound quality offered by PC/Mac-based solutions. Some of them have spent a considerable amount on physical media in the past and want a solution that allows them to listen to CDs in the original quality, and to access the growing HD music offerings in the market. They have dabbled with the iPod/iTunes solution, but deemed it only good enough for portable use, and not for their home stereo system.

"The second product category our customers analyzed before buying an Olive were streaming clients such as Sonos, SlimDevices (now Logitech), and others. Mostly they decided against this category due to the sub-optimal sound quality of these solutions, which is a result of low-end audio technology (DAC, power supply, no support for 24-bit/192kHz playback, etc.), but also caused by network limitations (drop-outs, bandwidth).

"Some of our customers also looked at a third category, namely using a PC/Mac and interfacing directly to their stereo system with a high-end USB DAC. They decided against this solution as the price/performance ratio (with some USB DACs) was not superior and that they did not want an ugly PC and cable-clutter in the living room.

"Both of the last groups also mentioned that they decided against a streaming client or

EQUIPMENT REVIEW - Olive O6HD Music Server

Remember! You can go to another section by clicking here.

PC+USB DAC solution due to its complexity. They had reservations about having to set up a network and/or install software on a PC (i.e., they wanted simplicity)."

I have to agree with many of Olive's points. From what I have heard to date, the answer for non-geek audiophiles is definitely to go to a dedicated music-storage system for sound quality, and generally for ergonomics and management and playback of a large music collection.

But this superiority is a matter of personal style and convenience, and the advantages in sound quality may be fleeting. I'm not a fan of USB options, but my sons already use cards with straight digital outputs. The storage and music-management software may now be awkward, but it will improve. Few audiophiles have the kind of home network and WiFi system I need professionally, but the number of home network users is growing geometrically, and IR remote-control systems are going the way of the Dodo as iPad-like devices and smartphones take over. It is just a matter of time before adequate music-storage and management systems are available, and manufacturers of high-end DACs tie their product to computerized music systems and remote controls. Some top firms like Boulder and PS Audio are already focusing on DAC/computer combinations.

This puts intense competitive pressure on stand-alone storage and streaming systems. This pressure is compounded by the fact that many users already are converting to audio and video streaming, and to systems with surround sound as well as stereo music storage. Moreover, there already are enough illegal Blu-ray Disc-streaming efforts to suggest that the future lies in both high-resolution music and video streaming and storage.

I would still go with a system like Sooloos or Olive, but (a) I can afford to; (b) I want the best in sound quality now, and I'm simply not prepared to wait a year or more to find out when computers catch up; (c) I have enough problems with hacking/cracking in running other databases to want stand-alone music storage; and (d) I'm perfectly happy to have a totally separate (and legal) audio-video surround-music setup.

At the same time, I feel that the trends in high-end audio mean that any stand-alone player like the Olive O6HD must meet the following tests: 1) It must have outstanding sound quality and be better than today's computer options as a DAC; 2) its ergonomics must be better in virtually every respect than competitive computer formats; 3) music storage must be practical for large collections; 4) there must be as simple an option as possible to deal with the nightmare of bad, partial, and missing metadata in CD recording (especially older classical, foreign CDs, and multiple CD boxes like long symphonies and operas; 5) reliability must equal or surpass a standard PC or Mac; 6) there must be a functional and affordable backup option; 7) there must be a non-proprietary upgrade path and an easy, error-free way to convert out of the storage system that

preserves all of the quality of the music stored, including high-resolution recordings.

Fortunately, the Olive O6HD meets the first six of these tests, and seems to be addressing the seventh.

SOUND QUALITY

There is no question about the sound quality of the Olive O6HD. No DACs or players sound alike, and each tends to be voiced to emphasize certain qualities. In broad terms, however, the sound quality of the O6HD is only surpassed by units that cost far more, and the sound of its analog outputs blows away that of the analog outputs of the Sooloos in every sonic respect.

The overall timbre of the Olive O6HD is very natural. The O6HD does an excellent job in reproducing the deep (and even the really deep) bass and upper bass; the midrange is well-defined without any touch of hardness in the upper mids; and soundstaging and imaging are about as realistic as a recording permits. The upper frequencies do not have all of the natural life and air of the very best DACs and CD/SACD units that cost more, but they are very good.

There is no exaggerated hardness in reproducing ordinary CDs; in fact, storing a CD consistently improves the realism of its sound without sacrificing detail. Moreover, the Olive O6HD clearly shows the superior quality of higher bit-and-sampling rates. The nuances and quality of high-resolution recordings from top labels like Reference Recordings and Chesky come through with only a touch less impact and information than the sound of the best stand-alone DACs.

The O6HD has a fully balanced differential DAC design, which seems to be an industry-first

for music servers. While the O4HD used one TI Burr-Brown PCM1792 24-bit/192 kHz DAC for both channels, the O6HD uses a matched pair of PCM1792 DAC modules for the inverted and non-inverted signals on each of the right and left channels. This considerably improves the signal-to-noise ratio (the Olive O6HD claims a signal-to-noise ratio of 124dB), and allows the music to come from the kind of noise-and-artifact-free, or "black," background I normally only expect to hear in much more expensive DACs.

SPECS & PRICING

Olive O6HD Music Server
Drive capacity: 2TB or 4TB
Interface: 10.1" touchscreen
Outputs: Unbalanced on RCA jacks, balanced on XLR jacks; headphone on stereo 6.4mm jack; digital output (AES/EBU) on XLR jack; HDMI (480p); USB 1.1/2.0
Features: Internet Radio, dedicated remote control, iPad/iPhone app available

U.S.
Price: $4999

OLIVE MEDIA INC.
555 Howard Street
San Francisco, CA 94105
(888) 654-8369
(415) 908-3870 (international callers)
olive.us

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EQUIPMENT REVIEW - Olive O6HD Music Server

I still feel the Meitner sets the standard in upper-octave, lower-midrange, and soundstage realism relative to any other unit I've auditioned so far, and that the Boulder 1021 and Meridian 808.3 Signature are close competitors that provide musically natural upper-midrange and treble detail that is superior to that of the Olive O6HD. I'd also choose the PS Audio Perfectwave DAC if I wanted more upper-octave detail.

But any such margins of superiority are very definitely in the diminishing-returns category. The Olive O6HD provides truly outstanding sound for substantially less money than the top DACs, and the O6HD stores and manages your music collection, to boot. It also really does merit comparison with the best. It handles dynamics with a realism and life that are often missing in the more "polite" DACs and players, and does so without adding the touch of hardness or softness that colors almost all DACs and players in its price range. Put simply, damned good, musically natural sound!

ERGONOMICS

The ergonomics of the Olive O6HD do take a little getting used to. Its instruction manual follows in the great high-end tradition of either being non-existent, damned near useless, or an exercise in post-modern, Dadaesque-deconstructionist, abstract minimalism. (I should note my Sooloos does not even have an instruction manual.) A day or two of experimentation, however, allows you to draw on all of its features in ways that quickly become second-nature. Moreover, even during the worst moments of my learning curve (and they were moments and hot hours), I never had the desire to kill that has characterized my experience with most complex electronics.

MUSIC STORAGE

The storage and handling features are good but not great. The O6HD's menu and control features are certainly adequate for most libraries, but the O6HD does not have the speed and detail that Sooloos and some computer management-systems provide. The ergonomics and display features of the Olive O6HD and its software are probably better suited to a "large" collection (up to 2000 CDs or albums), rather than an "ultra-large" (4000+) one.

The dedicated 10.1" touchscreen may be wide (and very attractive), but it is narrow and can't be seen at a distance. The album-cover display feature on the touchscreen is also more decorative than useful in rapidly seeking out recordings. Both the touchscreen and the front button controls are a bit slow to respond, and display relatively limited data compared to the Sooloos and some music-management software that runs on a PC.

It is hard to set up one-of-a-kind playback efforts to compare musical pages and different recordings, and the unit relies largely on playlists rather than a combination of playlists and one-time, easy-to-program queues. There is an option to get a readout of the exact sampling and bit-rate of recordings, but you have to create a playlist to select high-resolution recordings.

The Olive O6HD's ergonomics are less important if you use a device like an iPad (for which the Olive has an app) or the dedicated remote control and a TV screen to select and control your music. As a result, I'd plan to use the iPad for most music listening. It is quicker, and can be used anywhere in the house if you have WiFi. It allows you to see the cover art and status of a recording very clearly once you select it, and the app can almost be counted on to steadily improve with time. Olive seems to have open-ended growth potential in improving the iPad and TV options in the future.

As for storage, the Olive O6HD has a built in 2TB drive, which it claims can store 6000 CDs or 20,000 HD Tracks. This should be sufficient for most collections, although I have pushed over the 2TB limit with about 4100 CDs and some high-resolution tracks. Fortunately, Olive designs its products so they can be upgraded to larger internal drives as higher-capacity hard drives become available—and 4TB and 6TB standards seem likely in the next two years.

Olive does make a major effort to help you put your CD collection on the O6HD. It provides a service for storing your collection of CDs, and believe me, you do not want to hand-load more than a hundred or so if you can possibly avoid it. If you already have a computerized collection, it will load it automatically. If you are still in the plastic-box era, Olive offers buyers a service that allows them to send in their CDs and have Olive load them onto its HD music server. It will rip the first 100 CDs for free, and each additional CD is ripped for as little as $0.50 (depending on the size of the library). Olive uses special pre-load robots to rip each CD with multiple error-

TECHNICAL NOTES

Available in black and silver finishes, both the i-1 and CD-1 are cosmetically minimalist and quite solidly built. The i-1's feeling of power-in-reserve comes from the 320VA toroidal transformer and 20,000mF worth of capacitance that occupies most of the interior left side. The preamp section utilizes JFET inputs, while the amplifier uses but a single pair of bipolar output transistors per channel. The back panel offers five line-level inputs, while the faceplate provides an eighth-inch mini-jack for personal media players and a quarter-inch headphone jack. The all-aluminum chassis was designed to be rigid and vibration free, and the i-1 was designed to be powered up at all times, to ensure consistently optimal sound, but also to operate at low temperatures for a lengthy operational life.

Built on what appears to be the identical all-aluminum chassis, the CD-1 boasts a proprietary CD transport mechanism, as well as Simaudio's in-house software and hardware technologies. Burr-Brown PCM1793 DACs provide 24-bit/192kHz decoding and 8x oversampling, while an RS232 port provides communication links in custom-install situations and for any firmware updates. Like the i-1, the CD-1 was designed to be powered on at all times. And while the red LED front-panel display is nice and large, I wish Simaudio allowed for the thing to be dimmed or shut off. It's very bright in a darkened room at night, so when I was not using it, I put the unit in the "standby" mode. **WG**

EQUIPMENT REVIEW - Olive O6HD Music Server

correction. Olive reports that some 67% of its U.S. customers use this service.

As for Internet radio, the O6HD's features are good. Olive uses its own database (not Shoutcast or VTuner etc.). This allows it to groom the metadata and make sure the information is displayed correctly on the O6HD display (and it can show extended metadata about the station). Olive also "pings" all HD radio stations frequently to make sure that stations that are off-line are not shown in the list (a complaint that Olive says it has heard from customers about other Internet radio solutions).

METADATA

Olive does more than simply directly store the metadata on the CD. It pulls the metadata used to store, describe, and manage music from several databases to ensure the highest accuracy. Olive also provides what it calls "manual grooming." This can be critical because no one really wants to edit his entire collection CD by CD, and about half of my classical and 20% of my older jazz CDs needed some editing, and roughly 90% of my classical CDs needed additional editing so that I could sort by composer using uniform spellings of last names.

Instead of simply dumping the faulty or missing data on the CD, Olive tries to correct the CDs of its customers during the CD-loading process and make sure that all metadata is written correctly and uniformly (especially composers). This means that classical CDs will then show up correctly in the composer category and album view. Olive also is preparing to sell digital HD music (e.g. from Reference Recordings) and plans to extended metadata, which is not available through standard CD databases. Moreover, buyers will be

able to preload certain HD collections when they buy a HD music server, or have them delivered afterwards by USB stick.

Do not, however, expect miracles in fixing the metadata, particularly with CDs of older performances on small and amateur labels. If you are a truly serious music collector, and demanding about spelling, you will still need a computer and have to edit the metadata. Olive provides a link to Maestro to provide a program do this, and it works. I could not, however, find the kind of global-search-and-replace routines I would have liked. (They may be there, but they are all Geek to me!)

RELIABILITY

I simply have no way of telling how reliable the Olive or any similar unit will be over time. I did, however, have only one minor glitch during start up, after which the unit then worked perfectly during the review period. This is good performance for any truly complex electronic device, and I wish my Sooloos had proved as reliable.

As for service, Olive does offer a 30-day return-and-refund option. It also offers a two-year parts-and-service warranty for the O6HD. Given the uncertainties involved, particularly in a proprietary device, I believe two years is the minimum warranty such products should have, but I'm not aware that any other manufacturer offers more than Olive. And many offer less.

BACK-UP

The back-up system works well, and can be used with any high-capacity hard drive. My only reservation is that Olive understates the need for such a back-up, and does not recommend

specific hard drives it finds suitable and reliable. Buying a back-up hard drive is an absolute must, given the cost and difficultly of loading and editing your music collection, and the tendency of a number of high-capacity drives to die just beyond their warranty date.

Every manufacturer of such products must offer a non-proprietary upgrade path and an easy way to convert from the storage system used in a device like the Olive O6HD to another storage device, while preserving all of the quality of the music stored, including that of high-resolution recordings.

Olive has shown it has an upgrade path for its own products. Olive has regularly offered its customers the option of upgrading their stored music from products like the O2HD and O4HD to new Olive products like the O6HD at a nominal fee. In the past, the cost has been included in the price difference to the new product, but it charged a fee for such conversions in its last promotion.

Olive was less clear about what would happen if a customer wanted to transfer a collection to a different storage media or manufacturer system, and noted the problems copyright law present for simply providing a direct copy. After some back

and forth, however, Olive made it clear that it will offer a service to download a returned Olive unit to a standard digital format like FLAC, WAVE, MPEG, AAC, Apple Lossless, etc.

I hope that Olive will clarify this offer in writing in its comments on this review, and I would be extremely cautious about buying any music server or storage system that did not offer such options. Technology is moving too quickly not to be able to transfer your collection from one system to another in a standard format.

Speaking personally, I would not buy a unit that did not have a clear option for transferring its collection, and I prefer systems that can easily download a music collection directly into a replacement system. Given that proviso, I'd strongly endorse the Olive O6HD within its price range. It is not perfect, but the sound quality is exceptional and its operating and storage features, while far from perfect, are never less than good. I give a strong recommendation to anyone who does not speak Geek as a native tongue. tas

EQUIPMENT REVIEWS

Integrated Amps with USB DACs

Making iPods and PCs "Sing"

NuForce Icon Mobile Battery-Powered Headphone Amp/USB DAC

Chris Martens

Probably the first two questions most people ask me about the tiny NuForce Icon Mobile are "what is it?" followed shortly by "why would I need one?" The answer to the first question is that the Icon Mobile is a compact, rechargeable, battery-powered headphone amp/USB DAC. And the answer to the second question is that a device like this can help you achieve significant improvements in sound quality from your iPod, iPhone or PC, opening up new levels of performance and musical enjoyment.

USING THE ICON MOBILE

Since a number of people have asked me why one would need a device like the Icon Mobile, or how such devices are used, I thought I should describe two of the most common real-world use scenarios one might encounter.

THE PORTABLE SCENARIO

Say you've got an iPod, iPhone, or other portable music player plus a nice set of headphones (in-ear, on-ear, over-the-ear, etc.) but feel your player isn't tapping the full potential of your 'phones. Now imagine that instead of powering your 'phones directly from your player, you instead run a "jumper cable" from the headphone jack on your player to the input jack on the Icon Mobile and plug your 'phones into the Icon Mobile. Voilà, the NuForce now does all the heavy lifting in terms of powering your headphones giving you an immediate and obvious improvement in sound quality in the process. What is more, the Icon Mobile also gives you the option of using serious, high-performance, audiophile-grade headphones—headphones your portable player would not be capable of driving effectively on its own. Cool, no?

THE DESKTOP SCENARIO

You've got lots of music files stored on your PC or Mac and like to listen through headphones but find that sound quality isn't as good as you'd hoped it would be. Now imagine that instead of plugging your headphones into a jack on your computer and listening through whatever DACs and amplifier circuits the PC provides, you connect the Icon Mobile to a USB port on your computer and let it both decode your computer audio files and amplify the audio signals to drive
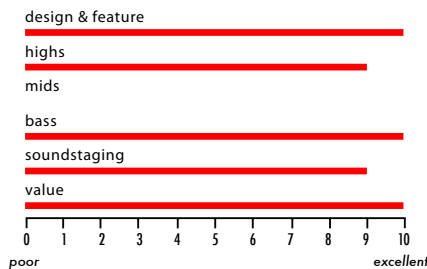
EQUIPMENT REVIEW - NuForce Icon Mobile Battery-Powered Headphone Amp/USB DAC

OVERVIEW

Consider this portable headphone amp/DAC if: you like the idea of an affordable, compact, rechargeable headphone amp/DAC that can make just about any in-ear headphone sound better than it ordinarily would and that has enough oomph to drive big, over-the-ear headphones, too. The Icon Mobile's USB DAC does a much better job of decoding computer audio files than most computers.

Look further if: you find it a pain to use add-on devices with your computer or portable digital music player (the Icon Mobile is not for everyone). Also look further if you require very long battery life per charge or seek products that have an elegant, upscale look and feel (certain higher-end competitors look/feel better than the Icon Mobile, and can play longer on a charge, but they cost quite a lot more).

RATINGS

design & feature		
highs		
mids		
bass		
soundstaging		
value		

0 1 2 3 4 5 6 7 8 9 10
poor   *excellent*

your headphones. Once you insert the Icon Mobile into the equation, you'll hear a very impressive jump in apparent resolution and focus (as if your music files themselves suddenly got better) and other sonic improvements. Once again, the Icon Mobile lets you plug in full-size, high-end headphones that your PC alone would not be capable of driving well.

Are you starting to catch the vision for what a good portable headphone amp/USB DAC can do? I certainly am.

FEATURES

- Compact chassis (about the size of a business card case) with an anodized, brushed aluminum top plate (in black, silver, blue, or red), with a wraparound rear housing made of soft-feel, no-mar plastic
- Controls:
  - High-quality, thumbwheel-style volume potentiometer
  - Recessed, two position gain switch (high gain is for full-size headphones, low gain is for in-ear headphones and headsets)
- Four status indicator lights:
  - Power On (a blue LED lights whenever a headphone is plugged in and the amp/DAC is thus turned on)
  - USB (a white LED lights when a USB connection is established, and blinks when the Icon Mobile's USB port is receiving data)
  - External Line In (a yellow LED lights when a line-level analog device has been plugged in to the Icon Mobile's analog input, and blinks if the input device is a headset whose microphone is muted)
  - Battery (a two-color LED lights up in green when the battery is charging, turns off when the unit is playing and the battery has power remaining, and lights up in red when the battery is low and needs to be charged)

- Two audio inputs:
  - USB 2.0 input jack
  - Line-level analog input via 3.5mm mini-jack (the analog input takes precedence over the USB input when both are connected)
- Two headphone outputs:
  - Dual analog outputs via 3.5mm mini-jacks, with one jack supporting 4-pin iPhone style headsets
  - Both output jacks can be used simultaneously
- USB DAC/ADC
  - USB port is USB 2.0-compatible
  - USB DAC supports 44.1kHz and 48kHz native sampling rates
  - Microphone input (for headsets) provides 16-bit ADC (analog-to-digital converter)
- Built-in Lithium Ion battery
  - Charges in 2.5 hours via USB port
  - Runs for 13 hours (at maximum power output) per charge
- Accessories:
  - A soft silicone band allows users to strap the Icon Mobile to an iPod, iPhone, or other portable player
  - High-quality cables, one mini-jack to mini-jack cable and one USB cable
  - Small non-conductive screwdriver allows uses to adjust the Icon Mobile's recessed gain control switch

SONIC CHARACTER

Many people think amplifiers sound more or less the same (or at least very similar), but in my experience, that isn't necessarily true—especially when you compare the intensely cost-constrained amps that are included in portable music players or PCs versus a more focused, dedicated design such as the Icon Mobile provides. In short, you can expect to hear readily discernible improvements with the NuForce in play.

The Icon Mobile headphone amp offers deeply extended yet very taut and well-defined bass and an impressive ability to delineate and resolve very fine, low-level midrange and treble details. Together, these qualities add up to a sonic presentation that is at once refined yet also robust and full-bodied. Note: you may find, as I did, that the Icon Mobile exhibits a touch of midrange/lower treble forwardness, which can be mitigated

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EQUIPMENT REVIEW - NuForce Icon Mobile Battery-Powered Headphone Amp/USB DAC

by giving the little amp/DAC some "run-in" time. After some burn-in, its sound becomes smoother and more neutrally voiced.

Does the Icon Mobile have enough grunt to drive power hungry full-size headphones? In my experience it certainly does. I tested the Icon Mobile with an oldie-but-goodie pair of full-size Sennheiser HD-580s (predecessors to the current generation HD-650s) and found that the Icon Mobile drove them easily and effortlessly.

While the Icon Mobile represents a worthwhile step forward when you use it purely as an amplifier, it brings even greater benefits when its built-in DAC is brought in to play—as would be the case if you plug into your PC or Mac. My main office computer is a current generation iMac, and

BOTTOM LINE

NuForce's Icon Mobile makes an affordable, versatile, and effective vehicle for those who would like to take their portable or desktop headphone listening experiences to the next level. Remember that to hear the unit at its best, you really must try it with your PC so that you can bring the NuForce's surprisingly good USB DAC into play. Potential buyers will, I think, appreciate the fact that the Icon Mobile puts primary emphasis on strong sonic performance at a reasonable price—not on flashy, "boutique" styling. The Icon Mobile, though attractive enough in its simple and straightforward way, is really one of those products where "beauty is as beauty does."

I tried listening to high-quality in-ear headphones (e.g., the Klipsch Images and Sennheiser IE7s), first through the Mac, and then through the Icon Mobile. With the NuForce connected, I enjoyed superior resolution of delicate musical textures and details and tighter yet also more potent bass. In fact, the illusion was that my digital audio files had magically improved in quality, though in reality the files stayed the same while the DAC and audio circuits got better—together revealing more of the music information that had been stored in the iMac all along.

MUSICAL EXAMPLE

One of the more impressive characteristics of the Icon Mobile involves its ability to make well-recorded tracks sound even better than they normally do. One such example would be the classic Tom Waits song "Falling Down" as performed by Holly Cole on *Temptation* [Metro Blue]. Right off the bat, you'll notice an increase in the delicacy, purity, and vividness of individual instruments and voices—especially the soulful, reedy moan of Howard Levy's harmonica and the melancholy, baritone rumble of David Piltch's acoutic bass. The subtle high frequency "splish" of gently struck cymbals and the "snap" of snare accents also sound terrific, perfectly complementing Cole's voice, which alternates between a lilting croon and full-throated

crescendos on the song's distinctive chorus lines.

One might reasonably ask, "Aren't these sonic qualities always present in this recording?" and the answer is that they are, but to a much less explicit and detailed degree. The difference the Icon Mobile makes is a little like the difference between a well-rendered drawing versus a photograph. Both try to convey the same shapes and spatial relationships, but the photograph (and by extension the sound of the Icon Mobile) offers much finer shadings and representations of finely filigreed details. The result is a musical presentation that invites you to explore the inner contours of each sound, instrument, and voice. tas

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Very Nicely Done

Hegel Music Systems H200 Integrated Amplifier

Kirk Midtskog

When I turned in my review of the smaller Hegel H100 integrated amp some time ago, I mentioned to Editor-in-Chief Robert Harley how much I also liked the H200. When Robert asked if I would like to write about the H200, I didn't hesitate. The H200 is a piece of audio gear that I can approve wholeheartedly.

Integrated amplifiers embody the "get the job done with the least cost and fuss" approach, and that appeals to my practical side immensely. Yes, there are drawbacks, such as the close presence of stray magnetic fields of the power-amp section interfering with the delicate circuits of the input or preamp section. There is also the prospect that airborne acoustical vibrations from loudspeakers, which are usually located fairly close to an amplifier, will affect the preamp section. On the other hand, a fine integrated amp is a perfectly legitimate way to hear good music in your home—even if you can afford more expensive separates. If you don't go the separates' route, the added cost of additional chassis, power cords, and interconnects can be applied to better speakers and cables—which may, in fact, yield better overall performance than spending more for a separate preamplifier and power amplifier.

Like the H100, the H200 has just four items on its elegant, slightly curved faceplate: an input-selector knob, a volume knob, a power button, and a status window. Slightly taller than the H100, the H200 has the same understated look; the feel of its well-proportioned anodized-aluminum faceplate and knobs are in keeping with the theme of simplicity. Most users tend to intuitively feel more at ease with straightforward interfaces. Hegel seems to understand this. When the unit powers up, it automatically sets the volume to 30 (out of 99) and defaults to the balanced input. This allows you to confirm that there is some signal (after selecting your target input) without damaging your speakers or ears. If you activate the mute, upon resumption it will quickly ramp up the volume from 30 to the previous volume setting, thereby giving you a chance to reactivate the mute if you happen to be playing a passage that is too loud—again, saving your ears and speakers. The blue display shows the volume setting and two-

characters (upper and lower case) to indicate the active input. Select input and volume, and you're ready to go.

Hegel designs and builds in Norway, a place not well known for manufacturing—although there is more than just small-scale industry and oil-resource-related commerce there (Volvo makes extreme tolerance jet engine parts in Norway, for example). In this era of wealth shifting from the West to the East—mostly through high-volume, low-cost manufacturing in Asia—Norway-based Hegel has still managed to grow since the late 1980s. By 2000, it was exporting CD players, DACs, integrateds, and separates to European and Asian/Pacific markets, and has recently expanded into the U.S. by continuing to offer gear that provides plenty of bang for the U.S. buck. Hegel pulls this off by using effective technologies, executed with cost-effective parts, to keep prices down.

For your hard-earned $4400, you get a clean-sounding, powerful 200W (350W into 4 ohms), remote-controlled integrated amplifier with just enough input and output pairs to be useful but not cluttered: one fully balanced XLR input, two single-ended RCA inputs, a power-amp input (or home-theater bypass), a tape-loop output, two preamp outputs for external amplifiers, and heavy-duty, gold-plated multiway speaker-cable binding posts. The H200 does not have an onboard DAC like the smaller H100. The H200 has beefier power supplies and higher-quality parts, and it electrically and physically separates the two channels (dual-mono) more completely. The H200 is more of a purist integrated whose emphasis is on optimal sonic performance and power reserves rather than on many different inputs and features. When I met Hegel's chief designer Bent Holter at the Rocky Mountain Audio Fest in October, 2010, he said, "The H200 is

EQUIPMENT REVIEW - Hegel Music Systems H200 Integrated Amplifier

the cornerstone of the whole Hegel product line." Apparently, it is Hegel's best-selling product.

Hegel uses a patented circuit called the SoundEngine that is its primary claim to improved performance through engineering. The SoundEngine is said to reduce distortion by isolating noise between various gain stages before it is passed on and amplified by subsequent stages. This is done by a "feed-forward" technique that compares the in-phase input of a given stage with the corresponding out-of-phase output and rejects what is determined to be noise, and so it goes on down the line with each stage. The SoundEngine is different enough in its particulars to qualify as a patentable solution and is not, apparently, like a traditional fully-differential balanced circuit. The preamp section and the input of the power amp provide the voltage gain and have their own power supplies. The main gain stages of the power-amp section increase the current, and also have their own power supplies. Keeping things separate, according to Holter, also reduces distortion brought about by large, simultaneous voltage and current swings in the two respective stages. By reducing interstage distortion and isolating voltage and current stages as much as possible, Hegel claims it is able to deliver Class A-like fluidity without having to bias the output devices at Class A levels—which would draw lots of AC power and dissipate lots of heat as wasted energy. The H200 becomes only mildly warm to the touch after several hours of play.

We have all heard manufacturers' marketing claims about new technologies, or superior parts quality, or better mechanical isolation, or maniacal attention to detail, but we have

to listen to the component in question to get an idea of what the results might be. In this case, Hegel has every right to be proud of its accomplishments. The H200 sounds clear, pure, relaxed, well-balanced, and extended at both frequency extremes. In performance areas that could lead some listeners to at least consider alternatives to the smaller H100—such as a less-than-cavernous soundstage width, an emphasis on airiness instead of fullness, and a buttery Class A quality—the H200 pretty much does away with any reservations. The H200 casts a wider and deeper soundstage—one that is populated with even clearer details—and it balances the qualities of light and airiness with rich fullness just beautifully. The H200 also has better bass extension and precision than the H100. (I use eight-gauge Wegrzyn power cords on my amplifiers; so I just transferred one of the Wegrzyns to the H200. Mileage will vary according to many factors.) While the H200 still has just a hint of that pure Class A creaminess, it only comes across as such when compared to my $20,000 Ayre and GamuT combo. On its own, I am not convinced I would notice it.

The odd thing about a lot of what the Hegel does is that it may actually not grab your immediate attention. Many of us have learned to listen for particular electronic artifacts as signifiers of elevated performance. A listener who heard my system a few years ago commented that he could not hear some of the background tape hiss in the recording that he was pretty sure he should hear more clearly. I later heard his system. To me, it had an overly processed, exaggerated top end. This echoes the "different strokes for different folks" principle when it comes to listening

priorities, but it also illustrates what I would call a focus on artifacts rather than listening simply for the unadulterated reproduction of music. I hear the H200's clean, relaxed quality very much as a positive. Absent are the slightly ragged edges around images and that little bit of extra zip on details that some listeners may mistake for "heightened resolution." The H200 does not reduce some of those electronic artifacts by covering them up through veiling or rolling off the high frequencies, nor is listening to it a boring or uninvolving experience. On the contrary, it is actually quite resolving and very musically engaging with plenty of dynamic drive at the same time. It just seems to reduce noise and allow more of the signal to come through without sounding stressed or forced.

How this all translates into a listening experience with the H200 is that musical details are always integrated into a larger picture rather than being hyped up so that your attention is unduly drawn to them. Via the H200, I tended to listen to music as a whole experience, the flow of the tune, the drama of the piece, the artistry of the musicians, rather than listening to imaging, extension, dynamics, and so forth. Those elements are more than satisfactorily covered—large soundstage, lots of space, fine images, etc. But the great bonus here is that the H200 actually plays music in a way that allows you to forget about audio evaluation and get involved as you might in a live concert. This is not to suggest that the H200 gets you close to live music in all aspects (that would be a tall order); it just seemed to shift my listening priorities away from "system listening" and more toward the musical performances.

Images boundaries are clear with a sense of

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EQUIPMENT REVIEW - Hegel Music Systems H200 Integrated Amplifier

solidity behind them, but they don't exist on their own; they are part of the space in which the music was recorded—recording permitting, of course. I don't think about image boundaries or pinpoint location when I attend live performances. Not just because I can literally see the musicians at a live concert, but also because the overall sound of musicians combining to make music generally includes a wash of direct and reflected sounds. I have heard a solo oboe playing in the middle of an orchestra and found it somewhat difficult to actually pick the musician out, even when looking directly at the wind section. Similarly, the Hegel H200 balances individual image information within the soundstage very well.

To transpose this "greater whole" aspect to non-acoustic music, studio pop and rock recordings are rendered a bit more clearly and less fatiguingly than one would expect at the H200's price level. The Hegel allows you to listen into the recording without a sense of discomfort when aggressive sections kick in. I could actually rock out with Tool's "The Grudge" [Lateralus, Volcano] even as Danny Carey's crushing cymbals and Adam Jones' gritty guitar really ramped it up toward the end. Similarly, on Alanis Morissette's "A Man" [Under Rug Swept,

Maverick], not only was her voice not as shrill as it can be on some systems, but also when drummer Gary Novak switched from his high-hat to keeping the beat on his ride cymbal for the rest of the song (about 03:24 mark), the song gained more momentum and became more exhilarating without sounding stressful. Again, I had no sense that things were made more listenable because of some obvious content omission; rather, recordings were rendered in a less harsh, noise-ladened way.

Compared to the April Music Stello Ai500 (150W, $3500), the H200 sounded more refined and had better bass control and extension. The Ai500 had good soundstage width but could not equal the H200's portrayal of depth and soundstage continuousness in all directions. In this regard, and in its touch of Class A-like liquidity, the H200 performs more like a good tubed amplifier than a typical solid-state integrated amp. Compared to my own Ayre K-1xe preamp and GamuT

M200 mono amplifiers, the H200 holds its own fairly well. The Ayre/GamuT combo ($20k) has more solid, more commanding presence than the H200. The Ayre/GamuT combo also expands the soundstage a bit and has better overall resolution. My separates have quick immediate presence, whereas the H200 has a more relaxed quality. In some ways, the H200 comes across as a bit more consistent in the way it tracks notes as they start, propagate, and decay than my pre/power combo. The combo tended to render the leading edges and then propagate steady-state tone just right, but doesn't quite hang on to the decay as well. Again, the H200 was somewhat reminiscent of a fine tubed amp in this regard. Could the H200 be even clearer, quicker, more expansive? Yes. Would I give up my separates for the H200? No, but I could easily live with it. It plays all kinds of music fundamentally well enough for me not to worry about the "could be

better here and there" stuff.

That sums it up well: I could live with the H200...happily. I submit that, on a limited budget, you are better off buying the H200 for $4400 and having more left over for speaker and cable upgrades (and maybe some room treatments) than stretching your budget to get into more expensive separates whose purchase would greatly restrict your speaker and cabling choices. The H200 strikes me as a fantastic performer, one that delivers sonic quality far beyond its price level. Nicely done, very nicely done. **tas**

SPECS & PRICING

Hegel Music Systems H200 Integrated Amplifier
Power output: 200Wpc
Inputs: Two unbalanced (RCA), one balanced (XLR), and one HT/power-amp (RCA)
Outputs: Two preamp (RCA), one record (RCA), speaker terminals
Dimensions: 17" x 4.7" x 14.5"
Weight: 55 lbs.

U.S.

Price: $4400 (includes RC2 remote control)

HEGEL MUSIC SYSTEMS
(U.S. DISTRIBUTOR)
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ben@hegel.com

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Peachtree Audio iDecco Integrated Amp/DAC

Chris Martens

When I reviewed the Peachtree Audio Nova integrated amplifier/USB DAC ($1199) in Playback 21 I was very favorably impressed and called it "a versatile, well-conceived and well-executed product that fulfills a number of roles with astonishing refinement, polish, and panache." Let me begin this review of Peachtree's new iDecco integrated amp/USB DAC/iPod Dock by telling you that the iDecco offers much of the flexibility and essentially all of the sonic excellence of its big brother, while adding a digital (not analog) iPod dock and selling for—get this—an even lower price ($999).

To really get what the iDecco is about, it is important to understand that it is really five different products in one:

- A hybrid solid-state/vacuum tube (6N1P) preamplifier with a Class A output buffer stage.
- A Class A headphone amplifier (when the tube circuit is enabled).
- A multi-input (USB, S/PDIF, TosLink, iPod) 96kHz/24-bit upsampling DAC with a solid-state output stage.
- A digital iPod dock with, of course, a built-in integrated amplifier.
- A 40Wpc MOSFET-powered, solid-state integrated amplifier.

Now all of the features and functions in the world aren't worth much unless they are well executed, and happily solid execution is one area where the iDecco really shines. As you'll see in a moment, the iDecco is a seriously refined audio component that is so good at each of its several roles that you might willingly pay its asking price to enjoy any one or perhaps two of them. But bundle all five functions together, throw in a generous helping of sonic sophistication, and the iDecco's value proposition skyrockets, pushing it up into "screamin' good deal" territory.

FEATURES

Preamplifier

Digital Audio Inputs: The iDecco incorporates an onboard "switching" DAC with four switch-selectable digital audio inputs (USB, Coax, TosLink, and the iPod dock), whereas the Nova provides five digital inputs (but no iPod dock).

iPod Controls: The iDecco remote control has dedicated buttons that allow users to control an iPod plugged into the iDecco's dock.

Analog Audio Inputs: The iDecco also provides a single stereo analog input, whereas the Nova provides three sets of inputs.

Analog Audio Outputs: The iDecco's preamp section offers both variable and fixed level analog outputs (the fixed outputs are driven by solid-state circuitry only), making it easy for you to use the Nova to drive an outboard power amplifier or subwoofer, if you wish.

Solid-state/vacuum tube switching: The iDecco preamp section is based on a circuit that is virtually identical to the one used in the Nova. As with the Nova, the iDecco remote control provides a switch that is simply labeled "Tube." When the iDecco is used as a preamp, integrated amp, or headphone amp, you can use the Tube switch to engage a Class A vacuum-tube-based output stage. Interestingly, the tube used in the iDecco is a 6N1P rather than the 6922 tube used in the Nova. Jim Spainhour of Peachtree audio explained that the 6N1P is essentially a higher-current version of the 6922, meaning that—if you wished to do so—you could substitute a 6922 tube in place of the iDecco's standard tube (but note: Spainhour advises that Nova owners cannot substitute a 6N1P tube in place of the Nova's standard 6922, since the Nova circuit is not set up to handle the 6N1P's current requirements). A blue LED illuminates the vacuum-tube viewing window in the iDecco's front panel whenever the tube output stage is engaged. One subtle yet significant difference between the Nova and the iDecco is that the latter uses a slow ramp-

EQUIPMENT REVIEW - Peachtree Audio iDecco Integrated Amp/DAC

OVERVIEW

Consider this amplifier/DAC/dock if: You like the idea of getting the sonic refinement and versatility of the Peachtree Nova, plus digital iPod dock functions, but at an even lower price (though you will step down from the Nova's 80Wpc amp to a smaller 40Wpc amp in the iDecco). The iDecco makes good sense in several contexts. You might use it as a fine standalone DAC, headphone amp, or preamp, or you could employ it is as an incredibly flexible front-end component upon which to base a superb yet sensibly priced high-end audio or computer-based music systems. One important note: Given its moderate power output, the iDecco shines brightest when driving smaller and/or higher sensitivity loudspeakers.

Look further if: You plan to use power-hungry, low-impedance, or otherwise hard-to-drive speakers (e.g., Magnepans). While the iDecco amplifier section is quite good, it simply does not have the muscle to handle those kinds of workloads. But note: As you'll see in this review, the iDecco can work beautifully when used as a DAC/preamp driving higher-end power amps.
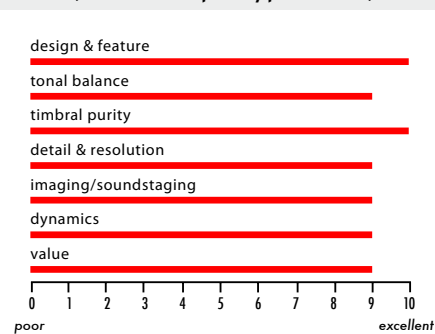
RATINGS

(relative to comparably priced integrated amps)
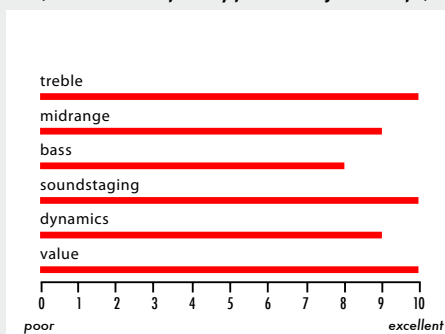
RATINGS

(relative to comparably priced DACs)

up circuit whenever the tube is brought into play. This allows you to hear a more gradual transition from the iDecco's solid-state sound to its tube sound.

Top-mounted digital iPod dock

The iDecco incorporates a top-mounted digital iPod dock that is similar in concept to the dock provided in the Wadia Digital Model 170 iTransport. By pressing the iDecco's iPod input selector button, users can send digital audio datastreams from iPods directly to the iDecco's built-in 96/24 upsampling DAC. Note that there is no digital audio signal pass-through since Peachtree's working assumption is that you will want to use the very high quality DAC built into the iDecco, rather than an outboard DAC.

A set of component-video outputs is provided on the rear panel of the iDecco, should you wish to play video content through your iPod.

24-bit/96kHz Upsampling DAC

Remote controlled input switching (four inputs: USB, Coax, TosLink, and iPod).

Features ESS 9006 Sabre DAC chip, which incorporates a patented jitter reduction circuit.

11 regulated power supplies for the DAC.

Transformer coupling for all digital inputs for ground isolation.

A "galvanically isolated" USB stage.

USB input accepts data at 16-bit/44.1kHz resolution, but upsamples to 24-bit/96kHz.

Rear-panel switch allows users to select "Fast" or "Slow" DAC filter slopes (the "Fast" setting generates better lab measurements, but Peachtree says many audiophiles prefer the sound of the "Slow" setting).

Rear-panel switch allows users to select "Narrow" or "Wide" jitter adjustments for the S/PDIF (coax) and TosLink (optical) digital inputs. According to Peachtree,

the "Narrow" setting sounds better with low-jitter sources, while the "Wide" setting is recommended for use with high-jitter sources.

Signal/noise ratio of 122dB.

Jitter: <1 picosecond as measured at the Master Clock (Super Clock).

Headphone Amplifier

The iDecco can be used as a standalone headphone amplifier with a Class A tube output circuit (the headphone amplifier shares circuitry with the iDecco preamp).

As a thoughtful detail touch, the iDecco is set up so that, when a headphone plug is inserted into its headphone jack, the iDecco's power amplifier section automatically disengages, effectively muting the speakers.

Amplifier

The power amplifier section of the iDecco is based on stereo MOSFET devices and puts out 40Wpc.

Art deco-inspired styling

Like the original Peachtree Decco and the Nova, the iDecco's faceplate features gently rounded corners and a rectangular "viewing window" through which you can see the iDecco's glowing 6N1P vacuum tube. The amp is housed in a svelte, round-edged sleeve finished in black lacquer, which gives the iDecco a decidedly upscale appearance.

SONIC CHARACTER, PREAMP/HEADPHONE AMP

Since the iDecco preamp and headphone amplifier share common circuitry, my comments here will apply to both functions.

Not surprisingly, the iDecco preamp sounds nearly identical to the Nova preamp. In fact, when used in solid-state mode they do sound identical, so that minor differences between the tube circuits of the two products

EQUIPMENT REVIEW - Peachtree Audio iDecco Integrated Amp/DAC

can likely be attributed to differences between the 6N1P and 6922 tubes. To my ears, the iDecco—with tube circuitry engaged—sounds perhaps a hair sweeter in the treble region, with a bit more harmonic bloom and greater three-dimensionality than the Nova, though some might interpret the Nova's sound as being a touch cleaner and therefore slightly more accurate. In any event, sonic differences between the preamp sections of the iDecco and the Nova are small.

I compared the solid-state versus tube sound of the iDecco and found that, as with the Nova, the solid-state output section was very clean, but also somewhat less rich, three-dimensional, and involving than the tube circuit. Candidly, if I owned the iDecco I would leave the tube circuit engaged probably 95 percent of the time. For this reason my comments, below, refer to the sound of the iDecco preamp with the tube circuit in play.

The core sound of the preamp has three defining characteristics. First, the preamp offers excellent clarity, with plenty of focus and definition. More so than many products at its price point, the iDecco offers lots of resolving power, meaning that it handles low-level textural, transient, and especially spatial or soundstaging cues in the music with remarkable acuity. Second, the preamp delivers bass that is very tight and well controlled, exhibiting none of the looseness or sloppy romanticism you might hear in other affordable tube preamps. Finally, the iDecco preamp does a great job of capturing the sheer richness of tonal colors in the music—in this respect sounding much more like an expensive standalone vacuum tube preamp, rather than an inexpensive integrated amp/DAC.

In my tests, I used the iDecco preamp to drive

a pair of NuForce Reference 9 v.3 Special Edition monoblock amps and though the power amps cost many times what the iDecco does, the Peachtree did not seem at all out of place. On the contrary, the match seemed a very good one, with the two products playing off of one another's strengths in a beautiful and musically satisfying way. But one thing the wide-bandwidth NuForce amps did reveal—and please consider this a minor nit—is that there is a bit of noise produced when switching between the iDecco's various inputs (or when turning the tube output stage on or off).

But let me be clear: Though there is obviously more to the iDecco than its preamp section, I would be very hard pressed to name a preamp at the iDecco's price that I would rather use in a high-end system. It's that good.

SONIC CHARACTER, DAC

When used as a standalone DAC the iDecco, like the Nova, provides solid-state outputs only. For the most part, the strengths of DAC parallel those of the iDecco preamp. The DAC resolves low-level sonic details beautifully, which helps the DAC create highly believable, three-dimensional soundstages. Through the Peachtree, for example, you'll hear long reverberation tails on individual sounds and can easily hear how those sounds interact with the acoustics of recording spaces. The DAC also captures both large and small-scale dynamic contrasts very effectively, letting listeners not only hear but also feel the living, breathing pulse and flow of the music.

If your reactions are anything like mine, you may be struck by the fact that the iDecco DAC doesn't conform to your mental image of a budget DAC.

In fact, it doesn't really sound like a "budget" anything, because it produces the sort of big, richly textured, wide and deep soundstages that are traditionally the hallmarks of higher-end audio components. In short, the iDecco DAC offer overall levels of sonic refinement and acuity typically experienced with DACs that cost as much as, if not more than, the entire iDecco does.

I switched back and forth between the iDecco's "Fast" and "Slow" filter settings and found that the "Fast" setting seemed to sap some of the iDecco's typical dynamic vividness and sense of life. The "Slow" setting, on the other hand, restored a more detailed and dynamically responsive sound.

I switched back and forth between the iDecco's "Narrow" and "Wide" jitter adjustment settings and found the "Narrow" setting gave a clearer and more focused sound. The "Wide" setting has a slightly softened and perhaps more forgiving sound that is appealing in its way, but a sound that also limits the absolute accuracy and vividness of the overall presentation. That said, I could see how the "Wide" setting might be just the ticket when using the iDecco with high-jitter sources.

Among DACs that provide both USB and S/PDIF inputs, the common wisdom is that the S/PDIF inputs will always sound better than the USB inputs, and in most cases the common wisdom holds true. But frankly, the iDecco DAC really surprised me in that its USB and S/PDIF inputs sounded essentially the same, which is pretty remarkable. I did numerous back-to-back comparisons, first feeding full resolution WAV files via USB to the iDecco, and then playing the same musical content via CDs in my reference

disc player and sending the resulting digital audio streams to the iDecco's S/PDIF inputs. The sonic results were so similar that I couldn't reliably characterize substantive differences (if any) between them. I've never had that happen when comparing USB and S/PDIF inputs in a DAC before.

During my listening tests, I compared the iDecco DAC both to a PS Audio Digital Link III DAC ($995, but currently offered at the special price of $700 in the U.S.) and to the output section of my reference Musical Fidelity kW SACD player (no longer in production, but a very costly player in its day).

I found that the PS Audio DAC produced a subtly sweeter, darker and subtly more "romantic" sound than the iDecco DAC, while the iDecco offered a more resolved, open, and transparent sound with—by comparison—an ever-so-slightly more lightly balanced presentation overall. I also found that the PS Audio DAC's S/PDIF input sounded better than its USB input, whereas the iDecco's USB and S/PDIF inputs, as mentioned above, sounded equally good.

In comparison to the DAC/output stage of my Musical Fidelity kW SACD player the iDecco offered many similarities, though I thought the Musical Fidelity offered slightly better resolution of low-level details and low-level dynamic contrasts, and a bit more defined and nuanced bass. On the other hand, you could argue that the iDecco offered a more relaxed presentation. In any event, the sonic differences between the iDecco DAC and the DAC/output section of the Musical Fidelity player were relatively small—especially in light of the big price differential between the products.

EQUIPMENT REVIEW - Peachtree Audio iDecco Integrated Amp/DAC

Given these results, I've come to think that the iDecco's DAC section alone could more or less justify the product's entire asking price, which is remarkable when you consider that there is so much more to the iDecco than just its DAC section

SONIC CHARACTER, AMP

Having listened to the iDecco DAC/preamp sections driving a powerful and accomplished pair of outboard monoblock power amps (the NuForce Reference 9 v.3 SE pair), I felt I was in a pretty good position to assess what the iDecco's own amplifier section could do by comparison. My conclusion, not too surprisingly, is that the iDecco's amplifier section is very good for what it is: namely, a high quality, mid-priced and moderately powered amplifier offered as part of an affordable, multifunction integrated amp. But that said, I would also observe that the iDecco amp is not quite the equal of a high-end standalone power amp, nor should we expect it to be.

On the plus side of the ledger, the iDecco amp delivers a rich, clear, and evocative sound with excellent soundstaging characteristics. When coupled with speakers that can be driven well by 40Wpc, the iDecco amp can produce huge, three-dimensional soundstages that leave the sound of many modestly priced integrated amps in the dust. During my tests, I used the iDecco in conjunction with a pair of Monitor Audio's superb (and quite easy to drive) Silver RX8 floorstanders ($2000/pair) and found the combination to be one of those rare instances of "sonic serendipity," where the whole was much greater than the sum of the parts. Think of it this way: You could buy an iDecco and the Monitor Audio speakers

I mentioned above for about $3000, then add either a PC-based music server and/or an iPod as source components, acquire an obligatory set of high-performance cables, and wind up with a music system that—I kid you not—could easily do battle with many of the five-figure systems I've heard at trade shows.

Good though the iDecco amplifier section is, however, I would say it is perhaps not quite as impressive as other elements of the product are. The main sonic differences you would observe between the iDecco amp and higher-end powerplants (such as the NuForce monoblocks I used in my tests) involve the iDecco's slightly reduced levels of resolution and detail from top to bottom and somewhat less tightly controlled and less deeply extended bass response. There is, too, a difference in sheer power output to be reckoned with (remember, the iDecco produces an honest 40Wpc at 6 ohms, while the NuForce monoblocks each belt out 335 watts at 4 ohms). In practice, this means you'll want to keep the iDecco's power output limitations in mind and plan your speaker acquisitions accordingly.

But let's keep things in perspective. While the iDecco's amp section may not enjoy the quasi-giant-killer status that its DAC and preamp sections do, it nevertheless offers very solid performance and—more importantly—unfailing musicality for the money.

MUSICAL EXAMPLES

I can't speak for you, but I sometimes enjoy playing well-made recordings that show unexpected combinations of instruments at play, partly because they draw your attention to the musical ideas being expressed, but also because

they seem like celebrations of the sheer beauty of sound, itself. One such recording is Marilyn Mazur and Jan Garbarek's *Elixir* [ECM], where two favorite tracks are "Bell-Painting" and "Talking Wind." Both tracks employ distinctive high-pitched percussion instruments of various kinds, highlighting differences in the attack, voicing, and decay characteristic of each instrument within a reverberant recording space. On good equipment, the sonic effect of hearing these tracks is not unlike running your fingers through a treasure chest full of variegated jewels—so many different shapes, textures and colors to take in at once. On both tracks the iDecco not only did not disappoint, but positively excelled.

On "Bell-Painting," the shorter and more delicate of the two musical selections, you initially hear a round of differently pitched small bells and chimes being struck, followed by a similar round of slightly deeper-pitched bells and gongs being sounded. The iDecco deftly captured the variations in attack between the bells, appropriately giving each its signature voice, and showing how decay characteristics help define the bells' timbres—with some fading quickly to silence as others continue to shimmer and ring for several seconds after being struck, their voices lingering and floating on the air. Most importantly, the iDecco captured—but did not overdo—the fundamentally metallic character of the bells, something that in practice is easier to say than to do on this revealing track (some amps, for example, make the instruments sound much too "dry," almost like bursts of white noise, which isn't right). The iDecco served up levels of realism and nuance that not many amp/DACs in its price range could muster.

On "Talking Wind," the longer and more dynamically challenging of the two tracks, the iDecco got an even tougher workout, as the performers unleash an array of high and mid-pitched cymbals, gongs, and bells, and then introduce a musical theme propelled by low-pitched drums. The iDecco impressed me with its ability to navigate gracefully the track's complicated combinations of pitches and wildly fluctuating dynamic envelopes (indeed, some of the percussion strikes captured on the track are downright violent). What's more, the iDecco simultaneously managed to catch the complex interplay between the instruments while also showing how their sounds interacted with, and reverberated within, the relatively live-sounding recording space. Faced with such vigorous musical demands, some amps lose focus and retreat into a region where their sound becomes diffuse and compressed, but not the iDecco. It hung right in there, tapping into and beautifully expressing the richness and dynamic liveliness of the song, while presenting the instruments on a wide, deep, and precisely delineated soundstage. Well done, Peachtree.

If you play music that demands very high levels of bass power and finesse at the same time, such as the bass guitar solos found on "Lil' Victa" from Stanley Clarke, Marcus Miller, and Victor Wooten's *Thunder* [Heads Up], you might observe one of the few limitations of the iDecco amp: namely, a tendency to run out of steam down low and to deliver bass that, while rich and nicely weighted, is not the last word in definition. Through the iDecco amp you can hear differences between the voices of Clarke's, Miller's, and Wooten's basses (Clarke plays an

EQUIPMENT REVIEW

SPECS & PRICING

Peachtree Audio iDecco

Power: 40Wpc @ 6 ohms

Inputs: Four digital audio (USB, S/PDIF-coax, TosLink optical, iPod), one stereo analog

DAC upsampling: 24-bit/96kHz

DAC signal-to-noise: 122dB "A-weighted"

Outputs: Two pre-amp outputs (one variable level, one fixed level), one headphone output (1/4-inch jack), main speaker taps

Dimensions: 5" x 14.75" x 14"

Weight: 25 lbs.

U.S.

Price: $999

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Alembic bass, Miller plays a highly modified Fender Jazz bass, and Wooten plays a Fodera bass), but the lines of distinction aren't quite as crisply drawn as they might be in higher-end amps. Similarly, there's a sense that the iDecco almost but not quite captures some of the finer textural and dynamic nuances of the three bass virtuosos' playing styles. But on the whole, the iDecco's sound is incredibly accomplished and refined—especially when you take its price and amazing versatility into account.

BOTTOM LINE

The Peachtree Audio iDecco is a worthy little brother to the firm's excellent Nova, as it combines remarkable flexibility (highlighted by the iDecco's signature digital iPod dock) with levels of sonic finesse and refinement so high that they really belie the product's modest asking price. As we observed at the outset, the iDecco can play many different roles, each at a very high level of performance. But whether you buy one to use as a DAC, a preamp, a headphone amp, or as one of the coolest DAC/integrated amps we've yet seen, the iDecco will more than give you your money's worth. tas

NAD M2 Direct Digital Amplifier

Inventing the Future of Audio

Robert Harley

The term "digital" is often erroneously applied to amplifiers with Class D (switching) output stages, but in the case of NAD's new M2 Direct Digital Amplifier that word is appropriate. In fact, the M2 represents a major rethinking of audio-system architecture, directly converting standard-resolution or high-res digital bitstreams into signals that can drive loudspeakers.

Functionally, the M2 is an "integrated amplifier" that replaces a DAC, preamplifier, and power amplifier. The M2 eliminates from a traditional signal path all the electronics of a DAC as well as the active analog gain stages of a preamplifier and power amplifier. It does this by converting the PCM signal from a digital source directly into a pulse-width modulation (PWM) signal that turns the M2's output transistors on and off. That's it—no digital filter, no DACs, no multiple stages of analog amplification, no interconnects, no jacks, no analog volume control, no preamp.

The conversion from the digital domain to the analog domain occurs as a by-product of the switching output stage and its analog filter. This is as direct a signal path as one could envision. (See sidebars for the technical details.)

NAD's M2 is a significant departure for the company that made its reputation building simple and affordable electronics. For starters, the M2 costs $5999, a new price level for a NAD "integrated amplifier." Second, the M2 is NAD's first amplifier to use a switching output stage. The company had previously rejected the technology

in favor of linear amplifiers because switching output stages just didn't sound good. But the M2's output stage is significantly different from any other currently offered (see sidebar). Third, NAD believes that the M2's technology could eventually become the basis for nearly all of its amplification products. In fact, NAD suggested that the M2 was not designed to capitalize on Class D's functional advantages, but rather to establish a new benchmark of performance in amplification, no matter what the technology.

Let's look at the M2 Direct Digital Amplifier

in operation. The unit looks and functions like one of NAD's upscale Masters Series integrated amplifiers, with a row of front-panel input-select buttons, a volume control, and a display. The rear panel, however, reveals that the M2 is not a conventional integrated amplifier. Five digital inputs are provided (two RCA, one AES/EBU, two TosLink, plus a TosLink loop) along with one single-ended and one balanced analog input. The digital inputs can accept any sampling frequency from 32kHz to 192kHz. Analog signals fed to the M2's analog-input jacks are converted to digital.

Remember! You can go to another section by clicking here.

EQUIPMENT REVIEW - NAD M2 Direct Digital Amplifier

TECHNOLOGY: NOT JUST ANOTHER SWITCHING AMPLIFIER

The M2 is different in two important ways from other amplifiers that use a Class D switching output stage. In a conventional switching amplifier, analog input signals are converted to a series of pulses that turn the output transistors fully on or fully off. The signal's amplitude is contained in the pulse widths (see sidebar "Pulse-Width Modulation"). An output filter smooths the pulses into a continuous waveform. But in the M2, PCM digital signals fed to the amplifier's input (from a CD transport, music server, or other source) stay in the digital domain and are converted by digital-signal processing (DSP) to the pulse-width modulated signal that drives the output transistors.

This difference might not seem that great at first glance, but consider the signal path of a conventional digital-playback chain driving a switching power amplifier. In your CD player, data read from the disc go through a digital filter and are converted to analog with a DAC; the DAC's current output is converted to a voltage with a current-to-voltage converter; the signal is low-pass filtered and then amplified/buffered in the CD player's analog-output stage. This analog output signal travels down interconnects to a preamplifier with its several stages of amplification, volume control, and output buffer. The preamp's output then travels down another pair of interconnects to the power amplifier, which typically employs an input stage, a

driver stage, and the switching output stage. In addition to the D/A conversion, that's typically six or seven active amplification stages before the signal gets to the power amplifier's output stage.

To reiterate the contrast with the M2, PCM data are converted by DSP into the pulse-width modulation signal that drives the output transistors. That's it. There are no analog gain stages between the PCM data and your loudspeakers. The signal stays in the digital domain until the switching output stage, which, by its nature, acts as a digital-to-analog converter in concert with the output filter. The volume is adjusted in DSP.

The second point of departure between the M2 and all other Class D amplifiers is the switching output stage itself. NAD partnered with the U.K. design team of the American semiconductor company Diodes Zetex, who had developed a novel switching-amplifier technology. NAD engineers worked with Diodes Zetex for more than four years to improve upon Zetex's basic idea before it was ready for the M2. Diodes Zetex calls its amplifier a direct digital feedback amplifier (DDFA). The primary innovation is the use of feedback around the output stage to reduce distortion. Feedback, used in virtually all linear amplifiers, takes part of the output signal, inverts it, and sends it back to the input. The technique

Once you've connected an analog or digital source to the M2 (such as a CD transport or music server) and loudspeakers via the output binding posts, the M2 functions just like a traditional integrated amplifier. You select the source from the front panel and control the volume with the large front-panel knob or from the remote control. The front-panel display shows the input sampling frequency and volume setting.

Purists will note that the M2 requires that analog signals, such as a phonostage output, be converted to PCM digital. Similarly, those who enjoy SACD will be loath to convert their SACD player's analog output to PCM, and then back to analog in the M2.

The M2 offers a number of features not found on a traditional integrated amplifier. Pushing the MENU button allows you to select the sampling frequency of the analog-to-digital converter (for analog input signals) as well as engage an upsampling feature that converts, for example, 44.1kHz to 96kHz. Analog signals are digitized at up to 192kHz/24-bit. You can also attenuate the level of the analog inputs by up to 9dB. A "Speaker Compensation" adjustment is a five-position adjustment that "allows fine tuning of the top octave to match the speaker impedance." An absolute-polarity switch rounds out the menu-accessible features. A rear-panel switch engages NAD's "Soft Clipping" feature, which limits the output to prevent audible distortion if the amplifier is overdriven. An RS232 port allows external control via a PC or control system such as Crestron or AMX. The full-function remote control selects between sources, adjusts the volume, dims the display, and can also control a NAD CD or DVD player.

The M2 doesn't seem like a switching amplifier in operation; it is heavier than most Class D amps and although it runs cooler than a traditional Class AB amplifier of comparable output power, it produces more heat than any other Class D amplifier I've had in my home.

LISTENING

I lived with the M2 for a couple of months, driving the Wilson Audio Alexandria X-2 Series 2 loudspeakers as well as the YG Acoustics Kipod Studio (review forthcoming). When driving the Kipod, the M2 could drive only the upper module, not the powered woofer that accepts a line-level input. I also heard the M2 with the Volent Paragon VL-2, a $5000 stand-mounted two-way employing a Heil Air-Motion Transformer (also on-deck for review).

I compared the M2 to my usual system of a Berkeley Audio Design Alpha DAC, Pass Labs XP20 preamp, and Pass Labs XA100.5 Class A power amplifier, all connected with MIT MA-X interconnect and MIT Oracle MA loudspeaker cable. Note that the M2 functionally replaces this entire Berkeley DAC/Pass preamp/Pass power amp/MIT system, and costs about one-tenth the price. The digital source for both systems was the AES/EBU output from a Classé Audio CDP-502 to play CDs. I tested the M2 with high-resolution bitstreams sourced from the fan-less, drive-less, PC-based music server built by Boston retailer Goodwin's High-End and described in Issue 189. When I connected the AES/EBU output from the server into the M2's AES/EBU input, the M2 instantly locked to any sampling frequency and was glitch-free.

I experienced two minor operating problems

EQUIPMENT REVIEW - NAD M2 Direct Digital Amplifier

lowers distortion. But feedback isn't practical in switching amplifiers because of the delay involved in sending part of the output signal back to the input. Switching stages operate on extraordinarily precise timing; a glitch of a nanosecond can cause the output stage to lock up. The Zetex innovation is to compare the actual high-level PWM signal (at the transistor outputs) to a low-level reference PWM signal. Any difference between the actual and reference PWM signals represents a voltage error. The actual PWM signal can deviate from the theoretical ideal because of power-supply noise or droop (a drop in voltage), slight changes in the pulse widths, transistor tolerances, or variations in the rise-time of the pulse edges. All these potential sources of errors affect the area under the pulses, which is how the analog amplitude is encoded. This error shows up as a voltage, which is digitized at a conversion rate of 108MHz, processed to compensate for subsequent modulation cycles, and then fed into a noise-shaper that adjusts the pulse shape, on a continuous basis, to compensate for errors in the output stage. In addition to decreasing distortion, this technique also lowers the amplifier's output impedance.

The reference PWM signal must be essentially perfect or else the system will correct "errors" that aren't present. The pulse widths must be precise to within five picoseconds, a level of performance commensurate with the lowest clock jitter in state-of-the-art digital-to-analog converters. In fact, you can think of the M2 as a DAC with gain and judge its technical performance using the same metrics as those employed in evaluating D/A quality. For example, at -120dB, the M2's linearity error is less than +/-0.1dB (an amazing spec, by the way), and the unit provides useful resolution down to an astounding -135dB.

The M2's topology has interesting ramifications for a system's overall noise performance. In a traditional system of digital source, analog preamplifier, and analog power amplifier, any noise introduced ahead of the power amplifier greatly degrades the system's signal-to-noise ratio (SNR). For example, if we start with a CD player with a SNR of 115dB, feed its output to a preamplifier with a SNR of 108dB, and then drive a power amplifier whose intrinsic SNR is 115dB (all great specs), the system's overall SNR is only 84.1dB referenced to 1W (all SNR numbers are unweighted). Noise at the front of the chain gets amplified by the power amplifier, no matter how quiet that amplifier is. In the M2, the only source of noise is in the DSP and the switching output stage, and the noise level is completely independent of the gain. That is, the SNR doesn't degrade at low volume. The DSP's noise is kept low in part because of the 35-bit data path. The M2 has an SNR of 91dB (unweighted, referenced to 1W) at any signal level. Indeed, I turned the gain all the way up and put my ear next to the tweeter of the sensitive Wilson Audio Alexandria X-2 Series 2 loudspeaker (95dB 1W/1m) and heard no noise.

There's no free lunch, however. Switching amplifiers require a serious output filter (typically a large inductor and a capacitor) to

with the M2. First, the protection circuit triggered a couple of times, even with no music playing. Turning off the power reset the circuit. Second, when I turned on the M2 on one occasion I heard noise from the right channel. Turning off the unit and turning it back on corrected the problem. This happened only once in dozens and dozens of power-up cycles.

Long-time readers will know that I'm no fan of switching amplifiers. They have their virtues—small size, very little heat dissipation, light weight, and usually a considerable amount of output power for the money. But when the music starts, Class D amplifiers have left me cold. They can sound very dynamic, but exhibit considerable variability in sound quality depending on the loudspeaker they are driving, the cables, and other factors. The switching amplifiers I've heard (admittedly, I have not heard many) have exhibited a mechanical character, along with a "chalky" coloration in the midrange that robs instruments of their distinctive tone colors.

But the M2 sounded completely unlike any other Class D amplifier I've heard. It had no characteristic fingerprint that identified its technology. Rather, the M2 tended to get out of the way, reflecting the virtues and verities of the recording. Unlike other switching amplifiers I've heard, the M2's departures from neutrality were subtractive rather than additive. That is, it commits sins of omission rather than sins of commission. The M2 sounded like a very high-quality conventional (linear-amplification) playback system in many ways, with one notable exception; this amplifier was dead-quiet at any listening level and with any loudspeaker—even the 95dB-sensitive Wilson X-2. Backgrounds were truly and totally black, a

SPECS & PRICING

NAD M2 Direct Digital Amplifier

Continuous output power: 250Wpc (8 ohms); 250Wpc (4 ohms); 300Wpc (2 Ohms)

IHF dynamic power: 300W (8 ohms); 450W (4 ohms); 600W (2 Ohms)

Peak output current: >60A

Signal-to-noise ratio: >120dB (A-weighted, referenced to 200W)

Digital inputs: S/PDIF on RCA jacks (x2), AES/EBU (x1), TosLink optical (x2) plus TosLink in/out loop

Sampling frequencies supported: 32kHz–192kHz up to 24 bits

Analog inputs: Unbalanced on RCA jacks, balanced on XLR jacks

Analog-to-digital converter: Fully balanced, 192kHz/24-bit

Dimensions: 17.12" x 5.24" x 17.87"

Weight: 44.45 lbs.

U.S.

Price: $5999

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EQUIPMENT REVIEW - NAD M2 Direct Digital Amplifier

remove high-frequency switching noise from the output, and to smooth the waveform. This filter is conceptually similar to the reconstruction filter in traditional digital-to-analog conversion. Switching amplifiers are also very susceptible to audible degradation if the power supply feeding the output transistors isn't perfectly clean. Switching amplifiers thus require an extremely quiet supply. Nonetheless, many switching amplifiers skimp on the power supply in an effort to keep size, weight, and cost low. The M2 has a more substantial power supply than I've seen in any other amplifier with a switching output stage. Three separate supplies are used, one for each audio channel and one for the control circuitry and housekeeping.

Each of the M2's amplifiers is contained on a roughly 6''-square circuit board and heat-sink assembly that attaches to a mother-board below it. It appears that each channel employs two pairs of output transistors. The rear panel is shielded, presumably to prevent radiated switching noise to get into the signal after it has been filtered. The chassis is segmented into two additional shielded modules, again to protect against switching noise pollution generated by the output stage.

quality that gave instrumental images a greater tangibility, both spatially and texturally. The dead-silent background seemed to throw instrumental images into sharper relief, enhancing the impression of three-dimensional objects existing in space. This palpability was also partially the result of the M2's somewhat forward spatial perspective which puts the listener around "Row E." The M2 also tended to "spotlight" the midrange to some degree, again adding to the impression of presence and the palpability of instrumental and vocal images. This was generally an appealing quality, although some forward-sounding and midrange-emphasized recordings, such as *In Other Words* from The Teodross Avery Quartet, were not complimentary to the M2. Conversely, naturally recorded vocals such as the outstanding ReVisions: Songs of Stevie Wonder by Jen Chapin, took on a "you are there" quality that was extremely involving.

The M2's bass was simply great—extended, rich, warm, powerful, and muscular. The bottom end was rich and densely saturated in tone color, wonderfully nuanced and articulate, and very fast and dynamic. I greatly enjoyed the M2's combination of weight and agility on acoustic and electric bass, particularly with virtuoso players— Stanley Clarke's acoustic bass on The Rite of Strings with Al DiMiola and Jean-Luc Ponty, for example. Left-hand piano lines were also well served by the M2's dynamic agility and powerful bottom-end (the Beethoven Piano Concertos led by Sir Colin Davis on the Pentatone label). The M2 conveyed the impression that it took tight-fisted control over the woofers, backed up by tremendous energy reserves. The articulation in the midbass was extraordinary; I could easily

hear the initial transient of plucked acoustic bass strings, followed by the rich resonance of the instrument's body.

When an audio product performs in many ways above its price class as the M2 does, there's a tendency to judge all areas of performance against its strengths. In other words, the product itself raises its own performance bar. Keeping that in mind, I noticed a trace of hardness in the upper midrange that manifested itself as a glare on certain instruments, particularly the upper range of trumpet. This is a common characteristic of amplifiers of this price, but it was different in the M2. Where most amplifiers impose this characteristic over a wide band that makes itself nearly always audible, the M2's coloration was confined to a relatively narrow band. Consequently, I heard it only occasionally when there was energy in that region. This slight coloration didn't bother me during extended listening to the M2 alone, but was apparent when I compared it to my reference system of the Berkeley Alpha DAC and Pass XA100.5 pure Class A power amplifiers. The M2 didn't have quite the timbral liquidity and midrange warmth of the reference system. Nonetheless, the M2's overall sound was smooth and relaxed.

The treble tended to favor ease over the last measure of detail. The top octave wasn't quite as open, extended, or transparent as my reference system. Listening to a straight-ahead jazz CD I had engineered live to two-track (Confirmation by the Chiz Harris Quartet), drummer Harris' cymbals were not quite as vibrant. Similarly, Conte Candoli's flugelhorn took on slightly more of a golden and burnished hue than it had in life. If a component departs from neutrality, it's better

that this departure be in the direction of slightly softening of the treble rather than emphasizing it. I should reiterate that you can adjust the M2's treble balance to match your system via the front-panel menu.

The M2 sounded quite detailed, although the very finest inner detail was not as nuanced as that heard in the reference system. The M2 didn't resolve the last measure of information that conveys the mechanism by which a sound was created. For example, there's a passage in "Sorceress" from Return to Forever's *Romantic Warrior* (on the newly re-mastered *The Anthology* CD) in which Lenny White overdubs an intricate percussion figure on timbales in counterpoint to his drumming. The reference system better revealed the nature of the timbales, making them

THE M2'S PREDECESSOR

The M2 isn't the first switching amplifier to convert PCM to PWM. That distinction belongs to the TacT Millennium, which I reviewed at its introduction in 1999. But the M2 isn't simply a more modern version of that topology. Rather, the M2 employs an entirely new and radically different switching output stage (see sidebar). In addition, the Millennium adjusted the volume by changing the voltage of the power supply rails feeding the switching output transistors. The M2 adjusts the volume in the digital domain with the same digital signal processing (DSP) chip that performs the PCM-to-PWM conversion.

EQUIPMENT REVIEW - NAD M2 Direct Digital Amplifier

PULSE-WIDTH MODULATION

How can a series of pulses represent the continuous waveform of music? In exactly the same way that Direct Stream Digital (DSD), the encoding format behind SACD, produces music from a bitstream. In fact, PWM and DSD are conceptually identical.

Fig.1 shows the relationship between a DSD bitstream and the analog waveform that bitstream represents. The bitstream is a series of pulses of varying lengths, with the pulse length encoding the analog signal's amplitude. The pulse-train generated by DSD encoding looks remarkably "analog-like." That is, you can look at the pulse train and get an idea of what the analog waveform looks like. The relationship between the analog signal and the bitstream is so close that in theory, a DSD signal can be converted to analog with a single capacitor (DSD-to-analog conversion is more complex in practice). The bit rate of DSD as used in SACD is 2.8224 million bits per second.

In a switching amplifier, the output transistors are turned fully "on" or fully "off" by the pulse-width modulated signal.

The analog signal's amplitude is encoded as the "area under the pulses"; longer pulses (longer "on" times for the output transistors) represent a higher analog-signal amplitude. This is contrasted with traditional "linear" amplifiers in which the output transistors are in a continuously variable state of conduction.

The output of the PWM stage is a series of high-level pulses that must be smoothed into a continuous waveform. Every amplifier with a switching output stage employs a large filter (an inductor and a capacitor) between the output transistors and loudspeaker terminals to perform this smoothing function and to remove switching noise.

In the Diodes Zetex amplifier module, the pulses are quantized at 108MHz. This frequency determines the number of discrete pulse widths available to represent the audio waveform. That number is 128, which appears at first glance to be too low to encode a complex musical signal. But even at 20kHz, there are many modulation cycles available within the period of a 20kHz waveform.
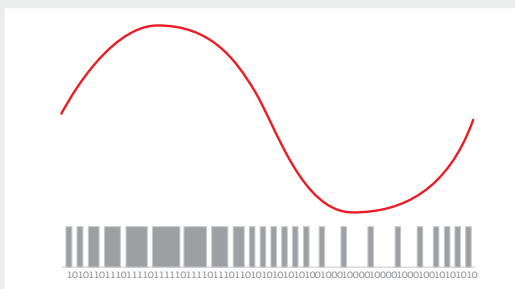
Pulse-Width Modulation represents an analog waveform with a series of varying-length pulses.

sound more like instruments being struck and less like mere transients.

The M2 was outstanding in its ability to unravel complex musical lines. Many amplifiers of this price tend to have a flat homogeneity that prevents one from hearing quieter instrumental lines in the presence of louder ones. This aspect of music reproduction is crucial to understanding the intent of the composer or performers. The M2 was the antithesis of smeared, congested, or confused. Instead, it laid out with exquisite resolution everything that was happening in the music. Moreover, it did this in a completely natural and organic way, with no trace of the analytical.

Partly as a result of this quality, and partly a result of the M2's fabulous way with dynamic contrasts and shadings, music always had an energetic and upbeat quality. I could feel the spontaneous music-making on the previously mentioned Confirmation disc I'd engineered and remembered from the session. The M2 had a rhythmic coherence and sense of life that thrilled me and riveted my attention on the music. Interestingly, I noticed this quality most on bebop; Freddie Hubbard's solo on his great composition "Birdlike" from pianist George Cables' Cables' Vision positively soared.

Finally, the M2's A/D converter (fed by the Aesthetix Rhea Signature phonostage) was very good, but not completely transparent. It shaved off a bit of resolution at lowest levels and very slightly hardened timbres.

CONCLUSION

Despite costing one-tenth as much as my reference system (all the components of which are outstanding), the M2 was extremely engaging

musically. Overall, I preferred the reference system, but not by as much as the price disparity would suggest. I usually wouldn't judge a $6000 product against one costing more than $50k, but the M2's outstanding performance in many areas invited the comparison. Moreover, the M2 represents a radically different approach to amplifier design, digital-to-analog conversion, and system architecture. As such, I evaluated how the M2 sounds not just in comparison with similarly priced conventional amplification and digital-to-analog conversion, but how its new technology stacks up on an absolute basis. (You should consider this when reading how the M2 falls short of a reference-quality system. I included those observations not to diminish the great achievement the M2 represents, but to put this new technology in context.)

As for the M2 as an alternative to a $3500 conventional integrated amplifier and a $2500 digital-to-analog converter, it's a slam dunk. I haven't heard, nor can I imagine, any combination of amplification and DAC at the price approaching the M2's performance. Moreover, the M2 delivers, in one chassis, decoding of high-resolution digital audio, the source-switching and control functions of a preamplifier, and 250W of amplification—all with outstanding ergonomics. I can envision the M2, or its descendents, as part of a three-piece playback-system: music server, M2-like product, and loudspeakers.

NAD's M2 is a triumph on many levels, not the least of which is that it points toward a new direction in amplifier design and system architecture. I predict that years from now audiophiles will look back on the M2 as the progenitor of the next generation of audio. tas