

the absolute sound BUYER'S GUIDE TO DIGITAL SOURCE COMPONENTS

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FROM THE EDITOR

Welcome to *The Absolute Sound Buyer's Guide to Digital Source Components*. In this edition, we've compiled reviews of the best digital products we heard in 2014 in an easy-to-navigate PDF. Herein, you will find products that range from entry-level, portable USB DACs to reference-quality separates that take your system to a whole new sonic level.

With the proliferation of high-res download sites and CD-quality streaming services such as TIDAL and Qobuz, digital source components have moved beyond the disc-and-transport era to become full-fledged, twenty-first century, computer-and-Internet-based music command centers. *The Absolute Sound Buyer's Guide to Digital Source Components* keeps you abreast of the latest digital-playback technologies, identifies those products (of the hundreds on the market) that truly deserve to be on your short list, and lets you make informed decisions about your next high-end-audio purchase—whether you're just starting out, or are a computer-audio expert.

In this Buyer's Guide you will find familiar features such as On The Horizon, which provides you with a sneak preview of noteworthy upcoming digital gear, as well as Top Picks, which lets you easily sort through those components that offer the most bang for your buck—no matter what your budget. With 27 full-length reviews of some of the finest digital products currently available, you will surely find the one that best suits your needs.

Also included in this Buyer's Guide is a feature article on digital and portable audio aimed at helping those of you just starting out unravel the mysteries of high-end digital playback, both portable and component, locally stored and streaming. Many of the misconceptions about digital audio stem from the incorrect use or misunderstanding of terminology; this article will help dispel that confusion, and give you a clearer basis upon which to make buying decisions.

In the end, all high-end audio is about your music, and reproducing it as enjoyably as possible within a given budget. Digital audio is here to stay, and we hope that this Buyer's Guide will be just that—a guide to your next digital source component.

Happy listening!

Spencer Holbert

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DIGITAL IN



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AKI20



Stars of 2014

WHAT HI-FI?

AK240

AK240





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AK240

NATIVE DSD(UP TO 5.6MHZ) DUAL DAC BALANCED OUTPUT AIRCRAFT GRADE DURALUMIN BODY WI-FI MQS STREAMING 256GB, PLUS I MICROSD SLOT





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DUAL DAC DSD PLAYBACK (DSD TO PCM) BALANCED OUTPUT WI-FI MOS STREAMING 128GB INTERNAL MEMORY MICROSD CARD SLOT OPTICAL OUT







AKI00 II

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HEADPHONES & EARPHONES







Your Guide to Computer and Portable Digital

Spencer Holbert

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Buyer's Guide to Digital Source Compenents

PREVIOUS PACE

ince the late 1990s, digital music downloads and streaming services have gone from a somewhat illicit network of peer-to-peer file sharing—e.g. Napster, Limewire, Kazaa, and BitTorrent—to a multi-billion dollar industry. Today consumers have an abundance of choices when it comes to how they access music—from major players such as Apple, Google, Amazon, Microsoft, Sony, and Beats (now owned by Apple) to grassroots startups such as Spotify, Pandora, Rdio, and Slacker.

The irony of the current state of the music industry is that, while sales of physical media are in major decline (aside from the resurgence in vinyl), more people are listening to more music than ever before in history. While digital downloads-starting with the infamous rise of Napster in the late 1990s-may have revolutionized the way we obtain music, streaming services like Spotify and Pandora have revolutionized the way we consume music. Pandora, the only publicly traded streaming service, delivers about 1.5 billion hours of music to its more than 70 million users-each month.

If music software and file exchange have drastically changed in the last decade, so has the hardware associated with converting those files into analog signals. Once a device found only in laboratories and recording studios, the standalone digital-to-analog converter (DAC) has been miniaturized to the point where smartphones now have DAC chipsets that allow users to enjoy high-quality digital playback.

But all this change and choice can be confusing, if not overwhelming, even for those familiar with portable audio devices and music streaming services. This guide will help you navigate personal audio (listening to music via your computer or portable device), music streaming services, and the hardware and software needed to maximize your listening experience. Whether you have been using portable devices and streaming services for years or are new to the world of streaming music, this guide has something for everyone.

GETTING STARTED

Personal audio falls into two main categories: computer audio and portable audio. Both means of accessing music can be used in isolation, but your experience will be much more satisfying when portable and computer audio are used together. Let's delve into computer audio first.

Computer Audio

Computer audio is simply the ability to play digital music using a computer. There are two main ways of playing music through your computer: 1) via locally stored digital audio files (whether ripped from a CD, stored on a flash drive, or

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downloaded via the Internet and archived on a hard drive); and 2) via streaming digital music from the Internet. Both means of digital audio playback have their advantages and disadvantages.

Locally Stored Digital Audio

Locally stored digital audio files are managed on a computer using music-management software such as iTunes, Windows Media Player, Foobar2000, WinAmp, MediaMonkey, and JRiver Media Center. There are hundreds of Freeware and paid music-management software programs, but most readers who have purchased a computer will be familiar with iTunes and Windows Media Player.

In addition to music, most digital files (including those on CDs) contain metadata-digital descriptions that allow computers to automatically acquire information about the artist, album, song titles, and album artwork. When you insert a CD into your computer's CD-ROM drive, your music-management software saves all this digital data (including metadata) to your computer's hard drive (a process called ripping), and can then automatically download additional metadata from the Internet using music catalog services such as Gracenote. All of this is automatic when you're connected to the Internet. The playback procedure is simple: just select a song, and that pick will be played back through your built-in laptop speakers or external desktop speakers. This is the most rudimentary breakdown of locally stored computer-based audio, and most of you are already well acquainted with this process.

The term MP3 has become a synonym for any type of digital audio file, but it is a misnomer. An MP3 is only one type of audio codec (the algorithm that encodes and decodes digital audio). In the early years of digital audio, lossy audio codecs (heavily compressed audio files that traded sound quality for smaller and easier-to-store files) were needed because of small hard-drive sizes and the additional expense of storing large numbers of digital files. Because large hard drives are now cheap and readily available, there is no need to use compressed files. Audio codecs for audiophiles to avoid are: MP3, WMA, AAC, and Ogg Vorbis, among others. (Understand that if you purchase music from the iTunes store, you are buying compressed music-about 320kbps, or roughly one-fifth the quality of a standard CD.)

When importing or ripping CDs to your computer hard drive—no matter what type of computer you have or which music-management software you use—ensure that your import settings use a lossless audio codec such as FLAC, WAV, AIFF, or ALAC. Unfortunately, most music-management software defaults to encoding your CDs in a lossy audio format,

so it is imperative to check your settings before importing.

Playing music from your computer is straightforward until you get into high-resolution audio. While there is no official definition for what constitutes high-res music, for the vast majority of people high-resolution is a digital audio file

with sample rates (frequency) higher than 44.1kHz and word lengths greater than 16 bits (higher, that is, than 44.1kHz/16-bit Red Book CD quality). The most popular high-resolution audio formats are FLAC (Free Lossless Audio Codec), WAV, AIFF, ALAC, and lately Direct Stream Digital (DSD), which is Sony/Philips' proprietary method for encoding SACDs. Since the vast majority of digital audio signalsincluding telephony and all Internet streaming services-are encoded using pulse-code modulation (PCM), we will limit our discussion to PCM-based digital audio. (For a complete breakdown of DSD, see Vade Forrester's excellent guide, "The ABCs of DSD," on-line or in Issue 238.)

PCM-based files can be labeled as 44.1kHz/16-bit, 88kHz/24-bit, 96kHz/24-bit, 176kHz/24-bit, and 192kHz/24-bit, all of which indicate the sample rate (frequency) and word length (bit-depth) at which the analog signal was sampled (in intervals of one second). Digital audio can also be described bit-rates, such as 196kbps, 256kbps, or 320kbps for lossy audio codecs such as MP3 and AAC. Bit-rate refers to the amount of digital information in each second of audio, expressed in bits. There are 8 bits

in 1 byte, and it is important to differentiate between the

referenced in kilobits per second (kbps) or megabits per second (mbps), while data that are stored are referenced in kilobytes (KB), megabytes (MB), and gigabytes (GB). Bitrate makes for a more shocking comparison when you evaluate lossy versus high-res files. For example, while most MP3s contain up to 320kbps of information, a 96kHz/24-bit digital audio file has a bit-rate of 4608kbpsover fourteen times more information per second. The more information contained in a sample, the better the sound quality, all other factors being equal. It is also important to remember that in digital audio, sample rate (frequency) has nothing to do with audible frequency, but rather refers to the number of times per second the analog waveform is sampled. Put another way, each sample is a tiny picture of the analog wave. If we have a 96kHz digital audio file, the analog audio signal was sampled 96,000 times per second. Many of the misconceptions about high-res audio stem from a misunderstanding of what these terms actually signify. Hopefully, this brief explanation will clarify things.

two. Generally, data that are transmitted are

If you are new to high-res music and digital downloads, I recommend downloading files in the FLAC format; not only is it a universal audio codec, but it also has great metadata capabilities, which means you will have album artwork and artist information. Currently, the majority of high-res audio is purchased and downloaded via on-line retailers such as HDtracks, Super HiRez, Sony, various smaller labels, and the PonoMusic Web site (which should be operational by the time



this article sees print). HDtracks is by far the most popular high-resolution retailer in the U.S., and offers a sizeable, though by no means exhaustive, catalog. Purchasing and downloading high-res files from HDtracks is easy, but there's a catch: You will need to have music-management software capable of playing back high-res audio, plus an external

audio into analog signals. (See the sidebars for software and hardware recommendations.)

DAC capable of converting high-res

While iTunes is an amazing music-management program, it lacks

the capability to properly play high-res audio. Three excellent software programs can be purchased and downloaded to circumvent this problem: Amarra HiFi (\$49), Audirvana Plus (\$79), and PureMusic 2 (\$129). These Macbased programs retain iTunes' excellent user interface, yet bypass its inferior audio processing. A highly recommended alternative to these iTunes add-ons is JRiver Media Center (\$50), which is used at many hi-fi shows, is Windows- and Mac-compatible, and is capable of playing not only PCM files, but DSD as well. The user interface (UI) is not as intuitive as that of iTunes, which can be frustrating for digital-audio newcomers, but if vou're already familiar with digital audio JRiver Media Center will be your best option. Depending on whether you're a Mac or Windows user, you can also explore various Freeware software programs, such as MediaMonkey. Once you're familiar with the programs' general UI, you can start to explore useful tools such as Memory Buffering, which allows digital audio files to be played from your computer's internal memory, rather than directly from the hard drive.

So...we have our computer, our digital files, and our music-management software; now we need the hardware capable of playing high-

> audio device-including computers, portable digital audio players, and CD players-has a built-in digitalto-analog converter (in computers

resolution audio files. Every digital

it is called a soundcard or audio interface). This is the device that converts the digital audio file into an analog waveform we can hear. Unfortunately, sound quality is generally an afterthought with most such devices, and therefore a high-quality external DAC is essential for the best sound quality. While there are dozens of excellent digital-to-analog converters (DACs), let's use the AudioQuest DragonFly V2 (\$149) as our example of a portable DAC, as almost all portable DACs, regardless of their performance capabilities, will operate in a way similar to the DragonFly. The Dragonfly plugs into your computer's USB port. Once the DAC is plugged into your USB port, navigate to your computer's Audio Preferences. I will use my MacBook Pro as an example, though Windows users will have a very similar experience setting up an external DAC. Under "System Preferences" select "Sound," then select the "Output" tab, and click on "AudioQuest DragonFly." Depending on your music-management software of choice, you will also need to verify that vour external DAC is selected as the Master Clock. This applies to portable DACs, as well as the component DAC you might have at home.

The benefit of an external DAC is that its only job is to convert digital audio files into analog, and possibly power a set of headphones. Instead of allowing your computer to convert digital audio files to an analog signal, an external DAC (generally a USB DAC) forces the computer to output raw data according to the external DAC's timing. This is called Asynchronous USB mode, in which the external DAC acts as the digital clock. The DACs found in the vast majority of computers are of inferior quality, and most cannot convert high-res audio files into analog signals. This means that even if you purchased and played high-res music on your computer, the software is down-converting those files to 44.1kHz/16-bit audio, and the internal DAC is then converting those lower-resolution data into an analog signal-a waste of money and sonic potential.

Now that you have all the tools in place to not only play digital audio, but high-res audio

Q Search

Browse

((●)) Radio

| Your Music

Settings

as well, the choice is yours as to how you listen to your music. Since this is an article on personal audio, we'll assume that you will use headphones or small desktop speakers. (See our Buyers' Guide for good starting points.)

Streaming Audio

The alternative to purchasing music and storing it on your computer or Network Attached Storage drive (NAS) is to stream music via the Internet using a musicstreaming service such as Spotify, Pandora, Google Play, or Sony's Music Unlimited. For a breakdown of the various services, see the sidebar. After testing the various services, my recommendation is to use Spotify, which offers higher-quality streaming than most of the others, has an incredible user interface and the ability to download music for offline listening, and tops everything off with a great mobile app that is available for iOS and Android smartphones. While the alternative services offer many of the features found in Spotify, none of them complete the user experience quite like Spotify does.

Spotify is easy to use and offers millions of songs to instantly stream. Download the software to your computer, create a user account, and try the service free for 30 days. Simple as that. One essential aspect to streaming audio via the Internet is to have a fast Internet connection. Download speeds of 3Mbps (megabits per second) or higher are recommended, though you may be able to stream with a minimum of 1.5Mbps. Most streaming services will compress music quality when connection speeds are too slow; streaming services in the U.S. currently

> only offer a max streaming quality of 320kbps (kilobits per second), or MP3-quality sound, and any further compression is anathema to our high-end goals. There CD-quality-and-higher streaming services that will

be available in the fall of

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this year, such as TIDAL (known as WiMP in the U.K. and Europe) and Qobuz, but these services are still a ways off from receiving my recommendation. While streaming in CD-quality has me very excited, I worry about the longevity of these services in the U.S. market. Will enough people subscribe to them to keep TIDAL and Qobuz streaming afloat? Only time will tell; but until then Spotify is your best bet, and it's not going anywhere.

Using the Spotify program is a music lover's dream: Under the "Browse" section you can look through "Top Lists" that feature the chart-topping hits of the day, lists based on genre, "Discovery" lists that feature new music, and more. Look under "Genres & Moods," and you will find compiled playlists based on what type of music you want to hear. Moods include: Party, Focus, Dinner, Sleep, Chill, Romance, Classical, and much more. Select one of the Moods, and lists of sub-Moods will pop up, all with cover art. I selected the "Dinner" Mood,

and then the "Country BBQ" sub-Mood, and instantly some rockin' country music started playing, which mated perfectly with a Texas BBQ. When the music starts, album artwork, artist, and title are all displayed, as well as volume controls, and fast forward; if you're not into the track played, simply skip forward.

Spotify also has a really great "Discover" feature, which pulls data from the music stored on your computer and the songs you have listened to in order to recommend music. And this feature isn't just a gimmick; it's an extremely accurate way to discover new music based on your taste. I went to the Discover tab and the recommendations were spoton. Because I have Valerie June and Sharon Jones & The Dap-Kings on my computer, Spotify recommended Gary Clark Jr. (a local hero here in Austin, TX), Laura Mvula, Lee Fields & The Expressions, Junip, Alt-J, and Stevie Wonder-all great music. You could spend the rest of your life delving deeper and deeper into these Discover and Mood features and barely scratch the surface of the over 20 million songs available (and Spotify adds almost 20,000 songs per day to this catalog).

One of the best features of Spotify is its ability to download music for "Off-line Listening" when an Internet connection isn't available. Simply create a playlist by dragging music you find into the playlist, then toggle the "Available Off-line" switch in the upper-right corner of the playlist. Now, even if you don't have an Internet connection, you can listen to your favorite music. The catch is that you are limited to 3333 songs (an arbitrary number), and you must log in to your streaming account

once every 30 days-Spotify's attempt to ensure that you remain a paid subscriber.

Streaming from a mobile device is also simple using Spotify, which we will discuss below. You can use unlimited devices to stream, though only one device may be used at any given time (again, Spotify's way to ensure that multiple people aren't streaming under the same username), and you can only download music for off-line listening to three devices. Still, for \$9.99 per month, Spotify provides a rich user experience and all the music you can hope for in one lifetime.

Because most streaming services only stream low-quality music, they aren't meant for ultimate fidelity or critical listening. The point of streaming services is access to immense amounts of music, which gives you the ability to discover new music, to preassemble playlists that can stream music for

a party or for any listening mood, and to listen at home or on the go. Music fans, rejoice!

Portable Digital Audio

What's better than having access to thousands (or millions) of songs on your computer? The ability to take all of that music with you. Portable digital audio has come a long way since the early MP3 players of the late 90s and early 2000s. Most people are familiar with portable digital audio

players like the Apple iPod, but since highres downloads became available in 2009, there have been an increasing number of portable players capable of high-res playback. Paired with quality headphones, a high-res portable audio player can radically change your listening experience. High-quality audio can now be consumed at home, at the grocery store, while traveling, or in any setting in which you want to listen to music, all without being tethered to your home stereo.

Most portable audio devices require a computer to transfer music to the device. To enjoy high-res audio, you will need to download music from a site like HDtracks, and then transfer that music to a portable player capable of high-res audio. There are now dozens of portable audio devices that play high-res files. Three recommended ones are the FiiO X3 (\$200), Pono Music Player (\$399), and the Astell&Kern

HTC ONE HARMAN/KARDON EDITION

The HTC One Harman/Kardon Edition is a complete rethinking of a smartphone's audio playback capabilities. Not only is it a superb phone, based on the Android OS, it's also backed by Harman International, a powerhouse of audio design. The HTC One Harman/Kardon is capable of playing high-res files up to 192kHz/24-bit, includes some pretty decent earbuds designed by H/K (\$80 retail value), and is only \$230 when purchased with a two-year contract. Like the possibilities of streaming music to your cell phone? The HTC One H/K comes with six months of free access to Spotify, which is loaded on the phone. The downside of the HTC One Harman/Kardon Edition is that the phone is only available through Sprint, but for audiophiles looking for a smartphone that can do anything and everything, including high-res music playback, the HTC One Harman/Kardon Edition is hard to beat.

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YOUR GUIDE TO COMPUTER AND PORTABLE DIGITAL

GLOSSARY OF TERMS

Bit-rate: The number of bits of digital information contained in one second of a digital audio file.

DAC: Digital-to-analog converter.

GB: Gigabyte.

High-Resolution Audio: Digital audio file with a sampling frequency and word length greater than 44.1kHz/16-bit (CD quality).

kbps: Kilobits per second.

MB: Megabyte.

Mbps: Megabits per second (there are 8 bits in one byte).

NAS: Network attached storage; generally an external hard drive connected to a network via Ethernet.

Sample rate: The number of times (frequency) an analog waveform is sampled each second expressed in kilohertz (kHz), e.g. 44.1kHz, 96kHz, 192kHz.

AK120 II (\$1699), each of which offers various storage and sonic capabilities. There are also smartphones capable of high-res music playback, such as the HTC One (M8) Harman/Kardon Edition (see sidebar), which consolidate all of your portable devices into one sleek package.

For those who have large CD collections and simply want the ability to carry all that music with them, the Apple iPod Classic is still the best option. For \$249, you get 160GB storage capacity, a user-friendly interface, and 36 hours of audio playback—far longer than most other portable audio devices. The downside to the iPod is that it is limited to 44.1kHz/16-bit

PCM files. However, if you aren't concerned about high-res audio, it will be the fastest and easiest way to take your music collection with you.

If you want the ability to play high-res audio on the go, you will need a device that can play those types of digital audio files. Before you purchase a portable music device, make sure that it's capable of at least 192kHz/24bit audio playback, and that it has adequate storage capabilities for your needs. Remember, an uncompressed four-minute song at 96kHz/24-bit quality will consume about 140MB, which means that a portable audio device with 128GB of storage space can hold about 900 four-minute songs. Most portable devices will have enough internal storage capability to keep you rockin' for many hours, if not days, but some require a microSD card (a small flash-based storage card) to expand their internal storage to 128GB.

A great starter digital player is the FiiO X3. At \$200, the X3 can play any high-res file, including DSD up to DSD64 (64 times the sample rate of a CD). Though limited to 8GB of internal memory, the X3 is capable of 128GB of storage capacity with an additional microSD card. For \$399, Neil Young's new Pono Music Player is compatible with all PCM files up to 192kHz/24-bit, has 64GB of internal memory and an included 64GB microSD card for a total of 128GB of storage. Designed in conjunction with Ayre, manufacturer of high-end audio components, the Pono Music Player offers a great mix of sound quality, storage capability, and cool factor.

If you're searching for the ultimate in port-

STREAMING SERVICES

Spotify: \$9.99/mo

TOP RECOMMENDATION

- 30-day free trial period.
- Download and save music for use at any time, including syncing with iPods and portable music devices.
- Audio quality is 320kbps for Premium subscribers in Ogg Vorbis format (don't be fooled by the high-definition claims; it's still compressed music).
- Great user interface.
- Mobile versions for most devices.
- Great discovery feature based on similar interests.
- Browsing based on 25 different main "moods," including Dinner, Sleep, Jazz & Blues, Romance, Focus, Party, Classical, and more.
- Sub-moods provide tailored playlists such as "Mixed Generation Party," which are perfect for selecting the perfect playlist.

Pandora \$4.99/mo

- Audio quality limited to 192kbps streaming.
- Simple interface is better for those looking to discover new music only.
- Very simple interface.
- Users create "stations" that play similar songs to the user-selected artist, title, or genre for which they search.

Beats: \$9.99/mo

- Free trial period (14 days).
- Curates music recommendations based on the user adding weight to various genres and artists; recommendations are accurate.
- Poor sound quality with overemphasized bass.
- Hip, young interface; geared toward younger generations.
- Streaming seems fast.

Google Play: \$9.99/mo

- Google makes you create a Google Wallet account to sign up for the one-month trial of Google Play All Access Music.
- Technical issues while using Mac Safari.

TIDAL (WiMP): \$19.99/mo

- Compatible with many high-end brands' proprietary iOS/Android apps
- Lossless streaming of CD-quality audio and HD music videos.
- 25 million tracks available for on-line streaming.
- iOS and Android apps.
- Curated music editorials writ ten by professional reviewers and artists.

Qobuz: \$19.99/mo

- Lossless streaming of CD-quality audio.
- Music streaming limited to Qobuz catalog (which is mainly classical and jazz).
- iOS and Android apps.
- Available offline.

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DESKTOP/LAPTOP AUDIO SOFTWARE

iTunes (Free; Mac/Windows)

Amarra Mac (\$49.99; Mac)

Audirvana Plus 2.0 (\$74; Mac)

Pure Music 2 (\$129; Mac)

JRiver Media Center (\$49.99: Mac/Windows) and

Media Center Remote (\$10: iOS/Android)

MediaMonkey (Free; Windows)

ability and high-res audio playback, look no further than Astell&Kern. The AK120 II (among other models) supports all high-res PCM formats up to 192kHz/24-bit, DSD64, employs dual CS4398 DAC chipset, a fully balanced output stage, and even has wireless capabilities, which allows you to sync music from your network and stream music to other devices, all in high-res. With a storage capacity of 256GB (128GB internal memory plus a 128GB microSD card), the AK120 can hold lots of high-res music. At \$1699, the AK120 II might be out of reach for some people, but it's the perfect choice for the ultimate in portable high-res audio.

Headphones, Headphone Amps, and Desktop **Speakers**

Of course, there's no point in investing money and time in DACs, software, and high-res audio files if you're going to listen to music with those free earbuds you receive on international flights. An essential part of the personal and portable audio experience

is a pair of high-quality headphones. Many would argue that quality headphones are the most important aspect of personal audio, but for our purposes we will consider headphones as an integral part of a larger whole-one more step toward great audio.

The two main types of headphones are in-ear monitors-or earbuds, popularized by Appleand over-ear headphones, which are the most common type of high-end headphones. (In my Recommended Products list on your right are two in-ear headphones and two over-ear headphones.) In-ear headphones are more sensitive (easier to drive and require less power) than their larger over-ear counterparts, and therefore are better suited for portable audio. Not only will they be easily powered by our recommended portable DACs and music players (all of which have a built-in headphone amp), but in-ear headphones are also ergonomically designed to form a seal inside your ear canal, which means that they will block out external noise while you're on the go. This will provide better sound quality than open-back, over-ear headphone designs, which are better suited for listening in quiet environ-

ments.

PRODUCTS

iFi iDAC/amp (\$299)

NuForce HA-200 headphone amp (\$349)

PSB M4U 2 headphones

(\$399)

HiFiMAN HE-500 headphones (\$699)

Cardas EM5813 in-ear (\$450) Westone ES5 in-ear (\$950)

Recommended Desktop

Audioengine A5+ (\$399, 50Wpc)

Audioengine A2+ (\$249, 15Wpc)

Audience The One

(\$999)

Playback Hardware

Audioengine D3 (\$189, Issue 241)

Up to 96/24

3.5mm headphone jack

Audioquest Dragonfly v1.2 (\$149, Issue 241) Up to 96/24 3.5mm headphone jack

Resonessence Herus (\$199.

Issue 245) Up to 192/24 USB 2.0 Input

1/4" headphone jack

Cambridge DacMagic XS (\$350. Issue 245) Up to DSD128 USB Input

3.5mm headphone jack

Hegel Super DAC (\$299, Issue 245) Micro USB input 3.5mm headphone jack

iPod Classic Limited to 44/16 PCM 160GB storage 36-hour playback \$249

FiiO X3 192/24, DSD64 10-hour playback WM8740 DAC

Line out and 3.5mm coax digi-

tal out

DSD, APE, FLAC, ALAC, WAV 8GB built-in memory 128GB Micro SD card expansion

Pono

\$200

Up to 192/24 PCM Available October 128GB 8-hour playback \$399

Astell&Kern AK120 II Up to DSD128; PCM up to 192kHz/24-bit Dual CS4398 DACs 128GB built-in memory; microSD 128GB \$1699

HTC One Harman/Kardon Edition Supports up to 192kHz/24-bit

\$229.99 (with two-year con-

tract)



Con Tolder 1,57 Jako Berry 1,55 Col Charder 1,55 St Jamp Jeff 1,55

YOUR GUIDE TO COMPUTER AND PORTABLE DIGITAL

A BRIEF HISTORY OF PORTABLE DIGITAL AUDIO

1976 - The Age of Digital

Recording Begins
First 16-bit digital recording in U.S.
made at the Santa Fe Opera.

1979 - IXI

British Scientist Kane Kramer designs one of the earliest digital audio players. The prototype was capable of one hour of playback. Kramer was later hired by Apple Computer.

1981 - Compact Disc

Philips demonstrates the Compact Disc.

1992 - MiniDisc

Sony introduces the MiniDisc.

1996 - First 24-bit/96k

Experimental digital recordings are made at 96kHz/24-bit.

1998 - The Audible Playeı

Audible.com releases its Audible
Player, designed for use with its proprietary digital audio book formats.
Capable of two hours of playback, the
Audible Player was the first production-volume portable digital audio
player.

1998 - Eiger MPMan

In mid-1998, the EigerMan F10 and F20 (32MB and 64MB, respectively) launched in North America, and were capable of storing up to 12 songs.

1998 - Diamond Rio PMP300

Considered the first commercially successful digital audio player, the Rio PMP300 essentially spurred American investment in portable digital audio devices. Much of the Rio's success was due to an RIAA lawsuit, which claimed that the Rio violated the 1992 Audio Home Recording Act.

June 1999 - Napster

Napster is launched, and becomes the catalyst for the computer-audio revolution.

1999 - Compan Personal Jukeho

Also known under the name Remote Solution, the PJB-100 Personal Jukebox had an initial capacity of 4.8GB. The Personal Jukebox was the first digital audio player to employ a 2.5" laptop hard drive.

2000 - Creative NOMAD

The NOMAD, which ran on AA batteries and looked akin to a portable CD player, was capable of 6GB.

October 2001 - Apple iPod

Apple Computer launches the first-generation iPod, "1000 songs in your pocket." With a 5GB 1.8" hard drive and 2" display, the iPod revolutionized portable audio by popularizing digital audio. Initially disregarded as a small market for niche Apple users, the iPod has sold over 350 million units as of 2012.

2006 - Microsoft Zune

Recognizing the market potential of digital portable music players, Microsoft launched Zune, which included music subscription services and a line of portable media players. In June 2012, Microsoft discontinued all Zune services in favor of Xbox music and video services.

2007 - Apple iPhone

Though the first touchscreen phone was IMB's Simon Personal Communicator, launched in 1994, the iPhone was the first multi-touch phone capable of playing not only audio, but video as well. As of March 2014, Apple has sold over 500 million units.

October 2008 – HDtrack

HDtracks begins to offer 88.2kHz/24bit and 96kHz/24-bit FLAC downloads, paving the way for the highresolution digital audio revolution.

Though in-ear headphones are practical for portable devices, offer great noise isolation, and are lighter in weight, the best sound quality will be had with an openback, over-ear headphone design. As aforementioned, open-back designs are open to the outside world and won't be suitable for noisy environments; they are also heavier, and require more power to operate at their full potential. Despite their lessportable nature, over-ear headphones can provide a more realistic listening experience, with much better threedimensionality and soundstaging than most in-ear designs.

The best headphone performance is realized with a separate headphone amplifier, particularly when driving headphones of low impedance or low sensitivity. Two high-quality headphone amps that won't cost an arm and a leg are the NuForce HA-200 and the iFi iDAC, which is actually a DAC and headphone amp in one small package. Both will provide more power for your larger headphones, which is needed for louder volumes and better sound quality. Driving headphones without enough power is like driving loudspeakers with a flea-weight amplifier; music just won't sound loud enough to have any kind of emotional impact.

If your personal audio revolution is taking place at home, an affordable

option for high-quality music at your computer is a pair of desktop speakers. Probably the best value in high-end audio, the Audioengine A2+ powered desktop speakers are what I recommend to family and friends. Great three-dimensional soundstage capabilities, surprisinglylow bass extension, the inclusion of an amplifier (remember, these are powered speakers), and 3.5mm RCA and USB (44kHz/16-bit) inputs make the \$249 A2+'s one heck of a bargain. If you need more power, and more bass extension, you can move up to the A5+ model, which offers 50W per channel versus the A2+ 15W capability. Both models will give you everything you need to enjoy highquality music, right out of the box and without the need for separate amps and DACs. tas

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ON THE HORIZON

Hot New Products Coming Your Way



Celsus Sound Partner P1

Celsus Sound, Inc. is a new company created by the founder of NuForce, Jason Lim. Its new Partner P1 is one of the most advanced and comprehensive portable high-res audio companions on the market. The P1 features a high-performance headphone amp with over 115dB signal-to-noise ratio, plus a USB DAC utilizing the class-leading ES9018K2M DAC from ESS that can decode up to 128x DSD and 384kHz PCM audio. The Partner P1 also includes built-in WiFi networking capability, which allows the unit to be used as a media streamer with PCM files up to 24bit/192kHz. Equipped with two digital outputs (coaxial and TosLink SPDIF) and one 3.5mm analog output, the Partner P1 can connect to most external devices. It fully supports Windows, Mac, Android (*OTG), and iOS (*camera adapter). Ships November 1, 2014.

Price: \$695

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ARIES Wireless Streaming Bridge

The ARIES serves as a "bridge" between music files on media server or high quality online streaming services and your DAC — enabling DACs for the first time to stream high-resolution music quickly and wirelessly in virtually any sampling rate, including DSD, Double-Rate DSD and DXD.



It is **not** a streamer It turns your DAC into a streamer!



AURALIC LIMITED | 声韵音响

1F Building No7, 1A Chaoqian Road, Beijing, 102200, China 12208 NE 104th St. Vancouver, WA 98682, United States http://www.auralic.com/ info@auralic.com





The AURALiC ARIES is a new network "bridge" that allows you to control, access, and stream all of your digital music to your DAC of preference. What this means is that you can stream any digital format-PCM up to 384kHz, DSD128, and DXD-to your DAC, and control everything via AURALiC's Lightning DS App. With dual-band WiFi (2.4GHz and 5GHz for faster operation), Gigabit Ethernet, USB inputs, and USB, AES/EBU, Coax, and TOSLINK outputs, the AURALiC AERIES is the ultimate digital interface for the serious audiophile. The ARIES can easily replace your computer and allow you the freedom to finally stream all of your music-including TIDAL, Quboz, and WiMP music streaming services—to your reference DAC in native resolutions. An upgraded Aries with dual discrete Femto Clocks and beefed-up power supply is also available for \$1599.

Price: \$999; \$1599 with upgrades. auralic.com





NuPrime DAC-10H and ST-10

The NuPrime DAC-10H is the world's first fully integrated digital headphone amplifier capable of PCM 384kHz and DSD256 decoding. Its balanced headphone amplifier is capable of driving headphones in either balanced or single-ended configuration, and can be used with up to two dynamic headphones. With five digital and two analog stereo inputs, the DAC-10H is also a fully-featured DAC and preamp for any high-end audiophile system. It delivers pure natural sound, while also providing the widest support for the latest high-resolution music formats. The NuPrime ST-10 is a new Class-D stereo amplifier with an incredibly high switching frequency of 70kHz, which provides definitively smooth and detailed sound.

Price: DAC-10H, \$1795; ST-10 \$1495. nuprimeaudio.com

Cocktail Audio X30

Professional video equipment manufacturer Novatron, Inc. has created Cocktail Audio, a new brand aimed at music lovers seeking modern musicsystem solutions. Cocktail Audio's first model is the new X30. Like a Swiss Army knife, the X30 is a single-box solution that incorporates an integrated amplifier (with color display screen), a UPnP-compliant Network Streamer capable of decoding 24-bit/192kHz files at native resolution, and a music server that can store up to 4TB of music or other files. The X30 can rip your CDs or vinyl collection, drive headphones, be used as a preamplifier, play local FM radio stations, store photo images, and link to an external display device. A comprehensive browserbased app can be accessed on your computer, mobile phone, or tablet/iPad. Price: \$1695.

cocktailaudio.com

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ON THE HORIZON



Simaudio Moon MiND 180

This Simaudio Moon MiND 180 allows you to skip the computer and go straight to the music by managing all of your digital files with a single app. Instead of trying to search through your computer's library, then switching to another device to stream high-quality music, the Simaudio Moon MiND 180 can control all of your digital music in one place, and supports resolutions up to 192kHz/24-bit. The MiND 180 has no internal DAC, amp, or traditional preamp, so you will still need a component DAC with which the MiND 180 connects, but this little device makes controlling your digital music collection a breeze. App is currently compatible with iOS devices only, but an Android version is in the works. Price: \$2000.

ReQuest Audio The Beast

With a name like The Beast, you would expect nothing less than something truly phenomenal. Using top-tier components from MSB Technology, The Beast combines beautiful design with state-of-the-art circuitry to create one of the most interesting, ultra-high-end music servers of the past few years. If you can afford the \$40k entry fee, The Beast will surely ignite your thirst for music.

Price: \$40,000. requestaudio.com



Reviewers agree...

... about the SPDIF and HDMI outputs from the **Baetis™** entry-level, all-format, 2-channel and multi-channel, audio and video media server, the **Baetis Revolution II**:

"... this was the best I've heard my system sound to date."

Kris Deering, **SoundandVision.com**, June 12, 2014 "... the best digital sound I have ever heard in my system."

Andrew Quint, **The Absolute Sound**, February 2014

Introducing the Baetis Reference Media Server -

the world's only 4-core media computer with a dedicated, galvanically isolated, and *fully shielded* AES/EBU output straight off the motherboard using no PCI card!

Specifically designed for the world's very best DACs.

You <u>must</u> hear the difference if you are a dealer or owner of a dCS®, CH Precision®, Berkeley Reference®, EMM DAC2X®, Esoteric D-02®, LightHarmonic®, MSB Diamond®, Audio Research Reference®, or DAC of similar quality.



www.baetisaudio.com

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EQUIPMENT REVIEWS

DACs

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AudioQuest DragonFly v1.2 USB DAC

Better Sound for Less

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Spencer Holbert

hen I purchased the original version of AudioQuest's DragonFly USB DAC in late 2012 for \$249, little tears of joy streamed down my face. For years I had suffered from separation anxiety every time I unplugged my home DAC in order to travel or work on my laptop at a coffee shop. It seemed so unfair to have to choose between high-quality sound and portability, but my on-the-go life dictated that I leave great sound behind. The DragonFly was my little savior and gave me the freedom to move about without sacrificing sound quality. Lo and behold, the progeny of the original DragonFly has arrived, and version 1.2 sounds better than ever, and costs \$100 less.

I wanted to test the differences between the original DragonFly and version 1.2 with some mid-grade headphones that a lot of people can afford, so I paid a visit to Brian at Whetstone Audio in Austin (whetstoneaudio.com), who was kind enough to lend me a pair of Grado PS500s for the review. After breaking in the headphones and acquainting myself with the original Drag-

reacquainting myself with the original DragonFly, I fired up version 1.2 and played several tracks from the Audiogon Wake Up Your Ears Sampler, a 24-bit/96kHz album specifically designed for testing headphone capabilities. First up was CC Coletti's "You Shook Me," a bluesy rock song that really highlights her voice, harmonica, guitar work, and the reverby live sound of the smallish recording venue. The main difference between the old version and version 1.2 that I noticed was the 1.2's ability to extend the center soundstage ever so slightly, which pulled Coletti's voice beyond my forehead. (With the original DragonFly, her voice danced just inside my head, which can be an odd sensation for many people.) There was also more of

that "I'm listening in a real space" feeling. Then came "Cantaloupe Island" by Lenny White, et al. I focused on this version's bass solo, which through the 1.2 sounded a bit more lifelike and a little less like a low-frequency bumblebee. That's a good thing, by the way.

Next was "Wandering Eyes" by indierock's Kopecky Family Band. With the original DragonFly I had always heard what I thought was Kelsey Kopecky shaking a tambourine; with the version 1.2 plugged in I realized it wasn't a tambourine at all, but a heavy metal chain which she drops on a floor tom. These kinds of micro-details started popping up across all the music I was listening to, and made listening through headphones all the more enjoyable.

The improvements that version 1.2 of the Dragonfly USB DAC made aren't extreme; if you've been using the older version, there won't be any monumental shift in soundstage, dynamics, or transparency, but certain elements of music will become more focused-a subtle yet noticeable enhancement, in other words. Maybe these details poke their head up because, as AudioQuest states, "The circuitry between the DAC chip and the analog output stage has been refined to create a more direct signal path, leading to even greater transparency and immediacy." Whatever the reason for the improvement, I like it, and so will the millions of laptop users who will benefit from using the DragonFly v1.2.

You know what I like more than the sonic improvements? The price: At \$149, even the most cash-strapped of music lovers can afford this DAC. tas

SPECS & PRICING

Input: Up to 24-bit/96kHz USB transfer mode:

Asynchronous (dual clock) Class 1 USB with Streamlength

protocol

Output: Analog audio mini-jack

Maximum driver output: 125mW

@ 32 ohms

Price: \$149

AUDIOQUEST

2621 White Rd. Irvine, CA 92614 (949) 585-0111

audioquest.com

audioquest.com

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Three Miniature Portable USB DACs

Have DAC, Will Travel

Steven Stone

or audiophiles who travel a portable DAC has become one of those "must-have" travel accessories, right up there with a toothbrush and an unexpired credit card. The first generation of portable USB DACs was big and had limited high-resolution capabilities in comparison to the current crop. But as technology marches forward, more capabilities and smaller footprints abound. I'll look at three small USB DACs in this review—Cambridge Audio's DacMagic XS, the Hegel Super, and the Resonessence Labs Herus.



About the size of a small box of wooden matches, the Cambridge DacMagic XS is one of the smallest and lightest portable DACs I've seen. It measures approximately 2 1/8" by 1 1/8" by 3/8" and weighs under 4 ounces. On one end vou'll find a micro-USB input and on the other end a 3.5mm stereo output. The top of the Dac-Magic XS has its own analog volume control, which "fully bypasses the soundcard and volume control of your computer." The two large buttons, plus and minus, are easy to locate and use even in dark or cramped spaces. Instead of plastic, the DacMagic XS is housed in a beveled brushed-aluminum case that should be capable of surviving a high level of abuse. The DacMagic XS has a small LED next to the headphone jack that glows purple or blue when the unit is operating properly and red when you try to boost the volume past its maximum level.

Inside the Cambridge Audio DacMagic XS you'll find an ESS 9023 24-bit DAC chip that supports PCM bit-rates up to 192/24 from a USB 2.0 input. Straight out of the box the DacMagic XS is set up as a USB 1.0 device, which will only support a maximum bit rate of 96/24. Switching over to USB 2.0 requires holding down both the + and - buttons for at least five seconds until the small light in the DacMagic XS flashes three times. Once in class 2.0 the DacMagic XS will remain a 2.0 device unless you switch it back.

The most difficult part of using the DacMagic XS with a Mac computer is finding the right kind of connector to attach it to the Mac. The DacMagic comes with a six-inch cable, but if you need a longer one, which I suspect many prospective owners will, the DacMagic XS shares

the same type of micro-USB connection as the Astell&Kern AK100, AK120, and AK240. A&K (and others) sell micro-USB cables on its site.

After attaching the DacMagic XS to one of my Macs (I tried it with a MacPro desktop, MacPro portable, and a Mac Mini), the AMSCP (Audio Midi Setup Control Panel) on each Mac recognized the DacMagic XS immediately. Once the DacMagic XS was set for USB 2.0 operation the AMSCP showed that it was capable of handling up to 192/24 files.

The only ergonomic quirk I experienced while using the DacMagic XS was that it was sensitive to static electrical shocks. All it took was a couple of strides across my office and back, then touching the DacMagic to generate enough of a static shock to disconnect the DacMagic from the USB buss—it would vanish from the list of DAC options in AMSCP. To correct the problem I needed to disconnect and reconnect the DacMagic XS from its USB connection, at which point it reappeared on the AMSCP DAC list and began playing as if nothing had happened.

DacMagic XS's Sonic Sorcery

I've seen the question posed on multiple locations on the Web, "Are thumb-drive-sized DACs a real sonic upgrade or merely convenience devices for accessing higher-definition music files?" In the case of the DacMagic XS the answer is clearly, "Both."

Since most prospective purchasers will want to use the DacMagic XS with headphones, I used a wide variety of different headphones and in-ear monitors with the DacMagic XS. With the most sensitive in-ears, such as the Westone ES-5 custom in-ear monitors (115dB sensitivity), the DacMagic XS did generate

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EQUIPMENT REVIEW - Three Miniature Portable USB DACs

some low-level hiss and background noise. With somewhat less sensitive in-ears, such as the Ultimate Ears In-Ear Reference Monitors, the DacMagic XS was guiet enough that the music came from a virtually silent background.

The DacMagic XS's headphone amplifier section had adequate gain and power to drive the Audeze LCD-2 and Mr. Speakers Alpha Dog headphones to satisfying volume levels with good bass extension. I was quite impressed by the combination of the DacMagic XS and the Grado RS-1 headphones, which can be guirky with portable gear. The bass sounded especially potent in this combination. I also enjoyed the venerable AKG K701 headphones connected to the DacMagic XS. While offering more of a leftbrained rendition of music than that of the Grados, the AKGs connected to the DacMagic had well-controlled upper frequencies that still had air and extension.

When connected to my desktop computeraudio system the DacMagic XS did a fine job of creating a believable three-dimensional soundstage that had all the weight, size, and imaging specificity of a "full-sized" DAC. When set to maximum output the DacMagic XS had enough gain to allow it be used like a fixed-output DAC into an analog preamp. While not quite as transparent and revealing as my reference DACs, including the April Music Eximus DP-1 or the latest version of the Wyred4Sound DAC2 DSD SE, the DacMagic did pass enough musical information to be completely involving. I never felt during my time with the DacMagic that it was limiting fidelity to the point of "grayness," which is the way some "entry-level" portable DACs sound.

Although it doesn't handle every audio format, and isn't DSD-capable, the DacMagic XS delivers a lot of functionality and sonic goodness for under \$200. For audiophiles looking for a road-warrior-worthy portable DAC that will be at home hooked up to any computer. portable or desktop, and successfully drive most headphones, the Cambridge Audio Dac-Magic XS DAC is a savvy and very affordable option.

Hegel Super DAC

Hegel gave audio journalists a sneak peak at the Super portable DAC during the 2013 Rocky Mountain Audio Fest. I was immediately impressed by the Super's solidity, both physically and sonically, and I looked forward to hearing the final consumer version. Flash forward six months and a Hegel Super DAC appeared at my doorstep. I'm happy to report the production version is just as solid as the pre-production version. Initially the Super was to be priced at \$399, but the current "street price" is \$299.

Hegel made some very specific design decisions for the Super DAC. First, it is a USB 1.0 device that needs no drivers with any computer. This makes it truly plug-and-play, but it also limits the Super DAC to a maximum sample/ bit rate of 96/24. For some audiophiles the Super's lack of 192/24 and DSD support will make it a non-starter despite its sound quality.

Hegel doesn't supply much in the way of "under the hood" specifications, such as the DAC chip used, but according to its literature the Super does not have an asynchronous USB interface, which Hegel considers to be more marketing hype than actual technologi-

cal advantage. Hegel's published design goals for the Super were "to be extremely silent, to be able to have flat frequency response regardless of the headphone's impedance, and to have sufficient power supply to drive even difficult headphones." The Super does have some "trickle-down" technology derived from Hegel's full-sized DACs including Hegel's proprietary re-clocking techniques, and

an output stage with an extremely low output impedance.

Physically the Super is simple, yet impressive. Its chassis is milled out of a single piece of aluminum that measures approximately 3 5/16" by 110/16" by 3/4" and features an engraved Hegel logotype on the top and a satin brushed finish. One end of the Super has a micro-USB connection while the other has a mini-stereo/optical-digital mini-jack output. The Hegel Super is capable of serving as either a DAC or a USB-to-TosLink interface. "Legacy" DACs that lack a USB connection can be used in a computer audio system via the Super. But if you do use the Super as a USB convertor, it will still only support a maximum sample bit rate of 96/24.

A Super Sound

If you favor a headphone that needs some juice to sound its best, the Super could be a perfect traveling companion. But if your go-to traveling earphone is a high-sensitivity in-ear, the Super isn't the right DAC for you.

I tried the Super with a variety of

headphones, and even with the lowest sensitivity ones in my collection, The Audeze LCD-2s, I still needed over 15dB of attenuation (using iTunes/Amarra) to bring the volume down to a comfortable listening level. With the Westone ES-5 custom in-ear I used over 40dB of attenuation. That's a lot of excess gain in the system.

The headphone that I enjoyed the most cou-

SPECS & PRICING

Cambridge Audio DacMagic

XS

Inputs: USB 1.0 and 2.0 supported

Outputs: 3.5mm stereo

headphone jack

Sample/bit rates supported:

USB 1.0 Mode: 16/24-bit, 44.1kHz, 48kHz, 88.2kHz, 96kHz: USB 2.0 Mode: 16/24bit, 44.1kHz, 48kHz, 88.2kHz, 96kHz, 176.4kHz, 192kHz Dimensions: 1.25" x 0.4375" x

2.125"

Weight: 3.5 oz Price: \$199

cambridgeaudio.com

Hegel Music Systems Super DAC

Inputs: Micro-USB 1.0 mode Outputs: Mini-jack headphone and optical digital Toslink

USB interface: 24-bit/96kHz, plug & play via USB 1.0 protocol

Dimensions: 1.6cm x 0.6cm x

3.2cm Price: \$299

(mini-jack)

hegel.com

Resonessence Labs Herus

Inputs: USB 2.0 supported Outputs: 1/4" TRS stereo headphone jack

Sample/bit rates supported:

USB 2.0 Mode: 16/24-bit, 44.1kHz, 48kHz, 88.2kHz, 96kHz, 176,4kHz, 192kHz, 352.8/24, DSD 64, DSD128

Dimensions: 63.5mm x 31.7mm

x 19mm Price: \$350

reseonessencelabs.com

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EQUIPMENT REVIEW - Three Miniature Portable USB DACs

pled to the Hegel Super was the Audeze LCD-2 (Bamboo version). The Hegel was able to propel the LCD-2s in an authoritative manner that I usually hear only from larger, AC-powered desktop headphone amplifiers. Bass was tight, controlled, but still powerful. Also the lack or electronic "grain," due in large part to the Super's 140dB S/N figure, contributed to the ease with which I could listen into any mix.

If your primary use for a portable DAC is with a desktop system or powered speakers, the high output of the Super will be a good thing. Hooked up to my desktop the Super sounded more like a "big boy" DAC than a portable USB device. If you listen for "pace" you'll appreciate the Super's ability to drive a system forward with alacrity.

In my desktop system the Super delivered a well-defined soundstage with precise lateral imaging. Depth was also clearly articulated, but with a hair less dimensionality than I've heard from my reference full-sized DACs such as the Wyred4Sound DAC-2 DSD SE. Bass extension and power through the Super, however, was equal to the best DACs I've heard in my desktop system including the Wyred4Sound Dac.

Although the Hegel Super does lack some features, such as DSD and 192/24 PCM capabilities, it makes up for it with its solid sound and ability to do double duty as a USB to TosLink convertor. Given the number of other portable DACs available at a similar price, the Hegel faces some tough competition. But for some prospective users, the Super's powerful output and easy setup might be deciding factors in its favor.

Resonessence Labs Herus

The Canadian-made Resonessence Labs Herus is the most expensive portable DAC in this survey at \$350 street, but it is also the most flexible in sample- and bit-rate capabilities. This lipstick-sized DAC supports PCM up to 352.8/24 as well as DSD64x, DSD 128x, and DXD files. So, regardless of how you like your high-resolution files, the Herus will play them.

Machined out of a solid block of aluminum, the Herus measures 2.5" x 1.25" by 0.75" and weighs less than a pair of CD jewel cases. On one end you'll find a full-sized USB B connection and at the other a full-sized 1/4" stereo connection. For those audiophiles who already have a premium USB cable, Herus' use of a regular as opposed to mini- or micro-USB could be a major advantage over some other portable DACs. Also the full-sized instead of mini-stereo plug means that you can use headphones with a standard 1/4" plug without needing an adapter.

The Herus puts out 2.4 volts from its headphone output at maximum output, giving it a slightly higher level than DACs set for the usual standard of 2 volts. Inside you'll find an ESS 9010-2M DAC, configured using Resonessence Labs' custom code and asynchronous algorithms that run in a generic Cypress USB interface chip. With its low 0.2 ohms output impedance the Herus should be able to handle any headphone from 32 ohms to 600 ohms with no issues.

Like the Cambridge Audio DacMagic XS, the Herus has its own volume control. But unlike the Cambridge Audio DAC, which has an analog control, the Herus adjusts its volume via the ESS 9010-2M DAC's internal 32-bit digital attenuation control. The Herus will also work as a DAC for your iPhone or iPad with the addition of an Apple Lightning-to-USB camera adapter to connect the iPhone or iPad to the Herus. Some Android devices are also supported, such as the Samsung Tab3.

A DAC of All Trades

During my time with the Herus I've thrown every file format in my music library at it with 100% success and playability. The only ergonomic issue I've had with the Herus is that when I changed headphones the Herus reverted to full output level, which can be quite loud with high-sensitivity headphones.

Resonessence includes the following warning on their main Herus info page, "Important-Please note: some (if not all) software on the PC, MAC, and Linux will, the first time Herus is connected, set the volume to OdB. That is, to the highest volume level. This may be very loud in the headphones. On subsequent connections the music player application will recall the last volume setting, but we have seen instances where plugging into a different USB port again sets the volume back to OdB. Consequently, we strongly recommend that you plug the Herus into any new port on your computer with the headphones unplugged, and set the volume to a reasonable level prior to plugging the headphones into Herus."

Occasionally when I switched headphones I didn't get music; instead all I heard was noise—loud digital-sounding noise. The solution was to close down iTunes with Amarra Symphony and then reopen them and the problem disap-

peared. I soon developed a standard procedure with the Herus when I switched headphones—never put on the headphones until I made sure that music, rather than noise, was coming through the drivers. [Resonessence says that this problem arises only if the user swaps headphones while music is playing. If the music is paused, this problem won't occur.—Ed.]

I used the Herus with a wide variety of headphones. Only with the 115dB sensitive Westone ES5 custom in-ears did the Herus produce some low-level hiss and background noise. With the 112dB sensitivity Ultimate Ears In-Ear Reference Monitors hiss was reduced to the point where it was almost inaudible. Combined with any headphones of less than 95dB sensitivity the Herus amplifier section was completely silent.

Because the Herus does produce an output that is slightly higher than the industry standard. I was concerned whether its volume could be adjusted and attenuated so that it would work successfully with a wide variety of headphones. To get an idea of how much volume variation was needed with different earphones I made note of the comfortable volume settings for a wide variety of cans. The Westone ES5 custom in-ears required the most attenuation, -40dB. In comparison, less sensitive earphones such as the Mr. Speakers Alpha Dog headphones needed only -15dB of attenuation. The most power-hungry headphones I had on hand, a Beyer-Dynamic DT990 600-ohm resistance earphone, required only -12dB of attenuation.

The first time I heard the Herus in my hotel room at the 2013 Rocky Mountain Audio Fest, driving my Audio-Technica ATH-W3000 ANV

EQUIPMENT REVIEW - Three Miniature Portable USB DACs

headphones, I was transported back to the moment the recordings were made. The Herus produced a level of sound quality that rivaled any DSD DAC I'd heard, regardless of price. During the many hours I've used the Herus since RMAF it has continued to impress me with its transparency and ability to impose little in the way of additive colorations onto the music.

I've been making DSD 128x recordings of live concerts since 2008, when I first started using the Korg MR-1000 DSD recorder, so I have plenty of DSD material in my music library. Whether the music is classical or acoustic folk, the Herus plays back my DSD files perfectly without a single odd noise, hesitation, or drop out.

The Herus proved to be as adept with high-resolution PCM files as it was with DSD. I usually make 192, 96, and 44.1 files from my DSD recordings using the Korg AudioGate application. Listening to the PCM files through the Herus I became aware of differences between the DSD and the PCM files. The DSD version sounded the best, followed closely by the 192/24 versions. Listening to the 96/24 versions compared to the DSD I was immediately noticed that the room sounds and trailing edges of the music were ever so slightly truncated compared to the 192/24 or DSD versions.

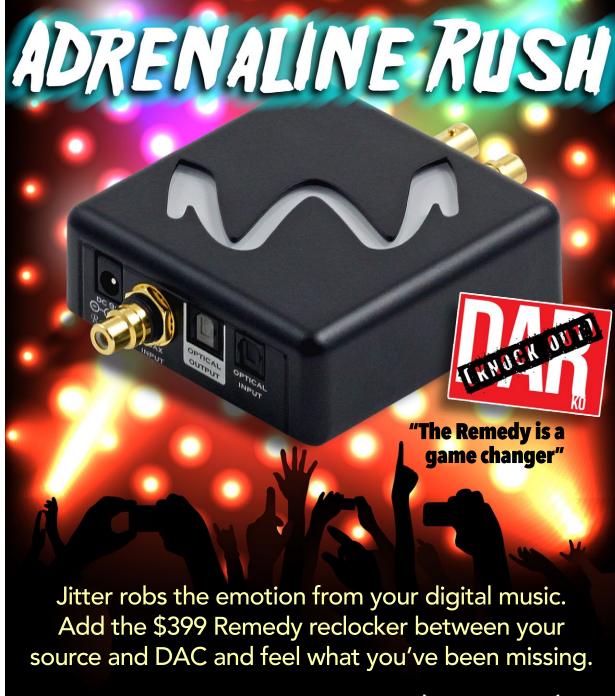
Switching the Herus over to desktop audio duties I was, again, impressed by its overall sound quality. Using the Herus as a source for my desktop system merely required attaching a 1/4"-stereo-to-RCA stereo pair adapter and then connecting it to a preamp via a 1-meter analog cable. Imaging was as precise as the Wyred4Sound DAC-2 DSD SE, as was depth recreation and low-level detail. When I set up a

matched-level A/B with both the Wyred4Sound DAC-2 DSD and Herus connected to the Nuforce MCP-18 using my own DSD recordings I was hard-pressed to tell which DAC I was listening to; they sounded that similar to each other.

Three DACs, Three Good Choices

Of the three portable DACS I reviewed, all three offered good performance at entry-level prices. The \$350 Resonessence Labs' Herus does seem to be the most "future-proof" of the three with its support for DSD, PCM, and DXD files, so in the long run it could prove to be the best overall value. Audiophiles who prefer an analog volume control and the ability to adjust the volume from the DAC itself may gravitate toward the \$199 Cambridge Audio DacMagic XS, which also supports USB 2.0 so it can play up to 192/24 files. If you have an older DAC that you still wish to enjoy with computer-audio sources, the \$299 Hegel Super offers you the option of converting USB to TosLink so that you can use "legacy" DACs with your computer-audio system. And while the Super is limited by its USB 1.0 protocol allowing only a 96/24 maximum sample/bit-rate, its dual functionality is a bonus that makes it a much better value than if it were only a DAC.

Whichever of these three portable DAC options you choose, you will be rewarded by better sound and greater flexibility in headphone options due to their ability to drive headphones with more power than your computer or smartphone. Any one of them will add only a few ounces to your traveling kit, yet make the time in your hotel room or in a plane far more pleasurable. tas



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Arcam FMJ A19 & airDAC

Dynamic Duo

Spencer Holbert

hough integrated amplifiers have been around a long time, they are often mixed bags. Some models try to do too many things at once, losing focus on what we listen for first—sound quality. Whether you are looking to purchase your first real hi-fi component or an affordable option for a second system, the Arcam FMJ A19 integrated amplifier delivers real high-end sound quality, without the high-end price.

Functionality and Sound

Though the A19's design is understated-downright minimalist, actually-don't let its lackluster exterior fool you. The A19 borrows heavily from its bigger and more expensive siblings, with features like a toroidal transformer for its 50W Class AB output stage and the same volume control as top-tier FMJ models. With seven single-ended inputs, the A19 makes plenty of room for those with lots of sources-enough for two turntables, two DACs, a tuner, and your dad's old tape deck. Even if you don't need all of them today, those extra inputs may come in handy down the road. Because of the resurgence in vinyl, Arcam has upgraded the A19's built-in phonostage to better reflect current listening preferences. The remote is a basic design, with the ability to control every feature except for a few user preferences that are adjusted via several buttoncombinations on the front panel. Despite some guirks that I will discuss later, overall this integrated offers everything you need for a mere \$999.

Now for the real meat—sound quality. Using components that I am very familiar with, I tried to determine exactly what the A19 does or does not bring to the listening room. After level-matching the A19, I was actually shocked to hear significant differences between the Arcam and several other integrateds of similar specifications. The A19 is incredibly quiet compared to many components. Even when I turned the volume all the way up there was no audible hiss coming from the speakers; so you don't need to worry about distortion and noise with this amp. I hate to be the one who brings up THD, because as we all know this is by no means a measure of sound quality, but the A19 has a harmonic distortion rating of 0.003% at eighty-percent power—and that's low.

Listening to Ludovico Einaudi's "Experience" from *In a Time Lapse* (CD and vinyl), I heard a smooth high end that never sounded overly bright or grainy. The A19 was convincingly realistic on Einaudi's piano, and when the violins—arguably the most difficult instrument to accurately repro-

duce-joined in with their unusually sonorous solemnity, I felt like I was listening to a genuine high-fidelity product (even though Arcam doesn't like to be associated with a "hi-fi" sound). So far so good.

I played the track several more times, then shifted my attention to the soundstage, which was on-par with what you would expect in this price range: generally wide, sufficiently detailed, with overall tight imaging (though somewhat misplaced locations compared to ultra-high-end systems). Soundstage depth was less deep compared to those more expensive systems, but nevertheless was plenty deep to satisfy all but the most demanding. What makes In a Time Lapse great for soundstage testing is that it was recorded in an Italian monastery with sound quality in mind, so it's very easy to tell when something is amiss. On "Experience," a harp placed behind and to the left of the piano is gently plucked amid the increasingly energetic violins. The Arcam A19 had sufficient resolving capabilities to allow the distant harp to be heard, though don't expect extreme soundstage depth with this-or any-integrated in this price range.

Maybe I'm being a little too tough on the A19's lack of soundstage depth. After all, imaging was fairly tight and was for the most part reasonably well executed. No "I'm there!" moments occurred, but nothing was egregiously wrong—complex soundstages are a difficult thing to resolve on such a tight budget. Then another thing announced itself: the slightly tubby bass. Low-end damping ability was a little lacking with the 4-ohm Endeavor E3 floorstanders. Yet, when compared to similarly priced integrateds, lowend handling was equal to or slightly better, so no worries here.

I don't want you to give you the wrong impression of the A19 by pointing out these things—they are meant to give you a realistic idea of what a \$999 integrated amp can accomplish. With regard to sound quality, \$999 buys you a musical, involving presentation with above-price-point performance in imaging. At 50W into 8 ohms and 90W

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EQUIPMENT REVIEW - Arcam FMJ A19 and airDAC

into 4 ohms, the Arcam has plenty of power to rock out with most dynamic speakers, and its build-quality is solid. Really, it feels like a tank.

Overall, the A19 is an integrated that I would want to own at this price point. In fact, a hi-fi newcomer friend of mine purchased it after a weekend of listening-that's how much he liked it. Just know that "best-sound-ever" claims can't be firmly rooted in the sub-\$1500 category; this integrated will help hook you on high end, but it won't be the end-all, which is exactly what you want when you're starting out-something that's so good you want even more.

British Quirks

The A19 has a few guirks that can be misconstrued as design flaws by those unfamiliar with the new British energy-consumption standards, so don't panic if you come across them out of the box. I generally leave new components on 24/7 during the first week or so, but this proved problematic with the A19 due to the integrated's auto-shutoff function. The first time this happened to me, it took a little while to figure out what had gone awry. I went in my listening room to find the A19 in standby, and pressing the power button and volume controls on the remote didn't seem to wake it up. I assumed the integrated just needed to be cycled, so I turned it off and back on, and had the same issue. It turns out that if you power off the unit via the front-panel power button, the A19 defaults to standby, supposedly in case of power outages. Finally, I turned the volume knob on the unit and presto-it came alive again. The solution is to press the "Aux" and "Balance" buttons simultaneously to adjust this feature, and disable autostandby altogether.

Arcam airDAC

Along with the Arcam FMJ A19, I also received Arcam's latest foray into networked DACs, the Arcam airDAC (\$699). I consider myself pretty computer-savvy, so I found setup was straightforward. But if you've never fiddled with a wired or wireless computer network in your house, the airDAC is going to throw some curveballs. The manual stated that it was possible to set up the airDAC wirelessly and provided an IP address to do so, yet I couldn't connect without first using a direct-wired connection between my laptop and the DAC. Like I said, this is nothing new for people who have set up a home network, but if you have a desktop computer and no network, you will need to connect the computer and airDAC directly via Ethernet cable in order to adjust initial settings.

After the airDAC was set up, it was pretty smooth sailing. Using the free Arcam Song-Book+ app for iPad, the airDAC found my RAID drive and other network-attached hard drives. The airDAC automatically indexed music from the hard drives, though it didn't distinguish between hard-drive partitions and displayed duplicate songs-a minor detail. The app is extremely fast, though it is a "light" version and doesn't display album artwork while scrolling through artist or album lists.

The airDAC features four input methods: Tos-Link, digital coax, Network Attached Storage (NAS) management, and Apple AirPlay. The first three methods worked just fine, sounded

great, and provided everything you would expect from a networked DAC in this price range. The airDAC was about ninety percent of the sound capability of standard (non-networked) component DACs of similar pricing, which was more than I expected. The Apple AirPlay feature was, well, underwhelming. It's limited to 16/48 (Apple's fault, not Arcam's), which is something I can live with, but there was a major delay between streaming from my laptop and the airDAC. When I hit Pause, almost three sec-

onds went by before the song would pause. When I attempted to stream Netflix movies, the video and sound were so out of synch that I switched back to some Bluetooth speakers after only thirty seconds. This lag persisted even after I restarted both my computer and the airDAC and checked my settings. I have gigabit routers and switches in my network, and such lags have never been an issue with other devices. The point is, forget the Apple AirPlay and stick with the other inputs.

But the airDAC has one huge advantage over the competition: Music streaming from a NAS drive. With the airDAC, gone is the need to have a noisy computer or a finicky Mac Mini in your listening room. Simply transfer vour music to an external NAS drive. plug it into the airDAC via Ethernet, and you're done. For \$699 plus a NAS drive (roughly \$100-\$200 depending on size), you get a music server and DAC that can be controlled from your smartphone or tablet, and that's huge.

The airDAC can build playlists from multiple drives, stream everything seamlessly and with great sound quality, and it's a bargain. Unfortunately, the UPnP network streaming is limited to 96kHz/24-bit, but most people looking for an affordable music-server solution will be just fine with the airDAC's capabilities. For those of you who have amassed an enormous number of digital music downloads, like I have, the airDAC will satisfy your music-manage-

ment needs. tas

SPECS & PRICING

Arcam FMJ A19

Inputs: Six line-level RCA, one moving-magnet phono input, one 3.5mm

Outputs: Record out, preamp

Power: 50W into 8 ohms: 90W

into 4 ohms

S/N ratio: 105dB

Frequency response: 20Hz-

20kHz +/-0.2dB

Dimensions: 17" x 11" x 3"

Weight: 19 lbs. Price: \$999

Arcam airDAC

Inputs: TosLink, digital coax, network UPnP (Ethernet), AirPlay

Outputs: RCA, digital coax

DAC chip: TI PCM5102 Frequency response: 10Hz-

20kHz +/-0.1dB

S/N Ratio: 106dB

Output level: 2.15V RMS Sample rate: Up to 96kHz/24-

Dimensions: 7.5" x 4.75" x 1.75"

Weight: 2.5 lbs. Price: \$699

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Rotel RDD-1580 Digital-to-Analog Converter

Great Digital Made Affordable

Spencer Holbert

t's an amazing time for computer-based audio. It wasn't too long ago that DACs connected to a PC or Mac were limited to CD-quality resolution and relied heavily on upsampling or multiple conversions to match the capabilities of transports. What's better than a computer that manages a seemingly unlimited number of highres songs at 192kHz/24-bit (or higher)? Here's what: the fact that you can now own an entire computer-based system at a fraction of the cost of components from just a couple years ago, without sacrificing sound quality. As with all facets of life there will always be \(\bar{u}ber\)-expensive gear that can do it better, but the new \$799 Rotel RDD-1580, with its myriad inputs and superb design, represents an affordable option that won't become obsolete in a few years. Plus the RDD-1580 is more than just a checklist of features; it's a true hi-fi component with gripping sonics that run with the best of 'em without running you into debt.

Back to Basics

First and foremost, a DAC should have the ability to handle any digital input you could ever hope to use. There's no point in purchasing one component for your transport, another for your computer, and then another for your iPod. The Rotel RDD-1580 has six inputs: two optical TosLink, two digital coax, one computer USB, and one iDevice USB on the front panel. It's easy to scoff at that last one, because the front-panel USB input is limited to 48k/16, but it's a great option when friends come over and want to play "that new song you just have to hear" without the hassle of ripping the music from their iPhone or iPad onto your computer. The front-panel USB input also doubles as a charger, which was superhelpful when my iPad-aka my computer-audio command center-ran out of juice.

For high-res computer audio, I connected the RDD-1580 via USB and TosLink to my iMac with an external 12TB RAID NAS drive, selected the Rotel under outputs, then fired up iTunes with Amarra Hi-Fi. It's nice that most Macs feature optical and multiple USB outputs, because that not only allows for easy A/B comparisons from the same source, it also allows for comparisons with multiple DACs. Like I said, it's an amazing time for computer-based audio.

Maybe I'm a bad reviewer for admitting this, but I no longer use a transport for SACDs-I rip all of my SACDs to my computer using a Playstation 3. Caveat: This requires an older firmware version that can read SACDs and convert them into an ISO file, then more software to convert the files into PCM that can be streamed to your DAC, all of which can be a little daunting for a newcomer to computerbased audio. If you have a large collection of SACDs, a transport is still the easiest option; but if you're up to the challenge it can be funyet very time-consuming-to finally transfer those SACDs to your computer and break free of the physical constraints of changing discs. This topic probably warrants an entire article, but let's get back to the DAC.

Rotel has long been known for high-quality components at an affordable price, and the RDD-1580 is no exception. Unlike most DACs in the same price range, the RDD-1580 features two Wolfson WM8740 converters—one for each channel—a Rotel-designed toroidal transformer, and slit-foil capacitors

to supply the DAC with great power. If you've been following DAC technology for a while, you'll know that sound quality is not just about the quality of the converters, but also the digital filters, output stage, and power supply; in this regard the "dual-mono" design of the RDD-1580 really shines. Unless you are getting into DSD, this DAC has everything you need to rule the digital world. Oh, and it has a remote! More on that in a bit (pun intended).

Bits, Bytes, and the RDD-1580's Sonic Capability

If the world of digital audio were simply eight bits in a byte, any ol' DAC would do. It's the aggregate design that counts, not just the mathematical sum of its parts. When I listened to the RDD-1580, it was obvious that Rotel always had high-quality analog sound as its goal. Sound quality seems to be an afterthought for many sub-\$1000 DACs that have the capability to handle 192/24 PCM signals; heck, there are \$30 DACs that can do this. For those of you who remember the early mindset when turntables were simply something that spins a record, this will be a little déjà entendu.

When testing DACs, my go-to music is always something from the Ultimae record label, purveyors of incredible ambient soundscapes from artists like Aes Dana, Solar Fields, Hol Baumann, and Carbon Based Lifeforms. This type of music is perfect because it's not only great to listen to, but also pushes the limits of a system in a controlled manner that orchestral movements just can't touch. Ambient music plays with soundstage width, depth, height, and extreme frequency response with lightning-fast speed. Such ambient music is

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EQUIPMENT REVIEW - Rotel RDD-1580



like a modern-day version of classical music in that it paints a landscape and takes you on a journey, except that the sound is phasey left and right, front to back, and top to bottom.

What's amazing about the RDD-1580 is that it took the massive amount of sound from Solar Fields' Movements and translated it into a beautiful soundscape that was far wider and deeper than that of my comparison DAC, which retails for about the same price. On "Sol," the first track of the album, the bass seemed to rip from the ground and leap into my chair, while simultaneously the high-frequency zipsand-zaps flew from beyond the outer edges of the speakers to land centerstage, dance in mid-air, then retreat well to the rear. With the comparison DAC the effect was "similar," but the soundstage was truncated, never extending beyond the edges of the speakers, and had about half the depth. This was using the same USB cable, the same computersame everything. For the same price, the RDD-1580 put the comparison DAC to shame, and was far more engaging in its ability to elicit a visceral response to the music. Several times during the track "Discovering" I caught myself clenching my fists and sliding toward the edge of my seat, all because the RDD-1580 made the music that much more gripping.

I wanted to throw another variable into this aural showdown and choose an album that I have on vinyl and digital. If you haven't

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heard Zero 7's When It Falls, it's an absolute must-own. This genre-bending album employs multiple "jazz" singers-both male and female-throws in violins, pianos, electric basses, and acoustic guitars, then interlaces everything with down-tempo ambient music to create an intoxicating sound. If you've seen the movie Garden State, or TV shows like Top Gear, CSI, or Smallville, then you've heard Zero 7. So I pulled out the vinyl version of When It Falls, threw it on an analog setup that cost the same as the RDD-1580, and A/B compared the digital to the vinyl. I'm going to get hate mail for saying this, but on the track "Somersault," underrated jazz singer Sia Furler sounded *much* better than with the vinyl setup of similar cost, not to mention that the instruments were more distinctly defined within the soundstage. Even though I liked the "vinyl sound" more than the digital, it couldn't compete with the RDD-1580's imaging, lack of smear, and superb dynamics. Before this, if someone would have asked me, "For \$800, should I go digital or vinyl?" I would have said vinyl all day long. Yet, the RDD-1580 made me reconsider that question, and then ultimately decide in favor of it over an analog front end for the same price. Yes, I'm going on record and saying that if you have \$800 and have to choose between vinyl and digital, buy the RDD-1580 first.

But maybe that was just a fluke, eh? Let's

try the same vinyl/digital comparison with James Blake's "Retrograde" from his second album, Overgrown. This track features Blake's incredible vocal range as he hums R&B-style up and down the octaves, backed by a simple beat and piano. Yet again, the RDD-1580 easily beat out the other DAC and comparable analog front end. The RDD-1580's soundstage was deeper, the piano was spatially separate from the vocals and the beat, and everything sounded tighter. I did the same test again with Portugal. The Man [sic], Neko Case, Wayne Shorter, Miles Davis, ZZ Top, and dozens more, and each time the RDD-1580 outperformed the "other DAC" and the analog setup.

I wanted to do this same "triple comparison" in another system located in an entirely

different room, so I went over to a fellow audiophile's house and began the process all over again. I didn't necessarily expect the same conclusions, but I was curious whether I simply preferred the sound of the RDD-1580 through my amp/speaker combination. Maybe the RDD-1580 better complements my system, I thought. After three or four hours of A/B/C testing, it was abundantly clear that the RDD-1580 still sounded better than the alternatives in my friend's system. A couple days later, I received an e-mail from this friend, who had gone out and purchased the DAC for himself. If you are in the market for a DAC and have a max budget of \$1000, you would be foolish not to audition the RDD-1580.

Other Likes, and a Few Minor Dislikes

Like I said earlier, the RDD-1580 comes with a remote, which when connected via USB controlled PLAY, SKIP FORWARD, and SKIP BACK; obviously this didn't work with the other inputs. But these controls were a little finicky: The Pause button didn't work via USB, but if you hit the PLAY button again it would pause the track. I could skip forward and back with the respective buttons, but I couldn't fast forward, nor was there any volume-control capability. I used the RDD-1580's remote mainly because it was faster than unlocking my iPad, letting the Remote app sync, and then trying to control the computer. But ultimately I preferred using the iPad to control the computer, rather than Rotel's remote.

SPECS & PRICING

Inputs: Two digital coax; two optical TosLink; one PC-USB;

one front-panel USB

Output: RCA; XLR

DAC: Dual Wolfson WM8740s Frequency response: 10Hz-

95kHz

S/PDIF LPCM: up to 192kHz/24-

Rear-panel USB: Asynchronous, rotel.com

192kHz/24-bit

Front-panel USB: Up to

48kHz/16-bit

Dimensions: 17" x 2 1/8" x 12

1/2"

Weight: 11.24 lbs.

Price: \$799

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EQUIPMENT REVIEW - Rotel RDD-1580

This next one might just be my personal preference, but the blue indicator light, which rings the circumference of the RDD-1580's power button, stays illuminated whether the DAC is on or in standby mode. Several times I thought the DAC was on when it was actually in standby, and vice versa. The only way to tell if the DAC is actually on is to look at the small input indicator light, or the sample-rate indicator. Again, this isn't a huge deal, though it is somewhat strange to not indicate on/ standby individually.

Another thing that might throw a lot of people off is the fact that you need to manually switch between USB 1.0 and USB 2.0 modes by holding the PC-USB input button for five seconds (this is a one time thing). Windows users will need to install a supplied driver in order to utilize USB 2.0. For Mac users, this is already taken care of, but I couldn't determine whether the switch from USB 1.0 to 2.0 actually made a difference in my Mac setup, because the 192kHz indicator light was illuminated before I read the owner's manual (I might have been overeager).

I really like the RDD-1580's sleek, slim design; the review sample I received came with the silver faceplate, which just so happens to match a lot of my other gear. Plus, the RDD-1580 ran surprisingly cool, which means that you could place a preamp on top of it without worries; this is most likely due to the fact that it only draws 25W when on, and less that 0.5W when in standby.

Another really cool feature is that you can stream music via Bluetooth when the supplied Bluetooth adaptor is plugged in to the frontpanel USB input. The Bluetooth dongle is tiny and unobtrusive, and was a lot of fun to use when I worked on my laptop and wanted to stream music from my favorite listening chair. You can also stream music from smartphones and tablets, but I didn't test out what would happen if multiple devices tried to connect via Bluetooth simultaneously, à *la* during a party where multiple people want to play phone DJ.

Aside from these few minor things, the RDD-1580 was flawless, both in features and in sound quality. It is by far the best DAC that I've heard in this price range, and probably would beat out most DACs double or triple its price. Does it beat out a \$10,000 DAC? Sorry Rotel, but the big boys still win in overall sonics (not to mention DSD capability). But if you are looking for a DAC that costs even \$2500, don't overlook the RDD-1580. I definitely hope Rotel will let me hang on to this one a while longer. tas





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inding an excellent digital-to-analog converter (DAC) for under \$1200 used to be an audiophile fantasy. But with advancements in DAC design and reductions in the price of the high-quality parts needed to make these designs a reality, the cost of entry into high-end digital sound has come down substantially. In Issue 228 I looked at three excellent under-\$1200 DACs: the Channel Islands Transient Mark II (\$699), the Lindemann USB-DAC 24/192 (\$1100), and the NuForce DAC-100 (\$1095). Now Musical Surroundings has an entry in the under-\$1200 DAC competition, its new MYDAC II (\$1199). The MYDAC II may be forgiven its less than totally original name because of the interesting technology under its hood. Does the MYDAC II's unique and soon-to-be-patented digital technology make it a game-changer? Let's find out.

MYDAC II Features

From the outside the Musical Surroundings MYDAC II doesn't look that special. Available in silver or black, the MYDAC weighs only 3.4 pounds and measures 9 7/8" square and 2" high. The front panel is clean but certainly not deluxe, with a single pushbutton that selects one of three inputs—SPDIF, USB, or TosLink. The

center of the front panel has three additional LEDs that indicate whether the sample rate is 44, 96, or 192kHz.

The MYDAC II's rear panel has one pair of single-ended RCA analog outputs, TosLink, USB and SPDIF inputs, and a barrel connector for an outboard wall-wart power supply. The RCA outputs are fixed-level so you will need to use

the MYDAC II with a preamp or an integrated amplifier with level controls.

The MYDAC II supports sample and bit-rates as high as 192/24 through its SPDIF digital input. The TosLink input can support up to 96/24, but its USB input can only handle up to 48/16 files. To play higher sample and bit-rate files from USB sources via the MYDAC II will require an external USB-to-SPDIF converter such as the M2Tech hiFace, Human Audio Tabla, or Empirical Audio Off-Ramp. The MYDAC also has no provisions to support DSD or DXD, but that capability can be added to the MYDAC (or any other DAC) through the acquisition of a Schitt LOKI (\$149) DSD-dedicated DAC.

MYDAC II Patentable Technology

So far, the MYDAC II seems like a fairly ordinary mid-priced DAC. Inside is where the MYDAC II differentiates itself from the competition. Designed by Michael Yee, who is best known for his analog and phono preamp products, the MYDAC II takes an idea from the analog world and adapts it to digital circuits.

Michael Yee's explanation of his new methodology makes it seem simple. "In a system with 16-bit ADCs and 16-bit DACs (CD audio), both ADCs and DACs have roughly the same contribution to quantization error. It is the high frequencies that need more resolution in a sixteen-bit system. If one were to emphasize the high frequencies in the digital domain (making the high frequencies represented by much bigger numbers), pass them through the DAC at a much higher resolution, then EQ them back down in the analog domain to make the system flat, the contribution to quantization error due to the DAC would be essentially

zero." Yes, you got that right; the MYDAC II uses equalization to improve the system's overall resolution, especially in the mid and higher frequencies.

According to Yee, "Our digital EQ is something akin to RIAA on LPs, though it starts at 40Hz and ends at 4kHz, rising at 6dB/octave. The boost at 4kHz is 40dB, where the curve ends. The analog de-emphasis looks very much like a phono preamp doing the exact complement to the digital EQ. The digital EQ is implemented in FPGA (Field Programmable Gate Arrays) in order to be an exact complement to the analog de-emphasis. This means that the higher frequency range is represented by numbers 100 times bigger when going through the DAC and the DAC appears to be 100 times more accurate."

The MYDAC II employs a 24-bit DAC so that Yee's design can support his EQ scheme. "For 24-bit audio, the MYDAC II 'throws away' eight bits of resolution in the bass and gives them to the high frequencies via noise shaping. For 16-bit audio, there is no reduction in resolution in the bass." The patent on Yee's new digital methodology, called MODR (Musically Optimized Digital Reconstruction), is currently pending.

Besides being used in the MODR process, FPGAs are also used in the MYDAC II for all internal signal processing and timing. Musical Surroundings claims that FPGA remove "another jitter source." The heart of the MYDAC II is a Texas Instruments TI PCM1798 delta/sigma chip. Low distortion op-amps from TI are used in the MYDAC II's output filter. This is the first time that Yee has used integrated op-amps in one of his designs, but they were

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EQUIPMENT REVIEW - Musical Surroundings MYDAC II

"a high-quality affordable implementation to introduce this new technology."

MYDAC II Sound

With this new whizz-bang digital technology does the MYDAC II sound different from all other DACs? Yes and no. At first listen the MYDAC II seems like many other good DACs-clean, dynamic, and harmonically well balanced, but further listening reveals its special sonic characteristics. Unlike many otherwise excellent DACs the MYDAC II sounds less "hard" without sounding mushy or soft. Details are every bit as clean and clear as through more conventional designs yet they lack that sometimes overly aggressive leading edge.

The MYDAC II also excels at preserving inner detail and low-level information. On Andy Statman's brilliant, recently released album *Superstring Theory*, the MYDAC II preserves the gutty texture and the upper harmonics of Statman's Kimble mandolin as well as the trailing edges of Jim Whitney's stand-up acoustic bass. On one particular tune, "French Press," it's easy to hear that Statman's mandolin was being routed through a less than pristine preamp because of the MYDAC II's excellent retention of inner detail. Like the late Glenn Gould, Statman "vocalizes" while he plays. His humming, although at extremely low levels, is easy to follow through the MYDAC II.

Listening to my own live concert recordings through the MYDAC II, I was impressed by its ability to preserve and illuminate the sounds of the concert hall—not merely the sounds on stage, but also the peripheral sounds from off-stage and in the audience. Instead of offering

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a "velvety black," monolithic, and artificialsounding background, the MYDAC II preserved the room's myriad low-level sonic cues so that the entire space seemed to "breathe" in a more realistic manner.

Dynamic contrast and differentiation, especially during quiet passages, rank up there with the best I've heard, but only when using the MYDAC II's best input which is certainly SPDIF. TosLink and USB don't have the same dynamic acuity or jump factor as SPDIF. The USB input should be considered more of a "convenience input" than a primary source. If you plan to use the MYDAC primarily for USB, I strongly recommend acquiring an outboard USB-to-SPDIF converter. The difference in sound quality between the MYDAC II's built-in USB solution and the Human Audio Tabla or Empirical Audio Off-Ramp is not subtle. The MYDAC II's USB input sounds flat, gray, and dynamically constricted when compared to the same source routed through an outboard USB converter. After several days, during which I did quite a few A/B comparisons between the internal USB and external converters, I did all my subsequent listening using the MYDAC II's SPDIF input.

Since the frequency range, especially in the upper registers, is boosted and then returned to pre-boost levels by the MYDAC II, I was especially curious about any noticeable negative effects from this resolution-enhancement and noise-reduction scheme. I spent quite a bit of time during my listening sessions trying to hear any augmentation to the upper midrange and lower treble that could be attributed to the MODR circuitry. After many hours of listening to a wide variety

of music at many different resolutions, I can state confidently that MODR had no noticeable additive sonic effects. Even on some of my ruder, upper-midrange-dominant mixes, such as the original release of the Clash's *London Calling*, the MYDAC II didn't add any steeliness or additional rasp to the sound. If anything the upper frequencies on more primitive recordings were slightly less harsh, but without any reduction in dynamic contrast.

I hesitate to call the sound emanating from the MYDAC II "sweet," since that implies a certain loss of upper-frequency incisiveness, a subtractive coloration that is not part of the MYDAC II sound. But the MYDAC II does have the ability to mitigate harshness and aggression in the upper midrange as no other DAC I've heard can. Some listeners would call this a "more analog-like" sound, but unlike some analog, which has a degree of built-in compression and limiting that mitigates exces-

sively hard leading edges, the upper midrange transients aren't blunted or reduced through the MYDAC II. They seem to have less odd-order distortion and sound more relaxed and natural.

Could a MYDAC II Be In Your Future?

Nowadays audiophiles have more highquality digital-to-analog converters to choose from than ever before. Most, even those priced at the entry level, can deliver a level of sonic quality that was unavailable in any digital product, regardless of price, just a few short years ago. The Musical Surroundings MYDAC II offers a unique solution to the problem of accurate digital sound reproduction by augmenting its performance through mid- and upper-frequency equalization. The result is a DAC that sounds "less digital" and reproduces upper frequencies with less of a hard, amusical edge.

The principal shortcoming of the MYDAC II is that its USB input is limited to 48/16 and can't support any higher sample- or bit-rate. The USB input is also not nearly as good as the MYDAC II's SPDIF input in overall sound quality. To hear the MYDAC II's true sonic potential with computer-based music files you will need to add a high-quality USB-to-SPDIF converter.

If you're an analog-only audiophile who has listened to a lot of DACs but never heard one that sounds right to your ears, you may want to give the MYDAC II a try. It could be the DAC that makes it possible for you to enjoy digital for the first time. Its

SPECS & PRICING

Inputs: Three digital (SPDIF up to 24-bit/192kHz, TosLink up

to 24-bit/96kHz,

USB up to 16-bit/48kHz)

Frequency response: 20Hz-

20kHz

Output voltage: 4V

Output impedance: 100 ohms

Distortion: 0.02%

Dimensions: 9 7/8" x 2" x

9 7/8"

Weight: 3.4 lbs.

Price: \$1199

MUSICAL SURROUNDINGS

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Wyred 4 Sound DAC-2 DSDse

Upgradeable Excellence

Steven Stone

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n 2010 I reviewed the original version of the Wyred 4 Sound DAC-2. Besides excellent sound and ergonomics the DAC-2's features included upgradeablity. Over the years I've seen many products that were supposed to be upgradeable, but never fulfilled their promise of a longer, more useful lifespan. In 2014, four years after its initial release, the Wyred 4 Sound DAC-2 is still aging strong with three versionsthe base model, a DSD model, and the DSDse. And if you purchased one of the original DAC-2s back in 2010, it can be upgraded to the latest, most advanced model for \$1299. That's kind of nice. don't you think?

All the DAC-2s use the same chassis with the identical number of inputs, outputs, and hardware connections. The base-level DAC-2 costs \$1495 and supports PCM formats up to 192/24 via USB or SPDIF. The DAC-2 DSD is only \$100 more and adds asynchronous USB support for up to 384/32 PCM and 128X DSD. According to Wyred 4 Sound, the DAC-2 DSD also offers "improved audio quality through USB input" and "galvanic isolation," which are two features not listed in the basic DAC-2's feature set. Given what you get for the extra \$100 upcharge, I can't imagine why anyone would still buy the base model.

Wyred 4 Sound's top model is the DAC-2 DS-Dse. It costs \$1050 more than the base model, which brings it to \$2549. Added features include custom Vishay Z-foil resistors with a 0.1% tolerance, a "Femto" clock that reportedly has just 82 femtoseconds of jitter, ultra-low-noise discrete regulators that are over 100x quieter than the regulators on the other two models, Schottkey diodes, premium-grade inductor coils, a rhodium-plated Furutech fuse, and improved galvanic isolation with a further refined USB interface.

The review sample used throughout the review was the original 2010 DAC-2 upgraded twice by Wyred 4 Sound. First it was changed into a DAC-2 DSD then converted to current DAC-2 DSDse specifications. These upgrades are not something that a local dealer or enduser can do "in the field," since they require firmware upgrades in addition to replacing circuit boards. But Wyred 4 Sound has a simple order form with complete instructions on its Web site to make the upgrade process as quick and painless as possible. Upgrades start at \$495 to convert an original DAC-2 to a DAC-2 DSD. For an additional \$125 the Femto clock option can also be included in that upgrade package.

Tech Tour and Ergonomics

The DAC-2 DSDse uses a 9018 ESS Sabre DAC at its heart, which is the same chip found in the base and DSD models. The "secret" to the DAC-2 DSDse's superior measured performance and sound isn't necessarily the chip, although the 9018 ESS Sabre is considered to be one of the top current options. No, the formula for better sound in the DAC-2 DSDse is all the other parts and design that surround the Sabre DAC. My recent experiences with the Antelope Audio Zodiac Platinum DSD with

its external rubidium atomic clock reinforced my opinion that the clock is a critical element in a DAC's overall sound. The new Femto-grade clock option in the DSDse lowers phase jitter by over 80% when compared with Wyred 4 Sound's original DAC-2 clock.

The front panel of a DAC-2 is simplicity objectified. It has three centrally located pushbuttons arranged in a triangle. Above the buttons is a two-line OLED display. That's it. On the back of the DAC-2 you will find an on/off switch, two RCA coaxial SPDIF inputs, two TosLink inputs, one AES/EBU input, one I²S input (via HDMI), and one USB input. The DAC-2 also has a pair of balanced XLR outputs, a pair of unbalanced RCA outputs, and a pair of "Bypass" analog inputs.

The I²S digital input technology seems to be making something of a comeback these days. The DAC-2 DSDse's inclusion of this input allows it to connect to I²S sources that also use an HDMI connector. PS Audio supports I2S output on its Perfect Wave Transport, as does Empirical Audio in its Off-Ramp USB converter. The primary disadvantage of using HDMI connection hardware is that a naive user could assume it's a standard HDMI connection and wonder why the audio output from his HDMIenabled Blu-Ray or DVD player doesn't work when plugged into the DAC-2's "HDMI" input. Fortunately, if you do mistakenly plug an HDMI cable into the HDMI connector on the back of the DAC-2, nothing bad will happen except silence, which has a history of being golden.

Setup and Daily Use

The DAC-2 DSDse is capable of accepting up to a 384kHz/32-bit signal. It accomplishes

Wyred 4 Sound DAC-2 DSDse

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 to be installed prior to operation. Mac users don't need to worry about downloading drivers.

Most of the time the DAC-2 DSDse remained in my nearfield desktop-computer audio system (see equipment box for specific list of gear), but it also spent some time in my large room system. During the time I've had the DAC-2 DSDse in my systems it never malfunctioned, but I did have one issue with my MacPro desktop (ver 1.1 running OS 10.6.7). Each time the computer woke up from an extended (more than twohour) sleep due to inactivity, the Mac would "see" the DAC-2 DSDse on the MIDI list of audio devices, but the DAC would not play. Any app that used sound would also not run successfully until either another audio device was selected or the computer was rebooted, at which point the DAC-2 DSDse was recognized and played without issues.

The only ergonomic problem I experienced with the original version of the DAC-2 vanished on the DAC-2 DSDse—the acceptance angle for the remote control is now wider. Now I had to move the remote way off angle for the DAC-2 DSDse to not respond to its signal. With the original I was forced to lower the remote so it was nearly parallel with the faceplate before its commands were acted upon.

Front-panel design has always come down to a battle between visual simplicity

(fewer buttons and knobs) and the complexity of commands needed to make a system function with the fewest number of sub-menus. The DAC-2 DSDse has only three buttons, so you need to do a double-button-push boogie 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 DSDse, but switches it between these two modes. Unfortunately I found it far too easy to be in the wrong mode and instead of adjusting the volume, I changed the input. My advice is to use the remote control whenever possible.

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 low-pass filter's cutoff frequency. DSD signals can contain large amounts of ultrasonic noise which can sometimes create problems with preamplifiers and amplifiers. The DAC-2 DSDse gives you a choice of three cutoff frequencies, 50kHz, 60kHz, or 70kHz as well as a slope adjustment. The set-up menu also includes a three-level brightness control for the front-panel display and an option for the TosLink input to be either a fixed or variable output source.

If you have an analog source that you want to use with the DAC-2 DSDse you have only one option—hook up an analog preamp to the DAC-2 DSDse's HT bypass inputs or use a source with its own volume control. Unlike Wyred 4 Sound's mPRE, which can

accept analog sources and control their volume via its analog volume control, the DAC-2 DSDse has no A/D to convert analog to digital and, thus, no way to adjust the volume of an analog source. Only by completely bypassing all internal circuitry can you listen to an analog source through it. The HT bypass is a true bypass with the signal going directly from the input to the output of the DAC-2 DSDse.

One product category that I think is severely under-populated is the stand-alone high-performance consumer analog-to-digital converter. The DAC-2 DSDse certainly isn't the only DAC/Preamp I've experienced that lacks analog inputs. A companion A-to-D component would be a welcome addition to the Wyred 4 Sound product line; maybe Wyred could even include an I²S HDMI connection on it.

The Sound of the DAC-2 DSDse

When I reviewed three portable DACs in Issue 245 I compared the Herus DAC, which received a Golden Ear Award, with the first version of the DAC-2 DSD (without the Femto clock upgrade or SE board, although I erroneously wrote DAC-2 DSDse). I found on DSD 64x sources that the Herus' "imaging was as precise as the Wyred 4 Sound DAC-2 DSD, as was depth recreation and low-level detail...I was hard-pressed to tell which DAC I was listening to; they sounded that similar to each other." I conducted similar comparisons of the Herus and DAC-2 DSDse using the NuForce MCP-18 as a switch and level control. With the latest DAC-2 DSDse

SPECS & PRICING

Typical THD+N: 20-

20kHz (A-weighted): <0.006%

Frequency Response: 20Hz-20kHz: +/-0.075dB

S/N Ratio: >115dB

Crosstalk: >98dB

Noise: (A-weighted): < 7uV

Noise: < 10uV

Balanced output level: **5.2V**

(14.2dB gain)

Unbalanced output level: 2.6V

(8.2dB gain)

Output Impedance: 100 ohms
Price: DAC-2 \$1495, DAC-2DSD

\$1599, DAC-2DSDse \$2549

WYRED 4 SOUND

4235 Traffic Way Atascadero, CA 93422S (805) 466-9973

wyred4sound.com

ASSOCIATED EQUIPMENT

Source Devices: MacPro model 1.1 Intel Xeon 2.66 GHz computer with 16 GB of memory with OS 10.6.7, running iTunes 10.6.3 and Amarra Symphony 3.1 music

software, Pure Music 1.89 music software, and Audirana

Plus 1.5.12 music software.

Oppo BDP-103, SimAudio

Moon 180 streamer

DACS: Antelope Zodiac

Platinum DSD, Resonessence Herus, PS Audio Perfect Wave

DSD DAC

Amplifiers: April Music Eximus

S-1, Wyred 4 Sound mAMP,

Accuphase P-300

Speakers: ATC SCM7 III, Role

Audio Kayak, Aerial Acoustics

5B, Audience Clair Audient 1+1,

Velodyne DD+ 10 subwoofer

Cables and Accessories:

Wireworld USB cable,

Synergistic Research

USB cable. AudioQuest

Carbon USB cables, PS

Audio Quintet, AudioQuest

......

Colorado interconnect,

Cardas Clear interconnect,

Black Cat speaker cable and

Interconnect, and Crystal

Cable Piccolo interconnect,

Audience Au24SE speaker

cable

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Wyred 4 Sound DAC-2 DSDse

I found that with DSD material the differences between the two DACs in matched level A/B comparisons were still so close that I could not reliably tell which was which. But with 44.1/16, streaming and MP3 sources I found the DAC-2 DSDse was more cohesive, nuanced, and listenable overall, and I could reliably tell which DAC was which during matched-level A/B comparisons. What I found most interesting was that the kind of sonic improvements I heard were very similar to the improvements I heard while I was reviewing the Antelope Audio Zodiac Platinum DSD DAC when I switched from its internal clock to Antelope's external "atomic" clock. While I certainly don't have a wide enough sample set yet to make any sweeping generalizations, my own experience so far using different or better clocks with a DAC is that I've heard greater sonic improvements with the lower bit-rate and Red Book material than I did with DSD sources.

Was it a shock going from the \$13k Antelope Audio Zodiac Platinum DSD to the Wyred 4 Sound DAC-2 DSDse? No. it was not. And while I think the Antelope was a better performer on lower resolution, streaming, and Red Book material, the DAC-2 DSDse wasn't far behind it, and on 64x and 128x DSD material I couldn't hear any difference between the two. They both made it easy to listen into DSD mixes without having to strain. On PCM sources I preferred the more relaxed and organic way the Zodiac Platinum DSD presented music. In comparison the DAC-2 DSDse delivered just as much musical information, but on lower-res material that information was a bit more strident and less suave in overall presentation. If I had to listen to MP3s all day I'd prefer to do it through the Zodiac.

During the review period I had an opportunity to listen to the new PS Audio Perfect Wave DSD DAC in my system and compare it to the DAC-2 DSDse. Once more the sonic differences between the two DACs were more pronounced on 44.1/16 and lower-resolution streaming sources than with DSD and higherres PCM. Like the Zodiac DAC the PS Audio DAC manages to upgrade all source material in a way that I haven't experienced on older, earlier-generation DACs, except for the Meridian 800 and 860 with their apodizing filters.

I recently had an opportunity to record the superb Brazilian ensemble Choro Dos 3, (www. facebook.com/chorodas3) made up of three sisters and their father. The band's orchestration is a seven-string guitar (which handles the bass lines and contrapuntal rhythms), flute, mandolin, and percussion (which is a single-miked tambourine). I recorded them during a live concert in a small church. The results went far beyond my sonic expectations and have become a very useful review tool. Their Choro music is rhythmically dense with multiple overlapping patterns. I record in 128x DSD and then made lower-resolution versions as needed-usually at 192/24, 96/24, and 44.1/16 sample rates. Through the DAC-2 DSDse I could easily tell the difference between the 44.1 version and the original. Even the 96/24 files lacked a bit of decipherability when compared to the originals on the DAC-2 DSDse. But when I moved up to the 192/24 files and compared them with the 128xDSD files I was hard-pressed to hear any differences in the overall presentation, including subtle micro-dynamics and low-level inner details. Also both the 192/24 and DSD versions had equal levels of decipherability through the DAC-2 DSDse.

In the past I've often had to resort to an outboard USB converter, such as the Empirical Audio Off-Ramp 5, to pull the best performance out of a DAC/pre. I connected the Off-Ramp 5 to the DAC-2 DSDse via its I²S connections and compared it to the DAC-2 DSDse's built-in USB implementation. I found that in matched-level A/B comparisons I could not reliably hear differences on 44.1/16, 96/24, or 192/24 material. Finally I've heard a mid-priced DAC that doesn't require an additional device to optimize USB sources.

Conclusion

After living with the latest version of the DAC-2 DSDse for more than a month, and having had some version of the DAC-2 in my possession for the past four years, I would be remiss if I didn't urge anyone who owns an original version of the Wyred 4 Sound DAC-2 to upgrade it to DAC-2 DSDse, or at the very least to the DAC-2 DSD with Femto clock. It's the equivalent of jumping four years ahead in time, from what was a very good DAC with an okay USB implementation to an excellent DAC with a well above average USB implementation.

Early adopters are usually the ones who suffer the most financially when purchasing components in a rapidly changing product category such as DAC/pre's. Some consumers are lucky to get a year's use from a DAC before the need for a newer, more capable one begins to manifest itself. And while there have been upgradable components such as NAD's line of preamps with MDC modules, few manufacturers have been able to make a DAC that can be updated to be competitive four years after its initial release. Bravo Wyred 4 Sound for keeping your promises and making it possible for early adopters to remain at the leading edge of DAC technology. tas



Wyred4Sound DAC-2

\$1495

The DAC-2 delivers high-quality digital to analog conversion; it is also a very well-thought-out preamplifier that can replace a separate preamp and DAC in an all-digital system. The DAC-2's obvious value and reasonable price makes it an enticing control center for either a high-end near-field desktop, or two-channel room-based system, or even in a multichannel system via its pass-through option. All the circuit boards can be upgraded to allow for some degree of future-proofing. And Wyred4Sound recently offered an upgrade for the DAC-2 to a DSDSE version that offers DSD capability along with improved PCM performance. SS, 239

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AURALiC Vega Digital Audio Processor

High Performance, Reasonable Price

Chris Martens

URALiC's Vega Digital Audio Processor (\$3499) is a powerful and versatile digital-to-analog converter that can also serve as a digital-input-only, balanced-output-capable preamplifier. Specifically, the Vega supports all PCM files from 44.1kHz/16-bit resolution to 384kHz/32-bit resolution, while covering all sampling rate/word-depth combinations in between. Moreover, the Vega is DXD- and DSD-compatible and can decode both DSD64 and DSD128 bitstreams via the DoP V1.1 data-transmission protocol. In short, the Vega is an ambitious, premium-quality DAC/ preamp that aspires to top-tier performance. Does it reach this goal? I think it does as I will explain in this review, but first let's first take a look at AURALiC's company background and at the Vega's underlying technologies.

As I mentioned in my review of the firm's Taurus MkII balanced headphone amplifier, AURALIC is a Hong Kong-based high-end audio electronics company co-founded in 2008 by President and CEO Xuanqian Wang and his business partner Yuan Wang. Xuangian Wang has had formal training as an electrical and audio recording engineer and is an accomplished classical pianist, while Yuan Wang has a background in sociology and management science. Both men share a passion for music and sound quality, having met (where else?) at a musical event-the 2008 Festival of Waldbühne Berlin. Not long thereafter, the men decided to launch AURALIC Ltd.

More than many DACs in its price class, the Vega is chock-full of advanced technical features, yet it is also informed by Xuangian Wang's thoroughgoing familiarity with classic analog-audio circuit designs. In practice, this means the Vega is a modern-as-tomorrow

DAC with stellar performance specifications, yet goes the extra mile not only to measure well but also to deliver sound that, first and foremost, holds true to the sound of live music. As I survey the Vega's rich set of technical features it is important to bear in mind that this is more a "music first" design than it is a "technology über alles" product.

As noted above, the Vega is a DXD- and DSD-compatible 384kHz/32-bit-capable DAC/ digital preamp. The Vega provides five digital audio inputs: one AES/EBU, one TosLink, one USB, and two coaxial SPDIF. The Vega provides single-ended and balanced analog outputs, with volume levels controlled by a 100step digital controller said not to compress dynamic range.

Digital audio processing is handled by AURALiC's proprietary Sanctuary Audio Processor, which the company says is based on a "multi-core ARM9 architecture" and provides a prodigious 1000MIPS (millions of instructions per second) of data-crunching power. Unlike many competing DACs, the Vega upsamples all incoming PCM audio data to 1.5MHz/32-bit resolution prior to decoding. Further, the Vega provides six user-selectable digital-audio filter modes (four for PCM formats, two for DSD formats). The PCM filter modes each comprise four individual filters optimized for a specific group of sampling rates. One can choose Filter Mode 1, which offers the best performance measurements; Mode 2, which reduces group delay while imposing minimal amounts of treble attenuation; Mode 3, which minimizes preecho and ringing effects but with a somewhat higher degree of treble attenuation; or Mode

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EQUIPMENT REVIEW - AURALIC Vega

4, which applies minimum-phase type filters and is said to allow "no pre-echo effect at all" with "very small group delay to eliminate ringing."

Filter Modes 5 and 6 are designed specifically for use with DSD files, and address the problem of the very-high-frequency noise that DSD bitstreams can entail, providing strategically chosen levels of ultrasonic treble roll-off. The concept is to preserve the music intact while getting rid of ultrasonic noise that could potentially damage wide-bandwidth amplifiers or speakers.

Significantly, the Vega permits users to switch between its various filter modes on the fly to compare their subtly different voicing characteristics and overall impact on the music. Xuangian Wang wisely observes that one's choice of filter mode might depend to a large extent on the recording quality of the material being played. Great recordings, he says, often sound best through Filter Mode 1, while customer comments suggest that Filter Mode 4 is the best "general purpose" setting for a mix of audiophile-grade and more commonplace recordings. The important point is that the Vega allows users to fine-tune the DAC's sonic persona to fit the musical material at hand.

Another signature feature of the Vega is its Femto Master Clock, which yields a spectacularly low 0.082 picoseconds (or 82 femtoseconds) of jitter—a figure few DACs at any price can match. The Vega provides three master-clock control settings: the default "AUTO" setting, which maintains "a balance between lock-in ability and jitter performance,"

plus "FINE" and "EXACT" settings (available only after the Vega has warmed up for an hour), which "force the (clock controller's) PLL bandwidth into a very narrow range to maximize jitter performance." Not all digital sources are precise enough to use the FINE or EXACT settings, but Xuanqian Wang notes that with the EXACT settings in play he sometimes hears "a significant improvement, compared to the AUTO setting, for certain sound tracks, such as well-recorded classical piano solo."

As expected, the Vega is compatible with both Macs and PCs and with most popular music-playback software. The Vega autoinstalls in Mac environments, but requires installation of an included Windows driver when used in PC-based systems. AURALIC does feel that music-software packages have a big impact on the DAC's sound and for this reason supplies a free copy of its recommended JPLAY software with the Vega. Accordingly, I used JPLAY software in conjunction with jRiver Media Center 19 music-management software in a PC-based system for my review listening.

The Vega's analog outputs are driven by a pair of AURALiC's signature ORFEO Class A output modules, whose design was inspired by the circuitry of the classic Neve 8078 analog recording console and whose sound is said to "share the same warm and natural sound of (the) Neve 8078." Perhaps as a result, the Vega claims vanishingly low THD and noise (just 0.00015%). Part of the performance equation, naturally, involves not only having high-performance analog output modules, but also addressing noise issues wherever

possible. To this end, AURALiC constructs the Vega's chassis of a highly EMI-resistant metal-alloy called AFN402 and coats the chassis' interior surfaces with a multi-layer electro-mechanical damping material called Alire, which is used in most other AURALiC components.

The Vega sports an easy-to-read OLED front-panel display that shows the input selected, the format and data rates of

whatever digital audio input has been selected, and the volume level (on a scale of 0-100) to which the processor is set. By design, the Vega can be operated from its faceplate or from an included remote control. The control menu offers options for adjusting absolute polarity and left/ right channel balance, or selecting preferred filter models. Users can also control the OLED display itself, turning illumination up, down, or off (for zero visual distractions at all). Overall, the Vega is an ergonomic delight, though it is sufficiently complex that it pays to read the manual to understand the scope of the control options at hand.

If the foregoing technical description seems promising, then please know that the sound of the Vega is fully as good as, if not better than, the description might lead you to expect. Frankly, I've been around the world of computer audio for years, but I never felt a keen desire to make a dedicated high-performance DAC a permanent part of my reference

system until I heard the Vega in action. Up to this point, most of the computer-audio/DAC-based systems I have auditioned seemed to me to fall short of the sound quality I was used to hearing from top-tier disc players. I also found that those DAC-based systems that were sonically satisfying tended more often than not to be astronomically priced.

In contrast, what makes the Vega so captivating to my way of thinking is that it is

SPECS & PRICING

Type: Digital-to-analogconverter/digital preamplifier Digital inputs: One AES/EBU, two coaxial, one TosLink, and one USB 2.0 buffered by ActiveUSB

Analog outputs: One stereo single-ended (via RCA jacks), one balanced (via XLR connectors)

Frequency response: 20Hz-20kHz, +/-0.1dB THD+N: <0.00015%, 20Hz-20kHz at 0dBFS Dynamic range: 130dB, 20Hz-20kHz, A-weighted Supported digital formats: All

PCM from 44.1kHz to 384kHz with word lengths up to 32-bit, DSD64 (2.8224MHz), and

DSD128 (5.6448MHz)
Important format information:

352.8kHz and 384kHz are supported through USB only; 32-bit word lengths supported through USB only; DoP V1.1 transmission protocol supported through USB only Output voltage: 4V RMS at maximum, with dynamic-loss-free digital volume

Dimensions): 11" x 2.6" x 9"

Dimensions): 11" x 2.6" x 9 Weight: 7.5 lbs.

Price: \$3495

AURALIC AMERICAS INC.

12208 NE 104th St. Vancouver, WA 98682 (360) 326-8879 auralic.com

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4.0

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EQUIPMENT REVIEW - AURALIC Vega

reasonably priced yet consistently supplies a rich panoply of audiophile virtues, while also demonstrating an uncanny ability to keep its focus on the musical whole. In short, the Vega represents the intersection of good value, great (and forward-looking) technology, plus terrific musicality—a compelling combination indeed.

If you asked me to cite several specific qualities that characterize the sound of the Vega, two that come instantly to mind would be transparency and resolution-effortless, elegant, and unforced openness and detail that sound more like the real thing than like hi-fi artifacts. On the track "Embraceable You" from The Larry Coryell Organ Trio's Impressions [Chesky] the Vega lets you listen deeply into the voices of each of the instruments at play and so to savor the round, ripe tone of Coryell's guitar, the reedy and breathy voice of the organ, and the delicacy of the drum kit's contributions, the cymbal work in particular. Moreover, the Vega shows you the worth of high-res files, helping you to appreciate how much fuller and more complete they make the music sound. The beauty of the Vega's presentation is that the additional layers of detail it provides are delivered in a relaxed and lifelike manner; additional music information is simply there-whole and complete without unwarranted spotlighting or pyrotechnics, so that the music is free to breathe and flow.

Other qualities that typify the sound of the Vega are its dramatic and at times explosive dynamics, which likewise unfold in a naturally expansive way. As with musical details, the dynamic qualities you hear seem to flow more from the music than from the equipment. Consequently, the music seems energized and illuminated from within, much as it does when heard live. To hear what I mean, try listening to Silvestre Revueltas' Sensamayà as captured on the Chicago Symphony Orchestra Brass Live recording [CSO Resound]. This exotic-sounding piece is full of lithe twists and turns as it progresses from one dynamic highlight to the next, with tension building as the composition unfolds. I've heard this piece through many digital source components, but none made Sensemayà sound as powerful or expressive as the Vega did; nor could any convey the tsunamilike force of the composition's final crescendo as effectively as the Vega.

Finally, I was struck on multiple occasions by the Vega's unfailing musicality, which I sometimes-tongue-in-cheek-called the "Neve factor." Neve recording consoles are known for pulling off a difficult but highly rewarding tightrope act of sorts; on the one hand, they deliver exceedingly high levels of transparency, clarity, and timbral purity, while on the other hand they preserve a naturally warm, organic, and lifelike sound. I think it is significant that Xuanqian Wang has chosen the classic Neve sound as his sonic model for the Vega and that the Vega strives (successfully) to strike a similar sonic balance. As a result, the Vega's sound is every bit as revealing, crisply defined, and informative as any "analytical" DAC would be, but without the drawbacks (sterility, a vaguely "mechanical" quality) that analytical products usually entail. Rather than dissecting or deconstructing the music, then, the Vega simply reveals musical



textures, timbres, tonal colors, and dynamics, and then gets out of the way.

I compared the Vega to my primary digital reference, Rega's superb Isis CD player/DAC, and found the Vega's sound competitive, though somewhat different. I had a slight preference for the Rega's sound on 44.1/16 material owing to the Rega's somewhat more coherent upper midrange and treble presentation, though in truth the contest was very, very close. But a key point is that the Vega is less than half the price of the Rega and is capable of exploring high-res PCM and DSD files, which the Rega is not. In particular, listening to DSD files through the Vega proved revelatory, because DSD files

as rendered by the Vega seemed to do a much better job than standard-resolution PCM files in enabling the presentation to sound more three-dimensional and realistic.

Although I'm not ready to part with my top-shelf Rega Isis CD player just yet, the Vega is the first sensibly-price DAC I've heard that I could readily embrace as one of my primary digital source components. For less than \$3500, the Vega takes discerning audiophiles and music lovers very far up the high-end audio performance ladder, providing them with a versatile and technically advanced digital playback solution they will not soon outgrow.

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MSB Technology Analog DAC

Just Like an LP?

Vade Forrester

ralog DAC? What the heck is that? Doesn't the "D" in DAC stand for digital? Yes, but what do all DACs try their best to sound like? Right: analog. Manufacturer MSB Technology went all-out to assure its new DAC sounded as much as possible like the fabled analog source. When I auditioned MSB's top-of-the-line DAC IV, I thought it was possibly the best source I had heard, analog or digital. But its lofty price might have spooked some audiophiles. I appreciate it when technological developments advance the state of the art in playback quality, even if they carry high price tags. That's because the technology used in high-priced gear often trickles down to equipment within my financial means. That's what's happened with the Analog DAC, which MSB bills as "The Most Technologically Advanced DAC in the World!" And even though the price couldn't be described as low, it's a lot lower than MSB's DAC IV.

So what makes the Analog DAC special? For one thing, its physical construction is unique. Instead of building a typical chassis-you know, a metal box to house the electronic components-MSB has made the Analog DAC modular. It's basically a machined aircraftgrade aluminum slab under which are attached removable input modules that give the Analog DAC its functionality. You start off with a basic unit with one input module (your choice) and a basic power supply for \$6995, then add additional modules to suit your needs. Want another digital input? \$995. Want a steppedattenuator volume control, with 78 1dB steps? \$995. The volume control is one option that's not modular, since it requires a hole in the top of the chassis for the volume control knob. It's \$995. Want a better power supply than the stock unit (which is a very respectable linear model)? \$2995. Want a different color than the stock matte-white or matte-black? That's \$700 for satin black, \$350 for custom colors (I think a red one would be neat; black and silver are boring). And there are quite a few other options like remote control, iPad control, Wi-Fi, and so forth. The review unit (\$6995) had a volume control (\$995), a USB input (included in the base price), a SPDIF input (\$995), and an Inter-IC Sound (I2S) input (\$995). An aluminum remote control was also included. which is an \$85 option; so if I add up all the options correctly, the total price of the review unit was \$10,065. See the MSB Web site at for a complete list of options and prices.

The different input modules connect to an internal I²S buss. If a need for a new input develops (maybe USB 3.0?), a module for that will be developed and can be installed with a

screwdriver. After it's installed, it becomes totally plug and play. The I2S buss connects to the DAC module, which is not an off-theshelf chip; rather, it's a 384kHz discreteladder DAC constructed from extremely precise resistors. The firmware that controls the internal operation of the DAC is easily upgraded (see the Setting Up section below). That makes it easy and fast to make changes. The firmware-update files are in WAV format, making firmware changes just like playing a song. That's very clever. The Analog DAC's output impedance through the RCA outputs is 53 ohms without a volume control; 38 ohms with the control. That's low enough to drive any amplifier or cable.

MSB Technology uses its precision "Femto Clock" technology to minimize jitter, and includes a large internal memory where you can set filters, upsamplers, and other DSP instructions. This is *not* an off-the-shelf DAC design using conventional parts and circuits.

The Analog DAC's optional power supply has exactly the same form as the DAC, and is usually pictured with the DAC stacked on top of it. The stock power supply is a linear power unit with two transformers.

The Analog DAC can play Direct Stream Digital (DSD) files in their native format without converting the files to PCM. These are the files used to make SACDs. DACs capable of playing DSD files in their native format may be the hot item in the hi-fi industry right now. We speak of DSD sampling rates as "DSD64" and "DSD128," where the numbers "64" and "128" denote multiples of CD's 44.1kHz sampling rate. DSD recordings created as masters for SACDs are DSD64, but it's possible to record

EQUIPMENT REVIEW - MSB Technology Analog DAC

DSDs at twice that rate. The Analog DAC plays both DSD64 and DSD128 recordings. And as I was pleasantly surprised to learn, it plays DSD files through both USB and SPDIF inputs. The capability to play DSD was added by a firmware update-no hardware changes were necessary. The Analog DAC also plays PCM files up to 384kHz/32-bits. That allows it to play the 352.8/24 Digital eXtreme Definition (DXD) files used for highresolution mastering by several recording companies. A few companies now sell DXD files, should you want to hear super-highresolution recordings. Be sure your music player can handle them before you pull the switch to download DXD files.

A Windows driver came on a CD, and can also be downloaded from the MSB Web site. Also on the CD were some set-up instructions. A manual can be downloaded from the MSB Technology web site.

Setting Up

I used a WireWorld Platinum Starlight USB cable to connect my laptop server to the Analog DAC. MSB Technology advised that since the internal circuit of the Analog DAC was unbalanced, the unbalanced connectors should sound better, so that's what I opted for. If you have the volume control installed, the Analog DAC is designed to be used as your system controller in lieu of a preamp, so it should be connected directly to the power amplifier's inputs and the very fine volume control operated by the MSB remote control. Since my Berning ZH-230 amplifier has unbalanced inputs only, the unbalanced

connection from the Analog DAC was ideal. If you have other analog sources, such as a phono preamp, you can plug it into the Analog DAC's analog inputs and select it from the MSB remote. I must say it would seem a little weird to plug a phono preamp into a DAC, but, hey, welcome to the digital age. I connected my tuner to the analog input just to see if it worked. It did, and I could select the tuner using the remote control.

I discovered the Analog DAC was sensitive to the cables used to connect it to the power amplifier. A Clarity Cable Organic interconnect sounded a bit bright and leannot the way it usually sounds in my system. Purist Audio Design Venustas interconnects sounded a bit fatter with more bass, but the best sound I found came when I used High Fidelity Cables' CT-1 interconnects. Operating via magnetic conduction instead of normal voltage conduction, the High Fidelity Cables interconnects are probably the best I've heard to date. I don't really understand how they work, but their sonic advantages are audible. With them, the Analog DAC sounded more balanced, with deeper bass.

When I used an Audience powerChord e power cord, the Wattgate IEC connector that plugged into the power supply almost completely blocked the on/off switch, so to turn the Analog DAC on and off, I had to plug/unplug the power cable. Wattgate IEC connectors are pretty average in size for aftermarket connectors, so I would expect the same problem with other aftermarket power cords. The molded IEC power

connector that comes with the power cord included with most components wouldn't have this problem.

I placed the Analog DAC on a middle shelf on my Billy Bags equipment rack and adjusted the cone feet until the DAC was perfectly stable. There wasn't room on a single shelf for both the power supply and DAC, so I placed the power supply, which is about the size of a hardback novel, two shelves above the DAC. That's about as far as the connecting power cord would reach. The Analog DAC has a huge display, the largest I've ever seen in a DAC, and it shows both the volume control setting and the input source; however, since it's only visible from the top, it can't be seen if you're sitting across the room, unless the Analog DAC is on a low shelf in your rack.

To use my Windows-based laptop server, I had to install a driver so Windows 7 would play Class 2 USB Audio, which enabled playback of high-resolution PCM and DSD input. After checking to be sure it was the current version, I installed the driver that was on the included CD. Installation was straightforward, i.e., not tricky.

The Analog DAC manual recommends Foobar 2000 as a music-server program, and included a file on the CD describing how to set up Foobar 2000 to work with the Analog DAC. I view this type of help as essential; DACs are seldom plug and play, and each computer-based music server has to be set to use a particular DAC. Most have different settings which need some tweaking to sound best. My preferred

SPECS & PRICING

Inputs: One digital input included in base price, additional inputs optional; one analog input on RCA jacks
Sample rates supported: PCM-44.1k, 48k, 88.2k, 96k, 176.4k, 192k, 352.8k, 384k up to 32 bits; DSD at 2.82M and 5.64M (DSD64 and DSD128)
Line output level: 2.62V RMS, balanced or unbalanced output Output impedance: 53 ohms without volume control, 38

without volume control, 38 ohms with volume control (unbalanced); 106 ohms without volume control, 76 ohms with volume control (balanced)

Dimensions: DAC, 17.63" x 1.5"

x 12.5" plus connectors; power supply, 6.7" x 2.25" x 8.9" Weight: DAC, 12 lbs.; power supply, 7 lbs.

Price: \$6995, \$10,065 as reviewed

Manufacturer Information
MSB TECHNOLOGY
CORPORATION
625 Main Street
Watsonville, CA 95076

(831) 662-2800

msbtech.com

REFERENCE EQUIPMENT

Speakers: Affirm Audio

Lumination speakers

Amplifiers: Berning ZH-230

stereo amplifier

Preamplifier: Audio Research

LS27 linestage

Digital sources: Hewlett Packard

dv7-3188cl laptop computer

running 64-bit Windows 7

Home Premium and J. River

Media Center version 19;

Interconnects: High Fidelity Cables CT-1

Audio Research DAC8

Auraliti PK100 music player;

Speaker cables: Clarity Cables Organic

Power cords: Purist Audio
Design Venustas, Blue Marble
Audio Blue Lightning, Clarity
Cables Vortex, Audience
powerChord e
Digital: Wireworld Platinum
Starlight USB cable, Gold

Starlight 6 SPDIF cable, and Gold Starlight 5 AES/EBU cable Power conditioner and

distribution: IsoTek EVO3 Sirius

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EQUIPMENT REVIEW - MSB Technology Analog DAC

software, J. River Media Center 19, was simple to set up-after I figured out what all the settings meant.

MSB Technology's position on burn-in is ambiguous. It begins by asking if burn-in is real or just a period of familiarization; then it says feedback (whose?) recommends 100 hours' burn-in, then it says customers recommend one month burn-in. Come on—it either needs burn-in or it doesn't; and it should be possible to specify a length of time. I burned in the review unit for about 300 hours.

Thanks to a recent software update, my Auraliti file player now plays DSD as well as PCM, and, somewhat unusually, plays DSD64 through both SPDIF and USB outputs. DSD128 is played through the USB output only; apparently, DSD128 exceeds SPDIF's limits. When I plugged the Auraliti into the Analog DAC's inputs, it worked without a hitch, providing plug-and-play DSD sound. I could get used to that!

During the review period, I received another software update, this time from MSB Technology. The firmware update took the form of a WAV file, and all that was necessary to apply the update was to play the WAV file. If the update doesn't "take," an audible message tells you so. How cool is that? I first tried playing the WAV file with iTunes, and heard the failure message. Then I tried it with J. River Media Center, and this time, there was no error message. When I restarted the Analog DAC, the correct firmware version was displayed. Most manufacturers don't ever update their firmware, and I don't know of any that makes it this easy.

Sound

While the Analog DAC's most advanced feature is its ability to play DSD files, it's just as important to assess how well it plays PCM files, even those ripped from CD. After all, what makes up most of your collection? So I enjoyed playing a wide variety of music through the Analog DAC.

In a word, the Analog DAC was detailed. I don't mean it was analytical, just that it revealed a ton of information about the music played through it. Since I had it on hand when writing the article about DSD downloads published in the previous issue, I used it to sample DSD files from various download sites. Unsurprisingly, the Analog DAC showed that not all DSD files are created equal; some were glorious, while some, well...not so much.

Dynamics were finely delineated when the music called for it. They weren't amped up, as may be the case with some components, but were fast enough to enhance the sense of musical flow. On Jordi Savall's La Folia 1490-1701 (ripped from Alia Vox AFA 98050) the track "Folia Rodrigo Martinez" is a dynamic minefield, requiring the audio system to play at continuously varying levels of loudness. Some components artificially divide the sound into discrete steps, but the Analog DAC showed the loudness changed continuously. The Analog DAC reproduced the bass drum. which descends into the mid-30Hz range, with extension and detail. Power and slam were good overall. The amount of detail revealed in this information-dense recording was amazing; it was not even slightly etched or bright, but very natural-sounding. I've never

heard a component extract this much detail from the recording—a recurring theme in my listening notes.

On Alex de Grassi's album *Blue Coast Special Event 19* (DSD64/DFF, Blue Coast Records), the cut "Shenandoah" exhibited exceptionally detailed guitar sound, with an unusual drone effect I had never heard before. But the picture of de Grassi's guitar revealed it to be a very unusual design, so maybe that explained the sound. The Analog DAC really showed how realistic a recording of solo guitar engineer Cookie Marenco had captured.

Piano recordings were spectacular. On Thomas Günther's performance of Schubert's Piano Sonata in A minor (DSD64/DSF, Cybele Records), the Analog DAC reproduced the piano's complete harmonic structure with a combination of delicacy and explosive power rarely heard on recordings. The sense of the hammers striking the piano's strings was captured superbly. It's odd that I noticed this particular detail on several piano recordings. Other finer details of Günther's performance were strikingly realistic, too. I'd never mistake the recording for a real piano, but it's getting closer to the real thing. Isn't that what the hobby is about?

Rebecca Pigeon's "Spanish Harlem" from her album *The Raven* has been an audio fave since it first appeared on LP. Remastered as a 176.4/24 FLAC album by Bob Katz for Chesky Records, it was eerily realistic through the Analog DAC. Pigeon's voice, in particular, had a "reach-out-and-touch-it" quality, creating the illusion of someone standing in front of me singing. Instrumental accompaniment was

equally detailed, especially the stand-up bass.

The Tallis Scholars' Miserere was their signature album, and on Gimmell Record's 96/24 FLAC download of Allegri's Miserere & Palestrina's Missa Papae Marcelli, the Analog DAC showed off its ability to throw a large soundstage. The "Miserere" piece is an a cappella work which has a small choral group at the front of the soundstage and a smaller solo group some distance behind it in the large church where the recording was made. The Analog DAC showed the separation between the main group and the solo group clearly, while making the words sung by the distant solo group unusually distinct.

DSD vs PCM

The 2L company offers several sample files in both DXD and DSD format. The company records its masters in DXD, performs whatever editing is necessary in that format, and then converts them to DSD. I expected that the original DXD files would sound better than the converted DSD files, but to my surprise, I slightly preferred the DSD-sourced files. On Beethoven's Sonata No. 32, the piano sound was more full-bodied and detailed. Once again, DSD portrayed the sound of the piano's hammers hitting the strings much more realistically than the DXD version. On the Allegro movement from Mozart's Violin Concerto in D Major, DSD string sound was richer and less mechanical, i.e., less digital. On a vocal selection, Vivaldi's "Recitative and Aria" from Cantata RV 679, "Che giova il sospirar, povero core," my impression was similar to the one I had with the Mozart

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EQUIPMENT REVIEW - MSB Technology Analog DAC



concerto—the soprano sounded more like a person singing than a recording of a person singing. The differences weren't night and day, but they established DSD as more analog-like. That doesn't mean DXD files sounded bad; they sounded good before DSD files came along, and still sound good. What's important for this review is that the Analog DAC made it easy to distinguish between the DXD and DSD versions of a recording, illustrating the MSB's transparency.

Comparison

My Audio Research DAC8 doesn't play DSD files, but it's what I have on hand, so I'll limit my comparison to PCM files. After all, the vast majority of my collection of downloaded and ripped music files are PCM, so that's not much of a limitation. The \$4995 DAC8 is a single-chassis unit which looks very conventional compared to the Analog DAC.

On "Folia Rodrigo Martinez," the Audio Research displayed its hallmark bass, the most powerful I've heard from any DAC. I've started to wonder if it's not actually too powerful, as peculiar as that concept may seem to some

audiophiles. Tonally, the Audio Research is similar to the Analog DAC, although the latter captures more details of the performance.

"Spanish Harlem" was a close match, but the Analog DAC's superior detail retrieval made Rebecca Pigeon sound just a smidgen more lifelike.

The "Miserere" track sounded a bit different on the two DACs. The Analog DAC's extra detail made the distant solo group more understandable, which had the effect of making them sound closer to the main group in front. The Analog DAC was squeaky clean, whereas the Audio Research sounded a little more smeared.

Bottom line

So does the Analog DAC sound like, well, an analog DAC? Maybe it does, if you can find an analog source as free from noise and distortion as the Analog DAC and if your idea of a quality analog source excludes any coloration from tubes or solid-state gain stages. If you want a flexible, top-notch DAC capable of playing any digital source currently available, in a unique, strikingly-designed package that can function as a line source as well as a DAC, the Analog DAC may be just your ticket. It's expensive, but its sound quality and functionality are hard to beat at the price. And its ability to replace an expensive linestage, connecting cables, and power cord can save you a bundle, reducing your overall system price and increasing the amount of free shelf space on your equipment rack. Viewed from that angle, maybe the Analog DAC isn't as expensive as it first looked, tas

ARIES Wireless Streaming Bridge

The ARIES serves as a "bridge" between music files on media server or high quality online streaming services and your DAC — enabling DACs for the first time to stream high-resolution music quickly and wirelessly in virtually any sampling rate, including DSD, Double-Rate DSD and DXD.



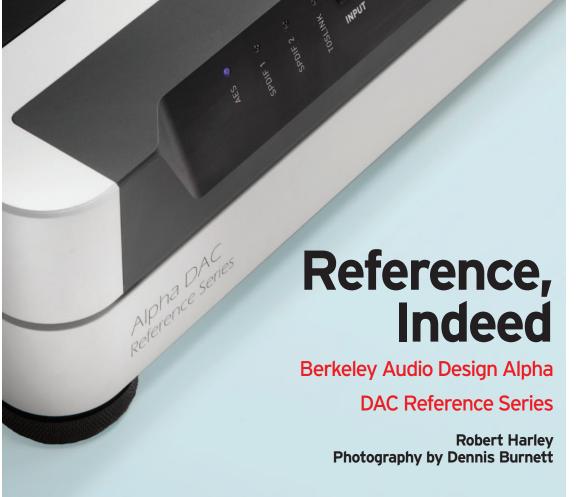
It is **not** a streamer It turns your DAC into a streamer!



AURALIC LIMITED | 声韵音响

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o understand Berkeley Audio Design's ambitious new Alpha DAC Reference, you need to know something about the company behind it. In its six years of existence Berkeley had produced just two products: the \$4995 Alpha DAC and the \$1895 Alpha USB, a USB-to-SPDIF converter. The hugely successful Alpha DAC established a new level of performance for digital products at anywhere near its price. I lived for several years with an Alpha DAC in front of some stellar electronics and loudspeakers, yet never felt it was the weak link in the chain despite its modest price relative to the rest of the system. Similarly, the Alpha USB was light years ahead of any other USB-to-SPDIF converter I'd heard. Given Berkeley's track record, I'd always wondered what this company could do if it aimed higher than the \$5000 price point.

The answer has arrived in the new \$16,000 Alpha DAC Reference Series, a vastly more ambitious effort than the venerable Alpha. Some potential purchasers will look at the Reference's lack of a USB input or its omission of DSD decoding, and consider the unit a nonstarter. That would be shortsighted. Both a USB input and integral DSD compatibility were omitted by design, which speaks volumes about the ethos of Berkeley founders Michael Ritter and Michael "Pflash" Pflaumer. Their approach could be summed up as "no sonic compromises." If including a USB input in the same chassis as the DAC circuitry shaved off even a sliver of sound quality, it was ruled out. If performing DSD-to-PCM conversion brought performance down a notch, the decision was the same. Berkeley Audio, more than any other company I've encountered, is engineering and performance driven. A USB input and DSD decoding could easily have been included for marketing purposes, but that approach wouldn't have accorded with Ritter and Pflaumer's fundamental values.

The Alpha Reference is considerably more upscale in look and feel than the original Alpha. Although the Reference shares the Alpha's front-panel display and controls, the Reference's chassis is milled from a solid aluminum block, giving this 30-pound component a solid, brick-like feel. Front-panel switching includes input selection (two SPDIF, one AES/EBU, one TosLink), volume control, absolute-polarity inversion, filter choice, a button to change the display (volume, input sampling frequency, filter type, left/right gain), and a display dimmer. All these controls are duplicated on the handsome remote, along

with a mute button and a balance control. LEDs indicate when the unit is locked to a source and if the input signal has been HDCD-encoded. The "Lock" LED glows amber when the Reference has established initial lock with the source, and then changes to green when the Reference locks to the source with a second, higher-precision clock. The Reference can drive a power amplifier directly with no need for a preamplifier in the signal path.

Both SPDIF inputs are on BNC jacks, not the typical RCAs. This is another example of Berkeley's "no sonic compromise" approach. BNC connectors are not only the correct impedance (75 ohms); they also form a much more secure mechanical connection between jack and plug. Berkeley recommends AES/EBU; it has ten times the voltage compared with SPDIF (5V vs. 0.5V), which reportedly confers a slight advantage in timing precision. Balanced analog output is on XLR jacks, unbalanced on RCAs.

Although you can't input DSD into the Alpha Reference, you can play DSD files by converting them to PCM in a Mac or Windows computer running the software playback engine JRiver Media Center. Buying an Alpha Reference gets you a license to JRiver. The rationale behind this approach is described in detail in the accompanying interview with Michael Ritter. If you want to drive the Reference with a USB output, you'll need Berkeley's Alpha USB.

Removing the heavy top panel and looking inside the chassis conjured up the image of a bank vault. The chassis' solid aluminum block has been milled out to create three separate isolated chambers—one for the power supply, one for the front-panel display and control

electronics, and one for the DAC, DSP, and analog output stages. This design confers several advantages, including isolation from outside noise and vibration, isolation between subsystems, and temperature stability.

I enjoyed using the Reference on a daily basis. The front-panel layout, labeling, and display, and the remote control are sensible and well thought out. The circuit design is similar in many ways to the original Alpha DAC, but implemented with new parts and build techniques impossible in a \$5000 DAC. After listening to the Alpha Reference and considering its design, I realized that this must be one of the most cleverly engineered products I've reviewed. By that I mean that every dollar of the parts budget was laser-focused on optimizing performance, with nothing wasted on superfluities. The Alpha Reference also upended several of my biases about what it takes to create a state-of-the-art DAC. As you'll see, the Alpha Reference sounds spectacular, and yet it realizes this unprecedented sound quality with what looks like a fairly conventional power supply (no outboard box filled with dozens of stages of cascaded discrete regulation), an off-the-shelf DAC chip (from Analog Devices), and an op-amp output stage. What you don't see are the extraordinary parts and the design techniques that have been applied to the subsystems that really matter, particularly the clocking and the handcalibration of the analog filter. Berkeley has figured out exactly where to spend its parts budget—and where not to.

Listening

The playback system in which I evaluated the Alpha Reference is as good as it gets, in my experience. All the components are extraordinarily transparent, resolving, and dynamic, with these qualities in abundance over a very wide band. It turned out, however, that rather than the playback system telling me how the Alpha Reference sounded, this DAC revealed to me, for the first time, the playback system's full capabilities.

The highest praise that reviewers can heap on a DAC is to describe it as "analog-like." The Alpha Reference is certainly "analog-like," but not in the way that term has been used in the past. This accolade has described a DAC with a slightly softish treble, good space and bloom for digital, and an overall presentation that favors ease over resolution. The Alpha Reference transcends such comparison, overturning the idea that digital can merely aspire to mimic analog's best qualities. Rather, the Alpha Reference stakes out entirely new territory with a presentation all its own that sounds like neither analog nor digital, but rather like microphone feeds. The Alpha Reference is the first DAC in my experience to cross a threshold in which digital reproduction is no longer judged

by how far it falls short of the analog benchmark. Make no mistake; the Alpha Reference is a watershed event in digital audio's long journey out of the Dark Ages. It's not just a little better than the best out there—it is significantly superior in every sonic criterion as well as in the musical involvement those sonic qualities engender. It's safe to say that no one has heard digital audio sound like this before.

The Reference's "un-digital" sound isn't achieved by masking digital shortcomings or by mitigating them with an overly smooth sound or by adding a bit of artificial bloom. Rather, the Reference presents a startlingand I mean startling-sense of tangible instruments existing in what is easily the most spacious and dimensional soundstage I've heard from digital media, from the best high-res files to older CDs. This vividness of timbre and image flows directly from the Reference's crystalline transparency. The Reference reveals that all previous DACs imposed a layer of opacity between source and listener, which diluted the sense of immediacy and realism. Hearing familiar recordings through the Reference is like taking several steps forward through the chain right to the microphone feed. As good as some digital has become, it has never guite engendered that same frisson of realism that comes so easily to analoguntil the Reference.

This sense of hearing nothing between

SPECS & PRICING

Input sampling rate: 32kHz-192kHz
Input word length: 24-bit

Inputs: AES/EBU, SPDIF on BNC (x2),

TosLink
Outputs: Balanced on XLR jacks,

unbalanced on RCA jacks

Output level: 6.15Vrms at OdBFS
(balanced); 3.25Vrms at OdBFS
(unbalanced)

Digital volume control and balance: 0.1dB steps, 0.05dB L/R balance, 60dB range

Remote control: Volume, balance, input selection, absolute polarity reversal

Digital filter: Custom, user selectable THD+N: <-110dBFS at maximum

output

Firmware: Upgradable through signal inputs

Warranty: Three years parts and labor Dimensions: 17.5" x 3.5" x 12.5"

Weight: 30 lbs. Price: \$16,000

BERKELEY AUDIO DESIGN

(510) 277-0512

berkeleyaudiodesign.com

ASSOCIATED COMPONENTS
Loudspeakers: Magico Q7

Preamplifier: Constellation Audio
Virgo II

Power amplifiers: Soulution 701 monoblocks

Analog: Basis Inspiration turntable, Air Tight PC-1 Supreme cartridge Phonostage: Simaudio 810LP Interconnects: MIT Oracle MA-X, Transparent Reference XL,

AudioQuest WEL Signature Loudspeaker cables: MIT Oracle MA-X SHD

Digital cables: Wireworld Platinum Starlight, AudioQuest Eagle Eye (BNC), AudioQuest Wild (AES/EBU)

AC power: Four dedicated AC lines, Shunyata DPC-6, Triton, Talos, Cyclops, and Typhon conditioners, Audience aR-6TSS, aR2-TSS

AC cables: Shunyata Alpha Digital,

Alpha HC, Anaconda; Audience Au24SE

Racks: Stillpoints Ultra
Amplifier stands: Critical Mass

Systems MAXXUM

Isolation: Stillpoints Ultra 5, Ultra SS, and Ultra Mini; Critical Mass Systems Rize; Shunyata cable elevators Acoustics: ASC 16" Full-Round Tube

Traps, ASC Tower Trap, Stillpoints

Aperture panels

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you and the instruments is heightened by the Reference's unprecedented timbral truth. The slightly grayish patina overlaying tone colors, the shaving off of fine micro-details that dilute vividness, and the homogenization of images we've become inured to in digital are completely absent. The Reference has an uncanny ability to reveal much more information about how a sound was created, and consequently to produce a more lifelike impression of the real thing. I was struck by this quality while listening to Skip, Hop, and Wobble, a wonderful acoustic trio album by Jerry Douglas, Russ Barenberg, and Edgar Myer. Although I've heard this standard-issue CD on countless systems over the years, hearing it through the Alpha Reference was revelatory. The picking of the guitar and dobro, the way each note bursts forth for an instant, the resonant instrument bodies, the fine texture in the dobro's unique timbre, and the sounds of fingers on the strings all came together to create a more convincing illusion of hearing the instruments themselves rather than recreations of them.

As you'd expect from this description, the Reference's reproduction of the human voice is startling in its naturalness. Jane Monheit's gorgeous voice on the 96/24 version of *Come Dream With Me* has a palpability and immediacy that are downright eerie. The close miking and minimal reverb on her voice make the illusion of someone singing between the loudspeakers that much more credible. Although this file can sound wonderful through other great DACs, it never quite crossed that threshold into making me believe, if just for a few moments during the unaccompanied passages, that someone

was actually standing in my listening room. The difference in the electrical waveforms output by the various DACs under consideration must be miniscule, but the musical effect is anything but. The Reference allows the music to create a sense of intimacy between listener and artist in a way that I've never before experienced from digital.

A large measure of the Alpha Reference's sense of realism comes also from the extraordinary spatial presentation. The Alpha Reference's soundstaging, dimensionality, and depth aren't merely spectacular "for digital"; they are spectacular, period. Instrumental images are tightly focused, but in a way different from other digital that has rendered a "sculptured" presentation. Rather, the image outlines are clearly delineated from the air around them in exactly the same way that real instruments sound in an acoustic space. There is no artificial edge to the outlines despite the tight focus. The way the sound expands around the image with each note-what Jonathan Valin calls "action"-is totally natural and lifelike. The spatial rendering is also extraordinary in the layering and bloom, with instruments positioned along the depth axis in a continuum rather than in discrete steps. I've described other DACs as exhibiting this depth-along-a-continuum phenomenon, but the Alpha Reference is clearly in a different league. The see-through transparency I mentioned previously combines with this spatial resolution to present even the lowest-level sounds at the rear of the hall with sensational vividness and clarity. I also enjoyed the manner in which the Alpha Reference "dehomogenizes" familiar music, presenting a collection of individual instruments, each distinct in tone color and space. The Reference is revelatory in the way it allows me to easily shift my attention between instruments or sections, and thereby to hear more of the composer's intent. I found myself experiencing familiar music from a different perspective as more and more musical information was unwoven by the Alpha Reference.

To hear all of these qualities at their zenith in a single musical example, look no further than Dick Hyman and the Swing All Stars at 176.4kHz/24-bit on the Reference Recordings HRx sampler disc. The distinct tone colors of the brass and woodwinds are richly portrayed, even during the unison phrases. The hi-hat "lights up" the acoustic in a completely natural way. The sense of transparent space is palpable. The piece includes an extended passage in which Frank Weiss plays a beautiful sax line that weaves in, around, and counter to the melody played by the brass and winds. The Alpha Reference, more than any digital I've heard, presents this playful counterpoint in all its glory-totally natural and unforced. I defy even the most diehard analog enthusiast to listen to this track through the Reference and detect the slightest trace of the flaws that have traditionally been assumed to be partand-parcel of digital audio.

For all the Reference's vividness and resolution, it has a completely non-aggressive, almost laid-back character. This may seem like a contradiction, but the Reference's lack of edge and glare allowed it to sound immediate yet relaxed-just like live music. The rapid-fire flamenco guitars on Paco de Lucia's *Live*

in America are beautifully delineated with tremendous transient speed yet without the etch that makes you want to turn the music down. Orchestral crescendos at high playback levels don't create that sense of physical tension or "cringe factor" as your ears prepare for the glare. In fact, the Reference allows you to listen at louder levels, for longer sessions, without fatigue because of this smoothness and liquidity.

It almost goes without saying that the Alpha Reference's resolution is simply stunning. This DAC reaches down into the finest micro-details of timbre, transients, spatial cues, inflection, and dynamic shading. Everything is right there, laid out in a completely natural way that doesn't call attention to itself as detail. The treble is ultra-smooth, silky, and richly resolved. Even compared with other digital that could be considered as having a smooth treble, the Reference is lacking the metallic bite that has plagued digital audio since its inception. This combination of rich detailing, massive resolution, and timbral liquidity in the top end is simply unprecedented.

The way in which the Reference portrayed dynamics is also unlike any other digital playback. Instrumental attacks jump to life with stunning speed and immediacy, much like one hears from a horn loudspeaker. Listen, for example, to the brass entrance in *The Firebird*, again from the HRx sampler, which will nearly lift you out of your seat. This quality just increases the Reference's vivid realism, but again, without the slightest trace of etch or artificial edge.

As if this embarrassment of riches weren't

SET-UP NOTES

The Alpha Reference's balanced output fed a Constellation Audio Virgo II preamp throughout my listening sessions. Although I tried driving the Soulution 701 amplifiers directly from the Alpha Reference, the DAC's highish output level combined with the Soulution's high input sensitivity to require several tens of dBs of (digital-domain) attenuation in the Alpha Reference. Nonetheless, the Constellation Virgo II was more than up to the challenge, fully revealing the Reference's transparency, resolution, and dynamics.

I listened to the Alpha Reference fed by SPDIF from the extraordinary dCS Vivaldi transport via an AudioQuest Eagle Eye BNC cable (both units have BNC connectors), as well as from a MacBook Pro running Pure Music. Berkeley's Alpha USB converted the Mac's USB output (via a 1.5 meter Wireworld Platinum Starlight USB cable) to AES/EBU. The AES/EBU output was connected to the Alpha Reference through a 1.5 meter run of AudioQuest Wild digital cable. Analog output was balanced on MIT Oracle MA-X interconnects.

For comparisons with other DACs, I evaluated those DACs' sound when fed through their USB input as well as with the AES/EBU output from the Berkeley Alpha USB. This enabled me to isolate the sound of the DAC itself from its USB implementation. Incidentally, if you own a USB DAC, adding the Berkeley Alpha USB to bypass your DAC's USB input is likely to be a significant performance upgrade.

I set the Reference's output level at 59 on the display, or 1dB of attenuation from full-scale. The digital filter apparently operates better when processing less than full-scale signals. Although the Reference offers several filter choices, the default filter is by far the best sounding.

In comparisons with other DACs I was careful not to use HDCDencoded titles, which would give the Reference an unfair advantage because of its HDCD decoding. The Reference sat on four Stillpoints Ultra 5 isolation devices, which in turn were bolted into the Stillpoints grid within the Stillpoints rack. At this level of performance, every change in isolation is easily audible, and sonically significant.

enough, the Alpha Reference's bass reproduction is in a league of its own. Believe me, no one has ever heard bass like this from digital. For starters, the overall bass balance is weighty, warm, and rich, but without the caveats that typically accompany those descriptors. "Warm" and "rich" often describe a softish bottom end that is pleasant, but that lacks dynamic agility and pitch definition. The Alpha Reference's fullbodied bottom end not only provides a solid tonal foundation, it is also exquisitely textured and nuanced. The sound of plucked acoustic bass, for example, is infused with rich microtexture and micro-dynamic details that, frankly, other digital simply smears. The attack of the string, the resonant body of the instrument, and the decay are all beautifully delineated in a way that sounds more like the instrument and less like a facsimile. I was surprised by how much more fine detail in the bottom end the Reference revealed. Despite the filigreed rendering, the bottom end has tremendous power and speed.

This synergy of muscular authority, resolution of textural detail, and dynamic agility is sensational on a wide range of music. Orchestral music is big and full-bodied in a way that you rarely hear from reproduced music. The "oomph" in the midbass, richness and density of tone color, and that thrilling visceral involvement you hear from live music are abundant through the Reference. The

sense of rhythmic propulsion on rock, blues, and some jazz is nothing short of addicting. The track "Trans-Island Skyway" from the 96/24 version of Donald Fagan's *Kamakiriad* has an unusual meter that gives it a powerful propulsive groove. The Reference better resolved this track's amazing bottom-end dynamics, and not just in pure impact but also in the lack of smearing of the closely spaced kick-drum attacks. The Reference takes this track up several notches in that powerful ability of music to involve your entire body.

Perhaps not coincidentally, all the components in my current playback system share the specific quality of muscularity in the power range. The



Magico Q7s, with their dual 12" woofers and 10" mid/woofer in a totally inert sealed enclosure, the mighty Soulution 701 monoblocks with their unprecedented bottom-end impact and resolution, and the MIT Oracle MA-X, known for its richly textured bass and midbass, teamed up to produce what is in my experience the most expressive presentation of what the British call pace and timing of any audio system I've heard. The Alpha Reference at the front end of these cost-no-object components revealed qualities in those components that had previously not been fully exploited.

I found myself astonished that these characteristics are apparent not just in super-high-resolution audiophile spectaculars, but across a wide range of music in my digital library. Standard CDs of my favorite recordings that I'd thought sounded hard, flat, and relatively low in resolution were "unwoven" by the Reference to reveal a rich panorama of musicality. That's a significant observation because it reveals that our CD libraries contain buried musical expression that can be released by improvements in digital-to-analog conversion technology.

Conclusion

The Berkeley Alpha DAC Reference Series is not only the absolute state of the art in digital-to-analog conversion, it also goes far beyond even this superlative to redefine what's possible

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in digital playback. This is a landmark product in that it crosses a threshold of sound quality and musical expressiveness that renders moot the idea that digital can only aspire to mimic analog rather than offer its own set of virtues.

I won't reiterate the Alpha Reference's merits, but can guarantee that you've never heard digital audio sound like this. This is a product that you have to hear for yourself to believe just how far digital has traveled. I'm also heartened by the Reference's price. Although not inexpensive, \$16,000 for the unquestioned state of the art in digital playback makes it seem like a bargain.

On a personal level, I can't tell you how thrilled I am to experience an entirely new and unexpected level of musical involvement from my library of standard-resolution CDs and files. After one particularly rewarding session, I reflected on how Berkeley Audio Design epitomizes the highest ideals of highend audio.

Ritter and Pflaumer toiled for years, researching the finest minutia of design details that affect sound quality to create a product that has allowed me to experience a deeper level of musical involvement and appreciation. Their single-minded pursuit of performance above all else exemplifies the ethos behind the landmark breakthroughs in the history of high-end audio.

And breakthrough the Alpha Reference is. I'm confident in saying that the Alpha Reference will be remembered decades from now as a turning point in digital audio sound quality.

Michael Ritter Discusses the Alpha Reference's Design

Robert Harley: Berkeley Audio Design has been in business for six years, yet you've introduced just two products before the Alpha Reference. Many other companies would have offered a range of products, including inexpensive portable USB DACs. Why have you taken this approach?

Michael Ritter Well, we're a reasonably atypical company. We're in business to be in business partly, but it really was a passion that led us to even start the company. The work we did at Pacific Microsonics [the company that invented HDCD and built the professional analog-todigital converters and HDCD encoders used in mastering studios] was as close to a pure R&D effort as you'll find in audio, where the company's sole effort was directed at sound quality. The concept of HDCD was to create technologies that provided a much higher resolution signal in a standard Red Book recording that could be played back on any CD player. Obviously, to do that you need to start with a high-resolution recording, so we developed professional high-res A-to-D's, encoders, and D-to-A's that operated at 176.4kHz. We spent many millions of dollars and had the best test equipment available, including some built in-house. And we had excellent first-generation Keith Johnson analog mastertapes as reference source material. But the final design optimization was done in the field using Keith Johnson's live mike feeds since reproducing them represented the ultimate test of fidelity. These weren't mike feeds of a guy sitting in a room with a guitar, but of large choral groups and orchestras. Just tremendously big and difficult material to reproduce. We had a process where we could compare the mike feed to the entire A-to-D and D-to-A chain. We could hear instantly if we were listening to the mike feed or conventional digital or even 30ips analog tape, but at the end of the development process for the Pacific Microsonics Model Two sometimes we would confuse the A-D-A chain with the mike feed. That was a big emotional experience that I never thought would be possible.

You know the story of how Pacific Microsonics was bought by Microsoft, who didn't do much with it. So Michael Pflaumer and I [Michael Pflaumer is the co-inventor of HDCD with Keith Johnson, and wrote the HDCD encoding and decoding DSP code—he also designed the clocking for the Pacific Microsonics Model One and Two] realized that it just seemed wrong to have arrived at this potential for audio reproduction and let it disappear. We had some brilliant engineering expertise so we concentrated on building products with the best possible audio quality.

Going back to what you said about introducing only a few products, we didn't want

to introduce products that offered only an incremental improvement, or feel compelled to change models for marketing reasons.

RH: Is the Alpha Reference a ground-up design effort or is it based on the original Alpha DAC?

MR Much of the technology we had already developed for the Alpha DAC Series 2, such as the data receiver and the digital filter, was pretty darn optimal. We knew there were areas we could do things better with certain parts, but those parts didn't exist. There were also areas where the implementation could be improved. The Alpha DAC Series 2 was the platform we started with.

RH: Why does the Alpha Reference lack a USB input?

MR There may be a time when it makes sense for us to introduce a lower-cost DAC with a USB input, but when you're shooting for the ultimate in performance, which is what we did with the Reference Series, not to mention the Alpha DAC before it, you absolutely don't want to connect the DAC directly to the computer or router. They have large amounts of electrical noise, and that noise gets injected into the DAC's ground, or the noise is capacitively coupled through the input. A separate isolation/reclocking device [a USB-to-SPDIF converter] for computer-audio playback is essential if you're going for the state of the art.

RH: Tell me about the DAC in the Reference and how it is clocked.

MR It's a highly optimized delta-sigma DAC, a topology we've used before. The DAC chip does nothing but D-to-A conversion; no other processing, no filtering, no DSD conversion. When you add other processing on the same piece of silicon while you're doing D-to-A conversion, it degrades performance.

The environment in which the DAC chip operates, and how it is clocked, is unique in the Reference. We use a clock with extremely low time-domain noise. We worked with a number of vendors before we found one that could deliver a part that met our requirements. The actual conversion clock in the Reference is very, very expensive but it's pretty stupendous. We're talking around 30 decibels superior to the competition. We're using really expensive analyzers like the Agilent to characterize each clock, but the clock has such low phase noise that when we measure it we're looking at the analyzer's performance as much as the clock's.

We think of maintaining the clock's performance as similar to maintaining a vacuum, meaning that everything in the environment around the clock—the signal path, the power supply, the digital input stream, the temperature variations—is trying to get in and degrade the clock's performance. It's very difficult to maintain the clock's extreme integrity, which is why we devoted so much design effort and circuitry to isolating the clock. The clock's ultimate performance has to arrive at the DAC chip or it's wasted. The clock is quite close to the DAC and goes through impedance-controlled lines. The circuit board's dielectric characteristics are absolutely the

state of the art—a ceramic aerospace material. It costs an arm and a leg, but it's worth it. The board material also pays dividends in the analog output section.

It's a big effort to A, have a phenomenal clock and B, isolate it from junk coming in and C, deliver it to where the payoff is at the DAC chip in pristine form. I don't understand those designs where the clock is some distance from the converter, or worse, in a separate enclosure so that the clock has to go through drivers, connectors, and cables. There's just no way that you can maintain the ultimate performance of a clock when it goes through a longdistance transmission system

like that. The more advanced and extreme the clock's performance, the more difficult it is to maintain that performance at the DAC chip.

You'll have seen that when an input signal is present the front panel "Lock" light comes on amber and you can hear audio and it sounds fine. But then when the high-precision clock engages and the green LED illuminates, the precision clock is operating in isolation. That's when the magic really happens.

It's amazing what you hear when you pay this much attention to the clock—things you might not anticipate, such as bass performance.

"We think of maintaining the clock's performance as similar to maintaining a vacuum, meaning that everything in the environment around the clock is trying to get in and degrade the clock's performance."



We've done a lot of work in this area since the late 1990s when we developed the Pacific Microsonics converters.

RH: Let's talk about the decision to offer DSD compatibility in software rather than building it into the Alpha Reference's hardware.

MR I have to confess that was the one feature that tested me a bit, for a number of reasons. By including DSD compatibility in the Alpha Reference we could advertise it as DSD-compatible without any complex explanations.

We could have run DoP [DoP stands for "DSD over PCM," a DSD interface standard] into the Alpha Reference and the front panel would say "DSD" and you'd hear DSD. It's extremely inexpensive to implement—it approaches zero cost.

But while that would have made a good marketing story, it would have compromised the Alpha Reference's performance. If I'm hewing to my highest purpose as a manufacturer, it's to give customers the most musical, satisfying experience I can. And that's in direct conflict with the DoP approach, because you're doing

the conversion from DSD to the multibit signal that drives the DAC within the DAC chip or adjacent to it. Virtually all manufacturers use DACs with multi-bit architecture. There are a few exceptions, but they are about 0.001 percent of all the converters out there. Almost every DAC that calls itself DSD-compatible, or even "native" DSD, is converting the DSD bitstream to multi-bit just before the D-to-A conversion.

Most DAC chips, including the one that we use, have a DSD input. Processing in the DAC chip converts it to multi-bit, but that's the worst possible approach because you're doing that processing and the digital-toanalog conversion simultaneously on the same piece of silicon. A step up from that is to do the multi-bit conversion with your own code in a separate DSP processor chip. And we could easily have done that. We have a lot of DSP processing power in the Reference Series-it would have been trivial. But then you've got the extra overhead of processing going on all the time in a chip that's contiguous to the DAC. And remember, we're being hypervigilant about the environment in which the DAC chip is operating in the Reference Series, because the intrinsic performance coming out of it, the way we operate it with those clocks and with everything else, is phenomenal. That processing noise would degrade that performance.

So, those are the two solutions for getting a front-panel DSD light to come on and make a DAC "DSD-compatible."

We could have rationalized including DSD in that way, but Pflash [Michael "Pflash" Pflaumer] and I both agreed that our sense of integrity required us to follow the path of providing the best possible audio quality for both DSD and PCM.

Fortunately, there's another way to provide DSD reproduction that doesn't compromise performance, and that's to do the DSD-to-multi-bit conversion outside of the DAC. Because virtually all DSD sources that feed external DAC's are computers, we can do the conversion in software in the computer. We did a fair amount of research on it and considered writing our own software, but we found a product that did the math right, and that's JRiver Media Center for Mac or PC.

For the tweaky types, you can optimize the low-pass filtering for DSD conversion in JRiver depending on the spectrum of the supersonic noise, which varies between recordings. The resulting upconverted 176.4kHz/24-bit PCM format has sufficient resolution in the frequency and time domains to contain everything that's in the DSD signal, including supersonic noise if you want it there. You can play the DSD files on your computer and have JRiver perform the conversion on the fly. Or you can convert the DSD file to PCM ahead of time and then just play the PCM file. That shouldn't make a sonic difference in theory, but it does in practice. The processing overhead to perform that conversion in real-time makes it slightly less good sounding compared with converting ahead of time. The computer then isn't doing any processing, just outputting data pulled from memory. From a technical and audio quality point of view that's the best way to reproduce DSD recordings if you have a multi-bit DAC architecture. You're not only performing the conversion outside of the DAC box; you're also performing the conversion outside of the time domain as well. By the way, including a license to JRiver with the Alpha Reference is vastly more expensive for us than implementing DoP.

RH: There's also the potential of upgrading the DSD- to-PCM conversion algorithm with a software update, something you can't do if the conversion is performed inside the DAC.

MR That's exactly right.

RH: Tell me about the Alpha Reference's physical construction and how that affects the sound.

MR The physical design of the Reference is all about minimizing noise and time-domain noise. As you've seen, it's a very solid device physically, and the main reason for that is isolation and stability. The mechanical mass reduces microphonic effects as well as temperature gradients. Once it fully warms up, which takes about 12 hours, it will stabilize at that temperature, which is important.

The top cover is over a centimeter thick and each circuit board is in its own isolated

THE IMPORTANCE OF CLOCKING

With so much discussion in this interview about the importance of a DAC's clock, I thought I'd offer a generic and simplified primer on what a DAC's clock does and why it's important.

In a typical multi-bit DAC, the DAC converts the 24-bit audio samples (called "words") at the DAC chip's input to an analog current at the output. The DAC chip performs this feat 352,800 times per second in a typical 8x oversampling DAC (352.8kHz is 8x the CD's sampling rate of 44.1kHz). The "word clock," a square wave with a frequency of 352.8kHz, tells the DAC when to convert each of those binary-encoded audio samples to an analog output. Each leading edge of that square wave triggers the DAC to perform the conversion of one audio sample to an analog current, and it does this 352,800 times per second.

If those clock pulses aren't perfectly uniform in time the definition of jitter—the reconstructed analog waveform will be distorted. Specifically, timing variations in the clock become amplitude variations in the analog waveform. The converted sample's amplitude may be correct, but if it's shifted in time from where it should be, amplitude errors are introduced in the analog waveform. The right sample at the wrong time is the wrong sample.

It turns out that humans are exquisitely sensitive to the most miniscule timing variations, probably because the type of distortion jitter introduces never occurs in nature. Moreover, evolution has finely honed our hearing mechanism for instantly identifying a sound's location and determining what is creating the sound. It is these very characteristics—spatial cues and timbral recognition—that are obscured by jitter. Given that our survival depended on correctly identifying the "what and where" of a sound, it's not so surprising that we are so attuned to any mechanism that confuses these aural cues. **Robert Harley**

chamber to reduce noise coupling. We have very tight RF shielding, and the top panel is machined to within a couple thousandths of an inch so that it forms a tight shield around the components inside. Everything about the design provides a quiet and stable environment for the DAC.

RH: Tell me about the hand-calibration process.

MR [Laughs] It's really not the best kind of product to make if you want to make millions of them and go to the bank. But once again, this is a hyper-precise device and the manufacturing process takes a couple of weeks. After testing we burn in each unit for seven days, 24 hours a day, with a digital input signal and loads on the analog outputs. Then it's taken to another location where it's tested and thermally stabilized for another day. Then we do the final alignment process which involves both measurement and listening. I didn't mention it before, but the digital filter is our own propriety design with very precise performance parameters. To preserve that performance we model the analog output filter as a precision cascaded part of the overall digital/analog filter system. It's the analog filter that we handadjust to the tolerances we require-1/100th of 1 percent. To be able to make adjustments with that precision we use the very best trimmers available with precious-metal wipers and operate them over a very limited range. These devices are not normally used commerciallythey're mil-spec. But for precision, stability, and repeatability, we had to use them. It's an iterative process of measuring and listening that takes three hours per unit.

If we tried to rush any aspect of the manufacturing or alignment process it would be just like putting a junk part in the device. Every aspect has to be done with full integrity or it's a wasted effort. We can build only two Alpha References per day.

RH: That's pretty limited production capability.

MR It is pretty limited production. If the whole world falls in love with this product, we could increase production with a parallel production path, but this is not trivial because of the final alignment. We have a brilliant, degreed engineer with well over \$100,000 worth of test equipment and about \$100,000 worth of playback components performing the alignment. If we wanted to have another final alignment station operating in parallel, we couldn't just hire somebody and have them align it by measurement.

The important thing is that it has to be right. We don't ship it until it's fully optimized. From an integrity point of view, we're giving the customer the performance he's paying for. And it's also self-interest; our success is based on our audio quality and we're not going to do anything to jeopardize that.

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Light Harmonic Da Vinci Digital-to-Analog Converter

The Vitruvian DAC

Scot Hull

first met Light Harmonic's chief designer Larry Ho in Atlanta at the AXPONA show, back in 2011. Of all the nifty bits I ran across that day, a few stood out, but the most striking was clearly his Darth Vader DAC.

Maybe it didn't really *look* like Darth Vader. Maybe it just *evoked* the Dark Lord of the Sith. But you take my point—it was a striking design.

He laughed at me when I mentioned Lord Vader, and quickly pointed out why his brandnew DAC, which he was calling Da Vinci and not Darth Vader (for copyright reasons probably), had that angular chic: I was looking at two distinct chassis, stacked in such a way that they could save space, eliminate the extra circuitry and external cabling that a separate chassis required, and still reap all the benefits that physical separation grants. All I heard was "it rotates," and I think I spent far too many minutes slowly pivoting the chassis top, which houses all the delicate conversion circuitry, back and forth over the power supply that sits in the lower box. You'll forgive me (and hopefully Larry will too) when I tell you that you really ought to try this out at the next audio show. It's a remarkable bit of machining, and the slip-snick as the top rotates is a bit akin to fondling the bezel of a Rolex. Slip-snick. Slip-snick...I think I might have hypnotized myself a little bit there. Anyway, the Da Vinci definitely made an impression but it would be another two years before I got the chance to get up-close and personal.

Just so you know, the Da Vinci will play all of the standard, run-of-the-mill lossy and lossless computer audio files like WAV, FLAC, and AIFF at all the standard sampling rates, including 44.1kHz, 48kHz, 88.2kHz, 96kHz, 176.4kHz, and 192kHz, across all inputs, which include a single USB-B interface, as well as AES/EBU and SPDIF. In a bit of foreshadowing,

it's the USB input that gets a bit of a bonusthe sampling *there* can also accept 352.8kHz and 384kHz files. And that, my friends, is pretty cool.

At \$20,000, the Da Vinci sits at the bottom of the ladder of Light Harmonic's scale. Sort of (we'll get back to that). One level up sits the DaVinci Dual DAC, an externally-identical box that adds the ability to decode DSD and double-DSD files along a completely separated and isolated path, which is the only way that the Light Harmonic team felt that DSD could be implemented without seriously compromising the sound. Two DACs, two paths, one chassis, \$31,000. And at the very top of the heap? The just-announced Sire DAC, at \$120,000, will compete with the very best on offer, and represents Light Harmonic's fullout, no-holds-barred assault on the high end.

An intimidating ladder, I'll admit it.

But for those of you prone to skipping ahead, let me completely spoil the surprise. The Da Vinci DAC is the best all-around performer I've yet heard. It isn't inexpensive and its looks will raise eyebrows, but the sound it's capable of weaving is the most comprehensively compelling I've yet heard out of my home system. Full stop. Done.

Listening to Da Vinci

So, let's be different and start squarely in the *middle*, with this: What do *you* listen for when auditioning a DAC?

Me, I tend to look for a couple of things. Maybe not first on the list, but somewhere near might be whether there's a sense of depth. A lack here covers a variety of sins-clarity and detail, for examples. When I'm truly happy with

a DAC, it's usually due to the sense that I can hear "more deeply" into a recording. Playback that's two-dimensional, with a soundstage that's abbreviated in any number of ways, is fairly common, and leaves me feeling like I'm peering through a window. This is something that's easy to get used to rather guickly, but a component can make that view more or less immersive, more or less immediate. This is where I start thinking about "veils" and whether or not they've been added or removed. Now, once in a great while, so many of these obstructions get removed that the hyperbolic in me tends to reach for something overblown. Like, say, imagining that the system has suddenly taken a sledgehammer to the window, removing it entirely, and providing something more direct in the way of a personal experience. Sometimes, hyperbole is really the only way to get across the fact that something is different. Really different. I guess it won't come as a shock that I'm going to paint the Da Vinci as one of those things.

Take an excellent recording, like the shockingly clean Stockfish release of Chris Jones' Roadhouses and Automobiles. I use the title track not for the music anymore because, quite frankly, I've played it so many times it makes my skin crawl, but instead for some of the sonic trickery embedded in the mix. To wit, there be bugs. Played back on a resolving system, you can guite distinctly hear crickets in the opening and closing sequences on the first track.

I have no idea why, but they're there. I've heard this odd bit of detail through many systems (and not heard it through more), and I've used it for a while now as a firstlevel barometer of how well a system can resolve detail in the soundstage. So, I ask: "Does the [insert component here] pass 'The Cricket Test'?"

With the Da Vinci, I found myself seated on my front porch at dusk, with the little buggers just swarming the soundstage. Interesting bit? It wasn't just the opener that was infested-it was the entire track. Every transition, Every change-up, Somebody clearly needed to call an exterminator. You can imagine me leaning abruptly forward in my listening chair, a studied look of snobbish boredom suddenly broken by a creased frown, darting eyes, and a stabbing jab at the replay button on the remote. That. Was. New. Cool!

Or rather, it wasn't. That sort of thing is always there; it's just whether or not you can get to it. Noise is a tricky thing, a great mask, but it's also one of those things-like that window-that's really only obvious when it's gone. Now, "black backgrounds" are about as cliché as you can get in high-end audio, and talking about them without sounding like a random-phrase generator that habitually spits out the word "inky" is apparently quite difficult. It's hard to know what the term actually means, especially out of context-silence is silence, after all. Which made the surprise visit from Inky Blackerstein and his Complete Ensemble of Deep-

Space Emptiness such an unexpected and completely revelatory treat. Apparently, I'd never been formally introduced, but be assured that he's guite a fine fellow to have over for a listening session. Anyway, what all this meant was a spill from The Cornucopia of 3-D Information pretty much all over my listening room.

Todd Garfinkle's excellent MA Recordings label, for example, is a marvelous way to explore these kinds of experiences. Todd is a "recording artist"-that is, he creates recordings in a downright artistic way. The idea of capturing music in "living spaces" is probably not new or unique, but it most definitely is unusual. La Segunda is the second outing for Sera Una Nocha, an eclectic group of musicians pulled together by Todd and his partner to play some equally eclectic music. This particular recording was made in a small monastery in the Argentinean countryside with only a pair of omnidirectional microphones, and the venue is the cradle that holds the haunting and delicate work. Played back at 24-bit resolution and 176kHz sampling, the sound is wildly open and airy. Playback shows the players clearly arrayed in a semicircle around the microphones, and the specificity of their placement is utterly transparent. Interestingly, Todd told me that the percussionists were actually sitting behind the mics; translation into my two-channel system places them behind the vocal-

SPECS & PRICING

Converter type: R-2R architecture with patent-pending 3-layer buffer Output levels: 2.05V unbalanced; 4.1V Sacramento, CA 95827 balanced

Digital inputs: One asynchronous USB 2.0 interface on standard USB-B connectors (will accept up to 32-bit PCM at 44.1, 48, 88.2, 96, 176.4, 192, 352.8, 384K S/s); one asynchronous AES/EBU on XLR connector (will accept up to 24-bit PCM at 44.1, 48, 88.2, 96, 176.4, 192kHz); one asynchronous SPDIF on one RCA phono connectors (will accept up to 24-bit PCM at 44.1, 48, 88.2, 96, 176.4, 192kHz)

Balanced outputs: One stereo pair on XLR connector

Unbalanced outputs: One stereo pair on RCA connector

THD +N (unweighted): Better than 0.0018%

Residual noise (unweighted): Better than -115dB @ 20Hz-20kHz Residual noise (A-weighted): Better than -125dB @ 20Hz-20kHz

Crosstalk: -142dBFS @ 10kHz Dimensions: 18.5" x 7.87" x 18.5"

Weight: 61 lbs. Price: \$20,000 LIGHT HARMONIC, LLC 3050 Fite Circle, Suite 112 (888) 842-5988

ASSOCIATED FOUIPMENT TIDAL Audio Contriva Diacera SE loudspeakers; Soulution 530 integrated amplifier; Vitus Audio RS-100 stereo amplifier: Vitus Audio RD-100 DAC/preamplifier; Pass Laboratories XA-100.5 monoblock amplifiers: Pass Laboratories XP-30 preamplifier; Berkeley Audio Design Alpha DAC and Alpha USB converter; Purist Audio Design's Corvus Line cables and interconnects; Shunyata Hydra Triton power conditioner; isolation products from Symposium AV; DeVore Fidelity Orangutan O/96 loudspeakers; BorderPatrol S10 EXD SET stereo amplifier; BorderPatrol Control Unit EXD preamplifier; Auralic VEGA DAC; signal cables from MG Audio Design; power cables from Triode Wire Labs; Silver Circle Tchaik 6 power conditioner: isolation products from Symposium AV; MacBook Pro used as a media server, configured with Audirvana; media files on external FireWire 800 hard drive

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ist. It's a fascinating effect and is especially clear with the Da Vinci, and another something I hadn't noticed before. Again, *neat*.

Okay, after detail and depth, I tend to look for bass "authority." For whatever reason, my experience with high-res-capable DACs has been dominated by a sense of litheness in the presentation. It's as if the tonal balance is anchored a bit high. A great deal of listener attention gets focused on sonic aspects like "air" and "detail," which is generally a rather pleasing effect, but doesn't necessarily mean baby's got back. So, I reached for another track from Chris Jones' Roadhouses and Automobiles. "No Sanctuary Here" enjoyed a year or two as the Most Overplayed Song At An Audio Show, due in no small part to the ominousness of the bass track. It is, in a word, Big. Like, that'sa-thunderstorm-and-we're-never-going-toreach-shelter-in-time Big. Played back on big speakers in a big room it'll shake the walls and everything between, which is probably why it was so popular-it'll stop show traffic out in the hallway, for sure. Anyway, I've found that many supposedly full-range loudspeakers don't handle the track with equal aplomb, so I like to use it when Looking For Mr. Big Bass. What I'm listening for is a deep, satisfying sense of harmonic rightness, of pitch definition, of... okay, you know what I really want? It's fear, plain and simple. If the sound doesn't make me dive for the floor screeching "holy crap," it isn't right. It's also not enough to sketch that tone, I want speed, precision, and decay, but perhaps most importantly, I want fullness and continuity, and they are rare. So, assuming the speakers can go there, the question is, will this DAC Thelma-and-Louise me right off the cliff and into an audio abyss, or will there be some kind of last second slide just short of the danger zone? The Berkeley Audio Alpha DAC does this zone particularly well, and quite frankly, it was the reason I bought it. The Da Vinci also gets this particular aspect "right," but with a slightly different take. Namely, there's the sense that there's nothing to catch you as you dive off the background into forever. Thank you, Inky.

Tracking bass speed and PRAT took me to Jem's Finally Woken and "Come On Closer." A sexy track, this, where the bass line is front-and-center and climbs and dives throughout the tune. Another one for testing the limits of a loudspeaker or setup, it's a matter of continuity and roundness to the notes as they drop like cannonballs onto a suddenly trampoline-like soundstage. With the backdrop the Da Vinci knitted out of the void, tracking the bass was an athletic exercise, arresting and explosive and altogether addicting.

Reference Recordings has a reputation for great recordings, and the Minnesota Orchestra's performance of Rachmaninoff's *Symphonic Dances* is dynamic and complex in just the right way to short-sheet a digital converter. I've heard this piece quite a lot in 24-bit/176kHz high-resolution PCM; it's included on the *HRx Sampler* that's available from Reference Recordings that came with the Berkeley DAC. The piece begins softly; the temptation will be to crank it up early, which makes the crescendos even more entertaining, the sudden climbs so stark and unexpected that my dog fled in a mixture of terror and

outrage and led to another argument with my wife over "proper listening levels" and whether or not I'd be allowed to have the remote back. Whoops. But the Deep Space Da Vinci's backdrop sets a really involving stage and tensions mounted swiftly as the woodwinds and strings began to struggle with each other, battling for supremacy, instead of battling for audibility. And even with the lightning crash of cymbals and the thunderbolt of the timpani, you had depth, placement, and delineation. This is the most coherent rendition I've heard of this piece, and played back at appropriate volume (i.e., loud), even a diehard classical skeptic (that would be me) was thrilled.

Note to self: Play demos when the kids are in school not when they're in bed.

There are those who would call the Da Vinci "very analog." I'm not sure I'm one of them, as Great Big Bass on a vinyl system is even more problematic than it is for a DAC. So, no, the Da Vinci is *not* analog-like. It's better than that. But, that said, there's an ease to the sound that is entirely non-fatiguing. Not to say that it's in any way treble-challenged, but tracks do not tend to go brittle when bad-bad recordings stay that way, which is to say, Adele's 21 still sounds horribly compressed, even with the Da Vinci's Duet Engine sorting it out (more below). But what I mean is that it doesn't sound worse. Some DACs, when fed crap, tend to either smooth out the hard edges or use them as an excuse to start swinging bags of broken glass at your ears. The Da Vinci does neither-compression sounds like what it is, which is "a horrible tragedy." Here, vinyl tends to do better, as whatever travesties usually

visited on a recording destined for silver discs and/or iTunes tend to not be visited on the vinyl version.

Again and again, I was tempted and taunted about the volume—a little voice kept saying, "Maybe you should turn it up." My wife loved that part. Ahem. But the grain-free view on the music was as transparent as I've been able to achieve at home, and that view was fully as three-dimensional as the source material allowed. I can't be faulted for throwing myself headfirst into such waters, now can I? Volume restrictions be damned! Ha HA!

Tonally, I found the Da Vinci to have a balanced presentation as no particular part of the sonic tapestry stands out. The bass is exemplary, and there is no tonal shift upward or downward that would mark the designer as overly celebrating some favored portion of the audio band. On the whole, the DAC's presentation is unremarkably excellent and nothing feels out of place; it's all of a piece. Organic. Which makes isolating its signature something of a nightmare, but there it is.

I'd say that it was, in a word, musical, but I'm pretty sure that's another one of those damned clichés, so let's just settle on "awesome."

Technical Bits

This DAC is almost annoyingly stuffed full of audio wizardry, but one thing you won't find is any digital upsampling or oversampling. Why not? Well, it's simple: Light Harmonic couldn't get that kind of design to come close enough to the sound of an analog front end. Bitperfect protection and preservation led the company to a shunt-regulated resistor-ladder

architecture, but the secret sauce might well be the patent-pending three-layer buffering system that feeds the bits into the converting architecture in the most jitter-free manner possible. Aiding this are three clocks, one for 44.1kHz (and multiples) and one for 48kHz (and multiples), and also one 13MHz clock dedicated to the USB input.

With 64-bits in the volume-control architecture, the inevitable degradation that bit-tossing will introduce is reserved for far lower on "the dial"-and I put that in guotes because while adjusting volume is simple, getting to the volume control isn't. Since the remote that comes with the Da Vinci has no volume buttons, you're going to need a second one-for the computer. That is, the volume is modified entirely and only from the computer that's attached to the Da Vinci; it's your player's software that sends the attenuation signal to the DAC (assuming your software supports volume control, but most iTunes add-ons like Audirvana do), and the DAC then handles the actual volume level.

A couple of other things: the Da Vinci does two things worth calling out. First is the least significant bit (LSB) correction. Without getting overly technical, it's this modification to the attenuation algorithms that allows the digital volume control to achieve an unexpectedly high level of performance. I'll say more about this in a moment, but at the risk of sidelining the review in favor of a treatise on the Promises and Pitfalls of Digital Attenuation, it's good to know the LH team is aware of the fact that "simple" isn't necessarily "better" in this sphere.

Speaking of digital manipulation, there's the Duet Engine. The Web site describes this as a way to improve on the sound of "regular resolution" CD-quality audio files without upsampling or oversampling. Um, yeah. I had to ask about this. What happens is "time-aligned analog interpolation"; with parallel output modules, Light Harmonic is able to take the signals from each and perform operations on them that yield a more accurate result. Since there are two modules, this in effect doubles the sample rate. Clever. In practice, I found that the improvement was subtle but pleasing, with an unusual transparency that was surprisingly non-fatiguing.

And that's the digital bits.

Now, if I had to categorize what it was that worries me most about DACs in general, it has nothing to do with the actual conversion. Sure, there are good and better ways to do that, and acceptable-to-bad ways of feeding those chipsets. My feeling is that these problems tend to be pretty well understood. Where most DAC designers tend to take naps is on the analog output. Here's another place where the Da Vinci steps up.

The design is a zero-feedback architecture and if this sounds like an amplifier, then I probably won't put you off when I say that the design is fully dual-mono, fully balanced, using JFETs with an output buffer, and there are no op-amps anywhere. The Da Vinci has both single-ended/RCA and balanced/XLR outputs, with 2.05V output on the SE outs and double that on the balanced. More specs: With better than -125dB residual noise across the audible band and -142dB of crosstalk,

the whole Blackety-Blackblack and the Great Empty Nothing make sense.

Then there are three beefy R-core transformers. It's an unusual design choice, but with separated windings it's also one that minimizes noise. And there are 40 different regulators deployed across the architecture, from the input all the way down to the DAC chipset itself, including regulators for both the USB input and the individual clocks.

Worth Noting

Did I mention that the Da Vinci has an odd shape? That's called understatement. Anyway. the look of this DAC is going to be polarizing. It's *not* ugly by any means, but with the angles and softly glowing racing stripe that traces across the edge of the top chassis, well, it's... eclectic. That look is a long, expensive way to go to make a Sci-Fi reference, but as I mentioned, the design choice has a purpose. The housing is actually two distinct chassis, one mounted directly on top of each other, and coupled with a rotating hinge. Yes, a hinge. The top "box," when ready for use, will sit at a 45° angle to the lower one. I suppose this could be another reference to the where the name Da Vinci came from. The Vitruvian Man is that line drawing by Da Vinci of the longhaired naked man superimposed over himself showing two different arm and leg positions while drawn inside a square and a circle (makes sense, but I still prefer the image of Vader's meditation chamber). Everything has a purpose, an optimal shape, is part of a well-conceived plan.

While technologically something of a marvel, the design choice means that getting

to the cable inputs/outputs is a problem—I had to lift the chassis up to get at that tiny rearmounted panel, and once exposed, it's clear that the cables are a bit crowded in there. Tilting a 60-pound chassis around with your fingertips is going to be problematic, so do yourself a favor and give yourself some room before having at it, or better still, connect everything before getting it settled into your rack.

A note about the footers: There aren't any attached, but there are two sets that came with my unit that you can rest the narrow bottom of the DAC on. I started out with none, simply resting the flat bottom directly on my rack's platform, but fiddling clearly showed that there's a positive difference with the feet instead of without. Pick one of the two supplied, or, as the Light Harmonic team suggests, find an aftermarket set that tickles your fancy.

About the USB cable. There isn't one that ships with the DAC, but you're obviously going to need one to use that input. I'm going to wave my hands at the "how can a USB cable make a difference" argument and simply state that it does and that the optional LightSpeed USB cable that Light Harmonic offers is the best I've heard. No, the differences are not huge, but this one is reliably good and full-frequency. There are two versions, one with the "client-side" connectors (USB-A) separated (power from signal) and one with them joined. I tried them both, and for my setup I found the separated-connector version to be preferable, though the difference was extremely subtle.

The Da Vinci has several digital inputs, USB, SPDIF and AES/EBU. I tried all three-

especially that AES connector, as that's what I use with my Berkeley Alpha DAC by way of the truly excellent Alpha USB-to AES/EBU converter. I've used that converter with every DAC that allows me to do so, and without fail, its addition dramatically improves the performance of every DAC I've attached it to. Tighter bass, airier highs, cleaner detail—the gains are almost always across the board.

Well, that was true until I used the Berkeley converter with the Da Vinci. Very clearly, the Da Vinci is to be used with the USB input. Yes, you can use whatever input you like, but the USB input is different. It uses the now-standard asynchronous mode, courtesy of the XMOS receiver chipset, and all the extra-special buffering and filtering is done on that interface, so bypassing it in favor of a "legacy" input is going to be a mistake, in my opinion. Better still, the on-board volume control options are restricted to the USB input, so if you have any curiosity at all about running this DAC directly into your amps, you're stuck with the USB input anyway. It should be obvious, but as a safety precaution let me note that if you do set this up to run amp-direct from any signal sourced from the other inputs, the resulting volume will be at full scale (i.e., insanely loud).

Another curiosity has more to do with form driving function: Given that the shape just has to be this particular shape to meet the design goal of a non-resonant chassis that minimizes internal reflections also means that the readout/display is cocked upwards at a 45-degree angle, and unfortunately, it's not a fancy-shmancy highly-visible OLED display like you'd find on the AURALiC Vega. If the

DAC sits on the top of your three-shelf rack in order to show its sexy self off, you're not going to be able to read the display from anything resembling a listening position. I solved this quite straightforwardly by placing the Da Vinci on the lowest shelf I had, and ta da! Done. This also gave me the side benefit of being able to use the included remote. Not that I really needed the remote. It doesn't actually control the volume; other than muting, it's really only for engaging features, and once you've set them you can pretty much put the remote back in the hulking Pelican crate everything came in.

Last bit on the amp-direct thing. I used the DAC in quite a few different setups, including without a preamp into both a single-ended tube amplifier from BorderPatrol and into balanced solid-state amplifiers from Vitus Audio, Pass Labs, and others. In general my systems tend to sound more transparent with no preamplifier, but I'm going to hesitate before universally and unequivocally calling that "better." To be fair, I suppose it depends entirely on the preamplifier. I found that the DAC-only sound tended to be more open, with a larger soundstage and higher level of detail retrieval, than what I could manage with an external passive preamplifier, but with a very high-quality active preamp, the presentation could be more robust and muscular, with little degradation in the soundstage (and, perhaps, an even more extended, precise one). Where things got really interesting was in the evening, when unrestricted dynamics invoked wrath. Here, the Da Vinci-as-preamp excelled-detail, dynamics, and soundstage all maintained their characteristically high levels of performance even as I accommodated the schedules of my Little People, and did so well past the points I was able to achieve with my other two DAC references.

What I'm hinting at is that the amp-direct thing is most definitely worth exploring. If you're like me and have an analog source that you have absolutely no intention of foregoing, then this is all moot, but if you're looking to simplify with an all-digital setup (and assuming that the amplifier has enough gain), chances are that the Da Vinci, with a rather low 12 ohms output impedance and a 2- or 4-volt output, will likely not only drive your system fully and expertly, but could revolutionize it entirely.

A Question of Value

In a world of wildly escalating pricing and irrelevancy to anything resembling the "common consumer," another \$20,000 product isn't terribly exciting. The problem with all this, as I see it, is a belief that audiophiles are only interested if products are at or beyond a certain price point, and titillating anecdotes aside, this is just crazy talk and may well be the leading cause of the decline of the industry as a whole. On the other hand, the high-end-audio industry is also the only one I know of that routinely prices and sells design concepts as if they were regular products. It's as if Ford's latest thinking about the future of automobiles, a design it called "Concept Future" and showed at the Detroit Auto Show, suddenly got a sticker price and an order queue. These cars aren't meant to be daily-drivers (or even driven, in many cases), they're just ideas that the company tests out to see what reaction they evoke. But in highend audio, that's exactly what happens with

the "concept" components—they're priced and positioned for sale, if only to the super-high-roller. Of course, there's little to no expectation that these concepts will move a ton of volume, but as test-beds of nifty engineering or shiny design ideas, they're superb.

Case in point is what Light Harmonic has done with the Da Vinci. This DAC wasn't conceived of as a built-to-a-price-point product. It was commercialized (eventually) in that it is currently priced in such a way as to make profit for the company, but it really started as Larry Ho's statement of vision. The finished result told him many things about what a DAC platform can do, and that education led him to the 100-times-less-expensive Geek Out USB dongle—a soon-to-be-released DXD/double-DSD-capable head-phone DAC/amplifier. It also told him things that his original design could not do, and that is leading to the 6-times more expensive Sire DAC.

The best a reviewer can hope for is to measure a product against his own references, for one, and to measure the product against the designer's goals, for another. For me, the Da Vinci is clearly, unambiguously, and obviously superior to my personal references. At the risk of putting words in Larry's mouth, I can still recall the giddy grin he wore the first time I met him, way back. He's so obviously proud of what he's put together that seeing his concepts made so stunningly, menacingly real has got to be a win. Of course, now that the Sire has been announced, I do wonder what it is he thinks he can do to top Da Vinci.

Anyway, if you want to take that as a measure of value, then so be it. But I won't ever be able to tell you if the Da Vinci is "worth it"

without a tediously detailed reference to my own, very personal calculus, and even if I could articulate that for you, the chances of it being meaningfully comprehended are slim. All I can offer is this: The Da Vinci DAC is an outstanding performer and I absolutely loved using it. When it left, it caused physical pain and a significant period of psychological withdrawal. I am still unhappy it is not here. And adding insult to injury, the Lotto Fairy is still not taking my calls.

Twenty-thousand dollars is a ton of dough. And that is for a product that has already been leapfrogged by the industry. DSD, like it or lump it, is the latest and stickiest buzzword, and any DAC that doesn't support it requires significant justification. The Da Vinci does not support DSD and will not. Which may be a problem. Of course, it may not be, and I know that many of my colleagues are adamant that the format is as irrelevant as SACD was—and for exactly the same reasons. Me? I'm agnostic. I have several hundred DSD albums, but admittedly I'm weird. If you do not see yourself as ever needing or wanting to explore that format, then awesome.

The Da Vinci, then, occupies that rarest of the rare when it comes to high-end audio—the Exit Ramp. It may be that there is better, or more refined, or whatever, when it comes to digital-to-analog conversion, but I'm really not sure I'd bother as beyond this point on the price/performance curve jumps become so incremental that value becomes an entirely alien notion. For me, the Da Vinci marks new territory and is my new high-water mark for what can be done, at least with non-DSD source material. In short, I've never heard better. Very highly recommended. tas

Q&A With Light Harmonic Designer, Larry Ho



SH: I like to think that designers are always chasing some kind of "formative sound"—something that shaped their preferences and guided the products they created and influenced. What was your "formative sound?"

LH I am a long-time vinyl and SET amp lover, which may "bias" my sound preferences somewhat. There are a few influential factors that I focus upon when I design or listen to designs by others, and some of those factors have a direct contribution to Light Harmonic's "house sound." First of all, I tried not to use the most-commonly used digital conversion or filter circuits, because I believe they are one of the major factors leading to listening fatigue and harsh high-frequency sound. Also, I believe in the formula "timing perfect x bit perfect = perfect digital music playback." To control the best timing, reduce jitter, and maintain the bit perfection from the beginning to the end of playback chain is a hard and challenging process. When we approach that ideal state, I think almost everybody would agree that the music is more natural and vivid.

How did you get into high-end audio? That is, how did it become a career and more than an interest or hobby? And what were you doing before that?

LH I'm a serial entrepreneur. I started my first Internet eCommerce company at age 22, then an on-line music-streaming company in 2005.

As a hobby (which turned first into an obsession, and then, a business) I've designed my own circuits and firmware/software

related to high-end audio for the last 15 years. In 2008, while working with Intel on a potential high-definition PC-audio project, my team demonstrated the highest sampling rate USB2 audio-to-SDPIF converter at CES 2009 (it's 24/192k). Then in 2010, I decided to form an expert team dedicated to ultra-high-performance digital playback.

Is this your first venture in audio's high end?

LH Yes!

Tell me how the Da Vinci DAC came about? What got the ball rolling?

H After we finished the 192k SPDIF converter in 2009, I wanted to listen to high-definition audio directly, rather than convert it into a SPDIF interface. Going through the SPDIF interface, the signal is more vulnerable to jitter. It came down to the question, "Where can I find a good 192k/24 DAC?" At that time, even the most advanced DACs on the market could only do 96k. After some research, I found one professor's article that said that, in his calculation, if we need to duplicate 100% of the music information from studio mastertape (or vinyl), we need to have at least a 550k sampling rate or more. The next day I thought, "Why not challenge myself to see how high digital audio could go?" Then I called in one of my good friends from Germany; I worked on the digital part, he worked on the analog part. Finally, we designed the power supply together.

At the end of 2010, after testing for more than 12 months, I decide to raise the bar and make the first 384k/32-bit DAC, and call it the Da Vinci DAC. Beta launched at AXPONA in Atlanta, on Leonardo Da Vinci's birthday, April 15, 2011.

Can you discuss the importance of the power supply (regulation, filtration, delivery, etc.) in your design?

LH This is a major topic. After much trial and error, I've concluded the following: (a) the power supply chain is within the music signal path—it's a part of the whole, and can not be separated; (b) shunt regulation always conducts current and has tremendously fast transient response, so generally, though not so "green," the sound is better.

We at Light Harmonic believe that multiple, local, in-place, in-time regulators, plus massive filtration is the best way to do it. In the end, our Da Vinci Dual DAC has more than 55 regulators! The only drawback is that it runs hot, almost like a Class A amp.

I think everyone is familiar with jitter—and why it's bad for digital audio. How do you address it in your design? Can you talk about the three-layer buffer and how that's implemented? What clocks are you using in the design?

LH Jitter is the enemy of perfect timing. One important thing to remember is that the digital clock signal is "analog!" Timing is analog! So how to fight jitter is not simply a O and 1 thing. Our patent-pending 3L (three-layer)

buffer-design ensures the best results in the audio industry. It needs 10+ technical pages to describe in detail, but in summary we use the first layer (L1) to deal with computer-speed fluctuation—which there is a 100% chance will happen. The third layer (L3) couples with core DAC conversion and core clocks. We add the intermediate layer (L2) to further reduce the coupling between L1 and L3, so that each layer can run at its optimal speed and size. And yes, these buffers are elastic.

Also, beside the 3L buffer, we have three dedicated clocks inside the Da Vinci. Because we don't use a PLL, we dedicate one clock to the USB interface. One best clock for 44.1k/88.2k~352.8k. One best clock for 48k/96k~384k. One thing needs to be emphasized: There is no integer multiple of 44.1k and 48k-based frequencies; therefore, the best solution is to have two dedicated clocks for audio.

Why is NOS (non-oversampling) important? Why does avoiding it matter so much?

LH Although a lot of "old-school" digital designers still don't agree, a lot of music lovers and audiophiles have already discovered the benefit of NOS. Even the use of slow-slope digital filters will make music better. Why? There are many good reasons. The most important one should be: Non-over-sampling won't destroy the bit-perfect presentation. Also, if no digital filter is used, it won't create the post-ring or pre-ring, which will "blur" the perception of phase and sound image position.

Ask around. There is no benefit to oversampling music with an original native sampling rate of more than 96k.

So our solution is simple: Don't do oversampling or upsampling and digital filtration at all. If users really want it for low-resolution music (e.g. 44.1k or 48k), then they can use the best software player to do that in the computer.

Can you talk about the chassis design? Why that shape?

LH Clarification: We never thought about anything related to Darth Vader, *Star Trek*, or *Star Wars*, although I'm a big fan of these films. One thing we truly did want to simulate was the "stealth fighter." The benefit? There is little direct-reflection soundwave surface and less vibration, resulting in a clearer soundstage.

The chassis is definitely distinctive, but some might say that the cost of the DAC is derived in large part from that design. Couldn't it have been done much cheaper in a simpler box, or even two boxes?

Da Vinci is our first product, and cost was really not our priority in the beginning. But I got tons of feedback that we should use the "easier" approach. I think we could take an easier route someday, in another product. But we would keep our philosophy there: Less direct sound-reflection surface, and a futuristic look.

One word about using two boxes: The noise

shielding in two boxes is not easy to implement, if you want the best. So Da Vinci's innovative design is "two boxes in one chassis." And the additional benefit is the upper "box" can rotate if you want.

The DSD option adds significantly to the cost of the DAC, but recent discussions from other designers have indicated that including DSD at all means a series of compromises. How have you addressed the challenges of adding DSD to the Dual DAC?

LH DSD in many ways is 100% the opposite of PCM encoding/decoding. But DSD has its potential. Just listen to it, and you will know. It's not a compromise—at least, not when it's played back through our Dual DAC. Sometimes, I'd dare to say, it seems more like vinyl.

The beautiful thing about our Dual DAC is you can have the best of two worlds. But the ugly fact is, we needed to add a completely new decoding engine, new power supply, new analog stage, new everything. The only thing shared are the chassis, OLED screen, and USB front end. Nothing else.

The funny thing is if anyone wants to buy a "DSD-only" DAC, I can customize one for him in few days. Just pull the PCM engine and related power/analog board out. Then it's done.

As far as I know, we are the only people in this industry to use this extremist approach but it's vitally necessary to achieve the best from both PCM and DSD.

The DAC seems to work best via USB. In fact, there



are many features (like volume control) that are only available via that interface. So, why include others? And why not design-in an on-DAC volume control knob or button?

LH AES/EBU and SPDIF have their legacy value. The first prototype of Da Vinci had a USB interface only. Then soon after that, we added the best AES/EBU and SPDIF approach. Some of our customers say they drive the Da Vinci through the SPDIF input. They still don't use their computer that much.

Why no vacuum tubes?

LH Well... you never know. I designed tube gear for myself, a long time ago.

The challenge of a tube DAC is we need to get very good tubes at the start, so the S/N ratio and dynamic range won't be compromised.

Now I am a serious tube collector, and I just got a bunch of what may be the best signal tube ever produced, the Telefunken C3g. If I really have some good free time next year, maybe... tas

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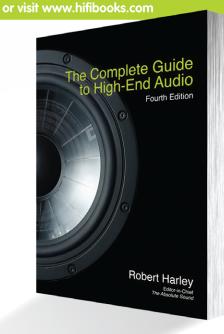
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PS Audio PerfectWave DirectStream DAC

Not Just Another DAC

Vade Forrester

n 2012, a group of Direct Stream Digital (DSD) enthusiasts within the audio industry (led by dCS) released the DSD-over-PCM (DoP) standard, providing a way to "trick" traditional PCM audio circuits into playing native DSD files on suitably-equipped servers and DACs without first converting them to PCM. Today, few new DACs dare venture into the marketplace without DSD capability. Designing a DSD DAC has become almost routine; just order up some DAC chips capable of handling DoP inputs and use them in standard digital and analog circuits.

There is, however, always someone who wants to toss conventional practices over the side and start from scratch. In the case of PS Audio's new DirectStream DAC, the original thinker is a computer scientist/audiophile named Ted Smith. A strong admirer of DSD recordings, Ted felt that the ideal DAC should

work entirely in the DSD domain, so he set out to build a DAC which was entirely DSD-based. Of course, it had to accept PCM files; there are too many of those (e. g., CDs) to ignore, but they would be internally converted to DSD upon input. On his own, Ted took seven years to design a working example of his DSD DAC. Almost by chance, his design came to the attention of PS Audio's CEO Paul McGowan, who liked what he heard (a lot), ultimately resulting in the subject of this review.

What's so great about DSD anyhow? Here are some of the advantages PS Audio cites for DSD playback:

- DSD is simple to convert to analog: just run it through a low-pass filter. It doesn't need a brick-wall filter like PCM, which can affect the sound. The DirectStream DAC uses a 24dB-per-octave low-pass filter, the same as you'll find in many loudspeaker crossover networks. [In theory, a DSD bitstream can be converted to analog with a single capacitor.—RH]
- DSD is inherently linear; it's hard to build a PCM DAC that always takes the samesized step in the output for any possible unit increment of the representative PCM voltage value.
- DSD soft clips when overdriven, more like magnetic tape.
- All bits in a DSD stream have the same weight; a single-bit error anywhere is barely measurable, let alone audible.

PS Audio describes the DirectStream DAC's circuit as follows: All incoming data, PCM and DSD, are upsampled to 30 bits running at ten times the standard DSD rate and then back down again to double DSD for noise-shaping. The ten-times DSD sampling rate was the lowest common rate attainable through integer upsampling of 176.4 and 192kHz PCM files. An internal volume control maintains complete precision. Except for the sigma-delta modulation process itself there is no rounding; a full 50 bits are used. Consequently, there is

no degradation from using the digital volume control. After the volume control, the signal is converted to DSD and downsampled to double-speed DSD (often referred to as DSD128). The double-speed DSD rate allows the low-pass filter to begin rolling off at 80kHz. A higher output rate would would have increased jitter.

There is no conventional analog output stage. The output of the DSD engine is fed directly into high-speed, high-voltage, high-current symmetrical video amplifiers and from there into the passive output filter. Rather than use an active output section, a balanced wideband transformer (which is part of the low-pass filter) drives either balanced or unbalanced interconnect cables. The output impedance is 100 ohms (unbalanced)/200 ohms (balanced), which should drive any reasonable load.

Off-the-shelf DAC chips can't perform the functions described above, so Ted used a Xilinx Spartan 6 field-programmable gate array (FPGA) chip instead. An FPGA is a computer chip that's a blank slate; you can program it to do whatever you want, and that's what Ted did. A single master clock is used, but it's unrelated to the input sampling rate.

I don't usually spend this much space describing the design and functions of gear I review, but since the DirectStream DAC is such an innovative design I thought it would be worthwhile; if you're interested in learning more about it, I highly recommend a visit to PS Audio's Web site. Suffice it to say that Ted Smith has completely rethought how a DAC should operate and has designed a unique and innovative DAC.

The \$5995 DirectStream DAC replaces PS Audio's PerfectWave DAC. The two DACs are

EQUIPMENT REVIEW - PS Audio PerfectWave DirectStream DAC

virtually identical, and that's not an accident; PS Audio has a program for updating existing PerfectWave DACs by converting them to DirectStream DACs. One of the options for doing that involves gutting the PerfectWave DAC and replacing it with the DirectStream DAC's interior parts. Apparently that's easy enough that an owner can do it, but since some owners won't feel inclined to take on that project, PS Audio has other options for updating the PerfectWave DAC. See the PS Audio Web site for details and pricing.

Like the PerfectWave DAC, a DirectStream DAC is a black or silver-gray box that measures 17" x 4" x 14" and weighs 19 pounds. Its finegrained metallic chassis has rounded corners, a color touchscreen towards the right end of the front panel, and a high-density fiberboard top panel finished in piano-gloss black. Its elegant and refined styling would look right at home alongside the fanciest components. In a nutshell, I'd describe its looks as classy. A plastic remote control is included. Some manufacturers provide remote controls hewn from ingots of solid metal, but the first time you drop one of those on your coffee table (or your foot), you'll really appreciate a light plastic remote-don't ask me how I know this. The PerfectWave Media Bridge, an optional expansion card that plugs into the PerfectWave DAC and enables you to connect it to a network, also works with the DirectStream DAC. The PerfectWave Transport, an advanced optical drive in an enclosure stylistically and dimensionally identical to the DirectStream DAC, is still available and makes a natural match with the DirectStream DAC. In other words, PS Audio has bent over backwards to protect the investment its customers have made in other PS Audio equipment.

Like the PerfectWave DAC, the new DirectStream DAC provides a wide variety of digital inputs: SPDIF on coaxial RCA and TosLink inputs, USB, AES/EBU on an XLR connector, and two I²S inputs on HDMI connectors. Although HDMI connectors are used for I²S inputs, these inputs don't carry HDMI video signals. Interestingly, while all the inputs will accept DoP-encoded signals, the I²S inputs will accept raw DSD signals direct, without DoP encoding. One raw DSD source is PS Audio's NuWave Phono Converter, which combines a phono preamp with a high-resolution PCM and DSD analog-to-digital recorder.

The color touchscreen on the front of the DirectStream DAC allows you to control most of its functions, duplicating the remote control; however, the remote control operates other items PS Audio manufactures, like the PerfectWave Transport, so it has a lot of buttons unrelated to the DAC. If, like me, you're suffering from remote control overload, it's quite convenient to be able to control all your PS Audio gear with a single remote.

Starting at the left end of the rear panel, there's the IEC input for AC power and the on/ off switch. To the right, towards the bottom of the panel, is the horizontal slot for the PerfectWave Bridge expansion card and an opening for an SD memory card. About halfway across, the rear panel is divided into two sections: input and output. The bottom section is the output section, where the XLR

and RCA output jacks are located. In the top section, you'll find the input jacks.

Setting Up and Using the DirectStream DAC

Although the DirectStream DAC can drive an amplifier directly, PS Audio recommends you not use both output jacks simultaneously. Because I use a subwoofer with my main speakers, I plugged the DAC's output into my Audio Research SP20 preamp, which

will drive my main speakers and subwoofer. Digital sources plugged into the DirectStream DAC included my HP laptop computer running Windows 7 and J. River Media Center 19, an Auraliti PK100 PCM and DSD File Player with its optional linear power supply, and a Meridian 500 CD transport. Hold on, a transport? Isn't that kind of Stone Age? Well, I still use a transport to play CDs inappropriate to rip, like those borrowed from my local library or from visitors. I also had the use of a PS Audio NuWave Phono Converter (reviewed by Anthony Cordesman in Issue 241), which when connected to the DirectStream DAC via the I2S connection, passes raw DSD formatted music converted from an LP. Music files used by both J. River and the Auraliti were stored on a NetGear ReadyNAS networkattached storage drive connected by an Ethernet cable through my home router to either server. The HP laptop was connected to the DirectStream DAC by Wireworld Platinum Starlight USB and AudioQuest Diamond USB cables (not at the same time), the Auraliti server used a Wireworld Gold Starlight 6 SPDIF cable, while the Meridian transport used a Wireworld Gold Starlight 5 AES/EBU cable. PS Audio includes a heavy-duty power cord, but a better cord should produce better sound, so I used an Audience Power Chord e cord. Clarity Cables Organic balanced interconnects connected the DirectStream DAC to my preamp. The manual

SPECS & PRICING

Converter type: Field
Programmable Gate Array
custom-programmed to serve
as DAC

Sample rate (PCM): 44.1kHz, 48.0kHz, 88.2kHz, 96.0kHz, 176.4kHz, 192kHz

Word length (PCM): 16, 18, 20, 24 bits

Data rate (DSD): Standard (2.8MHz) or Double (5.6MHz) DSD on PCM on all inputs as well as raw DSD on I2S inputs Synchronous upsampling, all

inputs: 28.224MHz
Analog conversion method:
Delta-Sigma, double-rate DSD
Output levels: "Low"

setting, 140mV RMS unbalanced/280mV RMS balanced; "High" setting, 1.4V RMS unbalanced/2.8V RMS Digital inputs: I2S(2), coax, XLR, TosLink, USB, Network Bridge slot Balanced outputs: One stereo pair on XLR connector

Unbalanced outputs: One stereo pair on RCA connector

Dimensions: 17" x 4" x 14" Weight: 22 lbs.

Price: \$5995; \$3995 with trade-in of PerfectWave DAC

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EQUIPMENT REVIEW - PS Audio PerfectWave DirectStream DAC

recommended plugging the DirectStream DAC into one of PS Audio's Power Plant power centers, but lacking one of those, I plugged it into an Audience aR6-T power conditioner.

It was easy to install the driver software necessary for the DirectStream DAC to work with Windows. However, as with any driver installation, a few basic computer skills are required: extracting files from a ZIP file and running the SETUP.EXE file. Once the driver was installed, I had to adjust the settings on my music server program J. River, so it would use the new driver. That too, was simple—if you're comfortable with J. River. My Auraliti server, being a Linux computer, didn't require a driver for USB, and no drivers are required for SPDIF connections.

PS Audio suggested breaking in the DirectStream DAC for two weeks. That's a good starting point, but actually, the DAC continued to break in for two months, running almost 24/7. I noticed that the highs, which were initially a little edgy, became smoother and sweeter, the bass more extended, and the overall sound more spacious. If you audition a DirectStream DAC, be sure it's well broken-in.

Sound

PS Audio claims the DirectStream DAC "uncovers all the missing information hiding in your digital audio media for all these years." That's a pretty tall claim; is it for real, or just hype? I'll let my listening buddy Carl answer that. When he entered the listening room where the DirectStream DAC was playing, he stopped, listened intently, and said, "That's a lot of detail!" And that was before I introduced

him to the DAC. Carl is pretty familiar with my system and room, so the fact that he noticed the increased detail before I even pointed out the DAC is a genuine testament to the validity of PS Audio's claim.

I had wondered if the additional information the DirectStream DAC claims to retrieve from a digital recording would be easy to hear, or would be subtle low-level information that I'd have to strain to discern. Well, the answer was: both. The first thing I noticed about the DirectStream DAC's sonic characteristics was its ability to capture a sense of space. Even recordings that had seemed a bit flat had some air around them, and those with already well-defined soundstages had those soundstages more precisely defined, with more information about the recording venue.

The DirectStream DAC also captured more mechanical detail, more information about the physical process of playing back music. That includes a variety of things, for example, the noises a guitar makes when it's playing music. And I could hear more clearly how a vocalist articulated words and phrases. In addition to the physical details, the DirecStream DAC captured a ton of harmonic detail that made instruments and voices seem more realistic, instead of cardboard imitations of instruments. If the recording contained lots of harmonic details. I could hear those reproduced in accurate proportions. Indeed, after the DAC was broken in, I'd describe its sound as sweet and relaxed, so there's no need to worry that you'll hear unpleasant threadbare sound. But wait, there's more: The DirectStream DAC also captured lots of information about dynamic contrasts—both macrodynamic and microdynamic. Finally, if the recording permitted, the DirectStream DAC put all the musical information into context, so it was easy to hear how the all those parameters—detail, harmonics, dynamics, and space—related to each other to portray a coherent musical event. It didn't just tell you how a violin sounded; it also told you how it sounded relative to the rest of the orchestra. The DAC was able to organize the information it retrieved to make its presentation more like a musical performance.

Does this sound like more insane reviewer ravings? I can certainly see how it might, so let me cite a few musical examples that led me to these conclusions. I made an effort to listen to old favorite recordings ripped from CDs, as well as newer high-resolution releases. I gueued up Chris Jones' "God Moves on the Water" from his CD Roadhouses and Automobiles [Stockfish] ripped to an AIFF file. The first thing I noticed was the subterranean bass this track is noted for presented with lots of detail and power. Then I observed that I heard more extraneous guitar sounds. Guitar harmonics were abundant. Jones' gravelly voice seemed unusually well fleshed-out harmonically. A visiting audiophile (not Carl) remarked that this track sounded like a highresolution recording-not a bad start for a listening session.

Another demonstration of how much information the DirectStream DAC could retrieve came when I queued up the cut "Folia Rodrigo Martinez" from Jordi Savall's CD *La Folia 1490-1701* [Alia Vox] ripped to an AIFF

file. The percussion instruments are quite vivid when played loudly, but often tend to fade into a background noise when played quietly. Or at least that's what I used to think; the DirectStream DAC raised them above the noise level and made them audible at all times. The clack of the wood blocks was clearly audible throughout the entire piece, as was the clatter of the castanets. The DirectStream DAC also generated a wider, more stable soundstage than I usually hear from this piece. The dynamic level is constantly changing, and the DirectStream DAC made it clear how band leader/viola da gamba player Savall was driving those changes. There was unusually precise placement of instruments in the soundstage, as well. The Direct Stream DAC gave my subwoofer a good workout as it delivered a deep extension of the bass drum, with lots of power and impact I could feel as well as hear.

The Tallis Scholars' recording Allegri's Miserere & Palestrina's Missa Papae Marcelli [Gimell] was recorded in a spacious church. On the cut "Miserere," a main choral group is up front in the soundstage, while a small solo group is further back in the room. A good system makes it obvious that the two groups are spatially separate, and decent DACs will give the impression of how far they are apart. The DirectStream DAC revealed not only that, but also gave a spatial picture of the recording venue. Likewise, while singers in the main choral group weren't exactly pinpointed, due to reverberation, their relative location was well-defined, as were their vocal characteristics. The vocalists weren't

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portrayed as a homogeneous blob, as they sometimes are with other DACs.

I think there's an unwritten rule that reviewers have to mention at least one female vocal performance in every review. So up next was Rebecca Pigeon's audiophile fave "Spanish Harlem," from her album *The Raven*. I had two recorded versions of the cut, an 88.2/24 FLAC and a 176.4/24 FLAC which had been remastered by Bob Katz. The DirectStream DAC made the differences between the two recordings easy to distinguish—the 176.4kHz recording sounded less mechanical and edgy, making Pigeon's voice quite human-sounding. I felt like I could hear how she vocalized each word.

Okay, we've established that the DirectStream DAC plays CDs and PCM material guite well, but does it do as well on DSD recordings? To find out, I switched to a DSD recording: Alex de Grassi's album Special Event 19 [Blue Coast Records]. Playing the cut "St. James Infirmary," the DirectStream DAC captured more detail about guitar than I thought was possible. Starting with initial transients, the DAC reproduced the pluck of the strings sharply but with the resolution that told me when each string had been plucked. In the sustain part of the note, each note displayed its full harmonic characteristics, and then decayed off into silence, guivering in space for several seconds. The DAC caught de Grassi's phrasing perfectly, giving the piece a bluesy tinge. While each note was individually captured in textbook fashion, they all blended together to form an organic musical whole, a song with a touch of swing. I don't think I've ever heard a better rendition of someone playing a guitar. I've never heard a DSD DAC play the cut with such abundant musical detail, either.

To see if the DirectStream DAC would fall apart playing a recording of a full orchestra, I queued up Michael Tilson Thomas conducting the San Francisco Symphony in Mahler's Third Symphony [SFS Media/Downloads NOW!]. The SFS Media DSD recordings of the Mahler symphonies may well be the most realistic orchestral recordings I've ever heard. The result: rich, accurate harmonics, well-defined spatial environment, dynamic changes ranging from barely perceptible to hammering blows that threatened my speakers' well-being-and the breathtaking performance didn't hurt, either. Instruments sounded spookily realistic and were scaled to create a believable impression of a large symphony orchestra. The DirectStream DAC played the recording effortlessly; the passive output stage never sounded strained or congested. After we'd listened to the Mahler Third, another audiophile buddy commented: "It really doesn't sound digital anymore." He's never said that about any other DAC

The above impressions were derived using a preamp between the DirectStream DAC and my power amp and subwoofer, for reasons I've explained. But I wanted to test PS Audio's claim that the DirectStream DAC will drive amplifiers directly, so I disconnected my subwoofer and used the DirectStream DAC to drive the power amplifier only. As I expected, the direct-drive mode yielded a slightly cleaner, more delicate sound, with even more spaciousness. Of course, absent a subwoofer, bass didn't extend as deeply. But if you only have one power amplifier to drive, I'd go with the direct-drive connection.

Comparison

My Audio Research DAC8 is a PCM-only DAC, so I could only compare it to the DirectStream DAC using PCM files. It's still in Audio Research's product line, selling for \$4995, not far from the price of the DirectStream DAC. I acquired the DAC8 back in 2010, but although I've reviewed several excellent-sounding

DACs, I haven't yet been motivated to replace it. Or have I?

On "Folia Rodrigo Martinez," instrumental detail was less distinct, and the percussion instruments tended to blend together in the background a bit. However, the dynamic contrasts and shifts which are so important to this performance were as forceful as with the Direct-Stream DAC. Instruments were well fleshed out harmonically, although they sounded just a tiny bit raw compared to the DirectStream DAC. As is usual, bass was very extended and powerful, one of the characteristics the DAC8 is known for. The DirectStream DAC's bass power and extension seemed every bit as powerful as the Audio Research DAC8. No other DAC has ever achieved that.

The "Miserere" cut sounded very spacious, but the details of the soundstage, the reverberant space, seemed a bit less distinct than through the DirectStream DAC. The tenor soloist in the main choral group sounded a bit grainier than through the DirectStream DAC. The DAC8's rendition was still well-defined and enjoyable, but the DirectStream DAC's version was better focused and smoother by a tiny margin.

EQUIPMENT REVIEW - PS Audio PerfectWave DirectStream DAC

Rebecca Pigeon sounded very good through both DACs. In "Spanish Harlem," the differences between the 88.2kHz and 176.4kHz versions were still discernible, but a bit easier to recognize through the DirectStream DAC.

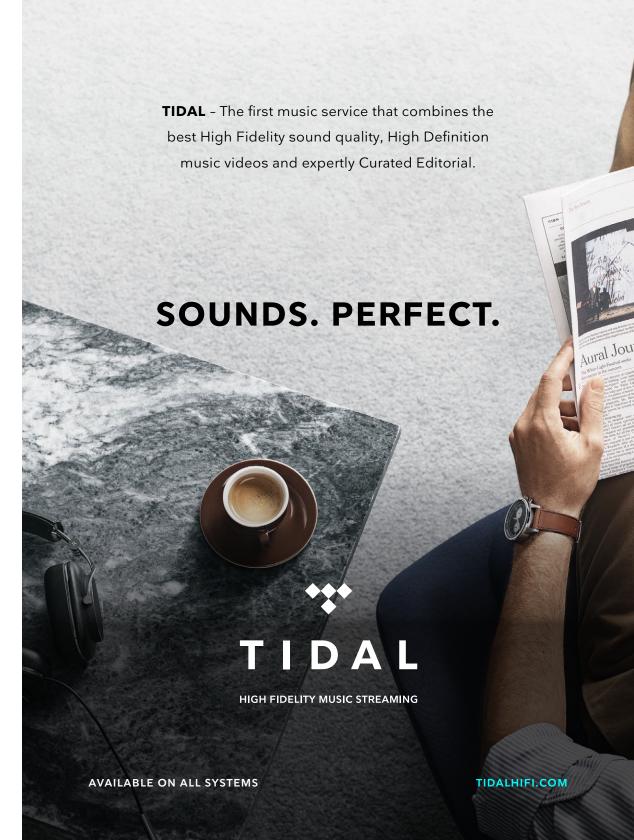
Audio Research's DAC8 is obviously blessed with a very good analog section; however, the DirectStream DAC's passive output section was a bit more refined-something I wasn't expecting.

Bottom Line

In this review, I've explored the performance of the DirectStream DAC and compared it to another DAC of roughly similar price. Now it's time to answer three important questions: 1) Does the DirectStream DAC live up to its claim of revealing hitherto hidden details in your CDs; 2) if the answer to the first question is yes, how much of an improvement in sound does the DirectStream DAC make; and 3) is it worth its price? Answer No. 1: Based on my listening sessions. I'd have to say that the DirectStream DAC does indeed retrieve more information from my recordings, from CD to the highestresolution recordings, than I had heard from other DACs. Answer No. 2: The differences in sound were perceptible, and contrary to my expectations, not really subtle. The effect of a lot of previously unheard information being added to previously audible information was often surprising. On the other hand, I wasn't surprised to learn that extracting more information from a recording is not always the same as making the recording sound better. Several times during the review period, I discovered that sometimes the DirectStream DAC made some recordings sound more obviously mediocre. As an audiophile, I suppose that's good; but as a music lover, sometimes less detail may actually be a benefit. An unexpected advantage, though, was that I learned that quite a few CDs and rips sounded better than I realized; so for wellrecorded material, it elevated the playback quality quite noticeably. I guess that's all you could reasonably expect. Now for the hardest question-answer No. 3: This answer depends to some extent on personal preference. Although my memory of other DACs has faded with time, I can't remember any DAC that impressed me as much with its overall sound quality as the DirectStream DAC. So my answer to third answer would be yes. Of course, your mileage may vary. Obviously, any purchasing recommendation must take into account your financial situation. A price of \$5995 is pretty substantial, but I don't know of another DAC at or below that price that sounds as good.

Whether you're a rabid DSD fan, or have strong convictions that PCM is the only way to go, PS Audio's goal for the DirectStream DAC was to make both types of recordings sound as good as possible. My personal take is that it substantially realizes that goal. I highly recommend putting the PS Audio DirectStream DAC on your must-audition list if you're considering purchasing a DAC in its price range—or even if you're willing to spend more, even a lot more. It's easily the best DAC I've heard in my system, making digitally-recorded music sound better than I've ever heard it.

Bravo, Ted and Paul. tas





Ayon Stealth DAC/Preamplifier

Long-Term Satisfaction

Dick Olsher

hat's in a model name? Well, sometimes not much at all when it's merely an arbitrary numerical designation and a potential source of confusion for my slightly dyslexic mind. On the other hand, Ayon's choice of model name strikes me as perfectly befitting a DAC, implying an unobtrusive delivery of the musical message by a digital device. The Stealth does just that by following a road less traveled. To be clear, the Stealth is more than just a DAC. Think line-level preamp offering two line-level inputs, an electronic volume control, as well as a world-class DAC.

Let's start by taking a peak at the analog side of this DAC/pre. The massive aluminum chassis hides several vacuum tubes which are allowed to "breath" through a number of small grilles in the top plate and sides. Ayon is a firm believer in tube technology, and in particular triodes, for reasons that have to do with inherent linearity under minimal- or zero-feedback conditions, benign distortion and overload characteristics, and capacity for simple circuitry. The analog output stage is amplified in balanced fashion to allow for both RCA and XLR analog outputs. Each signal leg consists of a Russian 6H3O dual triode, connected in parallel, and operated

as a pure, no-feedback, Class A voltage amplifier. That's as purist as it gets, and Ayon is proud of the fact that there are only five components in each 6H3O signal path. This is a great-sounding tube, but if and when it fails, replacement as I discovered is a tedious task. Several weeks into the review process, one of the left channel 6H3Os went bad. To remove the top panel requires loosening ten Allen-head screws through access holes in the bottom panel. As compensation, you get to ogle the beautiful hand-assembled internals.

All filament supplies are DC, and voltage levels are regulated. There is also an automatic tube soft-start during each turn-on cy-

cle. Voltage gain can be toggled on the back panel between low-gain (2.5V) and high-gain (5V) settings. The power supply features two power transformers and a choke-coupled filter bank. I was surprised to learn that the Stealth is tube rectified. A quartet of Chinese 6Z4 rectifier tubes is connected as a bridge rectifier. This is not the cheapest scheme to implement, and the question that comes to mind is why tube rectifiers in a digital device? The answer, according to Ayon, is because this approach sounds more dynamic than any other means of rectification. That alone should inform you about Ayon's commitment to the best technical solutions irrespective of cost.

A Cirrus Logic CS8416 192kHz receiver chip decodes incoming PCM digital data from the coaxial, BNC, TosLink, I2S, and AES/EBU inputs. An XMOS asynchronous USB input accommodates up to 24/192kHz PCM data. Indicators on the right side of the display show the incoming PCM frequency rate: 44.1, 48, 88.2, 96, 176.4, or 192kHz. Incoming SPDIF data can be upsampled to 24/192kHz. This feature can be "switched" on and off on the remote control. All samplerate conversion is performed by a Burr-Brown SRC4193 asynchronous sample-rate converter timed by an external crystal clock oscillator operating at 24.575MHz. DSD inputs include two BNC (DSD-L and DSD-R) digital inputs and one BNC for word clock. In addition, the Stealth is also outfitted with one optional I2S DSD-over-PCM (DoP) digital input. I'm not that sophisticated a digital user. Hence, the scope of the review was limited to PCM data inputs, either PCM digital out from a Sony XA-5400ES SACD player or USB input from my MacBook Pro computer running Amarra V2.6 software.

The DAC chipset is the highly regarded ESS Sabre ES9018, a 32-bit audio DAC that features ultra-low distortion levels, a timedomain jitter eliminator, and integrated DSP functions. Customizable filter characteristics allow for user-programmable filters with custom roll-off slopes. The Filter button on the remote allows selection of either Filter 1 (slow roll-off) or Filter 2 (fast roll-off). Filter 1 is said to sound a bit smoother than Filter 2, which is characterized by Ayon as being a bit more analytical. I suspect that your ultimate preference would be system-dependent; however in the context of the ENIGMAcoustics Mythology M1 loudspeaker, I had a clear preference for Filter 2. I found it to be simply more resolving of microdynamic nuances and more incisive spatially. Filter 1 did sound smoother but at the cost of homogenizing spatial outlines and dissipating some dynamic tension. I also had a clear preference for upsampling 44.1kHz PCM data to 24/192kHz. Taken together, the resulting presentation was not only more spacious and better focused, but also dug a bit more deeply into the mix and retrieved more ambient information. It's easy enough to click these buttons on the remote and experiment in real time to determine what sounds best in your system.

The volume control is based on the Texas Instruments (Burr-Brown Division) PGA2320 IC, a digitally controlled analog volume control designed specifically for professional and high-end consumer-audio applications. Internal operational amplifier stages are used to generate an attenuation/gain range of -95.5dB to 31.5dB. According to Ayon, this control is strictly used for analog

EQUIPMENT REVIEW - Ayon Stealth

signal attenuation with all of the gain being developed in the tube linestage. Thus, the display volume range is from 0 (max volume) to -60 (minimum volume) in 1dB steps. There is also a built-in balance function accessible from the remote that allows left-right channel attenuation in 1dB steps up to a total of 6dB.

I've run the Stealth directly into a power amp and have found its volume control to exhibit low coloration levels. I love its functionality; nevertheless, the question of how the PGA2320 compares to a conventional resistive potentiometer deserved an answer. It turns out that the Stealth's volume control can be bypassed by switching over to Fixed Volume mode on the remote control, in which case the audio signal goes directly to the analog output stage. I connected the Stealth to the PrimaLuna DiaLogue Premium line preamplifier with its conventional Alps Blue Velvet motorized conductive-plastic potentiometer. I was then able to switch between Fixed and Variable volume modes on the Stealth and at equal volume levels try to discern any sonic difference with the PGA2320 in and out of the signal path. Although not necessarily a definite test, I did prefer listening with the Stealth in Fixed (2.5V) volume mode. There were improvements in spatial presentation (depth perspective and image outline separation). Additionally, the treble range was a bit purer sounding. These results suggest that the Stealth's volume control does introduce a slight solid-state sonic signature.

It didn't take me long to realize that I was in the company of a superb DAC, one that was free from annoying digital artifacts and that could flesh out tone colors with startling realism. One of the hardest things for a DAC to get right is violin overtones, especially on recordings that are less than perfect. Many DACs don't react well to recording hot spots adding spurious grain, hash, and even sizzle to the reproduction. Not a pleasant phenomenon and one that had soured me on digital sound for many years. By contrast, the Stealth sailed right through recordings that had given me trouble in the past. Solo violin's upper range was reproduced with proper levels of sheen and textural purity. The sound of massed strings, and in particular that of violins, the most numerous orchestral string instruments. represents another difficult challenge for any digital device, and one in which most CD players have scored poorly. The sound of massed violins should be layered and the spatial impression ought to float like a feather within the confines of the soundstage with plenty of tonal warmth and textural purity. And in these respects, the Stealth was able to put a smile on my face. Its performance at the frequency extremes bettered that of all previous DACs I've lived with. including the EAR-Yoshino DACute I reviewed in Issue 238. Treble transients were exquisitely refined and bass lines left nothing to be desired, being well defined and pitch-perfect.

The Stealth wasn't just about tonal color fidelity and textural refinement. It lit a fire under the soundstage. Musical lines boogied with passion and drama. The dynamic range from loud to very loud was reproduced without hesitation or compression. A recording's ambient information was readily discernible, as was low-level detail often fuzzed over by lesser DACs. I'm fussy about soprano voice and like to discern vibrato

to the point of being able to count the number of pitch modulations per second. Not many divas out there can hit the ideal of seven cycles per second. With the Stealth in your system you'll be able to resolve this sort of detail.

After auditioning numerous DACs over the years I have come to realize that a large slice of the sonic difference between individual units could be ascribed to the analog stage,

and specifically tube versus solidstate designs. A tubed output stage appeared to present the soundstage more dimensionally. No matter how sophisticated the digital circuitry was, when mated to an op-amp buffer or gain stage, image outlines pancaked and depth perspective took a hit. And then there was the matter of textural grain. That's the stuff that my auditory system generally finds to be indigestible. Some solid-state buffers had achieved a respectable level of smoothness, but it usually came at the cost of dynamic sterility. It seems to me that the best approach to defanging a DAC is to introduce tubes into the mix as early as possible. And that's exactly what Ayon has done with the Stealth. It is perhaps a tad richer harmonically than the real thing, but it is far removed tonally from the sort of romantic, overly lush presentation that has been dubbed "vintage tube sound." The Stealth is about tonal accuracy, but what you think of it will depend greatly on the associated amplifier and speakers.

Mate it with reference-caliber gear and it will walk the line of neutrality.

In the pursuit of digital playback perfection, the Stealth ranks in the DAC elite. It's all about the music and the Stealth delivers a superb musical experience. You could do a lot worse at a higher price point, but I doubt that you would do any better at its asking price. It's a DAC that I could happily live with for years to come.

SPECS & PRICING

Inputs: One each coax SPDIF, BNC SPDIF, AES/EBU, USB, I²S, TosLink; three BNC jacks for DSD

Outputs: Unbalanced on RCA jacks, balanced on XLR jacks
Output impedance: 300 ohms

(balanced or unbalanced)

Total harmonic distortion: < 0.002% at 1kHz

Power consumption: 55W
Dimensions: 48 x 11 x 40cm

Weight: 16 kg Price: \$10,600

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MOON Neo 260D Transport/DAC

By Any Other Name

Neil Gader

hatever you do, don't go calling the Neo 260D from Simaudio just another CD player. That'd be like comparing Usain Bolt to a Sunday morning jogger or describing a McLaren P1 as just another road car. In today's audio climes, "player" implies a certain single-format finality. The Neo 260D is more than that, or at least it can be. As the "Neo" moniker implies, Simaudio has revisited this venerable and vulnerable segment and given it a swift and timely reboot. What this means is that for few extra bucks above the cost of the standard 260D player you can buy the MOON Neo 260D, which comes equipped with a four-input high-resolution DAC section that effectively transforms the unit into a transport/DAC and opens up a whole new world of digital connectivity. Unless you're planning to keep your head firmly buried in the sand for the next few years, you really need to consider the *all-in-one* flexibility the Neo affords. Soon after you'll thank yourself for having had so much foresight.

Under the hood the Neo 260D puts a lot of distance between it and its predecessor. It borrows much of its technology and smooth good looks from the more exotic and costlier 650D in the Evolution Series. The large frontpanel display is highly readable from rational distances and the pushbuttons engage cleanly. The disc tray operates without any lag.

I've got to hand it to Simaudio. I've reviewed this company's gear for years, and I have never failed to be impressed by the high quality of its products' construction. The Neo's chassis is top-notch-geared to minimize external vibrations. Its oversized power supply offers 13 stages of voltage regulation (five for the transport and eight for the DAC). Its proprietary CD drive system is mounted on Simaudio's now-familiar M-Quattro, gelbased, 4-point floating suspension for vibration damping. While the Neo 260D's isolation was good, it was not up to the level of Simaudio flagship players, as I could induce an occasional track-skip by moderately tapping my equipment rack.

The optional DAC section uses a 32-bit asynchronous converter with four rearpanel digital inputs (dual SPDIF, TosLink, and USB). There is no provision for a USB thumbdrive, however. Galvanic isolation of the USB input has been implemented to eliminate all ground current. As a result, there is zero electrical connection between the USB device and the Neo 260D, a feature that preserves the accuracy of the audio signal. Of course, key features you'd expect in a MOON Neo Series product are included, such as a SimLink controller port for two-way communications with other compatible

MOON components, plus RS232 and IR ports for custom-installation environments. Finally, the Neo 260D is available in three different finishes-black (standard), as well as a special order all-silver or two-tone (black and silver). Simaudio's CRM-2 remote control completes the package. In a nod to economizing on power usage, the remote automatically goes into standby mode in about twenty minutes, but this feature can be bypassed by holding down the program button for a couple seconds—a thoughtful touch.

Given digital inputs galore I immediately put them to good use. Apple TV found a home in the optical input. The USB was engaged for streaming from my MacBook Pro (iTunes/ Pure Music). And since I was experimenting with some new SPDIF cables, I used a Musical Fidelity USB/SPDIF converter.

Red Book PCM circa 2014 is a mature format. Anyone who's been paying attention over the last few years knows that standard 44.1kHz/16-bit discs can sound astoundingly good. The CD has guietly benefited from a steady diet of improvements that have manifested themselves at every stage of the recording chain. When you add to that the fact that MOON playback has always been one of the grown-ups in the room, you get rocksolid dynamics, sprinter-like transients, a low noise floor, and true-to-life tonality. Just a few bars of Valentina Lisitsa's fiery performance of Liszt's Totentanz [Naxos] provide ample evidence that the Neo 260D comes equipped to reproduce a concert grand piano and a bravura performance. From jackhammer bottom-octave excursions to searing treble arpeggios brimming with swirling harmonics,

EQUIPMENT REVIEW - MOON Neo 260D



the MOON brings its A-game. And it also brings its sensitive side, as I discovered listening to the delicate exchanges between Charlie Haden and Pat Metheny during their insightful duet rendition of "The Moon is a Harsh Mistress" from Beyond the Missouri Sky [Verve].

Vocals, such as Kasey Chamber's sexy "Pony," possessed an airy palpability that captured the intimacy and humor of the performance. And, as I listened to Holly Cole's cover of "I Can See Clearly" I had to conclude that the Neo 260D reproduces bass information with an uncanny balance of pitch and control. It's very specific in the way it conveys timbre-the tone and skin sound of a drumhead and the resonant character and decay of an acoustic bass. Plus it has an instinct for connecting images across an unbroken soundstage-a trait it more fully exhibits in high resolution. Cole's vocal was smooth and extended, and though a bit cooler and dryer than I've heard-not atypical for digital in this range-her voice never veered into peakiness. The piano solo midway through the track was reproduced with much (although not all) of the player's touch intact. However, the Neo 260D is a little less persuasive at drawing the eye onto the stage and into the dimensional space of a fine acoustic recording. For example, during the Tchaikovsky Violin Concerto with Heifetz and Reiner/Chicago [XRCD] the sense

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of instrumental interplay and continuous air and spaciousness between players was reduced due to the MOON's slightly more arid character. And on occasion I felt that the Neo 260D missed some of the more tactile aspects of music reproduction-something I could best describe as a certain fleshiness that the best recordings deliver, the sense of the complete performer materializing in space, fully physicalized and inhabiting the listening room. It's a beguiling impression that I have attained with significantly pricier rigs from the likes of dCS and MBL, to name two. Nonetheless the Neo 260D acquits itself more than satisfactorily. So no surprises here. The Neo 260D exemplifies the lion's share of what I've come to expect from Simaudio: musical, uncolored, and rock-steady. Standardresolution 44.1kHz/16-bit performance is uniformly very-good-to-excellent.

With high-resolution material, however, the Neo shows a whole new level of refinement. The 176.4kHz/24-bit HRx recording of Rachmaninoff's Symphonic Dances [Reference Recordings] is like a sonic elixir for microdynamics and air. Rather than appearing as uncorrelated segments on an ill-formed soundstage, images immediately register as elements of a fully realized acoustic venueparts of a fluid whole. The major difference as compared with standard-resolution CD is the added complexity of soundstage and dimension. No longer the relatively flat canvas of Red Book, high-resolution material almost seems to reinflate the stage. The micro-dynamic amplitude differences among instruments create a more convincing



representation of the distances between players within the symphony orchestra. There's a more focused sense of layering, beginning downstage with the first violins and moving gradually all the way upstage to the back of the percussion section. As a result

the Minnesota Symphony comes alive as a complex, multi-voiced entity breathing a continuous reverberant energy throughout the hall. The difference in the final analysis is that you feel more like an actual audience member than a mere listening-room observer. And that's really the point, isn't it?

The Neo 260D continues a tradition of fine Simaudio CD players. Its naturalistic and musical sonic performance is mated with resilient build-quality and ergonomics. As a

transport, or with the highly recommended DAC option, the Neo 260D won't lead you wrong. Fully equipped, it opens up an entire world of digital resolution that will preserve your investment for years to come. Dollar for dollar, a splendid component. tas

SPECS & PRICING

CD transport with optional four-input DAC

Digital inputs (with optional DAC): USB, TosLink, SPDIF (2) Analog output (with optional

DAC): Balanced and singleended

Digital output: S/PDIF and AES/EBU

Dimensions: 16.9" x 3.4" x 13.1"

Weight: 16 lbs.

Price: \$2000 (DAC option,

\$1000)

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alifornia-based Aesthetix has carved out an enviable niche producing veryhigh-performance products that, while not budget-priced, nonetheless offer extraordinary value. The company's Calypso linestage, Rhea phonostage, Janus full-function preamplifier, and Atlas power amplifiers pack a lot of innovation, superb build- and parts-quality, and great sound into the upper-end, though not stratospheric, price category. (Aesthetix does, however, offer two mega-priced, ultra-tweaky products, the lo Eclipse phonostage and Callisto Eclipse linestage.) Aesthetix has now complemented its more affordable line with the addition of the Romulus CD player/DAC. In keeping with the company's signature technology, the Romulus features a tubed output stage. In addition to spinning CDs, the Romulus can accept a digital input in a variety of formats including USB. For those of you who have dispensed with CD, Aesthetix's Pandora DAC is identical to the Romulus sans disc transport.

Technical Description

The \$7000 Romulus has a fixed output level for use with preamplifiers. For an additional \$1000 the Romulus includes a variable output for driving a power amplifier directly. The large front-panel display doubles as a volume control-push the right side to increase the volume and the left side to decrease. The circuit that realizes this variable output exemplifies Aesthetix's innovation and commitment to sound quality. Rather than simply attenuate the signal in the digital domain and accept the resolution loss (every 6dB of digitaldomain attenuation is equivalent to throwing away one bit of resolution), Aesthetix has combined analog and digital attenuation in a novel circuit. Decreasing the output level attenuates the signal in the digital domain just as in other DACs with a variable output. But when you reach 6dB of attenuation, a relay engages resistors that replace the 6dB of digital-domain attenuation with 6dB of analog-domain attenuation, resetting the digital-domain level to full scale. Voîlà!variable output without any meaningful loss of resolution at any output level. In practice, this transition from digital-domain to analogdomain attenuation is transparent to the user, save for a relay click when every 6dB threshold is crossed. The volume steps are 1dB, which is a little coarser than what is found in most DACs, but I didn't find it a problem. The variable output circuit is, incidentally, housed on a board that plugs in vertically to the horizontal motherboard.

The Romulus incorporates other interesting design techniques, including a SPDIF receiver and clocking circuit based on a fixed-crystal clock. Most SPDIF input receivers are built around a VCXO (voltage-controlled crystal oscillator), whose output frequency can be adjusted by an external voltage. By "pulling" the clock frequency slightly the DAC can lock to the incoming clock rate. A VCXO, however, isn't quite as precise as a fixed-frequency crystal. Aesthetix solves this problem by using a fixed-frequency crystal and running the data through a memory buffer. Presumably the buffer is large enough to "fill up" sufficiently before outputting the data so that the buffer doesn't run empty if the Romulus clock is faster than the incoming clock, or conversely so that the buffer doesn't overflow if the Romulus clock is slower than the incoming clock. In practice, there's no time lag between pressing "Play" and hearing music, suggesting that a larger buffer isn't required.

The SPDIF inputs are galvanically isolated so that ground noise on source components doesn't get into the Romulus. They can accept datastreams up to 192kHz/24-bit. The input circuits are housed on removable boards to allow future upgrades as new interfaces become available. The same crystals used in the SPDIF input receiver are also used in the

EQUIPMENT REVIEW - Aesthetix Romulus

USB input circuit. It goes without saying that the USB input is asynchronous, meaning that the Romulus serves as the master clock to which the USB source must lock. Other digital inputs include TosLink optical and AES/EBU.

The 8x oversampling digital filter software is written in-house by Aesthetix, and runs on a Motorola DSP56362. Creating a custom filter is considerably more expensive and time-consuming than buying an off-the-shelf filter chip (or using the filter built into most DACs these days), but allows the designer to employ more sophisticated filtering techniques, and to tailor the filter's sound to the context of the entire product. The digital filter has a large effect on the product's sound, which is why many DACs that use the same filter/ DAC chip sound quite similar.

The filter's differential outputs (+ and -, or the datastream and the datastream inverted) are converted to analog by a Burr-Brown PCM 1792 chip operated differentially. With four DACs in one package, the PCM 1792 can process the L+, L-, R+, and R- separately. The downstream signal path-current-to-voltage converter, gain stage, and output buffer-is also balanced all the way to the XLR jacks. This is the right way to create a balanced analog output signal. The less expensive, but more common, alternative is to convert the digital signal to analog with one DAC and one analog signal path per channel, and then create a "balanced" signal with a phase-splitter just before the XLR jacks. Not only does this latter technique add an additional active stage to the balanced signal path, but it also doesn't realize the benefit of differential DACs. One of these benefits is that any noise or distortion common to both DACs will cancel when the balanced signal is eventually summed, not to mention the increase in signal-to-noise

The output stage comprises a 12AX7 gain stage followed by a 6DJ8 output buffer. The circuit has zero global feedback. Tuberollers can supply their favorite 12AX7s and 6DJ8s, but I evaluated the Romulus with the stock tubes.

The power supply is large and elaborate, with multiple regulated stages for powering different subsections. Even the analog, digital, and clocking sections of the PCM1792 DAC are fed from independently regulated supplies, each with cascaded discrete regulation stages. "Discrete" regulation means that the voltage regulators that maintain a constant DC supply voltage to the circuit are built from separate transistors (along with the peripheral parts that make them work). Discrete regulation is contrasted with IC regulation in which the regulator is simply an inexpensive three-pin integrated circuit. "Cascaded" means that the output of a voltage regulator feeds the input of another regulator, further purifying the DC that powers the audio circuit. Cascaded discrete regulation is expensive and consumes circuit-board real estate, which is why it's usually only found in mega-priced products.

The input section, along with the Motorola DSP chip on which the filter runs, is powered by a separate power transformer and multiple independently regulated power-supply stages. The frontpanel display and control section are also powered from a dedicated transformer. The 12AX7 in the gain stage, as well as the 6DJ8 output buffer, is fed from multiple supplies, including regulated heater supplies.

It's impossible to overstate the power supply's sophistication, particularly considering the Romulus' \$7000 price; such implementations are usually reserved for cost-no-object products. I suspect that this approach to the power supply, if not to the circuits themselves, was derived from the development work on Aesthetix's flagship to and Callisto. Having designed the best possible products without regard to price, designer Jim White knows exactly what effect power-supply topologies and parts-quality has on the sound, and is able to make the most intelligent trade-offs. Incidentally, White spent many years at Theta Digital before founding Aesthetix.

I should mention that the Romulus is available in a "Signature" edition that features the identical circuit, but with upgraded passive components and isolation feet. The upgraded components include the same expensive capacitors found in the lo and Callisto. When I compared Aesthetix's Rhea phonostage to the Rhea Signature (whose only difference was passive partsquality) a few years ago I heard a substantial improvement in the Signature version. The

SPECS & PRICING

Inputs: USB, AES/EBU, SPDIF coaxial, TosLink Outputs: Balanced on XLR jacks, unbalanced on RCA

USB input: Up to 192kHz/24-

Tube complement: 12AX7 (x1), 6DJ8 (x1)

Dimensions: 17.875" x 4.25" x

Weight: 30 lbs. (net) Price: \$8000 with variable output (\$7000 fixed-level)

AESTHETIX AUDIO CORPORATION

5220 Gabbert Road Moorpark, CA 93021 (805) 529-9901

ASSOCIATED COMPONENTS Loudspeakers: Magico Q7 **Preamplifier: Constellation** Audio Virgo 2, Absolare Passion

Power Amplifiers: Constellation Audio Centaur monoblocks. Absolare Passion 845 Analog Source: Basis

Inspiration turntable with

Basis Vector 4 tonearm, Air

Tight PC-1 Supreme cartridge; Simaudio Moon 810LP phonostage

Digital source: dCS Vivaldi Transport, MacBook Pro running iTunes and Pure Music AC Conditioning and Cords: Shunyata Triton and Talos,

Audience aR6TS conditioners: **Echole Obsession Signature** and Omnia, Shunyata Zitron Anaconda and Audience Au24 AC cords

Cables: Echole Omnia interconnects; MIT MA-X SHD loudspeaker cables; MIT MA-X2 interconnects: AudioQuest Wild AES/EBU, AudioQuest Diamond USB

Equipment Racks: Stillpoints ESS, Critical Mass Systems amplifier stands

Acoustics: ASC 16" Full-Round Tube Traps, 10" Tower Traps Accessories: Stillpoints Ultra 2 and Ultra 5; Audio Desk Vinyl Cleaner: Mobile Fidelity record brush, cleaning fluid, stylus cleaner

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EQUIPMENT REVIEW - Aesthetix Romulus

Signature edition costs \$10,000 with fixed-level outputs, and \$11,000 with variable-level outputs.

Listenina

From the first CD, the Romulus impressed with its big, open, and expansive sound. It's interesting how source components can allow a pair of loudspeakers to disappear into the soundstage-or not. Playing a CD that has captured a tremendous sense of space such as Dick Hyman's Swing is Here on Reference Recordings made it immediately apparent that the Romulus didn't suffer from the common CD ailments of congealing images and sounding bright without top-octave air. In fact, the Romulus was among the most open and airy digital products I've heard-at any price. The soundstage had wonderful dimensionality and depth, coupled with a sense of being "illuminated from within," a wonderfully evocative phrase coined by Jonathan Valin to describe the classic Audio Research sound. Instruments toward the rear of the soundstage were presented with their tonal colors and spatial qualities fully realized, rather than blending indistinctly into the foreground.

Although the Romulus had tremendous bloom and air, it would be a mistake to interpret this as an artifact of a tubed output stage. On the contrary, image focus was tight and well defined, and the overall perspective was just a bit on the immediate and incisive side—not characteristics of a "tubey" sound. Nothing in the presentation suggested that I was listening to tubes, except the lack of a

metallic, brittle character in the treble. The contrast between the up-front midrange presentation and the ability to see way back into the hall produced a soundstage of remarkable depth. The musical benefit was an ability to easily follow individual instrumental lines, no matter how complex the music or how subtle those lines—one of analog's great strengths, incidentally.

The bass was remarkably rich, full, and weighty, yet still maintained a sense of speed and precision. The Romulus' big bottom end served as the anchor for the rest of the presentation, beautifully conveying everything from the dense textures of orchestral doublebasses to the purring of a Fender Precision. Again, the Romulus didn't sound like a tubed unit in its bottom-end impact, dynamic agility, and articulation. The midbass had a meaty quality that did wonders for an orchestra's power range and some bass-guitar playing. On the title track from Donald Fagan's Morph the Cat in 96kHz/24-bit the Romulus had tremendous low-end grip and power that conveyed the track's visceral physicality. The Romulus is the antithesis of thin, threadbare, lean, or bleached.

The more I listened to the Romulus the more I enjoyed it musically, and the more I admired its sonic achievement. That statement may sound like a tautology, but some products can sound stunning sonically but fail to connect musically. The classic example is the super-high-resolution, dead-quiet, light-ning-fast component that lays bare every last detail in a recording yet fails to engage the heart. It's like a musician who shows off his

astounding technical chops but has nothing to say. In digital products, this often results from a kind of mechanical quality that doesn't really convince your brain that you're hearing musical instruments rather than a collection of sounds. Such a product may hit many of the audiophile buttons, but there's something not quite right—the presentation is built on an artificial foundation. The Romulus' great appeal is that it avoids this pitfall completely. It sounds extremely "organic" and natural, without a hint of synthetic patina, particularly in the treble.

I'm glad that the review sample had the variable-output option installed, because driving a power amplifier directly revealed just how great this player is. The dynamic expression widened, the bass was fuller and went deeper, and transparency increased. It was comforting to know that no matter what the playback level (i.e., the amount of attenuation in the Romulus) I wasn't losing resolution.

Fronting a world-class reference system of Constellation Centaur monoblocks driving Magico Q7 loudspeakers via MIT's Oracle MA-X interconnect and MA-X SHD loudspeaker cables, the Romulus acquitted itself nicely under this ultra-high-resolution "microscope." In fact, this system's resolving power only emphasized what a terrific-sounding player the Romulus is; a lesser player's flaws would have been that much more evident. Moreover, I compared the Romulus directly to what is unquestionably the state of the art in digital playback, the dCS Vivaldi. Of course, the Romulus was not the Vivaldi's equal, but let's just say that my opinion of the Romulus went

up another few notches after this comparison when I reflected on the fact that the Vivaldi DAC and transport cost nearly ten times more than the Romulus.

Driving the Romulus through its USB port with a MacBook Pro running iTunes and Pure Music revealed the USB implementation to be excellent, although I preferred the sound of CDs played in the Romulus' transport compared to rips of those CDs. The bass was fuller and better controlled, the midrange and treble smoother, and the soundstage more expansive. Remember, however, that the USB cable and the computer setup are significant variables when using a computer source.

Conclusion

The Romulus is another home run for Aesthetix, combining innovative circuit design, premiumquality parts and construction, and first-rate sound. This player was never anything less than musically communicative and immensely enjoyable, even in the context of a cost-no-object system. The Romulus is a flat-out bargain at its price, and invites comparison with much more expensive players. In fact, I'd call the Romulus a giant-killer, offering one of the highest price-to-performance ratios in digital playback today.

Esoteric K-03 CD/SACD Player/DAC

Brilliant Concept, Great Sound

Alan Taffel



he Esoteric K-O3 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-O3 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-O1, the K-O3'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-O3. This model makes do with just two power supplies, a Duraluminum clamp, and a paltry eight DACs. Specifically, the K-O3 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-O3 is also unusual in that it requires a great deal of configuration before it can-or should-be used. Most digital players are plugand-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-O3 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-O3 has quite a few operating modes, and most-but not all-of them deliver

EQUIPMENT REVIEW - Esoteric K-03

reference-quality sound. Surprisingly, the least impressive (which is not to say unimpressive) 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-O3, 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-O3. 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-O3 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-O3 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-O3's SACD rendition simply annihilates it.

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-O3 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-O3'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-O3's DAC always proved more detailed and neutral—and therefore more enjoyable—when driven externally.

This should not be the case. Selfcontained 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-O3 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-O3 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

SPECS & PRICING

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.

Price: \$13,000

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EQUIPMENT REVIEW - Esoteric K-03

(sorry, Mac users). Using ASIO has the added benefit of enabling the K-O3 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 file of Keith Jarrett's Köln Concert and found it to be equally engaging. The real achievement of the K-O3'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-O3 can serve as a linestage. Like many recent digital players and DACs, the K-03 has a builtin 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-O3 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-O3'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-O3 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-O3 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.



Modwright-Oppo BDP-105 with "Truth" Modifications

Hot Rod!

Jim Hannon

he Modwright "Truth" modifications to Oppo's BDP-105 player were brought to my attention by Infinity co-founder Arnie Nudell when I interviewed him for *The Absolute Sound's Illustrated History of High-End Audio*. I asked Arnie which current products most impressed him, and he named two: the marvelous Constellation Audio Reference Series amplifiers, and the surprising Modwright-Oppo BDP-105. Since I regard Arnie as one of the greatest of all high-end speaker designers—having owned many of his stellar creations including the Infinity RS1, RS-1B, and Beta—his opinion carried a lot of weight. What was most intriguing was Arnie's contention that this set of Modwright modifications vaulted the Oppo to a reference level. He claimed, "With the right program material it does as good a job as anything conveying the musicality and emotion of a live concert."

I had to hear for myself what these modifications could do to enhance the performance of the already well-regarded Oppo BDP-105. In my experience, incorporating tubes in digital front-ends can certainly lead to better sound. Long ago, I purchased a California Audio Labs Icon II because its tubed analog output stage made those bits sound more natural to my ears compared to other moderately priced digital players of the day. Admittedly, many costly digital front ends have left me somewhat uninvolved, particularly with SACDs, whereas others that sound great are either far beyond my budget, such as the remarkable dCS Vivaldi, or are no longer available. Could the relatively affordable (\$2495) Modwright "Truth" upgrade to the Oppo BDP-105 be the solution to my "digital dilemma" and enable me to get more enjoyment from the latest high-resolution digital media?

In my experience, modifying stock products to improve performance can be a slippery slope. One assumes that since most products are designed to a price point, they can be improved, at a minimum, by replacing select components with higher-quality, more costly ones. However, there are several risks involved. First, the manufacturer's warranty is voided. Second, more costly parts do not always ensure better sound. Product designers spend a lot of time and effort listening to their products to carefully voice them, and monkeying around with the original design can destroy this delicate balance. Third, there is the issue of workmanship and reliability. Whereas the stock units typically undergo rigorous testing, too many modified units do not. Fourth, when you go to sell a modified

product, you generally recover only a small fraction of the cost of the upgrade. Fifth, you often take your chances on a modified unit without being able to audition it and without the support of your local dealer. If you don't like the results, you're stuck. Last, once you start down the modification pathway, where do you stop, particularly with tubed-based modifications? Besides several additional Modwright options to the basic "Truth" modifications-including the Bybee Music Rail and Audio Magic Pulse Gen ZX upgrades, which were not added-one can also spend a lot of time "tube rolling" to find the perfect sonic match. I did upgrade the Modwright-supplied Electro-Harmonic 6SN7EH driver tubes with some wonderful Sophia Electric 6SN7 tubes, as well as swapping the Sovtek 5AR4 rectifier tube in the external power supply for a taller and wider Philips 5R4GYS, recommended by tube-maven Kevin Deal. (Note: This latter change requires a top plate with a hole, which can be supplied by Modwright.) Both these tube replacements move the Modwirght-Oppo closer to the sound of a live performance with gains in openness, inner detail, image depth, truth of timbre, and dynamic explosiveness.

The key questions for me were: Would the Modwright "Truth" modifications to the Oppo BDP-105 be worth it, and what has Modwright done to help minimize the associated risks?

The Stock Oppo BDP-105

Modwright starts with Oppo's highly successful, award-winning BDP-105 universal player as its digital platform—TAS' 2013 Disc Player of the Year that was reviewed quite favorably by CM in Issue 232. He was

EQUIPMENT REVIEW - Modwright-Oppo BDP-105 with "Truth" Modifications

impressed by its "clean, clear, and detailed" presentation," commenting that "it is far more revealing than it has any right to be for the money." I concur and would add that with high-resolution sources this player is surprisingly good. However, during extensive listening sessions, I always knew I was listening to a digital source.

Modwright's Dan Wright selected this Oppo because "it is universal, state of the art, reliable, and lends itself well to significant improvement by way of our modifications." (The modifications are the same for Oppo's new BDP-105D "Darbee" edition that offers 4k video and a DSD input.) Fortunately, I was able to borrow a stock BDP-105 from Oppo's CTO, Jason Liao, for comparison. My listening observations mirrored CM's for the most part. I appreciated the Oppo's remarkable clarity and fine detail resolution; however, I thought that the stock unit was somewhat lean-sounding in the upper midrange when reproducing massed strings, a limitation I hear on most digital front-ends. This limitation has kept me from enjoying digital music as much as I do analog, and typically leads to aural fatigue during my extensive listening sessions.

Jason also volunteered to measure both units using his test software at the Oppo offices near my house. As I expected, the stock Oppo measured better than the Modwright on all distortion parameters, but as most audiophiles know, test results do not tell the whole story. Indeed, according to Jason, the higher level of second-order harmonic distortion of the Modwright is something Oppo is trying to design into some of its products to

help them sound richer and more natural.

The Modwright "Truth" Modifications

The Modwright "Truth" modifications to the Oppo BDP-105 are guite extensive and include a total redesign and replacement of the single-ended and balanced output stages with Modwright's tube analog output stage. The most noticeable physical differences are: the inclusion of two 6SN7 driver tubes rising above the top plate of the Modwright-Oppo; a separate external power supply with one 5AR4 rectifier tube and two 13EM7 voltageregulator tubes; and a Modwright "Truth" umbilical cord connecting the power supply to the main unit. The Modwright-Oppo also includes upgraded Cardas RCA connectors (for stereo outs), a Furutech cryo-teated IEC, upgraded resistors in the signal path, cryotreated solid-cord silver wire for the signal path, and damping mods for the chassis and transport. As mentioned, the external supply can accommodate taller, larger-diameter tubes with the new top plate from Modwright.

The digital stage is not touched by Modwright, except that the "Truth" modifications improve the existing supplies that power the digital circuitry. Modwright's Dan Wright said that Oppo's stock clock "is excellent and unique," and its DAC (employing two ESS Sabre ES9018 chips) "is exceptional." He added that because they could not improve upon the clock or the DAC, Modwright left them alone. My review unit did not include any other optional Modwright modifications, but in addition to "tube-rolling," I did add the outstanding Shunyata Alpha digital

power cord, which significantly increased the transparency and improved the naturalness of the timbre of both the Modwright and the stock Oppo.

While I hesitate to recommend any unit you cannot audition beforehand, Modwright takes the risk out of owning its modified Oppo BPD-105. Although the factory warranty is voided, Modwright offers a one-year warranty of its own and offers to service the unit should it fail in a way that would otherwise be covered by

the factory warranty, for cost of parts alone. Because the Oppo is so reliable, this is "very seldom necessary," according to Dan. Even Oppo's Jason Liao praised the workmanship of the Modwright modifications as "exceptional." One rarely hears this kind of praise about any modification from the manufacturer of the original equipment.

Listening Comparisons

The sonic comparisons among the stock Oppo, the Modwright-modified Oppo, and my turntable system were illuminating. As both digital units required extensive break-in, I did not conduct my listening tests until after both had time to fully settle down. I was able to switch between them on several discs, using all the same ancillary components. Since the upgraded tubes in the Modwright and the addition of Shunyata's Alpha Digital power cable to the stock and the modified Oppo units moved

me closer to the illusion of hearing a live performance in my listening room, I used both for my extensive sonic comparisons.

With the glorious Reference Recordings CD of the Rutter Requiem, the stock Oppo sounded surprisingly good for such a modestly priced component, with appealing clarity and bass extension. However, I noticed some sibilance and stridency in the voices, particularly on dynamic peaks. Switching to the Modwright-Oppo, the voices and instruments bloomed,

SPECS & PRICING

Price: \$2495 (for Modwright "Truth" modifications alone) Price of stock Oppo BDP-105 (user supplied): \$1199

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INC

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ASSOCIATED EQUIPMENT
Merrill-Williams R.E.A.L 101
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BDP-105 digital players; MFA Venusian (Frankland modified), PrimaLuna Dialogue Three, and Constellation Audio Virgo II preamps; PrimaLuna DiaLogue HP monoblocks and Constellation Audio Centaur amplifiers; Magnepan 3.7i and Quad ESL-57 (PK modified) loudspeakers; Silver Circle Audio TCHAIK6 power conditioner; Shunyata Research Alpha digital power cable. Nordost Valhalla interconnects and power cords, AudioQuest Niagara interconnects and Metro speaker cables, etc.

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EQUIPMENT REVIEW - Modwright-Oppo BDP-105 with "Truth" Modifications

and I found myself immediately more immersed in the music. There was still very good clarity and detail, but now without any of the digital stridency. The soundstage seemed to take on an added dimension, too, with layered depth separating the performers in the chorus and a really good sense of the hall. Voices sounded richer in tonal color and music just flowed with more natural ease. In short, the Modwright-Oppo opened the door more deeply into the music, and I found myself listening to the entire piece rather than a few sample tracks.

On the higher-res SACD of Reference Recordings' Exotic Dances from the Opera, percussion on the stock Oppo had appealing transient quickness and "snap." Soundstaging was also good, particularly in width. Turning to the Modwright-Oppo, the decay of the cymbals and triangles sounded more natural, woodwinds had more body, and there was greater separation among the performers on stage. The sonic gap between the Modwright-Oppo and my turntable system was surprisingly

narrow, and both left the stock Oppo in the dust. Admittedly, I preferred the enhanced bloom, air, soundstage depth, and more natural timbre of the vinyl to the Modwright-Oppo, but bass articulation, impact, and extension were comparable.

On Reference Recording's brilliant HRx DVD-R of Rachmaninoff's Symphonic Dances (recorded at 176.4kHz/24), the stock Oppo displayed more of its remarkable bass power and extension, dynamic

explosiveness, and clarity. Although massed strings had realistic shimmer and woodwinds sounded good, they were reproduced with some digital artifacts (edge), which detracted from the illusion of a live performance. Moving to the Modwright-Oppo, woodwinds had more body, and strings were more natural-verging on the lush. The timbre was more harmonically fleshed out, the instruments had more air, and the music breathed as it does in the concert hall. Yes, there was more warmth, but no syrupy or caramel tube coloration here, and the enhanced tonal richness didn't come at the expense of transient speed, inner detail, or dynamic explosiveness. Moving to vinyl, the differences in clarity and inner detail between it and the Modwright-Oppo were too close to call. However, with vinyl the sound was a bit more open, particularly in the highs, with slightly more delicacy and hall ambience, as well as more body and richness. However, I had to give a slight nod to the Modwright-Oppo in bass power and articulation, and its lower noise floor was more appealing.

As expected, the sound of both the stock Oppo and Modwright improved fairly dramatically as the resolution of the digital media increased (on good recordings). Arnie provided me with a 2.8MHz DSD recording (converted to 24/176 PCM) of Mahler's Symphony No. 7 in E minor from the San Francisco Symphony. The strengths of each unit increased and their shortcomings were less in evidence. For example, the sound on the Modwright-Oppo was stunning in balanced mode with see-through transparency, a deep, wide, and precise soundstage, fleshed-out timbre with no sense of digital artifacts in the pure harmonic overtones, and explosive dynamics with deep bass extension. The fine detail has a delicacy that is mesmerizing, perhaps lacking only the last bit of air one hears with top-notch analog sources, albeit with the benefit of a lower noise floor. Regrettably, in this instance, I did not have the vinyl for comparison, but I must concur with Arnie that on this recording, the Modwright-Oppo certainly conveys the musicality and emotion of a live concert!

One of my audio buddies and a frequent listener to my system said that he thought that I was playing my turntable when the Modwright-Oppo was in the system. I must admit that there were times when I was so lost in the music that I had to look up to see which source was playing. That is very high praise and has never happened to me before with a digital source in my home.

Conclusion

So are the Modwright "Truth" modifications to Oppo's BDP-105 player worth the cost and the risk? The answer to that question is a resounding, "Yes!" The build-quality is exemplary and the sonics are exceptional. It is the first digital player I have had in my listening room that didn't make me want to go back to my analog rig right away, and that's because it sounds so much like analog in many respects, without giving up the bass extension and control, clarity, fine detail resolution and retrieval, and convenience that can make digital so attractive. With outstanding sonics that can make you

forget you're listening to digital, and its remarkable flexibility and compatibility when playing discs from a player or a computer-audio setup, I suspect this is one universal player that you'll be hanging on to for a long time. While I'm unwilling to abandon my analog rig, I can see why others, like Arnie, have chosen to do take this path, particularly when one feeds the Modwright-Oppo first-rate, high-resolution source material. The Modwright Oppo is now my digital reference. Loss



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dCS Vivaldi Digital Playback System

Ne Plus Ultra

Robert Harley

here are few companies in high-end audio that can be considered unique. By that I mean a company that creates technologies unlike those of any other brand, and designs products that are fundamentally different from the innumerable "me-too" components that flood the market.

Take the important job of digital-to-analog conversion, for example. Virtually all DAC manufacturers buy from the same menu of available chips and configure them in slightly different variations of the same theme. Some are more successful than others in creating good-sounding products, but none builds a digital-to-analog converter from the ground up with technologies invented entirely inhouse.

An exception to this rule is the British company Data Conversion Systems (dCS). Every key element in every product it manufactures is designed and built by dCS using proprietary technologies. You won't find an off-the-shelf DAC chip, digital filter, or input receiver in a dCS product. What you will find are circuits, techniques, software, and engineering unlike that of any other product.

The ultimate expression of dCS' unique approach is embodied in the new Vivaldi,

the most advanced and ambitious digitalplayback system yet created. This \$108,496 four-box tour de force takes dCS' proprietary technologies to their ultimate realization. It is impossible to overstate the Vivaldi's technical sophistication-or its revelatory musical presentation. dCS has wrapped all of this advanced technology in stunningly bold and elegant metalwork that is as unique as the circuits inside.

Overview

Describing the Vivaldi, its capabilities, and how it functions is guite a challenge. This is by far the most complex, technical, intricate, and feature-laden product I've ever reviewed. I'll give you a brief overview here, with more technical detail in the sidebars.

The complete Vivaldi system is a four-box design, but not all four boxes are required. Those four units are the Vivaldi Transport (\$39,999), Vivaldi DAC (\$34,999), Vivaldi Digital-to-Digital Upsampler (\$19,999), and Vivaldi Master Clock (\$13,499). You could, for example, buy just the Vivaldi DAC and drive it with a computer source (it has a USB input) or connect your CD player (if the CD player offers a digital output). A big step up is the Vivaldi

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Transport and DAC combination; when used together the two units offer unique features including upsampling in the transport and the ability to transmit encrypted high-resolution audio (DSD or upsampled PCM) from the Transport to the DAC. In fact, a Vivaldi Transport and Vivaldi DAC will get you much of the way toward the sonic performance described later in the review. In addition to reading CDs, the Transport is compatible with SACD.

The Vivaldi Upsampler converts any common sampling frequency to any other sampling frequency, each user-selectable. Moreover, the Upsampler adds network capability, allowing you to integrate the Vivaldi with a music server, control the system via an iPad/iPhone/iTouch, and play music directly from a USB stick. The Vivaldi Clock simply serves as a high-precision master-timing reference to which all the other units are locked.

Each unit is housed in a gorgeous chassis featuring a three-dimensional sculpted front panel. Raised flowing lines grace the front panels like gentle waves. My review sample was silver, but black is also available. A full-color display toward the left side of each front panel shows, under normal use, the operating status (input selected, sampling frequency, whether the unit is locked to the Master Clock, etc.). But when you put one of the units into menu mode the display shows a myriad of details for setting up and configuring the component. The menu structure is so extensive that dCS provides a plastic-coated map in addition to the owner's manual.

Connecting and configuring a Vivaldi is

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best left to your dealer; a typical setup requires 12 digital cables, and that's without adding networking capabilities. The four units that make up the Vivaldi have collectively a mind-bending 56 rear-panel connectors. Moreover, the settings on each unit need to be optimized for the best sound. These include the upsampling options, whether the clock is dithered, and which of the six digital filters is selected, to name just a few of the many, many settings that are required to make the system work and realize its sonic potential.

Once the system is set up and has settled in, however, operation is quite simple. The DAC remembers the filter chosen for each input and sampling rate, for example, so there's no need to adjust these settings (although you may want to change filters for different recordings-more on this later). The only button you'll probably need to press on a daily basis is the DAC's input selector to switch between CD and SACD from the transport, or the Upsampler's input-select button to play music from a server. You can select DAC and upsampler inputs from the full-featured remote (supplied with the DAC). This hefty and well-designed unit controls transport functions, adjusts the DAC's output level, selects the DAC filter, fine-tunes the left-right balance, and changes the upsampler's output sampling rate, among other functions.

I received a very early production model that had a couple of minor glitches, but a software update (easy to do from a CD or USB stick) fixed them. Interfacing the Vivaldi with a music server via a wireless network was considerably more challenging. My setup

SPECS & PRICING

VIVALDI TRANSPORT CD/SACD TRANSPORT

Outputs: Dual AES/EBU with DXD or proprietary dCS encrypted DSD, AES/EBU, SPDIF (one RCA, one BNC),

SDIF-2, word-clock in, word-clock out

Dimensions: 17.5" x 7.8" x 17.2" Weight: 51.1 lbs.

Price: \$39,999

VIVALDI DAC

Inputs: Four AES/EBU (each can be used independently or as dual pairs to accept DSD or DXD); three SPDIF (two RCA, one BNC); SDIF-2; three USB Type B; word clock Outputs: One stereo pair balanced on XLR jacks, one

stereo pair unbalanced on RCA jacks, o

Output level: Variable (maximum of 2V or 6V output user

selectable)

Digital filter: Selectable, six for PCM and four for DSD

Dimensions: 17.5" x 6" x 17.2"

Weight: 35.7 lbs Price: \$34.999

VIVALDI DIGITAL-TO-DIGITAL UPSAMPLER

Inputs: Network (RJ45), USB (Type B connector), USB (Type A connector), AES/EBU, SPDIF (two RCA, one BNC, one TosLink). SDIF-2

Outputs: Two ES/EBU (can operate independently or as a dual pair to carry high-res PCM or dCS-encrypted DSD)

Dimensions: 17.5" x 6" x 17.2"

Weight: 31.3 lbs Price: \$19,999

VIVALDI MASTER CLOCK

Outputs: Two groups of four independently buffered

outputs on BNC connectors

Clock frequencies: 44.1, 48, 88.2, 176.4, 192kHz

Dimensions: 17.5" x 6" x 17.2"

Weight: 29.9 lbs Price: \$13.499

ASSOCIATED COMPONENTS

Digital Sources: MacBook Pro; AVA Media Zara Premium ripping server, Pure Music and Audivana playback software Analog Source: Basis Inspiration turntable with Basis

Vector 4 tonearm, Air Tight PC-1 Supreme cartridge;

Simaudio Moon 810LP phonostage

Preamplifiers: Rowland Corus, Constellation Perseus,

Absolare Passion

Power Amplifiers: Rowland 725, Lamm ML2.2,

Constellation Centaur monoblocks, Absolare Passion 845

AC Conditioning and Cords: Shunyata Triton and Talos,

Audience aR6TS conditioners; Shunyata Zitron Anaconda

and Audience Au24 AC cords

Cables: Shunyata Anaconda interconnects and

loudspeaker cables; MIT MA-X2 and MA-C interconnects, MIT MA-X SHD loudspeaker cables; AudioQuest WEL

Signature interconnects, Transparent XL Reference

interconnects; AudioQuest Diamond USB and WireWorld

Platinum Starlight USB

Equipment Racks: Stillpoints, Critical Mass Systems

amplifier stands

Isolation: Stillpoints Ultra SS and Ultra5

Acoustics: ASC 16" Full-Round Tube Traps, 10" Tower Traps

Accessories: VPI 16.5 record-cleaning machine; Mobile

Fidelity record brush, cleaning fluid, stylus cleaner

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included a wireless router and an AVA Media Zara Premium ripping server connected to the Upsampler via an Ethernet cable. Downloading dCS' iPad app from the Apple store allowed me to browse music on the server, choose music for playback, and assemble playlists. Although this sounds simple, it took many frustrating hours of troubleshooting to get the whole thing working. My experience might not be typical, particularly as dCS continues to improve its software.

Listening

The Vivaldi is built like no other digital source, and it sounds like no other digital source. Although there are obvious similarities between dCS' Puccini CD/SACD player and the Vivaldi, this new cost-no-object implementation of the company's best technologies vaults the sound quality into the stratosphere. The Puccini is the only other dCS product with which I'm familiar; many of you will want to know how the Vivaldi compares with dCS' former flagship, the Scarlatti. I asked our reviewer Jacob Heilbrunn, who owned a Scarlatti for many years (and bought the Vivaldi after hearing it) to make that comparison (see sidebar).

One of the key characteristics of any dCS product is the sheer density of information it conveys. By that I mean not just resolution of fine detail, but the impression that the fabric of the music is intricately woven from the finest silk. By contrast, other digital sounds somewhat less "continuous" in texture, as though, to expand on the cloth analogy, it were woven from coarser, less tightly woven

threads. With the Vivaldi, instrumental timbres have a richness and saturation that more convincingly create the illusion of hearing the instrument itself and not a digital reconstruction of it. This increased density of tone color and impression that the sound is organic and continuous-unique among digital playback systems in my experience-goes a long way toward narrowing the gap between digital and analog. The Vivaldi renders timbres with a vividness and immediacy that are startling, even from CD and 44.1kHz/16bit files. What the Vivaldi does better than any other digital source I've heard is to make a piano sound more "piano-like" and a sax more "sax-like." I could have chosen any two instruments for this example—the Vivaldi simply portrays every instrument more realistically and less synthetically. Part of this quality is due to the increased density of texture and tone color mentioned. but also to the lack of a mechanical tincture that makes digitally reproduced timbre sound slightly unnatural. When at the front end of a chain that includes the ultra-transparent Constellation electronics, MIT interconnects and cable, and Magico Q7 loudspeakers (along with state-of-the-art power conditioning and vibration isolation), the Vivaldi offers an absolutely gorgeous and lifelike portrayal of instrumental textures.

There's another reason why instruments and voices sound so "there" with the Vivaldi, and that's the system's tremendous clarity and transparency. The old cliché "veils were lifted" could have been coined to describe the Vivaldi's startling sense of nothing between

you and the music. Frankly, the Vivaldi makes other digital sound somewhat thick and opaque. This clarity and crystalline transparency not only increases the impression of timbral vividness and immediacy, it also contributes to the Vivaldi's astonishing spatial presentation. This transparency isn't just between you and the lead instruments; it extends to the far reaches of the soundstage. The ability to see deep into the hall and hear instruments at the back of the stage sound just as vivid and alive as instruments at the front is unprecedented in my experience.

Think all digital sounds flat and lacking in dimensionality? Listen to the Vivaldi and you'll hear just how much space, air, depth, and bloom is encoded in your music library just waiting to be liberated. I've repeatedly been amazed over the years to discover greater spatial fidelity and musicality from familiar discs after hearing those discs through better and better components, but never to the degree of the dCS. The Vivaldi represents a huge leap forward in rendering digitally reproduced music with dimensionality, depth, and a sense of transparent space between instrumental images. Listen to the first track on the amazing Playing With Fire from Reference Recordings. The wall behind the loudspeakers completely disappears; the low brass sounds as though it's twenty feet behind the loudspeaker plane. Not only does the Vivaldi recreate depth, but there's an organic continuousness to the presentation of depth along a continuum. I can vividly hear the placement of each instrument in the soundstage. Not only is the soundstage deep, it's also expansive, in

the sense of top-octave air riding "over" the music. I'm not referring so much to musical information in the top octave, but rather to the impression of a lid being taken off the top of the soundstage, with a resultant opening-up of the presentation. The impression of air around cymbals, for example, makes them more vivid and alive than a rendering that presents just the instrument itself. The latter sounds flat and lifeless, paradoxically being simultaneously bright yet lacking in treble extension. The Vivaldi's top-octave air and soundstage openness are unlike any other digital I've heard, and much closer to a great analog front end.

The Vivaldi's soundstaging pays tremendous dividends in the ability to hear individual musical lines and shift focus between instruments no matter how complex the music. The presentation is the antithesis of homogenized, thick, congested, and confused. This clarity of instrumental lines was apparent, and musically significant, on even small-scale music like the acoustic guitar and violin duet "Northern Lights" from the Dixie Dregs' Freefall, where the interplay between instruments suddenly became clearer, and the musicians' intent more palpable. Listen also to Joe Pass' comping during the muted trumpet solo in "Contractor's Blues" from the XRCD of Count Basie's 88 Basie Street through the Vivaldi and any other digital playback device. The dCS conveys every nuance of his expression and in doing so, restores the energy, the unbridled swing, and sense of contemporaneous music-making that other digital sources dilute.

I would characterize the Vivaldi's treble balance as leaning toward the incisive rather than the relaxed. I'm generally intolerant of a bright or forward treble-it's a deal-breaker in my book. Yet the Vivaldi manages to present a full measure of treble information with no sense of forwardness or aggression. That's partly because the treble is so well integrated with the rest of the spectrum, is finely woven in texture as described earlier, and is presented in its proper spatial perspective within the soundstage. Bright digital sources force the treble forward and make it sound like a separate component riding on top of the music. The Vivaldi's treble is at once delicate and refined, yet full-bodied and unrestrained. Moreover, the top end is richly and finely detailed, particularly at very low levels. It is also extremely clean and free from glare, another factor that allows the Vivaldi to sound fully alive in the top end yet not cross the line into brightness. When thinking about the Vivaldi's upper-midrange and treble I was reminded of Jonathan Valin's wonderful phrase "illuminated from within," which he first used to describe Audio Research electronics. That's exactly how the Vivaldi sounded-infused with light and air. The treble resolution sounded completely natural, devoid of etch, grain, glare, and mechanical artifice. Nonetheless, I should mention that in my system I preferred the slightly softer-sounding DAC Filters 4 and 5 rather than the brighter and "sharper" Filter 1 (which dCS claims is technically "correct"). As you scroll through the filter choices, moving to higher numbered filters, the sound becomes increasingly more relaxed. Filter 5

has no pre-ringing (the filter's ringing energy is shifted so that it occurs after the transient rather than before and after the transient) and is useful when listening to hard-sounding 44.1kHz/16-bit sources. This illustrates the utility of having multiple filters available at the press of a remote-control button.

In the reproduction of bass-extension, weight, dynamics, articulation, pitch definition—the Vivaldi is simply in a class of its own. The bottom end is big and powerful, yet fast and delicate. The entire bass region has a clarity that, again, is unlike any other digital I've heard. This quality alone makes the Vivaldi revelatory-it conveys the texture, the body, and life of low brass, cello, acoustic and electric bass, piano, and kick drum. The Vivaldi doesn't dilute the timbre, dynamics, power, weight, or clarity of bass-rich instruments. The thundering left-hand lines of Nojima Plays Liszt [Reference Recordings], for example, conveys the piano's size, power, and authority, as well as Nojima's commanding mastery of this music. I had the same goosebump-raising experience with this recording as when I stood a few feet from a 9' Steinway while pianist Fan Ya Lin performed with powerful ferocity at the last Rocky Mountain Audio Fest. These bass qualities combined with the transparency, bloom around image outlines, and textural finesse described earlier to create a more credible illusion of an acoustic bass in my listening room than any other digital I've heard-by a wide margin. Listen, for example, to Edgar Meyer's superbly recorded instrument on the CD Hop, Skip & Wobble [Sugar Hill], a trio album with Jerry Douglas

and Russ Barenberg. Rather than hearing a mere source of low-frequency sounds, I got an uncanny impression of the instrument's large wooden body resonating, and the attacks of each note expanding out into the room.

The Vivaldi's SACD performance was off-thecharts-great. Fabulous-sounding SACDs were sublime in their resolution, clarity, and lack of a "digital" signature. I recently discovered the Japanese label Eighty-Eights (distributed by Eastwind Imports). Judging from the first two discs I've heard from its catalog-The Great Jazz Trio (Hank Jones, John Pattitucci, and Jack DeJohnette) and Roy Haynes' Love Letters-the label is recording great music in state-of-the-art sound. All the titles are recorded originally in DSD and released as hybrid SACDs. The disparity between CD and SACD was not as great as I've heard from other players, but not because of any limitation in the Vivaldi's SACD performance. Rather, the Vivaldi's ability to make CDs sound so great vaults their performance closer to SACD territory than I thought possible. In comparisons between SACD and the same titles in 96kHz/24-bit PCM played from the server I'd have to give the edge to the SACD format for its smoother, more lifelike treble and concomitant greater ease. The PCM file, by comparison, has a hint of hardness in the treble not present in SACD playback. Cymbals reproduced in the SACD format are more delicate, airier, and lack the slightly hard timbre of PCM.

I was able to play native DSD files from a computer but didn't have the equivalent SACDs to make comparisons. I was also unfamiliar with the limited selection of music available. I can, however, report that the sound was superb and that the Vivaldi played DSD files with no problems.

High-resolution PCM files with which I'm very familiar were taken to another level by the Vivaldi. The low-level detail, spatial cues, and dimensionality of Exotic Dances for the Opera [Reference Recordings] at 176.4kHz/24-bit were stunningly portrayed. The Vivaldi revealed these spectacular recordings in all their glory. The qualities that make high-res better than standard-resolution—transparency, low-level detail, dimensionality—were fully realized via the Vivaldi's stunning resolution, timbral fidelity, dynamics, bottomend weight and precision, and particularly, its tremendous dimensionality and sense that the soundstage is infused with air and light.

If you're wondering whether it's worth spending the money for the Transport in this age of computer audio, my view is that the transport is essential for several reasons. First, the Transport will allow you to enjoy the large catalog of SACD titles. There are enough great SACDs to make it worth investing in hardware to access this format. Moreover, the Vivaldi represents the highest realization of the SACD format now and in the foreseeable future; I doubt that anyone will ever build a player that eclipses the Vivaldi. And when dCS inevitably discovers new techniques for extracting higher performance from the format, those techniques will likely be available via a software upgrade. Second, in comparisons between playing a CD in the transport and listening to a ripped file of that

CD I thought that the Transport had a small but noticeable sonic advantage. You wouldn't think that reading data from an optical disc on the fly could be preferable to accessing a file from a hard drive, but to my ears the CD had a slightly greater sense of musical flow and involvement. It was hard to pin down to a specific sonic attribute, but the file sounded very slightly mechanical by contrast. The difference was small—less than that between USB or Ethernet cables, for example—so the disparity could have been my particular setup.

You might also wonder what contributions the Upsampler and Master Clock make. The Upsampler gives you more options for upsampling (the Transport's upsampling choices are limited), a networking capability, and an improvement in sound quality similar to that offered by the Master Clock. Adding either unit sharpens the soundstage focus, deepens the spatial presentation, increases dimensionality, resolves more fine detail, and renders timbres with greater realism. Even without the Upsampler and Clock, the Vivaldi Transport and DAC pair deliver the bestsounding digital I've ever experienced. The Upsampler and Clock just take what is already a spectacular presentation to an even higher level. Note that when listening to SACDs the Upsampler isn't in the signal path; the Transport connects directly to the DAC via dual AES/EBU cables.

The Vivaldi sounds in many ways like the Magico Q7 loudspeaker: ultra-transparent, ultra-realistic in its rendering of timbre, precise without being analytical, wide in bandwidth, tremendous in clarity and resolution, with a

bass presentation that combines authority with definition. The Q7's transparency and neutrality allowed me to fully hear the Vivaldi's remarkable musical performance. The Vivaldi's clarity and textural density gave me a new appreciation for the Q7's resolving power and realism.

After listening to the Vivaldi for a month with it connected with generic digital cables (AES/EBU and BNC-terminated clock cables), I replaced the signal and clock cables (one at a time) with AudioQuest digital cables (Wild AES/EBU and Eagle Eye BNC). It wasn't a big surprise that the AudioQuest AES/EBU cables elevated the performance, but I didn't expect replacing the cables carrying the clock would make such a difference. The sound with the Eagle Eye clock cables became more coherent, relaxed, and more musical. It wasn't so much that the digital cables improved specific areas, but rather that the sound become more engaging and expressive. The Vivaldi is such a finely tuned instrument that it reveals everything, and at this level of quality, every improvement is significant.

Conclusion

After living with dCS' Puccini CD player, and then learning about the Vivaldi's technology, I had expected this new flagship to raise the bar in digital playback. I just didn't expect that it would raise it so far above the current state of the art. The Vivaldi is in a class of its own in every category—technical sophistication, capabilities, and most importantly, sound quality. It was mind-blowing to hear well-worn references brought to life with such

realism; I never thought that I would hear such dimensionality, clarity, and timbral fidelity, or experience such musical involvement, from standard-resolution digital.

In addition to reference-class sound from CD. SACD, and high-resolution files sourced from a computer, the Vivaldi also sets the benchmark in functionality. There's nothing the Vivaldi won't do-PCM to DSD conversion, PCM to DXD, play DSD files, upsample any sample rate to any other rate, connect to a network for music-server integration, play files from a USB stick-the list goes on and on. Moreover, the Vivaldi's hardware platform is overbuilt for the current software; new features and capabilities can be added with software updates without taxing the hardware's capabilities. This state-of-the-art functionality, however, comes at a price: The Vivaldi is a highly complex and sophisticated instrument that asks much of the user in terms of selecting the operating parameters and monitoring the displays to be sure that the settings are optimized. In addition, I was not entirely satisfied by the music-server integration function. It was a challenge to set up, and the dCS iPad-control app is not as intuitive as, for example, the free Apple Remote app for iTunes.

If you are in the fortunate position of being able to afford it, there's no better sounding, more capable, more technologically advanced, or more future-proof digital source than the dCS Vivaldi. There's simply nothing else like it. It is truly, and by a wide margin, the *ne plus ultra* of digital playback. tas



The Vivaldi's Technology

The Vivaldi Transport is built around the massive Esoteric VRDS Neo disc mechanism. I attended a detailed technical presentation on this mechanism several years ago at Esoteric's California office and can tell you that it's built like no other discplayback system in the world. The 14-pound device (conventional transports weigh a few ounces) is made from parts cut from solid-steel blocks. Most transports hold the disc at the center with a flimsy clamp; the VRDS Neo mechanism features a clamp just bigger than a CD that is made from machined Duralumin. The disc is clamped from above via a solid steel "bridge," securely holding the disc as it is spun and read. The laser pickup is mounted on a sturdy sled and only allowed to move in three directions (horizontal, vertical, and circular) to reduce vibration and servo activity. The sled is mechanically isolated from the rest of the transport mechanism. Each mechanism is made by hand in Japan, and undergoes two days of quality-control testing.

The Vivaldi Transport is unusual in that it can upsample 44.1kHz/16-bit data read from a CD to DXD or to DSD. DXD (Digital eXtreme Definition) is PCM data at 352.8kHz/24-bit. This format offers a data rate of 8.4672Mbs, considerably higher than DSD's 2.8224 Mbs. DSD is Direct Stream Digital, the encoding method of SACD. The Vivaldi Transport outputs DXD or DSD on dual AES/EBU jacks. When outputting DSD, the data are encrypted with a proprietary encryption scheme. To use the Vivaldi Transport's upsampled DSD outputs, you'll need to connect it to the Vivaldi DAC (the DAC decrypts the signal). For those of you who prefer no upconversion, you can connect the transport to the DAC via a single

AES/EBU connection, or better yet, with the SDIF-2 interface. Not to be confused with SPDIF, SDIF-2 is a three-cable interface developed for professional audio. The three cables carry, respectively, left-channel audio, right-channel audio, and word clock. This connection method, with the clock signal on a separate line, greatly reduces sonically degrading clock jitter.

The Vivaldi DAC is highly flexible and capable, able to convert digital data of any commonly found sampling frequency or signal format to analog. In a typical configuration one pair of the DAC's AES/EBU inputs is connected to the Transport for decoding DSD, and another pair to the Upsampler's outputs. The latter is selected when listening to CDs played in the transport, to an Apple device connected to the Upsampler, or to network attached storage under iPad control. The analog output is available on independently-buffered balanced and unbalanced outputs, and is variable via a front-panel knob or up/ down buttons on the remote control. In a nice touch, a menu setting allows you to set the full-scale output level to 6V or 2V to match your power amplifier's input sensitivity and avoid large amounts of digitaldomain attenuation. A useful display indicates the volume level. Of course, you can leave the volume at maximum and adjust the level with your preamplifier. Six filters are offered for PCM decoding and four for DSD decoding. The PCM filters include one with no pre-ringing. The DSD filters vary only in the filters' cutoff frequencies. Digital-to-analog conversion is performed by dCS' famous Ring DAC, a novel and ingenious solution to a fundamental problem in digital-to-analog conversion (see sidebar).

The Digital-to-Digital Upsampler fits between the Transport and DAC, upconverting audio from its native sampling rate to high-resolution PCM (up to 192kHz/24-bit) or DXD (352.8kHz/24-bit). The Upsampler's AES/EBU, RCA, and BNC connections can output PCM up to 192/24, with the dual AES/EBU jacks supporting DXD. It is also a digital hub with multiple inputs that can stream audio from a computer or a network-attached storage device such as a hard drive. The inputs include the network connection on an RJ45 jack, two asynchronous USB inputs (one Type A, one Type B), four SPDIF (two RCA, one BNC, one TosLink), and one SDIF-2. The Upsampler also supports the emerging DSD-over-USB protocol (which dCS developed and offered to everyone free as an open standard). This means that you can play DSD files from a server through the Vivaldi. The Upsampler also supports direct-digital playback from Apple devices, bypassing the Apple device's internal DAC. The Type A USB input will play files stored on a USB stick, with navigation and music selection provided by the

front-panel display. dCS offers an iPad app that allows you to browse and play your music library from a server or network attached storage device through the Upsampler.

The Vivaldi Master Clock isn't in the signal path, but instead sits outside it generating a reference clock to which the Transport, DAC, and Upsampler are locked. Without the Vivaldi Clock, the Transport, DAC, and

Upsampler run on their own clocks (which are themselves high precision), but those clocks are not perfectly synchronized with each other. In addition, the Vivaldi Master Clock improves the clock precision by an order of magnitude, from +/-10 ppm to +/-1 ppm. The advantage to the Master Clock is that all the digital operations occurring in the various chassis are all locked to the same, higher-precision, reference. Note, however, that when using the Transport and DAC alone, the Transport will lock to the DAC via the SDIF-2 interface's separate clock line.

Each of the four chassis has a display window showing operating status, including which input is selected, the upsampling ratio, whether the unit is locked to the Master Clock, etc. It took some experience with the system before I understood fully how it operated. The menu system is so extensive that the Vivaldi is, as previously noted, supplied with a menu hierarchy map. The Vivialdi is without question the most sophisticated and complex piece of consumer-audio hardware I've ever used. With one network cable and one

USB cable attached, a fully loaded Vivaldi requires 14 cable connections-not counting the analog output cables. In addition. there are so many different combinations of settings that it's easy to find yourself listening to one that is less than optimum. After a few weeks, however, I had it down to just selecting the input on the DAC or Upsampler and occasionally changing filters. RH



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Technology: Scarlatti vs. Vivaldi

The Vivaldi came to life as the result of dCS engineers asking, "What are the limitations of the Scarlatti?" That four-box flagship was designed in 2007 as a cost-no-object project, and has stood the test of time. Still, six years is a long time in today's world given changes in component parts, increasing options for accessing high-res music, as well as advances in dCS' thinking and expertise. The decision was made to examine every aspect of the Scarlatti and create a new reference that would transcend Scarlatti. The Vivaldi is the result of that three-year development effort.

For starters, the digital-signal processing platform in each of the four chassis was redesigned from the ground up to take advantage of new field-programmable gate arrays (FPGAs) and DSP chips. The new chips enabled dCS engineers to greatly increase the system's maximum processing capacity, speed of data transfer, and overall performance. The Vivaldi hardware platform has 200 times the processing power of its predecessor. The additional power not only streamlines operations, but allows for future capabilities via software updates. In addition, some parts of the code that were previously in firmware are now in software, allowing easier upgrades. An example is the software that converts PCM or DSD data to the five-bit Ring DAC code; when dCS improves upon this algorithm you can load the new code via a CD update rather than replacing ROMs. The control-system circuit boards are now eight-layer rather than four-layer. Eight-layer boards are quite rare in high-end audio products.

One of the biggest differences between the Scarlatti and Vivaldi is the latter's use of new single-element latches in the Ring DAC (see sidebar). The Scarlatti employed 22 latch chips, each containing four individual latches. The Vivaldi's 96 separate latches confer a significant advantage in that there's no crosstalk between the latches as occurred when four latches were contained on a single chip. In addition, the new discrete latches offer higher performance.

The analog output stage is all new. It is 6dB quieter and has 20dB greater channel separation than the Scarlatti (100dB vs. 120dB). The two halves of the balanced signal are also more closely matched in performance, as are the left and right channel characteristics. The analog board's traces are routed by hand, not by an automated computer program. This allows the designer to optimize the layout and parts placement for the best performance.

The user interface is also more intuitive and streamlined. Certain functions that the user once had to manually perform when switching inputs, for example, are now automatic. The Vivaldi's greatly improved industrial design and cosmetics are readily apparent to anyone familiar with





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The dCS Ring DAC

The Ring DAC, invented by dCS in 1987, is a brilliant solution to the challenge of converting digital data to an analog output signal. To understand the Ring DAC, let's first consider how conventional DACs work. You can think of a multibit DAC as a ladder, with as many rungs on that ladder as there are bits in a sample. A 24-bit DAC will have 24 "rungs," each one a resistor that corresponds to each bit in the digital sample. The resistors are connected to a current source through a switch; the digital data representing the audio signal open or close the switches to allow current to flow to the output or not. The currents of each rung are summed, with that summed value representing the audio signal's amplitude.

The resistor values are "binary weighted." This means that each resistor lower down on the rung must have double the resistance of the rung above it, and so forth, corresponding to the binary progression 1, 2, 4, 8, 16, and so on. Because each bit in the digital code represents twice the value of the next lower bit, each resistor must have a value exactly half that of the resistor on the rung below it. One problem with these so-called "R-2R ladder" DACs is that it's impossible to make resistors with the precision required for perfect binary weighting. The result is that the tolerances in resistor values introduce amplitude errors in the analog output. Moreover, those amplitude errors will occur in the same places on the audio waveform.

This problem becomes more acute the greater the number of rungs on the ladder. In a 16-bit resistor-ladder DAC the value of the lowest resistor rung should be exactly 0.0000152 the value of the highest resistor rung. In a 24-bit converter the lowest resistor value should be precisely 0.000000119209289550781 the value of the highest resistor. It is obviously not possible to achieve anywhere near this level of precision in resistor manufacturing, and any deviation from the resistor ratios translates to amplitude errors in the analog output.

The now-defunct UltraAnalog company addressed this challenge by driving its 20-bit DACs (which were composed of two off-the-shelf 16-bit DACs ganged together) with 100,000 different digital codes, measuring the DAC output at each code value, calculating the degree of error in each specific resistor, and then having technicians hand-solder tiny precision metal-film resistors on the ladder rungs to bring them closer to the correct value.

A DAC technology that doesn't rely on binary-weighted resistor ladders is the one-bit DAC. This device converts a multi-bit code into a single-bit data stream that has two values, one and zero. Unlike a multibit DAC, the one-bit DAC's amplitude precision is very high, but the one-bit DAC suffers from very high noise that must be "shaped"

(shifted away from the audioband). Onebit DACs are also very susceptible to jitter. dCS's solution is the Ring DAC, which can be considered a hybrid of the two approaches. It is based on a five-bit code that drives resistors of identical value. Because the resistors in dCS' Ring DAC are all the same nominal value their actual values are very close to one another. The five-bit code has a much higher signal-to-noise ratio than a one-bit datastream and requires an order of magnitude less noise shaping.

Digital signal processing first "maps" whatever datastream is coming in (192kHz/24-bit, or the 2.8224MHz 1-bit code of DSD, for examples) into a unique five-bit code. This five-bit code opens and closes latches connected to a current source that drives one of five resistors of identical value. Because these resistors can never have exactly the same resistance, the Ring DAC employs an array of resistors and randomly shifts the audio signal between resistors in the array. The Ring DAC gets its name from this "passing around" of the signal from one resistor in the array to another, as in a ring. The effect is to convert what would be amplitude errors in the analog output into a very small amount of random white noise.

The Ring DAC is brilliant in concept, and achieves its highest realization in the Vivaldi. The commonality in sonic character between all dCS products—the density of information, the resolution of fine detail, the unique spatial qualities—are probably attributable in large part to the Ring DAC. RH



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Inside the dCS Factory

I visited dCS' sparkling new factory last September to get the full technical briefing on the Vivaldi and to see first-hand how this extraordinary product is built. Last year dCS moved from an older building south of Cambridge to a brand new and very modern facility ten minutes north of the city. The new factory was built-out specifically for dCS to give the company greater efficiency of manufacturing.

dCS was founded in 1990 to build high-precision electronics for military applications, including ultra-precise analog-to-digital and digital-to-analog converters. The company then began applying its technology to professional audio, where dCS earned a reputation for extraordinary sound quality. dCS introduced its first consumer product, the Elgar DAC, in 1997. Since then the focus has been on pushing the envelope in high-end digital-playback systems. The company has 17 employees, five of them in engineering. The two leading engineers have between them more than 40 years at dCS. When visiting a company I always note the ratio of engineers to sales and marketing personnel; it gives you an idea of whether the company is technology-driven or marketingdriven. dCS is most assuredly an engineering-led firm.

The Vivaldi's industrial designer, Ray Wing, gave me a fascinating in-depth look at the design process via 3-D drawings on his computer. dCS wanted a distinctive new look for the Vivaldi and it achieved that goal. The front panels have gentle three-dimensional waves that are elegant and visually interesting, but exceedingly time-consuming to machine. Creating one

front panel for a Vivaldi component takes four hours of CNC machine time.

As with all dCS products, the Vivaldi is softwareintensive. All the control systems, upsampling, digital filtering, input receiver, and the algorithm that converts PCM or DSD to the five-bit Ring DAC code are created in-house. The metal work and printedcircuit board stuffing are performed by local outside vendors, with assembly in dCS' factory. Each board undergoes testing before assembly, and repeatedly during the build process. Some of these tests take four hours to complete on an automated test-jig. The critical clocks inside Vivaldi are individually calibrated by putting them in an oven for four days, varying the temperature, and monitoring the clock's frequency drift with the changing oven temperature. A support circuit is individually calibrated for that particular clock's characteristics based on the measured data. Each component of the Vivaldi system (Transport, DAC, Upsampler, Clock) is visually inspected by three different individuals before the unit is boxed-the test technician, the production manager, and either someone from marketing or the president of dCS himself. RH











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Jacob Heilbrunn On the Scarlatti and Vivaldi

Digital playback has long been the problem child of high-end audio. Suggest to a diehard vinyl lover that there might perhaps be some redeeming qualities about digital recordings and, more than likely, you will be met with a frozen or even pained smile. The implication is clear: For the true connoisseur seeking the audio truth and nothing but the truth, it is a foolish deviation, a trap and a snare, to listen to digital recordings. And for a number of years the disdain has not been wholly unjustified. Vinyl has always has had an inherent relaxation and warmth that digital can only envy from afar. And yet in recent years, the gap has been narrowing between the two formats.

One company that has been at the fore-front of that welcome development is dCS. A few years ago I sat up with a jolt when listening to the dCS Scarlatti playback system. There was a resolute quality to the bass and an abundance of detail that I had simply never heard before on digital. I bought it. Now dCS has upped the ante with its new Vivaldi system, which I first heard in New York at Ears Nova, where dCS introduced it to an American audience. Since then, I have had the opportunity to audition it in my own system.

Actually, that's baloney. I haven't been auditioning the Vivaldi. I've been reveling in it. While the Scarlatti was an excellent per-

former, the Vivaldi visually and audibly surpasses it in several important respects. For one thing, its casework is more impressive than the Scarlatti's-heavier and more inert, rendering it less susceptible to vibrations. It also looks more attractive than the Scarlatti.

But none of that would matter so much if the Vivaldi didn't offer superior performance. The differences with the Scarlatti are instantly apparent. It seems to be even tighter in the nether regions than its predecessor-on Christian McBride's sensational CD Conversations with Christian I was awestruck at the speed and energy of his bass. The notes seemed to fly into my room at warp speed. The Vivaldi also has more control and grip than the Scarlatti. It has more extended decays that seem to linger on into infinity. And it has a much more refined and extended treble. Slam and dynamics are second to none. I could keep going down the audio checklist, but it's like breaking down a fine painting and discussing its individual attributes without recognizing its overall beauty.

Which is to say that these sonic attributes result in the most significant aspect of the Vivaldi when contrasted with the Scarlatti. The Scarlatti was unable to efface the sense of a slightly aggressive treble region. Not so with the Vivaldi. Plunk a CD in the tray or stream a high-resolution file, and the music

seems to simply appear out of the ether with a sense of utter relaxation. The Vivaldi has. by and large, banished the sense of electronic reproduction and it has a nuance and filigreed sound in the treble that the Scarlatti lacked. There is an addictive and sensuous quality to the sound that approaches what, for better or worse, is usually called analoglike. Mind you, I'm not saying that the Vivaldi is superior to the Continuum Caliburn I use or other top-flight 'tables. What I do mean to say, however, is that given the quality of the Vivaldi, I am perfectly happy listening to CDs and that I don't find quarreling over the distinctions between the two formats particularly rewarding or edifying.

The blunt fact is that digital playback is reaching new heights. There is a gentleness and absence of grain, particularly in the mids and highs, that place the Vivaldi on an entirely different plane from the Scarlatti. Listen to the Lorraine Hunt Lieberson on Handel's *Julius Caesar* [Harmonia Mundi] and I defy you not to feel goosebumps at the pellucidity of the sound.

To my mind, the Vivaldi represents a revolutionary advance in digital playback. Over the past several decades, dCS has steadily refined its digital products. Each generation has represented an improvement over the previous one. Now dCS has surpassed itself with the Vivaldi. Indeed, when listening to it, I was reminded of the immortal remark by *Gilligan Island*'s Thurston Howell III: "No one can pull the wool over my eyes. Cashmere maybe, but wool, never." Friends, the Vivaldi will pull cashmere over your eyes.



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Oppo BDP-105 Universal/Blu-ray Player and DAC

Gives "Flexible Flyer" A Whole New Meaning

Chris Martens





n recent years Oppo Digital has followed a simple recipe for success: Just build universal disc players that offer greater versatility, more audiophile-friendly features, and more sensible pricing than the competition does, and then give them decisively better sound and picture quality than their peers. Naturally, this laudable goal is a lot easier to describe on paper than it is to achieve in the real world, but Oppo has made good on its promises, year after year and player after player, in the process earning a reputation as the nearly automatic "go-to" source for players that will satisfy discerning music (and movie) lovers on a budget.

Historically, many of Oppo's most popular players have sold for around \$499. But with the 2011 release of its BDP-95 universal/Blu-ray player (\$995), the firm began to explore a more upscale market. What set the BDP-95 apart was that it was not merely a "hot-rodded," sonically tweaked version of a standard Oppo player; rather, it was a unique, dedicated high-end model with a distinctive configuration all its own.

The award-winning BDP-95 sounded remarkably good both for its price and in a broader sense. Never a company to rest on its laurels, however, Oppo has recently announced the successor to the BDP-95; namely, the BDP-105 (\$1199)—a player that promises to do everything its predecessor could do and then some.

Like its predecessor, the BDP-105 can handle virtually any format of audio or video disc, including Blu-ray Video, Blu-ray 3D, DVD-Vid-

eo, DVD-Audio, SACD, CD, HDCD, and more. But with the BDP-105 the universality theme doesn't end with disc playback because the new player is also designed to serve both as a network-streaming player as a multi-input high-resolution DAC (complete with asynchronous USB).

To really "get" what the BDP-105 is about, think of it not so much as a powerful multi-format disc player (although it is that and more), but rather as a multi-function digital media playback hub whose bag of trick includes, but is in no way limited to, disc playback. In practical terms, this means the BDP-105 neatly resolves debates about whether it is better to listen to discs, to stream content from the Internet, or to enjoying audio files stored on computers, because it can guite happily do all of the above.

The BDP-105 comes housed in an all-new steel chassis said to be significantly more rigid than the chassis used in previous Oppo players (including the BDP-95), and it benefits from a fan-less architecture, meaning all internal components are convection-cooled (most previous Oppos required fan-cooling). Do such seemingly small detail changes like a more rigid chassis or a fan-free design make for meaningful sonic improvements? My opinion, based on extensive comparisons between the BDP-105 and 95, is that they do. Specifically, the new player offers a noticeably more solid and "grounded" sound with guieter backgrounds, improved resolution of low-level transient and textural details, and superior three-dimensionality.

Moving on, the BDP-105 uses a beefy toroidal power supply and provides both 7.1-channel analog audio outputs plus two separate sets of

Oppo BDP-105

stereo analog outputs (one single-ended and the other fully balanced). Interestingly, the BDP-105 (like the BDP-95) features not one but rather costly 8-channel ESS Sabre32 Reference DACs, one to feed the 7.1-channel outputs and the other to feed the two sets of stereo outputs. ESS's Sabre32 Reference DACs are used in some very pricey components, making it impressive that Oppo fits two of the devices into its sub-\$1200 player.

Another new touch is that the BDP-105 provides a built-in headphone amp that runs straight off one of the player's ESS Sabre32 Reference DACs. While the headphone amp offers relatively modest output, it has the undeniable benefit of being fed directly from one of the Oppo's ESS Sabre32 Reference DACs, so that it gives listeners an unusually pure, uncluttered, intimate, and up-close perspective on the music (precisely what you would want for monitoring applications, for example). I found the Oppo headphone amp had more than enough output to drive moderately sensitive headphones such as the HiFiMAN HE-400s or PSB M4U1s, though it might not have sufficient "oomph" for more power-hungry top-tier 'phones (for instance, the HiFiMAN HE-6).

While the original BDP-95 offered a reasonable range of Internet-content options and could play digital audio files from USB storage devices or eSATA drives, it was never set up to function as multi-input playback device or as a high-resolution audio DAC. The 105 changes all this by offering a greatly expanded range of general-purpose inputs, including two HDMI inputs (one that is faceplate-accessible and MHL-compatible) and three USB 2.0 ports (one that

is faceplate-accessible). Moreover, the BDP-105 also provides three dedicated DAC inputs: two S/PDIF inputs (one coaxial, one optical), plus one asynchronous USB input. Finally, to complete the connectivity picture the new player provides both Ethernet and Wi-Fi network connections implemented, respectively, through a rear panel-mounted RJ-45 connector and a handy USB Wi-Fi dongle.

To take full advantage of these network-connection options, the BDP-105 offers DLNA compatibility, complete with support for DMP (Digital Media Player) and DMR (Digital Media Renderer) protocols. In practice, this means the BDP-105 can access audio, picture, and video files stored on DLNA-compatible digital media servers (that is, personal computers or network-attached storage devices) that share a common network with the Oppo within your home.

From this technical overview, you can see that the BDP-105 is an extraordinarily flexible source component, but for most audiophiles the key question is, and always will be, "How does it sound?" Let's focus on that question next.

From the outset, the BDP-105 struck me as being a very high-resolution player—one that made child's play of digging way down deep within recordings to retrieve small, essential pieces of musical information that helped convey a sense of realism. To hear what I mean, try the track "O Vazio" from the Jim Brock Ensemble on a sampler disc (in HDCD format) from Reference Recordings. Throughout this track the Oppo did a stunning job of rendering the distinctive attack and action of each of the

instruments in the ensemble (accordion, bass, drum kit, guitar, trumpet, winds, and other more exotic percussion instruments), giving them a commanding sense of presence with precisely focused placement within a wide, deep, three-dimensional soundstage. In particular, the 105 showed terrific speed and agility on the leading edges of notes (especially on the drums), rendering them with the sort of clarity

and impact that reminded me of the sound of far more costly players.

Another song from "Jordan" from the Brock/Manakas Ensemble, contains a brief, quiet passage that reveals another important aspect of the BDP-105: namely, its impressive ability to maintain focus and resolution even when playing at very low levels. After the introduction of the song, which lasts about 35 seconds, the music comes to a dramatic pause that eventually is broken by the extremely faint sound of a cymbal (or small gong?) gently introducing the rhythmic pulse that will supply a heartbeat for the rest of the song. At first, the cymbal is heard so softly that its sound barely rises above the noise floor, yet even so the Oppo gets the sound of the instrument right, preserving all the essential elements of attack, timbre, and decay. This uncanny ability to resolve very-low-level musical information enables listeners to here all the little interactions between instruments and the acoustic spaces in which they are playing. While the original BDP-95 did

a fine job in this respect, I would say the BDP-105 sounds better still.

The voicing of the BDP-105 is generally neutral, with taut, deep, and well-controlled bass, transparent mids, and revealing, extended highs (highs that can, however, expose mediocre recordings for what they are). Pleasing though the Oppo can be, some might find it a bit lean-sound-

SPECS & PRICING

Disc types: BD-Video, Blu-ray 3D, DVD-Video, DVD-Audio, AVCHD, SACD, CD, HDCD, Kodak Picture CD, CD-R/RW, DVD-R/RW, DVD-R DL, BD-R/

Internal storage: 1GB
Inputs: Three USB 2.0 inputs
(one faceplate accessible),
two HDMI inputs (one
faceplate accessible and MHL
compatible), three dedicated
DAC inputs (one coaxial, one
optical, and one asynchronous
USB), one Ethernet port (RJ45), one Wi-Fi port (via USB
dongle)

Outputs: One 7.1-channel analog audio output, two stereo analog audio outputs (one set balanced via XLRs, one set single-ended via RCA jacks), two digital audio outputs (one coaxial, one optical), two HDMI outputs (can be configured for video output on one port and audio output on the other), one headphone output

DAC resolution: (USB Audio) 2 channels @ 192k/24b PCM, (Coaxial/Optical) 2 channels @ 96k/24b

Dimensions: 16.8" x 4.8" x 12.2"

Weight: 17.3 lbs.

OPPO DIGITAL, INC.

2629 Terminal Blvd., Suite B Mountain View, CA 94043 (650) 961-1118 oppodigital.com

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Oppo BDP-105



ing compared to the deliberately warmer-sounding offerings on the market. If you prefer components that give a voluptuous musical presentation then the Oppo might not be your cup of tea, but if sonic honesty and neutrality are your things you should get on very well with it.

Let me expand on my voicing comments by pointing out that the BDP-105 needs a of runin time to sound its best (some say as much as 200 hours or more). As playing time accumulates, traces of leanness and austerity gradually melt away, thus enabling the player to reveal a smoother, more full-bodied, and more forgiving sonic persona.

If you buy the notion that some source components try for a softer, smoother, and thus ostensibly more "musical" presentation, while others aim for maximum musical information retrieval, then I would say the Oppo falls squarely in the information-retrieval camp (as do a great many other high-performance solid-state play-

ers). Thus, tonal colors are rendered vividly through the Oppo, but without any exaggeration or oversaturation, so that there is nothing artificially sweetened, enriched, or "glowing" about the 105's sound. Instead, the Oppo is one of those rare "what you hear is what you get" sorts of players, whose primary mission is to tell you how your discs or digital music files actually sound, which in my book can be a beautiful thing.

As a disc player, the BDP-105 is more than good enough to show in palpable ways that well-recorded SACDs really do sound better than their equivalent CDs (there's greater smoothness and ease with SACDs, and simply more "there" there, so to speak). But as a DAC, the Oppo really comes into own, sounding much like it does when playing discs, but with subtly heightened levels of tonal saturation and warmth that make the music more engaging and intense.

Are there caveats? Apart from the extensive run-in requirements noted above, I can think of only a few. First, the BDP-105 is an inherently complex product that—at the end of the day—is simpler to navigate and control when it is connected to a display screen. Second, the player's sound is so unashamedly refined and sophisticated that you may feel inspired (if not compelled) to use top-tier interconnect cables that will wind up costing more than the player does. But trust me on this one: The Oppo's worth it.

If ever a product deserved to be considered the Swiss Army knife of digital media playback, the BDP-105 is the one. Whether you choose it for multi-format disc playback, for network-streaming capabilities, or to use as a DAC at the heart of a computer-audio system, the BDP-105 will consistently serve up levels of sonic refinement and sophistication the belie its modest price. Enthusiastically recommended. tas

Disc Player of the Year

Oppo BDP-105

The BDP-105 is more than just a disc player, though it will handle virtually any type of video and/or audio disc you'd care to name. In fact, depending upon your frame of reference, you might rightly consider it to be a powerful videoprocessing engine, a high-quality headphone amplifier, or a highly capable, multi-input highresolution DAC (complete with an asynchronous USB input). In addition to its versatility, the Oppo offers features geared specifically for audiophiles, including a beefy power supply, a fanless architecture for low noise, dual ESS Sabre32 reference DACs for ample decoding power and low jitter, separate multichannel analog outputs, and dedicated single-end and balanced stereo analog outputs. Clean, clear, and decidedly detail-oriented, the Oppo hews somewhat toward sonic leanness, but is far more revealing than it has any right to be for the money. (CM, 232)



EQUIPMENT REVIEWS

Music Servers

PEN & PAPER. THELMA & LOUISE. CHIPS & SALSA.

The MS-2 Music Server and new, limited edition DAC-1 LE. Some things are just better together.



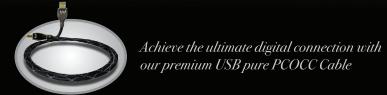
MS-2 MUSIC SERVER

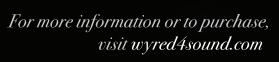
Quick & Easy One-Time Setup
Supports Up To 32 Bit/384kHz PCM & DSD 128
Built-in CD Drive Automatically Rips To FLAC
Wireless Control From Your Mobile Device
Drag & Drop File Uploading
2 TB Hard Drive, 2 GB RAM
\$2499 (Optional Solid State Drive Available)

New Limited Edition DAC-1 LE

Industry Reference ESS Sabre (ES9018) 32-Bit DAC
ESS Time Domain Jitter Eliminator®
Supports Up To 32 Bit/384kHz PCM & DSD 128
Proprietary Discrete Output Stages
Fully Balanced Design
Galvanic USB Isolation
\$999 (Same As Original DAC-1)











Bluesound Audio Ecosystem

So Long, Sonos!

Spencer Holbert

n the not so distant past, multi-zone audio systems were expensive, cumbersome, and time-consuming propositions that required a team of installers, the routing of cables through walls and attics, and a whole lot of patience. Add multi-zone rack preamps, multiple amps, multiple sources, multiple speakers, in-wall control panels, and hundreds of feet of wire, and installing a multi-zone audio system was akin to remodeling your house. Then a company called Sonos hit the boards in 2002, and since then multi-zone audio has become simpler and cheaper with each passing year. However, while the new wireless systems were easier to install and use, they were also limited to MP3 and CD-quality audio, and that quality was further limited by inferior parts and sonics. But what if you could combine the ease of wireless streaming, high-res sound, and legendary audio components, all in one system? Well, that's exactly what Bluesound has accomplished, and the result is true high-end audio with twenty-first-century connectivity.

What's Bluesound?

While Bluesound might be a new name in the hifi world, Lenbrook-the parent company of NAD, PSB, and now Bluesound-has a long history of designing high-performance components and speakers at affordable prices. At the heart of Bluesound are the design philosophies of Paul Barton, founder of PSB, and the engineers at NAD Electronics. With powerhouse audio companies backing Bluesound, the resulting line of products is spectacular. The Bluesound "ecosystem" consists of five components: The Node, the Powernode, the Vault, the Duo, and the Pulse. Because the Node and Powernode are essentially the same product, except that the Powernode has a built-in amplifier, the review samples I received included everything but the Node.

Bluesound Vault

The Bluesound Vault has been my new best friend for the past three months. The Vault is a music server, hard drive, and CD-ripper in one sleek box. The Vault sports 1TB of internal hard-drive storage, and ripping CDs to it is as easy as such things get. Insert the your CD into the front slot and give the Vault five minutes to rip; the Vault then automatically acquires album artwork and metadata, and ejects the CD when it's done. If you have a large CD collection and want to finally store everything on a music server that's affordable and functional, the Vault is far simpler to use than a computer-based system-and a whole lot cheaper. While the ripping process is slower than it is with a really fast computer drive, the fact that you don't have to do anything besides insert CDs is a huge time-saver.

But the power of the Vault isn't simply its

ability to rip CDs; the Vault also allows you to consolidate your digital music management in one sleek, easy-to-use system. I've had a computer-based music server in my home for years, so all of my CDs have long since been saved to Network Attached Storage devices (NAS drives). When I connected the Vault to my home network via Ethernet (the Vault requires Ethernet connection in order to deliver high-res music to all devices), it found my NAS drives, and after I selected "Re-index Music Collection" in the Bluesound iPad app the Vault went to work. The initial indexing took about 20 minutes for 10,000 songs, and when the process was finished I had full access to all my digital music-some of which was stored on the Vault, and some on multiple NAS drives. I now have the ability to search for music, create playlists, stream music to other Bluesound devices, output to an external DAC via the optical commnector, and even stream to multiple devices simultaneously-all from the iPad app.

The Vault has a built-in DAC capable of playing PCM audio files up to 192kHz/24-bit. While I found its analog output more than adequate for those just getting into highres audio, consumers with a component or reference DAC will want to use the optical output from the Vault to send music to their reference systems.

The Vault goes way beyond locally stored digital audio, though. It also allows you to stream Internet radio via Rdio, Slacker, and Tuneln, as well as streaming music services like Qobuz, Deezer, Wimp (not yet available in the U.S.), and Spotify. While Internet radio isn't high-res, it is still great to be able to stream NPR in the mornings, or use Spotify

EQUIPMENT REVIEW - Bluesound Audio Ecosystem

to discover new music and create playlists, or stream radio to multiple devices-again, all from your smartphone and iPad.

If having control of and access to all of your digital music-both locally stored and Internetstreamed-isn't reason enough to own a Vault. then maybe the fact that this little box does everything while maintaining a low profile is. You can place the Vault anywhere in your home as long as it's connected to your network via Ethernet, and if you want to use it in your reference system with a component DAC, the Vault is dead silent. Many NAS drives make noise when they spin up, and larger multi-drive systems like my Drobos have fans that can create unacceptable levels of noise for a listening room. The Vault never makes itself known, runs silent and generally cool, and is easily hidden away. There's no need for a keyboard or monitor for initial setup, as with a Mac Mini, and the ability to stream highres audio and Internet services to multiple devices is killer. In all honesty, now that I've had the Vault in my home, I don't think I can live without it. Finally, one device that does everything without blowing the budget!

Bluesound Powernode

Most people want the ability to play music in more than one room, but that generally means buying a complete second system with sources, cables, and all the accouterments of a stand-alone stereo system. Enter the Bluesound Powernode, an integrated amplifier and network-attached (wireless or Ethernet) music-management device. The Powernode has many of the same features as the Vault, without the CD drive and internal storage. Connect it to your home network, launch the Bluesound app, and the Powernode will find your NAS drives and index all of your digital music. Every Bluesound device is controlled using the same app, so the Vault and Powernode work in a symbiotic relationship (or individually) to provide you with music whenever and wherever you want.

The Powernode is a "just add speakers" kind of system. With 150W into 8 ohms and 80W into 4 ohms, the NAD-designed DirectDigital amplifier has plenty of juice to really rock out. The Powernode also has an RCA subwooferout, so connecting a 2.1 system is a breeze.

I connected the Powernode to the Bluesound Duo, which comprises a pair of bookshelf speakers and a high-quality 8" 280W subwoofer designed by none other than PSB. I set up the Powernode and Duo combo (you can use any speakers with the Powernode, not just the Duo) in several different rooms during the course of my audition, and it was simply amazing to just plug the Powernode in, connect the Duo, and then stream music via the Bluesound app. You can wirelessly control volume, listen to your favorite local or international radio stations, build playlists from the music stored on the Vault (or other NAS drives), stream Spotify, and do all of this with little to no work or bother. For parties, I streamed music via the Vault to my main system, to the Powernode, and to the Pulse (which I will discuss), and controlled everything wirelessly as I walked around the house. No running back and forth to adjust volume, change tracks, or even sources. You can even control multiple sources for each Powernode you install. Say you want jazz in the kitchen, blues on the back patio, and highres classical on your reference system. With the Bluesound Powernode, you can have an

independently managed music system in each and every room, or stream the same high-res music to all rooms simultaneously; the choice is yours.

While the Powernode is a really great digital integrated amplifier, I would by no means call it a replacement for a reference system. When it comes to comparably priced alternatives, though, the Powernode blows away the competition. Highquality amplification from NAD, wireless control and integration, and the ability to stream high-res audio in native 192kHz/24-bit will have you singing "So long Sonos!" At \$699, the Powernode is one of the cheapest and fastest ways to have a high-res system anywhere you want-just plug it in! And if you already have a system and need a way to stream and manage digital music, just add the Node, which only costs \$449. You get all of the features of the Powernode. minus the amplification.

Bluesound Pulse

The Bluesound Pulse isn't just a "lifestyle" product; it's everything you need to add music to a room in one box. With the same great Direct Digital amplification as the Powernode, a 25-bit/844kHz DAC, a built-in high-end speaker, and the capability of wirelessly streaming high-res music from the



SPECS & PRICING

BLUESOUND ECOSYSTEM

Vault

Storage capacity: 1TB internal (infinitely expanded with NAS drives)

Formats: MP3, AAC, WMA, FLAC, ALAC, WAV, AIFF

DAC: PCM audio up to 192kHz/24-

Inputs: Ethernet network Outputs: RCA, optical, Ethernet

Weight: 6 lbs.

Dimensions: 11" x 9" x 8"

Price: \$999

Powernode

Inputs: Wireless/Ethernet streaming up to 192kHz/24-bit Power: 50Wpc into 4 ohms

Weight: 4.5 lbs.

Price: \$699

Dimensions: 9" x 8" x 7"

Pulse

Inputs: Wireless/Ethernet

streaming up to 192kHz/24-bit

Amplification: Bi-amped DirectDigital 80Wpc

Frequency response: 45Hz-20kHz

Weight: 13 lbs.

Dimensions: 15" x 7" x 7"

Price: \$699

Duo 2.1 Speaker System

Frequency response: 36Hz-23kHz Powered sub: 280W peak, Class H

MOSFET

Weight: 24 lbs. (sub), 5 lbs.

(speakers)

Dimensions (sub): 9.5" x 15" x 14" Dimensions (speakers): 5" x 7"

x 8"

Price: \$999

BLUESOUND

633 Granite Court Pickering, Ontario L1W 3K1 Canada (905) 831-6555 bluesound.com

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EQUIPMENT REVIEW - Bluesound Audio Ecosystem

Vault or NAS drives, the Pulse sounds great for what it is. Are you going to be blown away by incredibly three-dimensional soundstages? No. But the Pulse is an instant means of accessing and playing back high-quality music. During BBQs, I brought the Pulse to the backyard and everyone loved it. Stream all that high-res music from your network, create playlists, stream Internet radio or Spotify, all from the same great Bluesound app.

Plug in the Pulse, connect to it via the Bluesound app, go through the painless set-up process, and within five minutes you will be playing your favorite tunes. And if you need to move to another room, just unplug it and move it with you; the Pulse automatically reconnects to your home network once it's plugged in again, and needs only 30 seconds or so to boot up. Again, you can "group" all of your Bluesound devices to play the same music in every room, or stream from multiple sources to have different music playing in each room.

Bluesound Duo

The Bluesound Duo 2.1 speaker system is another symbiotic piece of the Bluesound ecosystem. At \$999, the Duo isn't the cheapest option for the "just add speakers" Powernode, but what you get are high-quality transducers that work seamlessly with the NAD technology employed in the Powernode. As mentioned, the 280W powered subwoofer with 8" driver really fills out a room with added low end. (The system uses digital equalization with the DirectDigital amplification found in the Powernode to enhance the performance of any 2.1 system.)

I used the Powernode/Duo combo at my desk in a nearfield setup, and also in bedrooms and living rooms, and the sound quality was exceptional (which you would expect from a PSB-backed design). Especially in the nearfield setup, it was easy to achieve great soundstaging and imaging with the Duo, and the integration of the subwoofer was seamless using the Duo's crossover, phase switches, and volume controls.

One of my concerns with having all of these stereo components plugged in is energy consumption; however, the total standby power consumption of the subwoofer is only 0.5W, which allows me to leave everything "on" without watching my electric meter whirring away. Again, the Duo isn't for those looking for any ole cheapo speaker to play back music; these are high-quality, audiophilegrade speakers for your secondary system.

The Bluesound Ecosystem: Final Thoughts

Each of the Bluesound products works synergistically in a home ecosystem capable of wireless and Ethernet audio streaming, and all are controlled from a sleek tablet or smartphone app. When used in conjunction with a home network, audio can be streamed to each device using the same source or multiple sources. Each device's volume and input can be controlled individually and wirelessly, or devices can be grouped together and controlled as one. Highres audio can be streamed to all devices in native resolution, and all of your digital music can be managed from the Bluesound app. Even if you already have NAS drives full of music, the Bluesound ecosystem will be able to manage everything from one spot.

While the Bluesound ecosystem can be used without being hooked up to the 'Net, a high-speed Internet connection will transform it, allowing you to stream from Internet radio and music services like Spotify. In an all-wireless system, the Bluesound ecosystem can handle up to eight different devicesplenty for most people. But if you have a very large house, and want music streaming in every room-both inside and out-you can manage up to 34 different devices when everything is connected via

Ethernet to your home network. In essence, you can have a 34-zone audio system, capable of streaming from multiple sources. With such a large system, make sure you are using gigabit routers and switches, have an extremely fast Internet connection (50 megabits per second or faster), and are wiring with CAT6 Ethernet cables. The point is, though, that such a system would have cost an extremely large amount of money even ten years ago, and would be limited to CD-quality audio. With Bluesound, you can build a huge multi-zone/multi-source whole-house audio system that is easy to set up and easy to control, and has the ability to stream high-res audio, all for a fraction of the price (and hassle!) of professionally installed systems. Given that Bluesound is essentially a



twenty-first-century combination of the best of NAD and PSB, you know you're getting highquality products.

I can't stress strongly enough how amazing it is to be able to stream high-res music to all my rooms, and control everything from an incredibly intuitive smartphone and iPad app. I recently purchased a Victorian-style home built in 1910, and was worried about filling this beautiful 104-year-old house with wires and marring it with holes in the walls and floors. Now, everything can be accomplished wirelessly, set up in an hour (not days), and I don't have to do damage to my historic digs. I will be purchasing the review samples and a few additional Bluesound devices—that's how much I like these products.

Thank you Bluesound, and so long Sonos! tas



Sony HAP-Z1ES HDD Audio Player

Game Changer

Steven Stone

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

The Grand Tour

What is an HDD audio player? In the case of the HAP-Z1ES, it is a local network-aware device that plays digital music files. It hooks up via Ethernet or Wi-Fi to your local network and the Internet. The HAP-Z1ES contains a 1TB hard drive

for storing music files; it also has the ability to use external USB drives for additional storage. And what can the HAP-Z1ES store and play? It supports virtually any format audio file, including: DSD (WSF and DSDIFF), WAV, AIFF, FLAC, ALAC, ATRAC, MP3, AAC, and WMA files.

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

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

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

tell you more about the app later in the review.

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

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

A low-phase-noise liquid-crystal oscillator handles internal digital timing in the HAP-Z1ES, which acts as the master clock for all digital signals. According to Sony's measurements, the low-noise liquid-crystal

EQUIPMENT REVIEW - Sony HAP-Z1ES

oscillator delivers 20-30dB lower noise than conventional clocks.

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

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

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

it can operate with maximum efficiency and minimum noise.

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

The Setup

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

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

After connecting the HAP-Z1ES I turned it on and went to the "Network Settings" section of the main menu. There I selected "wired set-up" and "Auto" from the IP address page. After that, the HAP-Z1ES linked to my network and I saved the configuration. For

users who like reassurance, the HAP-Z1ES lets you check and confirm that the settings are "OK" before closing the network settings pages. The procedure is much the same for wireless Wi-Fi, except you have a page that lets you select your access points. If you live in a Wi-Fi-intensive environment you can pick the correct Wi-Fi network and enter your password. Near the end of the review period I switched over to Wi-Fi access and had no issues with changes to the installation or impaired Internet performance.

Once the HAP-Z1ES is connected to your home network, either via Ethernet cable or via Wi-Fi, you can transfer music files to its internal hard drive. Unlike many music servers

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

Initially I had some problems

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

SPECS & PRICING

Frequency response: 2Hz-80kHz +/-3dB

Dynamic range: 105dB or higher THD: 0.0015% or less

HDD capacity: 1TB

Supported playback formats: DSD

(DSF, DSDIFF), LPCM (WAV, AIFF), FLAC, ALAC, ATRAC Advanced Lossless, ATRAC, MP3, AAC,

WMA (2 channels)

Outputs: Unbalanced 2.0V RMS (50k ohms): balanced 2.0V RMS

(50k ohms), 600 ohms

External ports: Type A USB for

hard drive, IR Remote-Out jack

for IR blaster

Power consumption: 35W (on),

0.3W (off), 2.8W (standby)

Dimensions: 17" x 5 1/8" x 15 3/8"

Weight: 32 lbs. Price: \$1999

SONY ELECTRONICS INC.

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EQUIPMENT REVIEW - Sony HAP-Z1ES

flawlessly.

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

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

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

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

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

In addition to playing music from your music library, the HAP-Z1ES also has a built-in Internet radio tuner. Called the "V-Tuner," this feature includes the ability to search for Internet radio stations by genre or location. It also lists the bit rate of each station so you

can see exactly what quality level a station can deliver. I quickly found the local stations that I listen to regularly and designated them as "favorites" via a heart symbol icon, which added them to a special list that I could access more easily.

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

The HAP App and HAP-Z1ES Remote

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

The HAP control application will be available

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

Dav-to-Dav Use

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

While I didn't find Sony's SenseMe feature of particular value, I'm sure most users will find some use for it, if only to annoy significant

EQUIPMENT REVIEW - Sony HAP-Z1ES

others by selecting "lounge." One feature I did enjoy was the "Favorites" selection feature in the V-Tuner. I was able to assemble a very nice list of higher-bit-rate Internet radio stations in a short time by using V-Tuner's search features.

The Sound

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

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

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

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

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

The most difficult and least conclusive A/B test I performed during the review was comparing the DSD Remastering Engine's DSD conversion of PCM files with those same files played back without the DSD Remastering Engine engaged. When switched back and forth there was a pause followed by about a two seconds of playback of the last snippet of music before the switchover. During that two seconds the sound was slightly different,

seemingly warmer and rounder, but after that initial two seconds the sound reverted, and in blind A/Bs I could not tell whether I was listening to Remastering Engine or native output. I used both 16/44.1 and 24/96 PCM files for this test and didn't hear any differences when I switched between DSD and PCM on standard Red Book or higher-definition digital files.

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

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

The High Value HAP-Z1ES

In overall sonics and build-value for the dollar, the Sony HAP-Z1ES sets new standards. A Mac Mini with monitor, keyboard, mouse, and external drives attached to the MyTek Stereo192 DAC runs over \$2500, and if you use better quality cables the price could go substantially higher.

Even the Sonos ZP100/Mytek Stereo192 front end costs around \$2300 when you include a NAS drive. For \$1999 the Sony HAP-Z1ES supplies the computer, hard drive, DAC, and app to run it all. While this is a bit of a stretch, the HAP-Z1ES could be considered the iMac of HD music players—everything you need to acquire, store, and reproduce HD music files, regardless of format, in one carefully thought out and powerful box.

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

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

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



Naim NDS Network Player

Tradition Meets The Modern World

Robert Harley

f you'd asked me ten years ago to name the audio manufacturers best able to adapt to the seismic shift in the way people access music today, Naim Audio wouldn't have been on my list. If you asked me the same question today, Naim Audio would be near the top of the list, for adapt is exactly what this, the most traditional and venerable of British electronics manufacturers, has done.

Rather than clinging to the past and fading into obscurity, Naim has seized the opportunities presented by the file-based music revolution and led the way with its wildly popular Uniti Series of network players. These "do-it-all" components combine an integrated amplifier, DAC, network streamer, and Internet radio in one chassis, not to mention the option of adding Naim's UnitiServe ripping drive for full music-server capabilities. Unlike many similar "lifestyle" products, the Uniti Series also offers extraordinary, Naim-level sound quality. It's not surprising that the Uniti Series

accounts for fully one third of Naim's business these days.

Which brings us to the NDS, Naim's flagship network player. This is a no-holds-barred product whose features, ease of use, and sound quality are geared toward the most discriminating listeners in Naim's large and loyal customer base. It is designed to complement Naim's reference-level amplifiers and preamplifiers by employing the company's most advanced build techniques and digital-to-analog conversion technologies (see sidebar for the details).

Naim has designed in the NDS a network player that can handle a wide variety of sources and, with the addition of some ancillary equipment, becomes a full-fledged, highly capable music server. The best news is that Naim has created for the NDS (and all Uniti Series products) a fabulous music-management iPad app. To anticipate my conclusion, a fully configured NDS system offers an extremely compelling option for music lovers making the transition to file-based audio.

Newcomers to computer audio are typically offered two macro-level options: a turnkey music server or a do-it-yourself PC- or Macbased system. TAS writer Steven Stone has, half-jokingly, written that the turnkey server is for the ultra-wealthy and the do-it-yourself server for the ultra-geeky. That sums up the situation perfectly. Turnkey systems don't require any computer-audio expertise, but they often fall short in sound quality, are expensive, and are typically closed systems. By "closed" I mean that turnkey systems don't allow you to expand the product's capabilities

and, importantly, prevent you from moving your music library to another platform in the future-your music becomes a captive of a particular manufacturer's software and hardware. The do-it-yourself server suffers from none of these shortcomings, but you must possess considerable computer skills (and patience) to assemble and configure such a system. Moreover, there are so many arcane hardware and software settings in PC-based servers that sound quality is often a crap shoot. Traditionally, do-it-yourself systems have also lacked the elegant user interface of turnkey systems, though that situation is being remedied via recent innovations in iPad apps.

The NDS/UnitiServe combination offers the best of both worlds. Setting up an NDS is somewhat more involved than buying a turnkey server, but vastly simpler than building a computer-based system. But, unlike a turnkey server, the NDS is an "open" platform that allows easy hardware and software upgrades, and even the ability to transport your music library to another server platform in the future. I suspect, however, that most users won't want to move to another server after experiencing Naim's amazing music-management software and superb sound quality. The NDS also offers more ways of accessing music than either a turnkey or do-it-yourself server-and in a much simpler to operate package.

The NDS and UnitiServe SSD

You can opt for the NDS network player alone, which will play music from digital sources



SPECS & PRICING

NDS NETWORK PLAYER

Digital inputs: Three SPDIF (one each BNC, RCA, TosLink), one front-panel USB

Network connection Ethernet, supports

UPnP playback up to 192kHz/24-bit (WAV and FLAC)

Other inputs: Rear-panel USB mini-B (for software updates only)

Digital outputs: SPDIF (on a BNC jack)
Analog outputs: Unbalanced on RCA,
unbalanced on DIN jack

Other connectivity: Wi-Fi, remote input (3.5mm jack, RC5), remote output (2x 3.5mm jack, RC5), CAT5E

Power supply required: XP5 XS, XPS, or 555PS

Audio formats supported: Internet radio (Windows Media-formatted content, MP3, MMS); Playlists (M3U, PLS), MP3, AAC (up to 320kbs), M4A, ALAC, Windows Media formatted content-9 (up to 320kbs); WAV and FLAC (up to 192kHz/24-bit via UPnP or USB only), AIFF, LPCM 16/24, Ogg Vorbis

Control: n-Stream App for iPad/iPhone/

iTouch

Dimensions: 17" x 3.4" x 12.4" Weight: 29.7 lbs.

Price: \$11,000 (requires one of three power supplies); \$22,150 as tested

555PS DR POWER SUPPLY

Dimensions: 17" x 3.4" x 12.4"

Price: \$9650 (plus \$1500 for a pair of Burndy cables)

XPS DR POWER SUPPLY

Price: \$5800

XP5 XS DR POWER SUPPLY

Price \$2800

UNITISERVE SSD

CD RIPPING DRIVE AND UPNP NET-WORK CONTROLLER

Outputs: SPDIF (75 ohm BNC and TosLink) Audio formats supported: WAV, MP3, AAC, AIFF, FLAC, WMA, Apple Lossless, Ogg Vorbis

Disc compatibility: CD, CD-R, CD-RW

Dimensions: 8.15" x 3.43" x 12.36"

Price: \$4000 (UnitiServe SSD); \$3800

Composite Video

(UnitiServe HDD)

NAIM AUDIO

Southampton Road Salisbury, UK naimaudio.com

THE SOUND ORGANISATION (U.S.

DISTRIBUTOR) 159 Leslie Street Dallas, Texas 75207 (972) 234-0182 soundorg.com

ASSOCIATED COMPONENTS

Loudspeakers: Magico Q7

Preamplifier: Constellation Audio Virgo 2,

Absolare Passion

DAC)

Power Amplifiers: Constellation Audio Centaur monoblocks, Absolare Passion 845

Digital Source: dCS Vivaldi (Transport and

Analog Source: Basis Inspiration turntable with Basis Vector 4 tonearm, Air Tight PC-1 Supreme cartridge; Simaudio Moon 810LP phonostage

Remo

Digital S/PDIF Ou

2:05:39:C3

AC Conditioning and Cords: Shunyata

S/No: 329695

Triton and Talos, Audience aR6TS conditioners; Echole Obsession Signature and Omnia, Shunyata Zitron Anaconda and Audience Au24 AC cords

Cables: Echole Omnia interconnects; MIT MA-X SHD loudspeaker cables; MIT MA-X2 interconnects; AudioQuest Wild AES/EBU, AudioQuest EagleEye BNC clock cables, AudioQuest Diamond USB, AudioQuest Ethernet

Equipment Racks: Stillpoints ESS, Critical Mass Systems amplifier stands

Acoustics: ASC 16" Full-Round Tube Traps, 10" Tower Traps

Accessories: Stillpoints Ultra 2 and Ultra 5, Audio Desk Vinyl Cleaner; Mobile Fidelity record brush, cleaning fluid, stylus cleaner

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including any device with SPDIF. You can also access music via the integral Internet radio and its more than 17,000 stations. In addition, the NDS will play music from a USB stick, though it won't accept USB output from a computer. The NDS' functionality can be greatly expanded, however, by adding the UnitiServe, a CD ripper and universal plug 'n' play (UPnP) server. UPnP is a standard that allows devices to communicate with each other on a network. The UnitiServe will stream audio to any UPnP device on the network, including, for example, a Naim Uniti or UnitiQute in another room in the house. It will also scan the network for music stored anywhere (USB, NAS, for examples) and play those files in a wide range of formats (WAV, AIFF, FLAC, WMA-see Specs & Pricing for the full list) at resolutions up to 192kHz/24-bit.

Naim offers two versions of the UnitiServe, the HDD and SSD. The HDD version incorporates a 2TB hard drive for music storage; the SSD has no storage and requires an external network-attached-storage (NAS) drive. The SSD version is a bit more complicated to set up, but offers slightly superior sound quality and greater flexibility. With an external NAS drive, capacity can be expanded, drives can be swapped, and your music library can easily be moved to another server platform. Note that if you opt for the UnitiServe SSD, you'll need a NAS drive; a conventional USB drive won't work. Unlike a simple USB drive, a NAS drive is a computer in its own right, incorporating an operating system and software that manages files on the disk and talks to other devices on the network.

The NDS is a two-chassis affair; the audio circuitry is housed in one chassis and the power supply in the other. The \$11,000 NDS must be used with one of three power-supply options, the XP5 (\$2800), XPS (\$5800), and the top-of-the-line 555PS (\$11.150), which was included with the review sample. The 555PS has two umbilical cords feeding DC to the NDS, while the XP5 and XPS supply DC through a single umbilical. It is possible to supply the NDS with dual 555PS supplies for even greater isolation of the analog and digital sections, although this option adds considerably to the cost. This emphasis on the power supply, and the ability to upgrade sound quality by changing the supply, is a longstanding Naim tradition.

Although the circuitry inside the NDS and UnitiServe is highly sophisticated, the styling and chassis work hearken back to Naim's classic products. The NDS offers a bank of push buttons for selecting the input and navigating the menu, which is displayed in a window next to the buttons. The UnitiServe's front panel is equally minimalist, with only the Naim logo and a CD-loading slot.

The products' rear panels reveal their complexity and sophistication—they are packed with digital inputs and outputs, Ethernet jacks, a Wi-Fi antenna, and ports for a mouse and video monitor.

Once the system is set up, most of your interaction with it will be via the n-Stream and n-Serve apps. These apps, free at the Apple store, provide control over the entire system. N-Serve controls the UnitiServe, and allows you to access and edit metadata, such as

reclassifying a title's genre. N-Stream is the music-management software that you use on a daily basis to browse your library, select music, and create playlists. In my view, a music server lives or dies by its user interface (after, of course, sound quality). The whole idea of a server is to make a large music library easily and quickly accessible. I'm happy to report that the n-Stream interface is outstanding. In fact, now that I'm accustomed to using n-Stream, I don't want to use anything else. And if you'd like to play your music library from another room in the house, just add Naim's UnitiQute or other Uniti Series products and access your library via the iPad interface.

Listenina

Before evaluating the NDS' sound quality I experimented with Ethernet cables to be sure that I was getting the NDS' highest possible performance. I had never listened to differences in Ethernet cables in my own system and was interested in hearing their effect. I first listened to the system with the UnitiServe connected via a 30' generic Ethernet cable. (I located the UnitiServe in my office loft so that I could rip CDs while working.) After loading some music, I moved the UnitiServe to the equipment rack and connected it via a short generic Ethernet cable. I heard an improvement in spaciousness and resolution from this change. But it was replacing all the Ethernet cables with AudioQuest Forest that made the biggest difference. The soundstage widened and deepened, with more space between instruments. The sound had greater ease while simultaneously sounding more

resolved. Bottom line: It's worth investing in specialty Ethernet cables.

When I first heard the NDS in my system I was immediately taken back to one of the most startling and memorable experiences I've had evaluating audio equipment. It was May of 1989 and I was working on the first review of my career. My assignment was to evaluate three inexpensive British integrated amplifiers, one of which was the Naim Nait 2. I had never heard a Naim product, but looked askance at this shoebox-shaped minimalist unit, which was twice as expensive as either of the competing integrated amps (\$795) and offered half as much power (18Wpc). After two weeks of carefully comparing the other two units to each other (and to the system I owned at the time), I connected the Nait 2 and my world changed. The other two integrated amplifiers (not to mention the equipment I previously owned) were hi-fi. The Naim was music. It produced a profoundly different listening experience, one that engendered an engagement with the music, not the sound.

That's exactly the best way to describe the NDS-music rather than hi-fi. The NDS takes a decidedly different path toward musicality, one that eschews sonic fireworks in favor of musical communication. The NDS won't strike you as vivid, present, detailed, or particularly dynamic on first listen—in fact it won't strike you at all, which is the point. Instead the NDS offers an understated and relaxed presentation that seems to render moot the audiophile sonic-checklist approach to evaluating sound quality.

But sit down with the NDS and your favorite

music, as I did, and something truly special happens-the music flows with a natural coherence and complete lack of electronic tincture. I found myself relaxing into the presentation and opening up more fully to the musical expressiveness. A presentation that sounds "spectacular" in a hi-fi way never really lets you sink into the listening chair and develop an intensity of connection with the music. There's always a bit of arm's length detachment that makes you more of an observer than a participant. The system's performance is something you admire rather than enjoy. The NDS/UnitServe reverses that all-too-common paradigm, going right to the heart of what an audio system should do.

Since this is a product review, I'll dissect the sound and offer my view of why the NDS has this ability to fully engage me musically. First, the NDS has an utterly liquid and voluptuous rendering of instrumental timbre. It's not slow and thick-sounding like classic tubed amplifiers. Rather, it achieves this timbral richness and realism by banishing electronic artifacts. Textures are completely and I mean completely-devoid of etch, grain, excessive brightness on transients, and that metallic patina that so often overlays timbre. The lushness comes not from some sort of euphonic coloration, but from reducing the synthetic glaze that we've become inured to in reproduced music.

Similarly, the treble tends toward ease, rather than realizing the last measure of extension. The top end isn't as extended and open as the dCS Vivaldi, for example, but I never found myself wishing for more

in that department. This slight trade-off in top-octave air is repaid many times over in the NDS' stunningly beautiful reproduction of strings-especially violins. Solo violin and massed strings as heard in the concert hall have a completely natural ease and lack of etch that the NDS replicates in a way that is remarkably like the real thing. Listen to the Bach Violin Concerto No.1 (English Chamber Orchestra) from the HDtracks high-res sampler and you'll hear an unforced liquidity, along with an emphasis on the instrument's resonant body rather than the sheen of the strings. Cymbals are portraved with a similar character, revealing the burnished "gonglike" component rather than sounding like the hissing of spray cans. In addition to fostering a relaxation, the Naim's smoothness allows long listening sessions without fatigue. If you've listened to a DAC with different digital filters, there's usually one that offers a slightly softer and smoother treble than the rest-that's how the NDS sounds. I suspect that a large degree of the NDS' character is due to the custom filter, a subsystem that has a great influence on a DAC's sound.

Another factor that contributes to the NDS' musical ease is the way it presents dynamics, particularly transient leading edges. Acoustic guitar, percussion, the upper registers of piano, and cymbal strikes are fully present and alive, but don't have a hyped leading edge. On the beautifully recorded acoustic-trio album (violin, guitar, bass) *The Rite of Strings*, Al DiMeola's guitar had a full measure of life and dynamic attack, but the leading edges of hard-struck notes in the upper registers were

devoid of the edge and hardness so often heard from digital. Don't interpret this as a suggestion that the NDS doesn't fully preserve the dynamic integrity of steep transients; the Naim is extremely dynamic, yet thankfully missing the synthetic leading-edge hype that gives an artificial impression of dynamics.

The NDS' portrayal of dynamics is different from other digital I've heard; rather than calling attention to transients, the NDS conveys the music's underlying pulse and flow with a powerful physicality that must be experienced to be appreciated. A perfect example is the brand new Chick Corea album The Vigil and the track "Legacy" that closes the set. The tune is underpinned by the exuberant, pulsating bass-playing of the terrific young bassist Hadrien Feraud. The NDS conveys this piece's rollercoaster of complex polyrhythms, driven by the amazing drumming of 22-year-old Marcus Gilmore (the great Roy Haynes' grandson, incidentally), with a power and physical involvement that other digital sources dilute.

The new high-res download of *Kind of Blue* from HDtracks (I selected 192kHz WAV) is stunningly great, and in my view the definitive digital version of this classic. On this file the NDS conjures up an eerily realistic impression of musicians in the studio, making the CD release sound like a pale imitation. The NDS is adept at portraying individual instruments as separate entities rather than as less-differentiated elements within a homogenized whole. As a result it is easy to shift focus among various instrumental lines, just as one does with live music. This isn't to say the NDS has a

huge and sculpted soundstage that spotlights each instrumental image; rather, this clarity of line is the result of an understated resolution that allows the music to unfold in a natural and "unspectacular" way. These impressions were consistent over a wide range of sources and resolutions, from ripped CDs to high-res downloads (88.2/24, 96/24, and 192/24).

The NDS reminded me of the extraordinary BALabo BP-1 MK-II amplifier (\$88,500) Jonathan Valin reviewed in Issue 201. Like the BAlabo, the NDS has a voluptuous sound which may not be the last word in treble extension or transparency, but which brings something else that is unique to the table—an ease and musicality, coupled with a powerful bottom-end grip, that produces a compelling intimacy with the music.

The NDS and UnitiServe also gave me a valuable opportunity to compare USB to audio streamed via Ethernet, with high-res files and no other variables. I downloaded some high-res files from HDtracks (the HDtracks Sampler, Joni Mitchell's Hejira, and Kind of Blue) and transferred them to the Synology NAS and to a USB stick. The files on the NAS were streamed via Ethernet by the UnitiServe to the NDS. The files on the USB stick were played directly by the NDS via the USB interface. Note that in this comparison there's no USB cable, only the USB interface. As we all know, a cable can only degrade sound, not improve it.

The USB was decidedly inferior to streaming over Ethernet in every way. I'll start with the treble, the most noticeable area of divergence. Through USB, the upper midrange

and top end sound synthetic and mechanical by comparison. Listen to the gentle cymbal work that opens "So What" from Kind of Blue, or the tambourine in "Dance of the Tumblers" from Reference Recordings' Exotic Dances from the Opera. In both cases, treblerange instruments via USB lacked the natural textures of Ethernet, sounding more like highfrequency noise and less like a stick striking metal on "So What" or like a tambourine's zills chattering against each other on "Dance of the Tumblers." The Ethernet revealed more about how the sound was created, and in doing so, created a greater sense of realism. Tonal colors were similarly affected, with the midrange sounding a bit glazed over through USB. I also heard a reduction in transparency and clarity through USB. Instrumental lines were easier to follow with the Ethernet streaming connection. Finally, the soundstage was a little congealed and "thicker" through USB.

I suspect that these impressions would have been amplified with a USB cable in the system, but this experience points to the conclusion that USB is perhaps not the ideal interface for audio. I don't know how well implemented Naim's USB interface is, but given the level of execution of the rest of the system, I suspect that it is high.

Conclusion

With the NDS and UnitiServe SSD, Naim Audio has married its hard-won expertise in analog circuits with highly advanced digital decoding techniques, and then added extensive musicaccessing capabilities. To top it off, the system includes a wonderful user interface that is a joy to work with on a daily basis. It is impossible to overstate how brilliantly this machine achieves its goal of making a large array of music easily accessible to its user.

The NDS/UnitiServe is the perfect solution for many music lovers who want to transition to computer-based audio without the limitations of turnkey music servers or the confusion of do-it-yourself systems. The only drawback for some listeners may be the lack of SACD compatibility.

Fortunately, Naim has imbued this highly capable system with one of the best-sounding DAC sections I've heard in any product. The NDS is high in resolution but without hyped "detail," voluptuous and rich in tone color without euphonic coloration, and musically vivid without being sonically vivid. The NDS also has a particularly powerful and appealing expression of music's rhythmic flow. It's the kind of sound that lets you become quickly and deeply immersed in the music, not the sound.

The NDS/UnitiServe brings the power of computer-based audio to the company's long legacy of building musically involving products. It's a startling fusion of the old and the new.



HAVING YOUR CAKE AND EATING IT, TOO - OVERCOMING THE METADATA LIMITATIONS OF WAV FILES

Naim has made it possible to enjoy the uncompromised sound quality of WAV files without WAV files' metadata limitations. WAV files typically don't offer the ability to store metadata—the information about the album, artists, and genre that allows you to search and find music. That's why many computer-audio users select the FLAC format, which adds metadata as an integral part of the audio file. Unfortunately, FLAC doesn't sound as good as WAV. Here's why.

A WAV file is an exact representation of the PCM datastream on CDs and high-resolution professional editing workstations. FLAC files are "losslessly" compressed so that they consume less disk space (it's about a 2:1 reduction). The term "lossless" refers to the fact that the uncompressed datastream from a FLAC file is bit-for-bit identical to the WAV file that created it. This is contrasted with "lossy" coding systems such as MP3 that intentionally discard information to reduce the bit-rate. Claiming a sonic difference exists between two "identical" bitstreams may appear to be the height of audiophile lunacy, but there's a rational explanation for why FLAC files sounds inferior: The file must be uncompressed on the fly during playback, a process that many computer-audio experts believe degrades fidelity. Note that a FLAC file may be converted to a WAV file, stored, and played back as WAV without compromise because there's no decompression during playback.

But back to Naim's solution to the problem of the lack of metadata with WAV files. When you rip a CD to WAV with the UnitiServe, the UnitiServe downloads Internet metadata from All Music Guide (AMG) and appends a "sidecar" metadata file to the WAV file. Note that this metadata file structure is unique to Naim, and can be read only by other Naim products. If you rip a CD collection to a NAS drive with the UnitiServe and then move that NAS to a Mac server, for example, you won't be able to access the metadata.

Although Naim's metadata scheme is a big plus, the real magic is in the linking of the NDS/UnitiServe to the Internet service Rovi. Rovi supplies a wealth of information about the album and artist, which is displayed on an iPad/iPhone/iTouch as pages that you "turn" just as you would a CD booklet. (The Rovi graphic interface in the n-Stream app was developed by Naim.) The additional information goes far beyond what you'd find in even the most generous liner notes, with up-to-the-minute listings of other titles by the artist, substantive reviews, artist bios, artist photos, background on the musical genre, and other data. There are also listings of similar artists, with live links to information about them and their albums.

The lack of liner notes has been a major shortcoming of file-based music in my view, but Naim's implementation of Rovi in its iPad app not only matches liner notes; it far exceeds them. The information is a treasure trove, one that encourages exploring other artists and music you may not have been aware of. Note that the Rovi-supplied information isn't stored anywhere; it's a real-time Internet connection run through the UnitiServe and presented via n-Serve.

NETWORKING AND CD RIPPING

Connecting and setting up a full NDS/UnitiServe system requires a few ancillary components and downloading two apps from the Apple store. Your Naim dealer will set up the system for you; my self-installation was atypical. Fortunately, I had expert help in Chris Morris of The Sound Organisation, Naim's U.S. distributor, who guided me through the process (Chris trains Naim dealers).

If you choose the UnitiServe SSD, which has no internal storage, you'll need an external NAS drive. Chris suggested a NAS from a company called Synology. I bought the Synology DS213 dual-bay along with two Western Digital 2TB hard drives (\$524 for the package). I configured the system as a RAID, meaning that the second drive is purely a back-up. The Synology is UPnP compatible. Incidentally, the UnitiServe's UPnP implementation was written in-house by Naim. Naim also created its own CD-ripping software, something it calls the Naim CD Ripping Engine.

The NDS, UnitiServe, and NAS must all be part of the same Internet-connected network. Although the NDS has Wi-Fi capability, Naim recommends a wired connection for both sound quality and reliability. The company has designed its Wi-Fi implementation so that if there's any compromise in signal integrity, it simply mutes rather than allowing the sound to be degraded—it's all or nothing. I discovered this myself; I got the NDS to join my wireless network, but it wouldn't stream Internet radio. I then ran an Ethernet cable from my router to the equipment rack and added a gigabit switch (Netgear GS105, \$45) so that I could add the UnitiServe and the NAS drive to the network.

The next step is to download Naim's n-Stream and n-Serve apps from the Apple store. This music-management software allows you to control the entire system from an iPad or iPhone, as well as see and select music on devices attached to the network. A second app, called n-Serve, is purely for accessing the UnitiServe. N-Serve allows you to access your music library's metadata and correct mistakes, or to reclassify the music genre, for example.

The whole process looked a bit daunting at first, but it couldn't have gone more smoothly. In fact, Chris Morris guided me through the entire installation—including configuring the NAS—from his hotel room while on the road, without the benefit of an owner's manual or any hardware in front of him.

The UnitiServe rips to WAV format unless you tell it to encode in FLAC. Just pop a CD into the UnitiServe's slot and a few minutes later the CD appears as a title in your music library, complete with album art and metadata. I put the UnitiServe on my desk in the upstairs office/loft, and the rest of the system in the equipment racks downstairs in the music room. Because they were all on the same network, I could rip CDs conveniently while at my computer, storing the data on the Synology NAS downstairs. I subsequently moved the UnitiServe next to the NDS and Synology with short Ethernet cables from AudioQuest.

When it's all set up, the n-Stream app displays your entire music library on your iPad as one integral unit no matter where that music is stored-multiple NAS drives, USB, etc. I can't say enough about the fabulous user interface and convenience of accessing your music library through the iPad app.



Visiting Naim's Factory

The building that houses Naim Audio speaks volumes about this storied company's history and legacy. Located in the charming town of Salisbury about two hours southwest of London, the building itself reflects the company's start-up roots, multiple expansions and add-ons over the past 41 years, and its present-day transition into a modern, high-technology design-and-manufacturing operation. Founded in 1972 at the site of the current factory, Naim simply added more and more factory and office space as the company grew. The juxtaposition of the quaint location and venerable original parts of the building with the modern factory sections housing a large and advanced R&D division was striking—and a parallel of the NDS itself.

I was surprised by the number of products in Naim's line and the manufacturing scale. You can see in an accompanying photo one of every product Naim currently makes (the Plexiglas covers are for show). The factory floor was bustling with activity, with a huge variety of products in various stages of construction. I was also greatly impressed by the large and modern R&D department, much of it devoted to software development. Despite Naim's relatively large size and high production output, the focus is on quality. For example, I saw a test station that measures transistors for grouping into matched pairs in differential circuits. Another station shakes cables and power cords before testing to weed out any bad connections.

In the factory's listening room I heard in succession perhaps 15 different ascending "levels" of Naim gear, including integrated amplifiers, preamplifiers, power amplifies, and DACs. Some of these "levels" were power supply upgrades to the same product. Significantly, every single step up the Naim ladder

produced a clear and well-defined improvement in sound. The power supply upgrades were particularly striking, in some cases sounding like a component upgrade. I got the impression that the designers, and critical listeners in the company who guide them, have precise control over every aspect of their products' sound. tas











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Lumin A-1 Network Player

New Kid

Neil Gader

know that I don't speak for everyone, but when I sit down to enjoy an evening of music I prefer to leave my entire workday behind. And that includes keeping my laptop as far from my listening room as possible. Don't get me wrong. I'm grateful for my laptop in incalculable ways, but it's still primarily a business tool, a reminder of looming deadlines and balancing my checkbook. Audio in contrast is pure pleasure. I suspect I'm not alone in this feeling either. Yet, computer media is playing a vastly stronger role in the high end than just a few years ago. The question arises then, in this age of computer media and digital downloads, how does one untether a bespoke audio system from the ubiquitous computer? One option is a network player like the Lumin A-1.

In the parlance of the day the \$7200 Lumin A-1 Network Player is technically a "renderer/ DAC" not a player. It doesn't have an optical drive or internal storage; rather, it plays back what it is being "served" from outside digital sources. (By the way, I think the industry needs a more user-friendly descriptor than renderer, which is just a bit too Black Ops creepy for me.) In any case, let's define the digital environment that it's designed to prosper in. In a nutshell, the A-1 pulls audio media from external sources like a USB stick, a USB hard drive, Lumin's L-1 media server library, or ideally a network with a NAS and UPnP server. Its DAC then snaps into action and plays back a multitude of formats, including the current crop of high-resolution files up to and including 32-bit/384kHz DXD and standard DSD. Additionally it will enable PCM-to-DSD conversion. It does all this by operating wirelessly through its own controller software loaded to an iPad.

So who is Lumin? It's a new brand to TAS's pages, but its roots go back to Pixel Magic Systems Ltd, a Hong Kong firm whose core business is developing high-definition hometheater products with advanced software programming and design. More recently Pixel Magic has been a leading supplier of highdefinition digital TV products under the Magic TV brand. Lumin was formed by a group of Pixel Magic's commercial engineers, who wanted to leverage the company's video reputation into the nascent high-resolution-audio market. Its first product was the MediaBox, which played all type of media files and included an audiophile version, the MB200. In 2012 Pixel Magic launched the Lumin Audiophile

Network Music Player, a DSD-compatible audio streamer.

The A-1 is actually Lumin's mid-tier network player in an expanding lineup that now includes the entry-level D-1 (pricing TBD), the standard \$5000 T-1, and the more advanced \$12,500 S-1, which is equipped with, count 'em, four ESS Sabre DACs. Machined from a heavy block of solid aluminum, its gracefully curved faceplate houses only a lighted display. There are no buttons or knobs to clutter the clean lines. Internally, the vault-like chassis isolates crucial components and circuitry, but Lumin takes isolation one step further by extracting the dual toroidal power supply-a common source of noise-from the main chassis and putting it in its own external case. Digital-toanalog conversion in the T-1, A-1, and A-1Black is handled by Wolfson WM8741 DAC chips, one per channel. The internal layout is fully balanced, while the analog output connectors are coupled with dual Lundahl LL7401 output transformers. The back panel has inputs for Ethernet, two USB drives (thumb drive or external HD/SSD). HDMI and SPDIF outputs. plus unbalanced RCA and balanced XLR outputs. The Lumin PSU connects via a locking multi-pin plug. My only issue with the layout involves the overhang at the rear of the top panel. Meant to hide the cables, it creates a clean overall look, but it also makes it difficult to access back panel inputs if you're reaching around from above, which is the normal approach if you don't have the room to stand at the back of the equipment rack.

Among the Lumin's features are gapless playback, DSD and PCM upsampling options, and an ultrasonic filter for DSD playback.

EQUIPMENT REVIEW - Lumin A-1

Radio aficionado that I am, I particularly relished getting my fix of classic rock and NPR via the A-1's Internet Radio option. It's accessed via the TuneIn.com website. Just open a free account, browse Tuneln's radio list, and add some radio channels to an onlinecreated Favorites folder. Then return to the Lumin app, input your TuneIn sign-in info, and all the selected stations appear, complete with graphics. Pretty straightforward with often excellent sound quality depending on the station's compression stream.

The Lumin app is a sensitive and highly intuitive navigation interface. Its visuals are attractive and adjustable, plus it's very responsive to the touch. It supports multiple servers and Lumin players, as well. Playlists are displayed on the left side of the screen, and drop-down menus allow users to choose inputs, typically between a NAS drive, the L-1 and USB inputs. The user can also configure the "look" of the display to some degree, and engage filters. Album cover view can be easily resized with just a pinch of the display between thumb and forefinger.

A couple minor gripes. The search box is too small and the input selection could be more easily accessible. Deleting a playlist while simultaneously listening to Internet Radio will lock the app, requiring a quick reset. Also the user needs to reboot the NAS after adding a new batch of music files. This is no biggie, but I'm told Lumin is working on the issue. I should add that the company appears to be keeping its pledge of upgrading the app when necessary-during the course of this review the app received a wireless upgrade one morning without a hiccup.

Once up and running, the Lumin A-1 was pretty close to trouble-free in everyday use. Engage the Lumin app from an iPad touch screen, power on the Lumin with a guick swipe, and the NAS drive springs to life from sleep mode. Seconds later, a couple thousand files of music are available at the touch of a finger. And, wonder of wonders, except for moving files to the NAS my laptop is now out of the loop and out of the room!

Sonically, smooth sailing describes the character of the A-1. From Red Book PCM to high-resolution PCM or DSD, music is more settled and continuous-a familiar trait that is consistent with being untethered from a spinning optical drive. Symphonies possess an enhanced fluidity across the soundstage, which goes a long way in enhancing dimensionality. Backgrounds are stunningly deep and silent on the Lumin, allowing the full range of ambient energy to emerge. A lowlevel image like the gently insistent concert harp that underscores one of the themes during The Wasps Overture is presented with virtually no veiling or smearing. Soundstage information, hall sound, various ambient cues stand out as if in greater relief. The stage expands with greater dimensionality. As I listened to Graham Nash's "I'll be There For You" the backing vocal harmonies were less etched and more smoothly integrated with the lead vocal, rather than bucking up against it. Holly Cole's "I Can See Clearly" had a more rounded, less edgy quality, particularly as regards sibilance.

The Lumin seems to settle music into a

more relaxed state, as if all the tension that timekeepers.

accompanies a digital recording is suddenly released, an effect that actually enlivens the performance. ("Tension" in this case stands in for a sense of dryness and constriction that often is part and parcel of the digital experience.) I noted this difference in varying degrees while listening to Jane Monheit's "Waters of March" and Rosanne Cash's "If I Were a Man." Both are very good recordings in any format, but the A-1 has a warmer almost velvety texture without any vestige of digital rigidity. The sound acquires the compliancy of great analog—the give, the sonic elasticity. Bass timbre, like the standup that introduces Jen Chapin's "It Don't Mean Nothing," has more character, air, and bloom. Bass guitar and drums have more individual personalities rather than just playing the role of robotic

As many of us have already discovered, sonic nirvana is far from assured when entering the world of high resolution. High-res transfers alone won't save a sub-standard recording. However, stalwarts like Reference Recording's HRx discs-Stravinsky's Rite of Spring and Rachmaninoff's Symphonic Dances, for examplecan be revelations with shocking dimensionality and orchestral layering. Over and above these traits, it's the sheer liquidity of the upper octaves that distances these recordings from most others. This said, my most jaw-

dropping moment came from the most unlikely of DSD music files, The Carpenters Greatest Hits. As I sat back and listened to the megahit "Close to You," the clinical precision of this reverb-happy track was such that I felt like I was sliding into the chair of the recording engineer and feeling my fingers moving across the mixing board, with the power to isolate each musical element at will. From the carefully tuned drum fills to the spotless piano accents and layered vocal harmonies, nearly every song on this recording could be dissected into a collection of sound modules and analyzed. Had I not known better, I would have sworn I was listening to mastertapes.

As the deadline for this article approached, I attended a concert with the Dallas Symphony at The Meyerson Symphony Center. The program of Haydn and Beethoven further clarified a general impression I was forming

SPECS & PRICING

Supported formats: DSF (DSD),

DIFF (DSD), DoP (DSD), PCM lossless: FLAC, Apple Lossless (ALAC), WAV, AIFF

MP3. AAC (in M4A container). Supported sample rates: PCM,

44.1kH-384kHz/16-32-bit; DSD

Dimensions: 13.78" x 13.6" x 2.4"

SOURCE SYSTEMS, LTD.

\$5000

Weight: 17.6 lbs.

San Clemente, CA. 92672 (949) 369-7729

Price: A-1 \$7200; T-1 (same

electronics w/ metal casing),

sourcesystemsItd.com luminmusic.com

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E-Luminating

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EQUIPMENT REVIEW - Lumin A-1

of the Lumin A-1. It was an impression of resolution, soundstage integrity, and clean transient speed not unlike what I experienced with the dCS Puccini player (SACD-capable) of a few years ago. And, as I listened to young violinist Augustin Hadelich play the Beethoven Violin Concerto I recognized what I was hearing in the Lumin and dCS was more complicated than just a reduction of upper-octave edge or coarseness. In Hadelich's hands the Strad that evening could, indeed, sound aggressive, even edgy, when prodded. The difference was that those transients were blazingly fast and didn't overhang or blur the trailing harmonics. High-frequency distortions tend to brighten a component. It's this lack of distortion that gives the Lumin its warmly weighted presence and overall neutrality. An equally significant "Ah, ha" moment occurred during the Haydn, when the harpsichord came in. Its delicate but steady presence was heard at a level so quiet as to suggest something subliminal. Yet rather than become submerged when the music welled up in intensity, it easily cut through the fabric of the orchestra. These instances were almost identical to my experiences with the Lumin and aforementioned dCS-the microevents in music, a harp or a harpsichord or even the dynamic progression of a steadily building drum pattern, are always revealed unsmeared and complete.

The charms of the Lumin are pretty seductive even if at the end of most days I still prefer popping on a choice piece of vinyl. But that's just me. The truth is I like the way the A-1 factors into my current system like just another source component. I like the way it operates

and performs with a level of musicality that makes me forget its computer roots. I like that I have about 1200 titles (and g-r-o-w-i-n-g) on the NAS. I'm even surprised that the ritual of selecting songs or symphonies at will from my iPad is far more satisfying than I'd assumed it would be. And finally, I love the fact that I can indulge my passion for music with not a computer in sight. Thank you Lumin, and hats off to the new kid on the block.

NETWORK CHALLENGED? NO WORRIES!

For those not especially interested in getting their hands dirty in the world of NAS drives and home networking, Lumin recently debuted the L-1, Lumin's own UPnP server with internal storage capacity of 2TB. It's roughly the size of the Lumin power supply and also sheathed in aluminum. The L-1 was developed to be the simplest way to store music for a Lumin DAC-designed to take the sweat and angst that the novice might encounter setting up a NAS drive. It meets UPnP standards and requires zero configuration. Simply plug it into your computer, drag and drop your music, just as you would from a simple USB thumb drive, and then connect it to your network. Used in conjunction with the Lumin app, it can also unlock additional browsing features. Like the player, it serves up everything from DSD128 (5.6MHz) through high-res PCM to WAV, FLAC, Apple Lossless, and MP3. Price: \$1200.

WHAT YOU'LL NEED

To begin you'll need a home network with Wi-Fi (pretty common today) and an iPad (Gen 2 or later, Retina displays welcome) to run the Lumin App for Apple computers. (Kinsky is suggested for non-Apple OS. And other tablets will work but with limited functionality. Lumin is currently developing Android versions.) To get up and running the typical configuration connects the Lumin and the NAS to a router using Ethernet cables (I use a Netgear GS605 and AudioQuest Cinnamon), and the Lumin to your preamp or integrated amplifier inputs. The Lumin power supply then connects to the Lumin. Once everything is switched on simply confirm that the UPnP media server (MinimServer) is operating on the NAS drive. (Lumin recommends Synology drives for their ease of installation, legible set-up instructions, and compatibility with MinimServer, a free download. No arguments here. I use Synology's 1TB DiskStation, and it works like a charm.) The Lumin App will then automatically detect the media server and the Lumin player, and the music selection on the NAS drive will be displayed in the main Lumin App window. Once the Lumin A-1 is up and running, the only role the laptop plays is uploading music to the NAS. And even then it can be done wirelessly over the network.



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EQUIPMENT REVIEWS

Integrated Amps with DACs

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OPTI 0dB 20 40 60 80 -100

NAD D 3020

Reinventing a Classic

Neil Gader

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

Now, the 3020 is back with a "D" prefix for clarification. A capital "D." As compared with the all-analog original, the new D 3020 is a digital animal designed primarily for computer/USB sources. Power output is a solid 30Wpc thanks to NAD's ultra-compact

Class D topology. True to NAD tradition the amp's power rating is deceptive in that it can output bursts up to 100W (into 4 ohms) during dynamic peaks. In digital connectivity, it offers aptX Bluetooth music streaming—an efficient alternative to Wi-Fi—plus a USB input

that plays back computer-based music in up to 24-bit/96kHz resolution, and operates in asynchronous mode to ensure low jitter.

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

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

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

Operationally I've only got a couple of nitpicks. The lack of a mute button seems a weird oversight. Also the iPod-style IR remote is all flat-black, including the navigation buttons. The only way to see what you're doing is to

angle the remote so that it catches a glint of light to illuminate the markers. Most of us will memorize the six key buttons (on/off, volume +/-, and source select arrows), but really!

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

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

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EQUIPMENT REVIEW - NAD D 3020



double-kick-drum rhythm figures flying off the feet of Metallica's Lars Ulrich.

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

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

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

Perhaps the biggest surprise I encountered

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

Although I'm an infrequent headphone user, whenever I don my AKG K501 cans (still terrific after all these years) I am always impressed by the gorgeous midrange tonality and intimacy these 120-ohm 'phones produce. As a headphone amp, the D 3020 does its job noiselessly and is musically satisfying. The tonal characteristics that make it so appealing with conventional loudspeakers translate fully to the more intimate world of earspeakers. Frankly I haven't ever appreciated headphone listening as much I did during the time I spent with the D 3020.

If computer audio is your primary source for music, and Blue Tooth capability is a must, then the D 3020 makes a compelling argument. The other argument is, hello, its price tag of \$499, making it by most standards a small miracle of packaging and portability, and with few exceptions a delight to use and listen to. Too small for you? NAD has you covered with a

bigger cousin in the new D 7050-a streaming integrated with more power, advanced topologies, plus AirPlay wireless at \$999. For many, however, the D 3020 will be just what the digital doctor ordered. Faithful to the original 3020 but totally dialed in to our times. tas



SPECS & PRICING

Power output: 30Wpc into 8 ohms Inputs: Three digital (USB, SPDIF, TosLink); one analog

Dimensions: 2.3" x 7.5" x 8.7" Weight: 4.6 lbs.

Price: \$499

NAD ELECTRONICS INTL

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

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Hegel H80 Integrated Amplifier

High Performance, Reasonable Price

Kirk Midtskog

egel Music Systems has been on a roll. Since my review of the H100 integrated amplifier in September 2010, the Norwegian company has released three DACs, a preamp, a headphone amp/DAC, a power amp, and two integrateds, as well as updating a power amp already in the line. Hegel strikes me as a company driven by original engineering aimed at offering the highest possible sound quality at reasonable prices. The company's \$15,000 H30 may raise some eyebrows on that score. It is worth noting, though, that given the H30's high performance level, Editor-in-Chief Robert Harley said in his Issue 223 review, "The Hegel H30 is not just a great-sounding amplifier; it's also a tremendous bargain." Elsewhere in TAS, Associate Editor Neil Gader had some very nice things to say about the 250Wpc H300 integrated in Issue 233. Hegel's H200 integrated amp, which I reviewed in 2011, won Product of the Year, and the H300 received two Golden Ear Awards in 2013. Hegel has been busy indeed, and its efforts have been well received by consumers and the audio press.

In general, Hegel products are user-friendly, offer good value, and hew toward understated cosmetics, as if to say, "We let the music do the talking." The 75Wpc, solid-state H80 integrated amplifier with onboard DAC is a case in point; it allows a lot more of the music

to "do the talking" than I thought possible for \$2000. On the nuts-and-bolts side, it has three analog inputs (one balanced, two unbalanced-one of which can be configured as a home-theater bypass), and five digital inputs (two coax, two optical-both types supporting 24/192-and one 24/96 USB. The small supplied plastic remote operates normal preamp functions and also includes buttons to skip

functions and also includes buttons to skip, go back, play, and pause through the attached computer's playlist—with most media players and only via the USB port. A much nicer metal remote is available as an upgrade for \$180. I recommend it.

In a way, the H80 is a perfectly ordinary-looking, average-sized, minimalist integrated amp. Closer inspection reveals a nicely finished product, weighing about 24 pounds with a gently curved, glass-blasted faceplate and control knobs—input and volume. In a departure from other Hegel integrateds, the H80's power switch is located on the bottom of the chassis in the front left corner instead of in the center of the faceplate just below the display. This makes more room on the H80's faceplate for a larger display which, by the way, can be easily read from across a fairly large listening room.

The sound of the H80 is not ordinary at all, though. It delivers a nice measure of musical verve, accompanied by a lack of listener fatigue that one rarely encounters in \$3000 integrateds—let alone in one priced at \$2000. Conversely, many integrated amps near its price with a low listener-fatigue factor too often also sound overly polite or reserved. The H80 is musically involving, well balanced, and surprisingly powerful for its power rating.

While I realize that an amp's nominal output figure doesn't necessary tell the whole story when it comes to its ability to drive real-world speakers, I really wasn't guite prepared for the sense of power the H80 can deliver-even while driving the 85dB-sensitivity, 4-ohm Dynaudio C1 II. In a word, it sounded more "commanding" than I expected. It imparted commendable bass extension and control, maintained its baseline tonal balance during difficult music passages, and served up plenty of rhythmic drive. Some of my sense of its outsized power delivery may be the result of a greater-than-1000 damping factor. (Damping factor represents a measure of an amplifier's ability to control a connected woofer and is related to the amp's output impedance.) When pushed beyond its output power envelopeand at fairly loud volume levels, mind youthe bass-heavy synth lines in Bjork's *Greatest* Hits version of "All is Full of Love" [Elecktra] or the dense climaxes in various movements of Stravinsky's Rite of Spring [RR] could become grainy and unstable. Even so, the H80 sounded considerably more composed than I had a right to expect from a 75Wpc, solidstate integrated amp.

The overall tonal balance of the H80 is very similar to all the other Hegel amplifiers I have used in my own system: H100, H200, H300 integrateds, and H30 power amp. That is to say, the H80 sounds neutral without glare, harshness, or graininess—unless, as already noted, the amp is pushed beyond its overachieving power limit. In general, Hegel amps have a marvelously clear and smooth quality but do not realize that smoothness by sounding rolled-off or veiled. The H80 is no exception. It sounds tonally even-handed and texturally

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EQUIPMENT REVIEW - Hegel H80

smooth while transmitting enough resolution to allow a wide selection of musical nuances to come through with their "essence" intact. Predictably, you will notice better resolution, refinement, power output, and soundstagingespecially the rendering of depth-as you move up the Hegel amplifier line. As such, the H80 still offers a commendable level of the company's characteristic neutrality and smoothness at a relatively low price.

I omitted the H100 in the above comparison because I no longer had one on hand, but I recall the H100 I reviewed in Issue 206 as sounding very smooth and beautiful but also noted some "reticence" in its delivery, as if it were a bit hampered in rhythmic timing. Happily, I can report the H80 is not at all reserved or reticent. In fact, I consider its agility and deftness of timing to be among its greatest strengths. The H80 is just plain fun to listen to. It ably communicates much of the natural liveliness in music and does so without any associated leanness or "presence region" emphasis, which wear poorly over time. For example, Alanis Morissette's "That Particular time" on *Under Rug Swept* [Maverick] retained the recording's forward emphasis of Morissette's upper register but without veering into piercing territory, as some amps do. The forward momentum of the next cut "A Man" was also well served as drummer Garv Novak switches from hi-hat to ride cymbal at about the 3:24 mark. The clangy sheen of Novak's ride cymbal came through but did not become strident. Essentially, what you forgo by opting for the H80 over a H200, H300, or one of Hegel's pre/power combos amounts to

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some omissions of overall resolution, power reserves, and rendering of depth rather than intruding sins of commission like an unnatural tonal emphasis, a fatigue-inducing glare, or some other characteristic that registers as anti-musical.

The H80 creates a soundstage of respectable width and depth for an amp of its power rating and price. In my setup, its listener perspective was roughly in the front section or mid-hall, and the soundstage started just behind the speakers and filled in rearward from there. I mentioned a perception of depth-foreshortening compared to Hegel's more expensive offerings more to illustrate what you get when you move up the product line, rather than to draw attention to a shortcoming in the H80 per se. I consider the H80's depth portrayal to be better than most other solid-state integrated amps in its price category. I believe it is unrealistic to expect truly fleshed-out depth presentation from a solid-state integrated amp that includes a DAC for \$2000.

Speaking of the DAC, it's a really good performer. I compared it to Hegel's standalone HD20 (\$2000) and could not discern appreciable differences. The HD20 may have a bit more body and weight, but my impression could be influenced by cabling differences iust as much has anything else. This is truly impressive performance from the H80's DAC, apparently a scaled-down version of the DAC found in the H300 integrated amp reviewed by Neil Gader in Issue 233. I tried both the USB and SPDIF input on all three DAC sections (H80, H300, and HD20) and preferred SPDIF, in all cases, for its greater liquidity and clarity. The H80's USB input supports 24/96 files and, as mentioned, allows the remote to control most media-player functions like play, skip, back, and pause. The two coax/SPDIF and two optical inputs support 24/192 resolution files but do not allow the remote to control any playlist functions. (I kept all playback set to 24/96 to maintain the same resolution as that of the USB port for my comparisons of USB vs. SPDIF). I didn't try the optical inputs. Hegel DACs are about as easy to set up as they come; "plug and play" really does sum it up. My PC recognized whichever DAC I plugged into within a second or two, and I could then resume music playback for fairly quick sideby-side comparisons.

Hegel has leveraged some new technology derived from its P20 preamp into the H80 and employed a price-scaled implementation of Hegel's patented SoundEngine technology

in the power amp section. SoundEngine uses a feed-forward technique (instead of feedback) to reduced distortion as the signal passes from one amplifier stage to another, and, apparently, also greatly reduces crossover distortion (as the positive and negative halves of the signal switch over to each other). The isolated voltage input gain stage and output current gain stage have their own power supplies, and Hegel uses a rigorous partssorting protocol to make sure complementary device pairs are

closely matched. Chief designer Bent Holter told me at CES 2014 that Hegel is taking a relatively low profit margin on the H80 to keep the price at \$2000. Judging by the H80's build and sound (and Holter's straightforward manner). I have no reason to doubt him.

The H80 represents much of what is right in the high-end-audio scene. Those who are (sometimes justifiably) frustrated with escalating prices, take heart; the Hegel H80 answers the call for high-performing audio kit at a very reasonable price. No, it does not have the seamless liquidity, high resolution, and fundamental solidity of the more expensive stuff, but it gets you enough of the high-end essence to be more than a great place to start. I hope more people will participate in the deeper enjoyment of music in their homes because products like the H80 make it more accessible. The H80 is the real deal...and a sweet deal, too. tas

SPECS & PRICING

Power output: 75Wpc

Inputs: Analog, two RCA, one XLR; Weight: 26.4 lbs.

digital, two SPDIF, two optical (both types 24/192), and one USB upgrade, \$180)

(24/96)

Outputs: One of the RCA inputs configurable as HT by-pass

(power amp in), speaker terminals (508) 405-0910

Dimensions: 16.93" x 3.94" x

13.80"

Price: \$2000 (RC8 remote control

HEGEL MUSIC SYSTEMS USA

david.cohen@hegel.com

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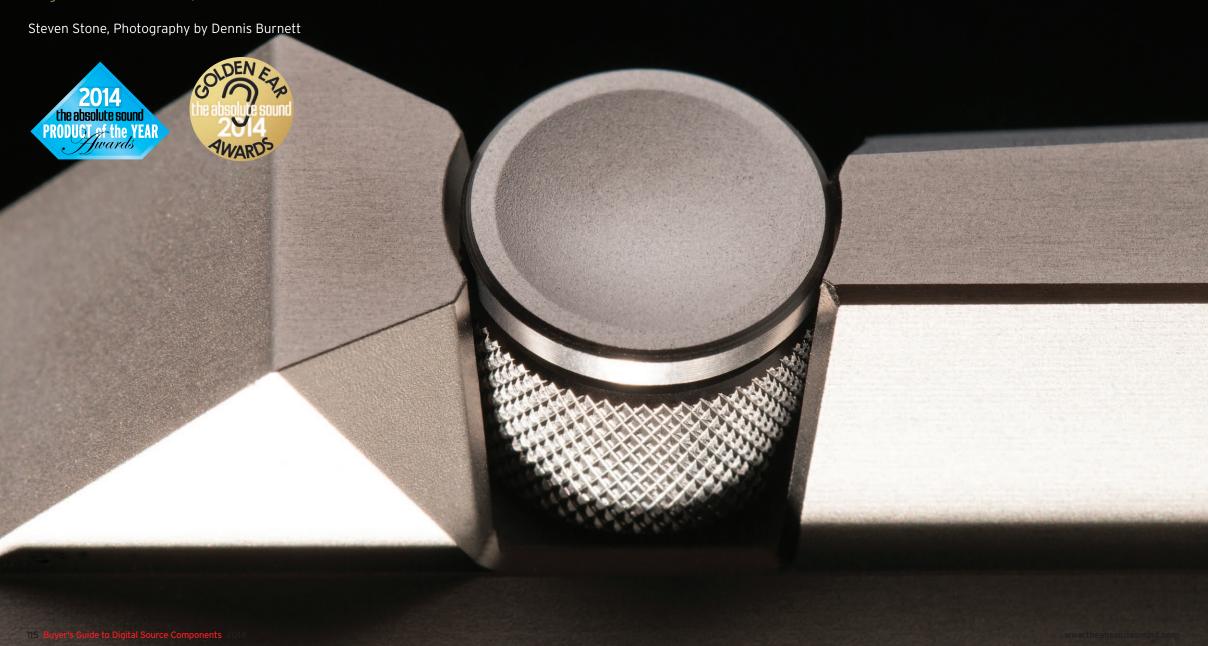
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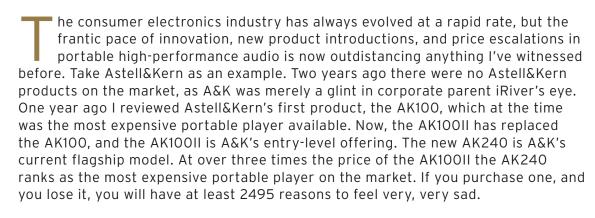
Portable

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High Performance, Reasonable Price

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TechnicalTour & Ergonomics

The AK240 crams an amazing amount of technology into its duraluminum chassis. The chassis itself begins as a 435-gram billet and goes through a twelve-step process that includes laser-engraving the finished enclosure. This level of attention to detail carries through to all aspects of the AK240's design, including its shape. Two of the AK240's four corners are cut off. At first I thought this might have been a sly nod to the TV series Battlestar Galactica (the new one, not the original) where all the papers have cut corners. But the cut-corner design is purely ergonomic-when you pick up an AK240 your pinky naturally wraps around one cut corner while your thumb rests on the other. The AK240 feels great in your hand with just enough heft to feel substantial without being overly heavy.

Inside the AK240 A&K you will find not one, but two, Cirrus Logic CS4398 DAC chips, one for each channel. The AK240 supports PCM rates up to 352.8/24 and DSD up to 128x. It also supports all formats including FLAC, WAV, ALAC, AIFF, MP3, OGG, APE, AAC, DFF and DSF files.

The AK240 is not only a portable player, but also a USB DAC and Wi-Fi and Bluetooth portable streaming device. What's interesting about this dual-streaming feature is that unlike most streaming via Bluetooth, which goes from your smartphone or player to your home music system, the AK240 also streams from your home computer's music library to the AK240. The idea, I assume, is that while you mop the floor or vacuum your bedroom you can be listening to anything in your main

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music library via headphones, not merely what is in the AK240's 256GB internal memory or removable micro-SD card (provided that the AK240 shares the same Wi-Fi network as your computer).

The AK240 has a 3.32-inch AMOLED WVGA (800x480) touchscreen display that functions not only as a display that supports full-color graphics but also as a multifunction control surface. In addition to the screen's controls the AK240 has a single pushbutton on top for wake-up and on/off, plus "forward," "pause/play," and "back" mini-buttons on the side opposite the volume control knob. The volume knob is large and more sculpted in shape than the AK100's volume control. The case has protection on either side of the knob that reminds me of a Panerai wristwatch.

I found the AK240's volume control to be much twitchier than the AK100's control. If you want to increase the volume rapidly on the AK100 you merely turn the knob quickly, but if you try to increase the volume on the AK240 the same way, the control often takes the volume down instead of up. Only a smooth, slow touch would consistently yield "up" when I wanted up. But down was never a problem.

Unlike the original version of the AK100, which came with no case, the AK240 comes with a custom-fitted leather case available in a variety of colors. I know that it is expensive because I bought one made by the same company for my AK100, and it set me back \$80 for a much simpler design.

When I used the AK240 as a portable player I inserted the same 32GB micro-SD cards into the AK240 that I created for the AK100. The AK240 can take a while to read a fully populated card. Some patience is required when you put in a new card because it will not play any of the files on the card until the card has been fully scanned. Unlike the original AK100, which has two SD card slots covered by a sliding click-lock door, the AK240, AK120, and AK100II all have a single SD card slot with no door. I understand that since the current series of A&K devices can accept a much larger card than the original AK100 the need for two slots isn't as great, but having two slots and the sliding door was and is a nice feature on the AK100 which I miss on the AK240.

There are many unique playback options in the AK240 and some can trip you up. If you select MQS or DSD instead of "All" from one of the AK240's sub-directories, the AK240 will not show you anything (or play anything) but MQS or DSD files. When this happened the first time, after I had just finished using the AK240 in streaming mode, I thought I had "broken" another component under review (which occurs more than you would think), but after a couple of e-mails with A&K's technicians we figured out I had somehow gotten into DSD mode. My bad.

Like the AK100, the AK240 has builtin EQ features. But unlike the AK100 you can create and save your own EQ settings. If you wish, you can have a unique EQ setting for each headphone or music genre that you regularly use. Frequency points include 30, 60, 120, 250, 500, 1k, 2k, 4k, 8k, and 16kHz. I don't recommend trying to set your EQ on a moving train or busthe touch controls are quite sensitive, and it's far too easy for a 0.5dB adjustment to turn into 5dB one if you aren't careful where your fingers go. Also, if you listen primarily to DSD material you will be disappointed to discover that the EQ features are not active with DSD files.

One ergonomic area where I found the AK240 to be glitch-free was firmware updates. The AK240 has built-in Wi-Fi. When Wi-Fi is turned on from AK240's setting menu, it also activates the automatic detection of firmware updates. When A&K releases new firmware, the AK240 will let you know and give you the option of downloading it. (This sure beats having to download new firmware to the main directory of your device by USB connection from your computer for the firmware to begin the update process, as is necessary with the AK100.)

You have several output options with the AK240. First and probably most-often used is the single-ended headphone output via the mini-stereo plug. For most headphones or DAC/pre applications this is the go-to connection. This mini-stereo input can also serve as a digital TosLink output, so you can use the AK240 as a USB-to-TosLink digital converter if the need arises. The TosLink output supports PCM up to 192/24, but will not pass a DSD signal, at least not yet. You never know what future firmware updates will

SPECS & PRICING

Supported audio formats: WAV, FLAC, WMA, MP3, OGG, APE

(Normal, High, Fast),
AAC, ALAC, AIFF, DFF, DSF,
DSD64, DSD128

Sampling rates: 8kHz-192kHz (8/16/24 bits per sample)

Output level: Stereo unbalanced 2.1V rms/ balanced 2.3V rms

Output impedance: 1 ohm DAC: Two Cirrus Logic CS4398

Decoding: Support up to 24-bit/192kHz

Input: USB Micro-B input for charging & data transfer (PC & Mac)

Outputs: Phones (3.5mm); optical (3.5mm); balanced (2.5mm, only

4-pole supported)

Wi-Fi: 802.11 b/g/n (2.4GHz)

Bluetooth: v4.0

External memory: micro-SD

(128GB max)

Supported OS: Windows XP, Windows 7/8 (32 & 64 bit): Mac

OS X 10.7

Dimensions: 2.59" x 4.21" x 0.68"

Weight: 6.5 oz. (185g)

39 Peters Canvon Road

Price: \$2495

ASTELL&KERN/IRIVER INC. (U.S. Distributor)

Irvine, CA 92606 (949) 336-4540 astellnkern.com

ASSOCIATED EQUIPMENT

Source Devices: MacPro model 1.1 Intel Xeon 2.66 GHz computer with 16 GB of memory with OS 10.6.7, running iTunes 10.6.3 and Amarra Symphony 3.1 music software, Pure Music 1.89 music software, and Audirana Plus 1.5.12 music software

DACS: Astell&Kern AK100, Colorfly C4, Calyx Audio M, Oppo HA-1, Wyred4Sound DAC-2DSDse

Amplifiers: April Music Eximus S-1, Wyred4Sound mAMP, Accuphase P-300

Speakers: ATC SCM7 III, Role Audio Kayak, Aerial Acoustics 5B, Audience Clair Audient 1+1, Velodyne DD+10 subwoofer

Cables and Accessories: Wireworld USB cable, Synergistic Research USB cable, AudioQuest Carbon USB cables. PS Audio Quintet, AudioQuest Colorado interconnect, Cardas Clear interconnect, Black Cat speaker cable and Interconnect, and Crystal Cable Piccolo interconnect, Audience Au24SE speaker cable

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bring. The last output on the AK240 is the balanced analog output using the four-conductor 2.5mm connector. It was included so you can drive some power-hungry headphones in balanced rather than single-ended mode.

Day-to-Day Use

When I received the AK240 initially I had some issues with streaming. Playing DSD, high-res, and Red Book WAV files was no problem, but the AIFF format was a no-go. The Version 1.15 firmware update fixed that problem completely. With the current firmware in the AK240 and the latest version of the streaming app on my main computer, the only issues I had were a result of signal-strength loss. Sometimes as I wandered around my house with the AK240 in streaming mode the music would stop, but if I moved to an area with higher Wi-Fi signal strength the music would begin once more. The incidence of dropouts was not related to the density of the music file-dropouts with DSD files were no more frequent than with a 44.1/16 file.

Used as a DAC the AK240 had no problems decoding everything I threw at it. When I used Audirvana Plus as the playback app with the AK240, I had the option of either DCS or DoP [DSD over PCM, a method of transmitting DSD data) 1.0 over PCM, and both supported up to 128X DSD with no down-conversion. I noticed only three potential issues when the A&K was used as a DAC. First, the AK240 gets hotter in DAC mode than it does when used as a portable player. I would recommend removing the leather case when it is in DAC mode to improve cooling. Second, the AK240's volume

control is inactive in DAC mode. If you need to attenuate the volume level, you must do so by some other means, such as the software-controlled volume in your player app. (With high-sensitivity in-ear monitors, such as the Westone ES-5, the amount of attenuation required for comfortable listening can be rather severe, which may result in some loss of resolution due to extreme attenuation.)

Third, when using Audirvana Plus with the unit in DAC mode I could "trick" the AK240 into getting stuck in DSD mode. After playing a DSD file in Audirvana Plus and then closing the program and opening another playback app, such as iTunes or Amarra Symphony, I would only hear high-frequency noise, not music, when I tried to play a PCM file. To "re-set" the AK240 to play PCM files, I needed to reopen Audirvana Plus, play a PCM file, and then close Audirvana. After this, iTunes and other apps would play PCM files correctly again.

The AK240 has a special 2.5mm balancedoutput connection, which is such a new scheme that few cable manufacturers have off-the-shelf adapters available for it. Like 99% of those early-adopters who have an AK240, I had no adapters either. Given A&K's usual level of attention to detail I was somewhat surprised that a balanced adapter wasn't included as a standard accessory, but it was not. By the time you see this review some cable fabricators, including Double Helix and Moon Audio, will have custom-made adapters available.

When used as a player via its single-ended headphone output the AK240 could be paired with a wide variety of headphones. Noise and low-level amplifier hiss weren't a problem even with the most sensitive in-ears in my possession, the Westone ES-5s. On the other end of the efficiency spectrum, the AK240 had sufficient gain to drive a pair of Beyerdynamic DT-990 600-ohm headphones past satisfying levels into the spectrum of sound I call "really darn loud." Easy-to-drive headphones such as the Oppo PM-1 sounded dynamically alive and were so well matched to the AK240's output that I would question whether anyone really needs to use an accessory headphone amplifier with the AK240, at least with the vast majority of headphones.

Obviously one of the AK240's strengths is its flexibility and portability. But for owners of large music collections who want to access their entire collection on the go, the AK240 remains inadequate, as do all portable music players. With 256GB in permanent storage plus one slot with a current maximum storage-capacity of 128GB, the AK240 has 374GB maximum storage. Most mature music collections are substantially larger than this. To bring your entire collection on a trip or vacation would involve multiple mini-SD cards. Not only does each card take a while to be scanned when inserted, but also as you add more music to your main library you will need find some way to keep your portable music library on SD cards updated and current. This can be a lot of extra work and will also require a systematic way of doing the updating. It's too bad that Astell&Kern doesn't have a companion app for the AK240 which can find new music and move it onto your AK240's spare internal storage or SD cards similar to the

HAP music-transfer program that Sony developed for its HAP-Z1ES music player.

Sound

What does the AK240 sound like? Like music should sound. When you're not doing critical A/B comparisons with other players, it's very easy to get lost in the music coming from the AK240. I've been making some recordings during the summer that provided me with a good deal of new high-resolution music for evaluating gear. Unlike in the recent past, when the only way I could listen to my DSD recordings was through PCM conversions, now I can listen to the files at their native rate anywhere.

I loaded my latest recordings of Choro Dos 3, Tarka, Matt Flinner Trio, and Bryan Sutton with Chris Eldridge into the AK240 via the Astell&Kern-supplied Android Transfer program for the Mac OS. It's basically a drag-and-drop application that makes adding files to the AK240's internal 256GB storage relatively easy. (The app had a habit of working for a while and then stopping mid-file, after which I needed to disconnect and reconnect the AK240 to USB to get the app to work properly again.)

Once my high-resolution DSD files were loaded into the AK240's internal storage I could compare them with the same files on my computer. The AK240's sonics did not change one iota when it went from playing the files stored internally to files stored on the computer. Even though they're identical bits, they had to travel very different paths to arrive at the AK240's DAC chip. My inability to hear any differences between the two digital signal

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paths shows that Astell&Kern's USB interface is very good. Both had equal levels of information, musicality, and pace.

Most of my latest recordings were made "in the field" in non-traditional performance spaces such as outdoors, in homes, and small public spaces. Invariably I must use in-ear monitors to listen while making the recordings. I used these same in-ear monitors with the AK240 during playback. And while I could not do any comparisons between the AK240 and a live mike feed during the recording sessions, I do feel the AK240 did an excellent job of recreating exactly what I heard in the original session. All the subtle spatial cues on my recording of Bryan Sutton and Chris "Critter" Eldritch playing vintage Martin dreadnaughts in a tent, outdoors, at the RockyGrass Academy came through the AK240 with remarkable fidelity. I could hear the water cascading down the St. Vrain River fifty feet away, as well as children playing in the river far in the distance between songs. Also all the subtle tonal differences between Sutton's 1942 rosewood-bodied D-28 and Eldridge's 1937 mahogany-bodied D-18 were as obvious through the AK240 as they were at the original recording session.

During the review period I had several USB-capable DACs to crosscheck with the AK240. I wanted to compare the AK240 with the Wyred 4 Sound DAC-2DSDse, but since the DAC-2DSDse lacks an analog input I couldn't merely hook up the AK240's analog output to an analog input and then switch between USB DACs via the Apple MIDI control panel. However, I could do this very thing with the just-arrived Oppo HA-1 DAC/pre/headphone-

amp. On high-resolution material (any files greater than 96/24) I couldn't hear any differences between the two DACs, but on 44.1 and Internet MP3 radio streams I felt the AK240 did have a very slight edge when it came to resolving low-level inner detail.

At the time of this review I had five other portable players in-house for comparison—a 160GB iPod classic, an iPhone, an Astell&Kern AK100, a Colorfly C4, and the Calyx M player. In sound quality the Apple iPod 160 was not competitive with any of the other players. The AK100 and Colorfly C4 were close but not quite equal to the sound quality of the AK240. Only the Calyx M matched the AK240's sonics. Also all other players lacked the features found in the AK240.

The AK100, which has been solid and completely glitch-free during the year I've had it, was as quiet and noise-free as the AK240 with sensitive in-ears, such as the Westone ES-5s. The AK100 also drove high-impedance, low-sensitivity earphones, such as the Beyerdynamic DT-990 600-ohm version, as well as the AK240 did. But the AK240 plays 128X DSD in native format, which the AK100 can't do. The AK240 has a "bigger" and slightly larger overall soundstage presentation than the AK100. And the AK240 also sounds more in control on dynamic peaks.

The Colorfly C4, while limited in that it can't play any DSD files, does have a powerful headphone amplifier that remained silent with high-sensitivity in-ears, while also having just enough juice to drive anything I tried with it, including the Audeze LCD-2 Bamboo headphones. I did find that on my own high-



res 192/24 recordings the Colorfly had barely enough output to play loudly with the Audeze. Given the radical disparity in looks between the Colorfly's retrosteam-punk styling and the AK240's sheer slickness, it's hard to imagine that someone would be attracted to both physical designs equally. Sonically the differences between the AK240 and the Colorfly on 44.1/16 Red Book files were minor. Harmonically, the Colorfly had a bit of additive warmth compared to the AK240, but the AK240 remained more linear than the Colorfly, which became slightly "overcooked"-sounding when pushing inefficient headphones.

The Calyx M (\$999) proved to be the most sonically competitive with the AK240. Both produced excellent sound on anything I threw at them. The Calyx doesn't have all the capabilities of the AK240-it's "only" a portable player and a USB DAC, and its internal storage is just 60GB, but it also has a very refined interface and ergonomics. Although blessedly silent with sensitive in-ears, the Calyx M had barely enough gain on my own high-res recordings. I had to push the sliding side-mounted volume control all the way up to max to get enough output. But during A/B comparisons of 44.1/16 as well as high-res 128x DSD music files, I couldn't reliably tell one player from the other. Both the AK240 and Calyx M had an equal level of control and finesse.

Final Thoughts

I'll admit I've been intentionally hard on the AK240, but given its place at the top of the price-hierarchy in portable players, potential owners would consider me remiss if I glossed over any of the ergonomic or performance areas where it might not be perfect. Of course, perfection is a difficult-to-obtain goal. After 35 years of reviewing audio gear rarely has any audio component been "perfect," but the AK240 comes far closer to that status than any portable player I've used so far.

Whether the AK240 is the best player for you depends on whether you require any or all of its unique features. If you already have a high-quality USB DAC, streaming device, and smartphone that uses Bluetooth for streaming, some of the capabilities of the AK240 will be redundant and perhaps the new Astell&Kern AK120 or AK100II would be better options.

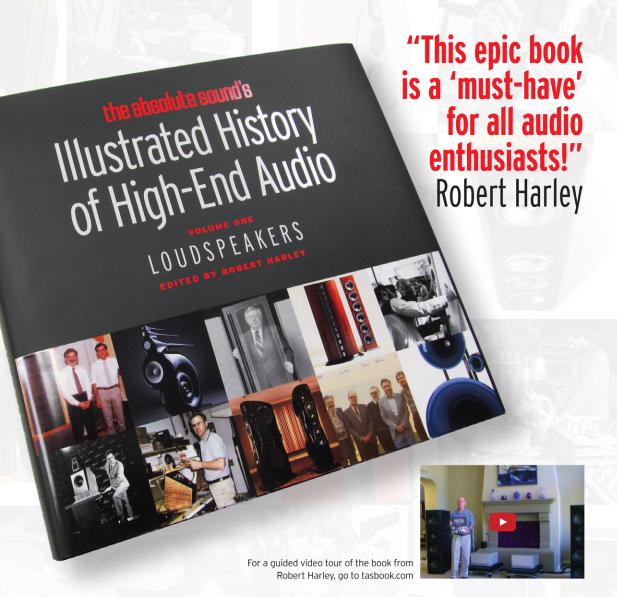
As I said at the start, the pace of new portable player introductions has been phenomenal. And if Astell&Kern and others continue releasing new players at this clip the AK240 will most certainly find itself challenged by something new in the near future. But right now, in Fall 2014, the Astell&Kern AK240 is the best-sounding, most fully featured portable player currently available. If you demand "the best" the AK240 should, inevitably, be at the top of your musthave list. tos



Portable Audio Player of the Year

Astell&Kern AK240

The AK240 is Astell&Kern's current flagship. At over three times the price of A&K's entry model, the AK100II, the AK240 ranks as the most expensive portable player on the market, but it manages to cram an amazing amount of technology into its duraluminum chassis. Anchored by a pair of Cirrus Logic CS4398 DAC chips, the AK240 supports PCM rates up to 192/24 and DSD up to 128x. The unit is not only a portable player, but also a USB DAC and Wi-Fi and Bluetooth portable streaming device. Is the AK240 the best player for you? That depends on whether you require any or all of its unique features and capabilities. It certainly ranks as the best sounding and most transparent portable player SS has heard, and A&K's regular firmware updates have added additional features since his review (such as more EQ options.) Although there are less expensive portable playera, such as A&K's AK100II and AK120, if you want it all in one compact and beautifully made chassis, the AK240 is the player to own. (248)



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THE BEST CD/SACD PLAYERS, DACS, MUSIC SERVERS & PERIPHERAL PRODUCTS, PORTABLE MUSIC PLAYERS

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2015 BUYER'S GUIDE

THE BEST CD/SACD PLAYERS, DACS, MUSIC SERVERS & PERIPHERAL PRODUCTS, PORTABLE MUSIC PLAYERS



Rotel RCD-1570 \$999

Long ago Rotel demonstrated that high-end sound need not come at a highend price. Exhibit A was its now-legendary CD player—costing a mere \$400 that outperformed units ten times its price. Rotel's new stack carries on that tradition, with three components that—aesthetically and functionally—were obviously designed to be deployed in tandem. First in line is the Wolfson DACpowered RCD-1570 CD player. This slot-loading unit has both single-ended and balanced analog outs. (There are also RS-232C and Rotel Link connections for external control.) A nice additional touch: The RCD-1570 has a digital out so it can be used as a transport in the event its owner decides to spring for a more expensive DAC. But even when used as a stock player, the RCD, like its now-famous forebears, makes few sonic compromises. rotel.com (242)



Oppo BDP-105 \$1199

Few disc player/DACs can compete with Oppo's BDP-105 at its price point (or even near its price point), because the Oppo offers a seemingly unbeatable combination of versatility, flexibility, and serious high-end sound quality. Clean, clear, and decidedly detail-oriented, it hews somewhat toward sonic leanness, but is far more revealing than it has any right to be for the money. With the BDP-105 what you hear is what's on the record, with no comforting (but perhaps sonically misleading) infusions of softness, warmth, or bass enrichment. In sum, the do-all Oppo is a multi-format disc player and multi-input DAC with which your system can grow (and it is also the vehicle of choice for many firms offering ultra-high-performance upgrade mods). Finally, did we mention the Oppo sounds terrific when heard through top-tier headphones? oppodigital.com (232)



Moon Neo 260D \$2000 (\$3000 w/DAC)

The Moon Neo 260D continues a tradition of fine CD players from Canada's Simaudio. However, unless you are a CD-only loyalist, you really need to consider adding Simaudio's \$1000 high-resolution DAC section to the 260D. With a 32-bit asynchronous converter and four rear-panel digital inputs (dual SPDIF, a TosLink, and a USB), this optional DAC effectively opens up a whole new world of digital connectivity. Standard CD playback, though expectedly excellent, pales next to the level of refinement that the DAC brings to the table on high-resolution material—an added dimensionality that almost seemed to re-inflate the soundstage. The DAC's superior reproduction of micro-dynamic gradations also more convincingly recreates the distances among the players in a symphony orchestra. With or without the optional DAC, the 260D offers natural sonics elegantly mated with resilient build-quality and good ergonomics. simaudio.com (244)



Modwright-Oppo BDP-105

\$2495 (with Modwright Truth modifications only)

Modwright offers a superior modification to Oppo's highly regarded universal player, the BDP-105, replacing the stock analog stage with an incredible tube design and external power supply that elevate the 105's performance dramatically. The build-quality is exemplary, and the sonics exceptional. It is the first digital player JH has had in his listening room that didn't make him want to immediately return to his analog rig-and that's because the Modwright sounds so much like analog in many respects, without giving up the bass extension and control, clarity, fine detail resolution, and convenience that can make digital so attractive. Whether you'll recover the cost of this upgrade when you sell the unit remains to be seen, but with its outstanding sonics and remarkable flexibility this is one universal player you'll likely hang on to for a long time. modwright.com (Review forthcoming)



Marantz Reference Series SA11S \$3999



Marantz products almost always stand out from their competitors for a very musical sound that is notably free from harshness, glare, or anything remotely abrasive. Such is the case with the SA-11S. The tonal balance is neutral, which means that nothing calls attention to itself up and down the spectrum. It has state-of-the-art control and resolution yet also an elusive naturalness and musicality that banish all memories of the digital sound of yore. On SACD sources especially, the SA-11S is one of the best PS has heard anywhere and unquestionably the best he has had in his house. If he were in the market for an integrated player to handle both Red Book and SACD sources, this is the one he would buy owing to its lineage, its perfect mediation of musicality and neutrality on CD sources, and its absolutely magnificent SACD performance. marantz.com (233)

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Aesthetix Romulus CD Player/DAC \$7000

This all-tubed CD player and DAC is one of the great bargains in high-end audio today. What makes the Romulus

special is that it sounds so "non-digital." Rather than being flat and congealed, it opens up the spatial presentation, giving instruments and voices room to breathe. The Romulus couples this expansiveness with an unusual (for digital) sense of top-octave air and openness. The tonal balance is rich and warm in the bass, which, when added to the player's treble smoothness, results in an immediately engaging and fatigue-free presentation. The Romulus isn't "tubey" in the classic sense, but neither does it sound like solid-state. The design and build-quality are beyond what's expected at this price. If you have no analog sources, the Romulus



can serve as a preamplifier and DAC with multiple digital inputs, provided you purchase the variableoutput option (\$1000). Thanks to an innovative hybrid analog/digital volume control, there's no loss of resolution no matter the volume setting. aesthetix.net (243)

MBL Corona C31 \$9200

\$21,000

The Corona line from MBL may well be just about the most purely beautiful electronics on the market with sonics to match. However, in today's computerdriven marketplace, if you needed further proof that the CD player is alive and well, look no further than the C31. A slot-loading CD player at heart, it includes a high-performance DAC with inputs for USB, SPDIF, and TosLink. At times NG found himself giving the CD player a slight nod for superior image focus and the reproduction of low-level detail. But moving to 24-bit/96kHz material, he preferred USB hands-down. The classic MBL signature—the bloom and analog warmth that informs all its gear—is built into the C 31's DNA. A fabulous twofer. mbl-northamerica.com (228)



Esoteric K-03X \$12,000

A brilliant concept beautifully executed, the Esoteric K-03X is much more than a CD/ SACD player. It is also a full-fledged DAC, with ample inputs, multiple upsampling and filtering options, and even provisions for an external word clock. Its USB interface is state of the art, supporting the highest resolutions and asynchronous clock control. As a DAC, the K-03X has few peers. Both the S/PDIF and USB interfaces are among the best AT has heard. In either case, rhythms are unflagging, details emerge clearly and naturally, and listener fatigue is non-existent. Dynamics are superb as well, and the

sound is always open and airy. The K-03X also excels as a disc player, especially when playing SACDs. CD sound is not quite up to the K-03X's benchmark in other modes, but it is ravishing nonetheless. Though it is not cheap, the K-03X delivers a level of versatility, build, sound quality, and operational smoothness that fully justifies its price. esoteric.jp (213)



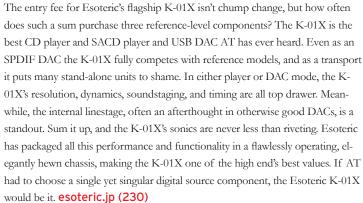


dCS Puccini and U-Clock

\$18,999; dCS Puccini U-Clock; \$5499

If you're among the 99.99% of audiophiles who can't afford dCS' \$110k Vivaldi digital playback system, the company's Puccini CD player is the next best thing at less than one-fifth the price. As with the Vivaldi, the Puccini is packed with dCS' proprietary technologies, including the brilliant Ring DAC and a custom softwarebased digital filter. The sound from CD is extraordinarily dense in detail and tone color, with a solid bottom end and spacious soundstaging. The icing on the cake is the Puccini's fabulous rendering of SACD. The U-Clock is an outboard clock that not only improves the Puccini's sound quality but also serves as a USB interface so that you can access the Puccini's DAC with a computer-sourced signal. Recently updated to support DSD on USB and Puccini digital inputs. tempohighfidelity.com (183)







CH Precision D1 CD/SACD Transport/Player

From \$37,750, depending on configuration

Although a formidable CD/SACD transport/player in its own right, the D1 comes into its own when paired with the companion CH Precision C1 DAC/Pre. The two communicate via the company's proprietary CH-Link, and once you hear the Link's open, natural sound, you will never go back to SPDIF. Furthermore, the CH-Link can carry raw DSD straight from an SACD to the C1. The combination of D1, C1, and CH-Link results in the best SACD sound AT has yet heard. The bottom line is: Buy the D1 with the C1. ch-precision.com (239)

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dCS Vivaldi Digital Playback **System** \$108,496

The terrific-sounding four-

box Vivaldi is unquestionably the state of the art in functionality and technical sophistication. This flagship from dCS incorporates technology unlike that of any

other digital product, with all the key subsystems designed and built by dCS using proprietary hardware and software. It sounds unlike other digital products as well, with a density of information, saturation of tone color, bottom-end authority, and highly spacious yet precisely rendered soundstage that outdo the competition. Although the complete system comprises four separate chassis, not all of them are required. The pairing of the Vivaldi Transport and Vivaldi DAC (\$74,998) will get you most of the way there. The Clock and Upsampler are nice additions, but not required to realize the Vivaldi's extraordinary sound quality. Note that the Vivaldi is a highly sophisticated instrument that requires more user involvement than most digital-source components. tempohighfidelity.com (223)



DACS

\$149



How can you not absolutely love an asynchronous USB DAC packed with audiophile-grade design elements in the form factor of a USB stick that costs \$149? The variable output level allows you to drive headphones or a power amplifier directly from its 3.5mm stereo mini-jack output. Amazingly, the DragonFly features an ESS Sabre

DAC, analog-domain volume adjustment, and separate clocks for different sampling frequencies for better sound. The DragonFly has a remarkably sophisticated and relaxed presentation, yet excels at conveying dynamics and drive. Great sound from a computer has never been this simple or inexpensive. A brilliant product.



Audioengine D3 \$189

A perfect complement to the Audioengine A2+ powered desktop speakers, or to your portable laptop, the Audioengine D3 DAC can handle PCM audio files up to 96kHz/24-bit. Sound quality is on par with or above that of comparable portable USB DACs, and the sleek design matches nicely with Apple products (the Audioengine engineers are former Apple designers). If you need an affordable option to play high-res audio, take a listen to the D3 DAC.

audioengineusa.com (241)



Cambridge DacMagic XS \$199

About the size of a small box of wooden matches, the Cambridge DacMagic XS is one of the smallest and lightest portable DACs we've seen. It measures approximately 2 1/8" by 1 1/8" by 3/8" and weighs under 4 oz. On one end you'll find a micro-USB input and on the other end is a 3.5mm stereo output. Although it doesn't handle every

audio format, and isn't DSD-capable, the DacMagic XS de-

livers a lot of functionality and sonic goodness for under \$200. For audiophiles looking for a road-warrior-worthy DAC that will be at home hooked up to any computer, portable or desktop, and successfully drive most headphones, the Cambridge Audio DacMagic XS DAC is a savvy and very affordable option. cambridgeaudio.com (245)



audioquest.com (226)

Hegel Super DAC

Hegel made some very specific design decisions for the Super DAC. First, it is a USB 1.0 device that needs no drivers with any computer. This makes it a truly plug-and-play device, but it also limits the Super DAC to a maximum sample/bit rate of 96/24. Although the Hegel Super does lack some features, such as DSD and 192/24 PCM capabilities, it makes up for it with its solid sound and ability to do double-duty as a USB-to-TosLink converter. Given the number of other portable DACs available at a similar price, the Hegel faces some tough competition. But for some prospective users, the Super's powerful output and easy setup might be deciding factors in its favor. hegel.com (245)

Meridian Explorer USB/DAC \$299

Packaged in a chic, four-inch-long, extruded-aluminum ovular case, the Explorer is one terrific fully asynchronous, USB-powered streamer. It handles files up to 24-bit/192kHz resolution while supplying smooth touches of



analog-like warmth and fluidity-factors regarded as essential in the otherwise arid landscape that describes much entry-level digital. Even more important was an impressive ambient bloom that elevated the acoustic ceiling of venues rather than tamping them down like a lid over a stew pot. The music was spacious, detailed, and transparent, inviting comparisons to more expensive DACs. While the Explorer's spectral balance is modestly lighter, even so, perspective please! Hi-res "to-go" for less than the price of a decent set of headphones? Amazing.

meridian-audio.com (234)

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DACS

arcam.co.uk (245)

Resonessence Labs Herus \$350

The Canadian-made Resonessence Labs Herus is one of the most flexible USB-powered DACs in sample- and bit-rate capabilities. This lipstick-sized unit supports PCM up to 352.8/24 as well as DSD64x, DSD128x, and DXD files. So, regardless of how you like your highresolution files, the Herus will play them. Machined out of a solid block of aluminum, the Herus measures 2.5" x 1.25" by .75" and weighs less than a pair of CD jewel cases. On native 128X DSD it offered sound quality that rivaled that of any DSD DAC SS has heard, regardless of price. resonessencelabs.com (245)



\$399

DSD DXD

Micromega's MyDAC is entry-level in price only; its sound is far more refined and sophisticated than its modest cost would indicate. The unit looks very much like an Apple AirPort Extreme, but with a front-panel wheel to select between TosLink, coaxial, and asynchronous USB inputs. The Micromega gives you some sonic attributes usually found in much more expensive DACs—qualities like air around instruments, a sense of three-dimensional space, and a laid-back ease. Through the Micromega, instruments don't sound like flat cardboard cutouts; they are fully fleshed out three-dimensional images surrounded with a wonderful bloom. Timbres are remarkably smooth and free from grain. The bass is solid and tight, although the very lowest bass lacks ultimate authority. audioplusservices.com (228)



The Sony PHA-2 portable headphone amplifier and DAC was created to be a digital "bridge" product, designed to improve the sound from smartphones, iPods, iPads, and computer USB sources. The 270g (.595 lbs.) PHA-2 is housed in an aluminum enclosure that features a zinc-alloy bumper as well as a unique rail/edge design. The Sony PHA-2 offers a lot of capabilities and excellent sound for under \$600, but since no one device can do everything, prospective owners should look at the PHA-2's feature set carefully to ensure that it does what they need it to do. sony.com (245)



Tired of being limited to high-res audio only in your dedicated listening room? The Arcam airDAC is the perfect solution for those who have second systems in other rooms, but no way of connecting them to the main system or home network. Via Airplay or a wired Ethernet connection, the Arcam airDAC will stream highres music to your other systems. A sleek app for smartphones and the iPad allows you to control the unit, while an optical-out lets you connect to a separate DAC if you choose. Airplay and Ethernet UPnP-enabled, the airDAC harnesses modern connectivity to bring your system into the twenty-first century.



Meridian Direct could best be described as an Explorer with a larger portfolio. Compact, yet designed for the home rather than the street, its mission is digital media—from computer audio via USB to virtually any device with an optical or SPDIF input. However, unlike its smaller USB sibling, Explorer, Direct uses a pair of unbalanced RCA output jacks, permitting audiophiles to exploit the potential of interconnect options. Included in the bargain are Meridian resolution enhancements such as upsampling and an apodising filter. A sonic knockout that captures much of the realism, depth, and dimensionality of live music, the Direct is a cost-effective way of inoculating a system against digital obsolescence. meridian-audio.com (240)



The NuForce DAC-100 marks NuForce's first foray into the product category of DAC/preamps. With a feature set that should work equally well in a computer desktop/headphone system or a computer-based



system, the NuForce DAC-100 packs a lot of features and technology into its svelte chassis. NuForce calls the DAC-100 a DAC/preamp because it performs the functions of a DAC and a preamp. It has four digital inputs-USB 2.0, TosLink, and two for SPDIF RCA digital. For outputs the DAC-100 includes one pair of single-ended variable-output RCA connectors and a headphone jack on the front panel. The DAC-100's headphone output is designed to support headphones with an impedance range from 120 to 600 ohms, so it may not be suited for all 'phones, especially highsensitivity low-impedance in-ear models. Still, NuForce's entry at this hotly contested price point delivers excellent sound combined with a useful feature set, making it one of the DACs that should be on anyone's "must audition" short list if he's in the market for an under-\$1500 USB DAC. nuforce.com (228)

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DACS



Rotel RDD-1580 \$799

If you're in need of a high-quality DAC capable of PCM audio up to 192kHz/24-bit at an affordable price (and who isn't?), Rotel has designed a DAC capable of producing audiophile-quality sound at big-box-store prices. With six digital inputs, including USB, coax, and optical, plus the ability to stream Bluetooth audio from your favorite portable devices, the Rotel RDD-1580 is a DAC that will blow you away without blowing the budget. rotel.com (243)



Wyred 4 Sound DAC-2

wyred4sound.com (247)

\$1499 (\$100 to add DSD, \$1000 for SE boards)

The Wyred 4 Sound DAC-2 combines a rich feature set with remarkable performance at a price that makes it hard to beat. Its overall sound has a solidity and weight that are both arresting and involving. While SS hasn't heard every available DAC in its price range, he has yet to hear any USB DAC under \$1500 that outperforms the Wyred 4 Sound. Factor in the basic DAC-2's 192kHz high-resolution capabilities, small upcharge for DSD support, and the ability to convert to SE anytime you wish via built-in circuit-board upgradability, and you have a DAC that will remain *au courant* long enough to make it a savvy and satisfying purchase, regardless of how much more you can afford to spend.



Mytek Stereo192-DSD DAC \$1595

Manufactured in Poland, the Mytek Stereo192-DSD DAC was designed by Michal Jurewics, who is also the founder of Mytek Digital. The Stereo192-DSD DAC comes in three versions. Fortunately for consumers, all three have the same price. The "standard" version is available in two finishes, silver and black. The black chassis 192-DSD DAC has front-panel volume-level LEDs below the LCD panel, while the silver version has none. Input and output options on the black and silver 192s are identical. The "Mastering" 192-DSD DAC has a similar front panel to the black version, but instead of an analog pass-through it substitutes a dedicated DSD input for 128x (5.6MHz) files (currently only accessible via a PC-based computer). If you want or need a DSD-capable DAC with a FireWire interface (which can be attached to any Thunderbolt connection via an adapter), the Mytek is the only game in town, so far. mytekdigital.com (234)



Benchmark DAC2 HGC \$1995

The DAC2 HGC employs new high-

efficiency low-noise power supplies with each sub-system using its own dedicated low-noise regulation. The UltraLock2 digital clock replaces the older UltraLock clock that was used in the DAC1 for jitter attenuation. Like most current-generation DACs, the DAC2 uses an asynchronous interface, which Benchmark, with a knack for verbal invention, calls its "multi-mode asynchronous USB." The HGC moniker stands for "hybrid gain control," which is a "dual domain" attenuation system that combines digital with analog gain controls for an optimal result. By using a 32-bit digital system along with a servo-driven analog potentiometer, Benchmark claims that the HGC design "outperforms traditional analog or digital volume controls, including the two-stage DAC1 HDR system." The Benchmark DAC2 does make a strong case that the current-generation digital-to-analog interfaces are no longer the weakest link in the reproduction chain, if indeed they were in the past. Also available in an L version without headphone amp, \$1795; and a D version with only digital inputs, \$1795. benchmarkmedia.com (234)



Bryston's BDA-1 builds on the success of its highly regarded predecessor, the BDA-1. The new unit adds a greatly improved asynchronous USB interface capable of handling sampling rates up to 192kHz, along with a pair of top-of-the-line chips from AKM. The USB input, merely a convenience feature on the BDA-1, is now a state-of-the-art implementation that is fully competitive with other top-class USB DACs. Because of its particular strengths, the BDA-2 will be especially appealing to listeners seeking to maximize enjoyment from USB sources. The overriding impression of music played via the BDA-2's USB input is one of relaxed ease and unflustered composure. KS found that the crowning achievement of the Bryston BDA-2 can be described in the simplest of terms: It makes the best of every recording that you play through it. A great bargain. bryston.com (233)

Cary Audio DAC-100 and DAC-100t \$2495 and \$2995

Available with either a silver or black front panel, the DAC-100 and DAC-100t look virtually identical. The "t" on the second DAC-100 stands for tube. This is what differentiates the DAC-100t's analog output stage from that of the all-solid-state DAC-100. Except for their output design the two are technological twins. Overall the DAC-100t has more additive colorations, especially in the bass, while the DAC-100 is slightly subtractive in its upper frequency ranges. Which is "better" will depend on the rest of your system. WG encourages readers to listen to both and make their own choice. The fact is you can't go wrong with either one.

caryaudio.com (242)



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DACS



Auralic Vega \$3495

\$5995

If you are looking to take the plunge into the world of DSD and need a high-quality DAC/preamp capable of handling all your digital sources, look no further than the Auralic Vega digital pre/DAC. With AES/EBU, two coax, optical, and USB inputs, the Vega is highly versatile. Because it is also capable of acting as a preamp, all you have to do is add an amp and speakers and you're ready to start rocking. The Vega supports all PCM-based audio up to 384kHz/24-bit and DSD up to DSD128. As good as it gets for the price. auralic.com (240)



Berkeley Audio Design Alpha DAC Series 2 \$4995

The Golden Ear and Product of the Year Award-winning Alpha DAC is not only one of the best-sounding digital-to-analog converters, it's also an amazing bargain. In addition to world-class decoding of CD sources, the Alpha DAC can handle any sampling rate to 192kHz and word lengths to 24 bits. Its robust analog output stage and variable output level allow it to drive a power amplifier directly. This feature is significant, because the Alpha DAC is capable of such resolution, timbral purity, and dynamics you'll want to hear it without the limitations of a preamp in the signal path. When used at its best-fed by true high-res sources from a music server, and driving an amplifier directly—the Alpha DAC delivers stunning resolution of the finest musical detail, throws a spectacularly large and well-defined soundstage, and plays back music with gorgeous tone color and purity. A reference-quality product at a moderate price. berkeleyaudiodesign.com (189)

EAR-Yoshino 192 DACute Digital to Analog Converter \$5895

Tim de Paravicini's DACute, really a DAC/pre, represents a bold attempt to equate digital performance to good analog practice. High-frequency noise generated by the DAC chipset is filtered by an analog filter, the same sort of filter Tim has always used on analog tape recorders for bias and other ultrasonic noise filtering. The Cirrus SPDIF receiver accepts up to 24/192 digital data from USB, coaxial SPDIF, and TosLink inputs. The internal line preamp uses one 6922/ECC88 twin-triode per channel, while the output stage is transformer-coupled. For best results, avoid the DACute's USB input and purchase an external asynchronous data converter. The end result is a DAC that sounds more analog than most DACs and is responsible for restoring DO's faith in digital audio. ear-yoshino.com (238)





Sometimes it's good to start from scratch when designing a new component. That's what software guru Ted Smith did—he started from the premise that DSD recordings sound good and built a DAC around that premise. Using a field programmable gate array (FPGA)—the digital equivalent of a blank slate—he created a DAC that converts all incoming PCM files to DSD128, then decodes them with a 24dB-per-octave low-pass filter (LPF) with far less harmful sonic impact than typical brickwall PCM filters. The transformer that's part of the LPF filter is also the output section, so there are no tubes or transistors to be seen (or heard). PS Audio's Paul McGowan heard a prototype, loved it, and agreed to build it. VF thought it was easily the best digital sound he'd heard, but the DAC needs lots—probably 500 hours—of break-in. psaudio.com (245)

MSB Technology Analog DAC \$6995

An innovative high-tech DAC that handles PCM files up to 384kHz/32-bit (DXD) and DSD128 files, the Analog DAC can play DSD through all of its digital inputs, not just through USB. Its sculpted industrial design lets you add modules to tailor the unit to your needs. For instance, an optional \$995 volume control (which must be purchased with the initial order) allows the Analog DAC to serve as the system controller, with both digital and analog inputs. (Other options can be added by the user as needed.) With a sound that is clean, harmonically full-bodied, and dynamic, the Analog DAC competes against the best DSD-capable DACs out there. msbtech.com (239)



The least expensive DAC from England's digital specialist, the Debussy nonetheless makes use of virtually the same circuitry and technology as its far more expensive stablemates. Further, its generous feature list includes plentiful source-format options, single-ended and balanced outputs, and a front-panel sample-rate display. Most importantly, though, the Debussy's sound is pure dCS, with a density of musical information that sets it apart from the competition. Nor is there any sense of frenetic digital machinations; AT found that sound winds out of the Debussy like thread from a spool. Moreover, this DAC's USB interface is one of the industry's best-sounding, and was recently upgraded to accommodate 24/192 and DSD over a single cable. AT did not care for the Debussy's sound when directly driving a power amp, but otherwise the lack of a front-panel alphanumeric display is about the only drawback of this superb DAC. Despite being about \$10,000 less than the next "cheapest" model, the Debussy boasts a sonic and musical imprimatur that unquestionably identifies it as a true dCS. Recently updated to support DSD on USB and all other digital inputs. tempohighfidelity.com (209)

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Berkeley Alpha DAC Reference Series \$16,000

Berkeley's Alpha DAC Reference redefines what we can expect from digital playback. The Reference is simply

stunning in its ability to render instruments as real-sounding objects in threedimensional space. But it doesn't just nail dimensionality; it also excels in

timbral vividness and delivers extraordinarily high resolution of the tiniest micro-details. What's more, the Reference performs this magic trick on all instruments simultaneously, even in the most dense and complex passages. This unprecedented resolution allowed RH to easily follow individual musical lines in a way he'd never thought possible from digital. The build-quality is many steps up from the original Alpha DAC, including a chassis milled from a solid aluminum block. The Alpha DAC Reference is an unqualified triumph. berkeleyaudiodesign.com (246)



MUSIC SERVERS & PERIPHERAL PRODUCTS

Audirvana \$50

If you have so far resisted buying any third-party music-

playback software for your Mac, Audirvana offers some compelling reasons to reevaluate that decision. Especially if you use multiple DACs or listen to higher-res files and DSD, Audirvana Plus is a more ergonomically elegant and sonically superior alternative to iTunes. And for readers who need to see and hear for themselves, you can download the trial version of Audirvana Plus for free. For fifteen days you can use the full version with no restrictions. I'd be very surprised if, by

the sixteenth day, you haven't anted up that \$50 to become a licensed user.

audirvana.com (225)





Channel D Pure Music2

\$129

Pure Music is a great piece of software at a price that even a flea market-scrounging hobbyist audiophile can afford. Combine Pure Music with any recent Mac computer and you have a front end that will play back any digital file from FLACs to lowly MP3s on up to 192/24 high-resolution files with ease. Mate this front end with a top-flight DAC and you have a digital playback system that will catapult you to the forefront of the new computer-playback revolution. **channid.com** (211)

Amarra

\$189, \$99, \$85, \$49 (price varies depending on the feature set)

When Amarra first appeared on the scene it was pricier than most of its software competition. But with current pricing that barrier to ownership has been greatly reduced. While there are sonic differences between Amarra and its competitors, the nature, scope, and perceptibility of those differences will vary drastically depending on the other components, both hardware and software, in your



system. Still, after all these years, if you want to hear how good a Mac-based system can really sound, Amarra is one of the few playback programs you must have. In the end, it's that simple.

sonicstudio.com (225)



Berkeley Audio Alpha USB Interface \$1895

The Alpha USB solves an apparently simple problem: how to connect a DAC to a computer's USB output. The Alpha USB connects to your computer's USB port, outputting a coaxial signal (on BNC jack) or AES/EBU (on an XLR jack) to your DAC. The Alpha USB's sonic magic is the result of heroic measures to isolate the "dirty" USB signal from the "clean" S/PDIF or AES/EBU output. This state-of-the-art USB converter can greatly improve the sound of virtually any USB DAC by virtue of its reclocking and noise-isolation. berkeleyaudiodesign.com (214)

Sony HAP-Z1ES \$1999

As the poster boy for Sony's "High Definition Music Initiative" the new Sony HAP-Z1ES defines what Sony sees as the future of two-channel audio. It attempts to be easy for a new user to operate, yet also be capable of the highest audio quality. As SS put the HAP-Z1ES through its paces he looked for reasons it might be not be considered a true high-performance component—and found none. If you plan to spend more than \$2000 on any digital front end—be it an audio-computer, CD player, DAC, network player, or any other front end that uses digital files as a source—and you don't audition a HAP-Z1ES, you are ignoring what may well be the benchmark digital product of 2014. sony.com (242)



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MUSIC SERVERS & PERIPHERAL PRODUCTS

Bryston BDP-1 Digital Player \$2195

The astonishing BDP-1 Digital Player is a technological tour de force that bridges

the divide between the Old World CD player and the New World of high-resolution files and music-library management. The Bryston BDP-1 performs the same function as a CD transport, but it plays data files from removable USB storage media, capable of recording thousands of hours of music, rather than from optically encoded discs. It plays files of all industry-standard sample rates at their native resolution—from CD-quality 16/44.1 up to mastering-grade 24/192—in a multiplicity of file formats too numerous to list. As with CD transports, the BDP-1 offers digital-only output in both S/PDIF and AES/EBU formats for connection to an external DAC. The Bryston BDP-1 doesn't merely "sound better"; the experience of hearing music through it is



qualitatively different. It plays music with an unprecedented purity closer to the real thing. **bryston.com** (216)

Meridian Streaming System \$7500 (Price varies with configuration)

When RH had the Meridian music server (formerly called Sooloos) for review, he gave a visiting speaker manufacturer a two-minute crash course in how to use it. Five minutes later the manufacturer exclaimed: "I'm getting one!" Such is the power of having instant access to your entire music library with the tap of a finger on the album art.

But the Meridian also anticipates from your browsing what you might want to hear.

browsing what you might want to hear and suggests alternatives. That's just the tip of the iceberg in how Meridian's server revolutionizes the way you interact with your music library. The Meridian system offers the state-of-the-art in user interface. After you've lived with a Meridian, it's hard to go back to searching for CDs.

meridian-audio.com (204)

Naim NDS

\$13,800–\$22,150 depending on power supply

In the NDS Naim has designed a network player that can handle a wide variety of sources and, with the addition of Naim's UnitiServe and a NAS drive, can become a full-fledged, highly capable music server. The best news is



that Naim has created for the NDS (and all Uniti Series products) a fabulous music-management iPad app. The NDS/UnitiServe is the perfect solution for many music lovers who want to transition to computer-based audio without the limitations of turn-key music servers or the confusion of do-it-yourself systems. Fortunately, Naim has imbued this highly capable system with a terrific-sounding DAC section. The NDS is high in resolution but without hyped "detail," voluptuous and rich in tone color without euphonic coloration, and musically vivid without being sonically vivid. The NDS also has a particularly powerful and appealing expression of music's rhythmic flow. It's the kind of sound that lets you become quickly and deeply immersed in the music, not the sound. naimaudio.com (240)

PORTABLE MUSIC PLAYERS

Astell&Kern AK100 II and AK120 II \$699, \$1299

These portable players are best thought of as iPods on steroids. With their finely brushed black aluminum cases and intuitive controls, they give up nothing to Apple in industrial engineering. But iPods max out at a tepid 48/16 resolution, whereas the AKs go to

But iPods max out at a tepid 48/16 resolution, whereas the AKs go to 192/24. The AK120 will even play DSD files! Sonically, these players simply stomp modern-day iPods and iPhones, which sound dull and dreary by comparison. Even on moderate-resolution material the AKs deliver high-end qualities like timbral richness, airiness, detail, and pace. And once you have held high-res in your hands, you will never settle for less. The AK120 II boasts dual Wolfson DACs and twice the memory capacity (a precious resource when storing hi-res material) of the AK100 II. The 120 II also has marginally more air, a smidge less grain, and stronger bass. Both players constitute wild successes, bringing true high-end sensibility and performance to portable music. **astellnkern.com (236)**



Astell&Kern AK240 \$2500

Hard to believe, but the Astell&Kern AK240 improves upon the already brilliant performance of its highly regarded predecessors. Like them, the AK240 brings true high-end performance to portable music. Finally, audiophiles can enjoy music at the sonic level they're used to—without, physically, being anywhere near a reference system. Unlike iPods or iPhones, the AK240 can play high-res and native DSD files, which can be either locally sourced or streamed across a network. That gives it a distinct sonic advantage, but even with lower-res material the AK240 delivers resolution, timbral nuance, dynamic inflection, ease, and authority previously unheard of in portable players. Compared to the AK100 and AK120, the AK240 boasts a significantly quieter background, greater purity, and even greater resolution.

astellnkern.com (248)



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