PRO AUDIO REVIEW

Gear & Software Reviews For The End-User

Cheap Trick
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July 2006

Live Sound mplifier Issue!

QSC

3602

CREST X AUDIO

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dbx DriveRack 4800

dbx

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40 SERIES:
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From the outside, the side-address condensers in Audio-Technica's acclaimed 40 Series look pretty much identical: One elegant, acoustically ideal case design. That's the way our engineers wanted it — so they could concentrate all resources under the hood, where the distinct character of each microphone is born. This left us with the challenge of differentiating near look-alikes in our ads. Solution: We've opened up the hoods to give you a tour through five unique interior landscapes. Check our web site for a closer look.

LISTEN AND COMPARE ONLINE

But there's an even better way to appreciate the warmth, power, clarity and sensitivity that make each 40 Series model so individual: Listen for yourself. We've posted audio samples on our web site so you can experience the distinctive sound of each microphone.

Inspired sound. It's what the 40 Series is all about.
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Back in 1992 when Audio-Technica introduced the AT4033, it was the microphone that began the revolution in price/performance standards for professional side-address condensers.

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AT4060
Modern Tube Technology

Audio-Technica's AT4060 cardioid vacuum tube condenser offers exacting detail, superb clarity and superior dynamic range in a classic tube design.

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A blend of technical precision and artistic inspiration, the AT4040 offers exceptionally low noise, wide dynamic range and high-SPL capability for greatest versatility.

(4o) SERIES

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HS50M

HS50M
- 5" white polypropylene cone
- 3/4" dome tweeter
- 70-watt biamplified power
- XLR and 1/4" connectors
- Room Control and Frequency Response Switches

HS80M
- 8" white polypropylene cone
- 1" dome tweeter
- 120-watt biamplified power
- XLR and 1/4" connectors
- Room Control and Frequency Response Switches

HS10W
- 8" long stroke 120-watt woofer
- Dual XLR and 1/4" inputs
- 3 balanced XLR outputs (Mix, L&R)
- Phase switch
- Low/High cutoff

HS10W

HS SERIES POWERED MONITORS

The new HS Series powered monitors were designed to be true studio reference monitors in the tradition of the famous NS10MS. That means, mixes that sound good on Yamaha HS speakers will sound good on anything. In fact, that's the ultimate test of a reference monitor. Even better than that, HS series speakers not only sound good, they look great, too.

The HS10W powered subwoofer complements the HS speakers and easily handles today's bass-enhanced music or the most dramatic surround effects. The HS10W subwoofer uses a bass reflex design cabinet that maintains high efficiency and low distortion. You can combine HS50Ms or HS80Ms with the HS10W subwoofer to create different 2:1 (stereo) and 5:1 surround sound systems. So check out the new standard in near-field reference monitors at a Yamaha dealer near you.

THE NEW STANDARD
IN NEAR-FIELD MONITORS

⊛ YAMAHA

Ode to the Record Store

By John Gatski

A paradigm shift is not only going on in the music recording side of the business, but also the distribution side. With web site ordering of music and music downloads, the days of the "big" standalone walk-in CD/record stores appear to be waning. Walking into the CD store on a Saturday and browsing away the hours through potentially thousand of titles to find that right album is on the endangered list.

To test my hypothesis, I recently went to a national chain book and CD store and was

stuff that sells. I suppose the trend is to cater to the teenage crowd.

Maybe its progress, evolution or convolution. I don't know, but this trend saddens me. I always thought the music store inventories (record, CD, etc.) were like libraries. If Mile Davis had a hundred albums - you ought to make them all available. Doesn't mean you have to stock 300 copies of *Bag's Groove*, but at least one until it sells. That is how the music stores did it for years.

Call me old fashioned, but I always thought it was an adventure to spend hours rummaging through the racks of music titles that were not the current hits. I have done it

> ## I always thought the music store inventories (record, CD, etc.) were like libraries. If Mile Davis had a hundred albums - you ought to make them all available.

browsing through what seemed liked a vanishing quantity of vintage country music CDs. I had noticed for some time that that genre of music seemed to be disappearing at this and other stores. It is getting harder to find music from the late Buck Owens, or the German import of David Allan Coe, that were easy to find in the store 10 years ago.

The nail in the coffin came when I heard a store manager telling his underling that this store was no longer going to carry all the scores of Miles Davis CD titles. After all, the wiser boss said to the earnest understudy, all of Davis' albums could not be good. "We are only going to sell the 30 or so that people will buy," he wisely explained.

This overheard conversation made me angry. As I checked out, I mentioned to the cashier what I had overheard and asked if this was official store policy. She confirmed my fears, and said "you should see the number of titles that already have been eliminated."

That is just great. The record stores are becoming like fast food restaurants." No depth to the menu selection; give 'em the

my whole adult life. From my weekend forays to the bargain bins at my college town's lone National Record Mart to the three-hour bin-prowling sessions at Tower, I found some great music.

I know these big music chain stores are hurting from competition from Internet sales and downloading, and yes, I have bought a few titles from Amazon.com — but only when I could not find them at the CD store. And yes, I use my Music Gremlin Wi-Fi MP3 downloader to sample hard-to-find songs that may have been remastered. Once I find the song or album exists, however, I still go and buy the CD or the high-resolution version if available.

If the fast food transformation formula is considered progress for music distribution, how come I don't feel good about it?

*John Gatski is publisher/executive editor of **Pro Audio Review**. Despite his best efforts, he still can't find that elusive first album by Sue Sadd and the Next. If you have a spare, email him at jgatski@aol.com.*

World Radio History

No Resolution

In John Gatski's editorial "Resolute Observations" (April, 2006) he writes about the alleged sonic benefits of high bit/sampling rate recording. Although he is expressing his personal opinion, his position as publisher gives a certain authority to his argument.

I hope readers realize that there is a significant body of opinion, backed up by scientific tests, that disputes this claim. In fact, in a series of ABX (double blind) tests conducted by the Boston Audio Society (soon to be published) listeners were uniformly unable to hear any difference between a "high resolution" source and a copy using 16-bit, 44.1 kHz processing, unless listeners were auditioning very quiet passages at 20 dB or more above normal gain settings.

Mr. Gatski assured his readers that he was able to make valid comparisons between his high-bit recordings and 44.1kHz, 16-bit versions because they were made simultaneously. But the BAS tests quickly showed that unless extraordinary care is taken in level matching, which must be better than 0.1 dB in both channels, the level differences will quickly become audible, spoiling the experiment.

When you know which is A and which is B it is impossible to avoid a bias in hearing what you want to hear. In the BAS tests, with double blind testing, the "obvious" differences disappeared, even for expert listeners.

David Hadaway
Rindge, New Hampshire

More Pros for iPods

I saw John Gatski's article ("Pros for iPod", March, 2005) and thought I should mention that here at the Seattle Center we have a number of music festivals and we are seeing iPods use more and more for dance and music tracks for festival performers. Especially handy instead of cueing up someone's homemade cassette or CD.

Bill Droege
Seattle Center
Sound Department Head

Clocking CDs

As a long time reader of *Audio* magazine from the late 1960s before John Gatski reviewed for them in the final days, I would

remiss in not commenting on his editorial (most recently 'Resolute Observations,' April, 2006). I believe he is correct that professionals should record in the highest sampling rate or format and not down to poorest quality such as MP3 from which original data is missing or even 44.1 kHz sampling.

But, some of his comments regarding his listening sessions to CD, imply to me that he is not hearing the best that CD has to offer. I understand that the Esoteric is mechanically one of the best available transports for the CD, clamping it so that there is no vibration. But a CD player is the sum of all of its parts. In the case of this player, it should have been used for its optimum playback, with its standalone clock unit. Clocks are an important part of CD players, and most of them have a simple crystal laying in or on the CD player

NEW PRODUCTS

Inter-M PA-2312 Public Address Amplifier

School AV geeks, get your inhalers out! Inter-M's PA-2312 public address amplifier has arrived. The 2312 is a two-channel amp with 4 ohm, 70V and 100V outputs. A high-pass filter, dual volume controls, XLR inputs, link, 24V DC operation in emergencies and a two-speed cooling fan – to cool the amp after a hot speech by the principal! – round out the feature list. Price: $667.
Contact: Inter-M at 61-874-8870, www.inter-m.net.

SLS RLA/4 Line Array Module

These days line arrays are the battlefield for sound reinforcement and installation speaker makers. Everybody's got a line array or is building one. SLS Loudspeakers is a grizzled vet. The latest weapon in its arsenal is the RLA/4 line array module. The driver complement is made up of twin 6.5-inch woofers and one of SLS' signature PRD planar ribbon HF drivers. The RLA/4's cabinet is 13-ply Baltic birch and includes rigging. Price: $1,895.
Contact: 5LS Loudspeakers at 417-883-4549, www.slsaudio.com.

Lectrosonics DMPA12 Amplifier/Processor

Bulking up its DM line-up, Lectrosonics is trotting out the DMPA12, a 12-channel amplifier/mixer/processor. The DMPA12 utilizes Lectro's LecNet2 DSP/mixing system. Amplifier power is Class D 10W per channel. DSP includes delay, filtering, compression and limiting. A USB interface provides communication with a computer. Price: approx. $2,665
Contact: Lectrosonics at 505-892-4501, www.lectrosonics.com.

Listen Technologies Stationary IR Wireless Tour Systems

Listen Technologies latest wireless transmission system for tour-style applications is the counterintuitively named Stationary system. The Stationary is a wireless infrared system offering up to 48 receivers and up to four channels of stereo transmission through stetho-style receivers and lanyard receivers. A half-rack transmitter drives paddle emitters/radiators via CAT5 cabling. Onboard transmitter DSP allows for stereo, delay and other processing to aid signal distribution. Price: basic system (transmitter, radiator, four receivers, charger) starts at $2,270.
Contact: Listen Technologies at 801-233-8992, www.listentech.com.

Can the performance features of Neutrik's XX series connectors improve your love life?

NC3FXX

Possibly. You see, research has long established that the key to a better love life is greater confidence. And that's where we can help.

While our X series connectors are the industry standard, our next generation XX series provides even greater performance. The increased durability, ease, and speed of XX brings a fresh new sense of self-assurance to every area of your life.

Exactly how do we improve performance? Notice the ingenious "cage" type female contact. It creates multiple contact points that increase conductivity and reduce wear.

"CAGE"
CONTACT POINTS
SOLDER STAYS WITHIN THE SOLDER CUP

Now check out the aptly named "solder cup." It keeps the solder away from the contact, for faster, easier assembly. You're feeling remarkably in sync with your mate, and you're not even sure why.

How else does the XX help? Our new ground contact enhances conductivity between the chassis and cable connector. And our unique latch design improves security, ease, and speed. Your co-workers keep asking you why you're smiling.

GROUND SPRINGS

But can the XX stand the test of time? No problem. Our unique guide flaps ensure easy, secure alignment. And our chuck type strain relief has higher traction teeth, for increased retention under the tougest operating conditions. You're going to have a great time on that cruise to nowhere.

CHUCK & INSERT ALIGNED
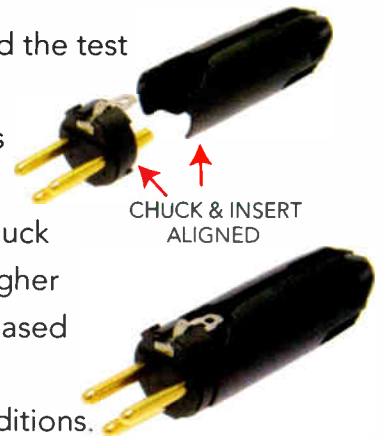
The moral of this story? If you want a better love life you need a better connector. The XX series from Neutrik.

BY DAVID RITTENHOUSE

dbx DriveRack 4800

A few years ago, I reviewed the DriveRack 480, so in turn I will review the new flagship for dbx, the 4800. The 480 was a great unit breaking new ground for dbx; but has dbx changed and fixed some of its limitations? The bigger question here is, as most manufacturers tend to update their products from year to year, has dbx actually improved on their product or just put it in a prettier box? Can the 4800 turn heads and budgets like its predecessor? Let's find out what the hype is all about.

FEATURES

The 4800 ($4,999) starts off with a new, powerful 96 kHz DSP (which is switchable down to 48 kHz) engine. Dbx has kept the standard configuration of four inputs to eight outputs found on the 480 and from other competitors. These are improved upon by adding fully digital ins and outs via AES/EBU XLR connectivity. You can also order the 4800 with the optional CobraNet if desired. One of the hippest new features is the QVGA display that shows, in color, all the routing and processing in the signal chain. This, along with the new accessibility brought on by the integration of the HiQnet/System Architect, allows full labeling of all in and outs. No more chart sheets of program numbers and which output does what; it is all right on the screen in front of you. New to the 4800 is an output gain control knob on the face of the unit below each output meter. This knob also doubles as the mute for its output with a lighted ring denoting red for

Fast Facts

■ **Applications:**
Live sound, sound reinforcement, installation

■ **Key Features:**
Four in/eight out; 96 kHz; onboard DSP; Ethernet HiQnet/System Architect control

■ **Price:**
$4,999

■ **Contact:**
dbx at 801-568-7660, www.drive-rack.com, www.dbxpro.com.

mute and green for on.

Within the 96 kHz DSP you have full band-pass filters, plus crossover and routing configurations with Bessel, Butterworth, and Linkwitz-Riley filters. Every input also has a 31-band graphic and a nine-band parametric EQ. You also have two selectable insert points per input and output for dynamics processing such

as the classic dbx compression and limiting, noise gates, de-essers, AutoWarmth (dbx says – 'AutoWarmth is a dynamic enhancement designed to compensate for the ears' loss in low frequency response as the signal level decreases'), subharmonic synthesis and advanced feedback suppression. Dynamic range has been rated at 113 dB A-weighted, with a frequency response of <10 Hz –50 kHz (+0/-3 dB at 96 kHz). Available input delay is selectable up to 682 ms, and output delay is selectable up to a total of 1,365 ms (shared between the outputs).

Controlling the DriveRack 4800 is quite a bit different than the old 480. Dbx no longer offers the remote like they did with the 480R, but it can now be controlled via Ethernet (HiQnet) to your own tablet PC. You can even go wireless by using a wireless router. The use of HiQnet/System Architect also allows access to other units like Crown I-Tech series amps. All of this is now integrated and controlled at your fingertips (tablet pen) for your entire system.

IN USE

The 4800 has been with me for months now. It was only supposed to be on loan for a couple of shows but I could not let it leave. I have taken it out on everything from gospel acts to Baltimore Symphony Orchestra performances. It functions in a much more logical way than the old 480, with control and accessibility features

> Now, it is great to talk about the control and functional ease of the 4800; but how does it sound? Well, in a word, great!

that are unrivaled. The ability to control the entire system wirelessly is great for when you have a multi-zoned set-up. You can walk into the zone mute, un-mute and EQ for that zone instantaneously. I say instantaneously but when working from the wireless Ethernet there can be some lag so for show time application it is good to revert back to a hardwired set-up.

One of the most improved features is the speaker processing and pre-EQ functions. Rivaling the old 480, on which you could run out of filters or have filters that were not variable, all of your processing needs are taken care of by the new 4800. What this means is you can accurately match the manufacturer's specs for each speaker, including driver alignment delays. So your system can now sound like it was meant to out of the box.

The addition of the HiQnet/System Architect control protocol is fabulous. You can now, in real time, fully integrate your entire system. No more multiple programs running on your laptop (e.g. one for the

amps, one for the processing, one for the digital EQ – now it all comes together). And no more weird adapters like a RS232-to-USB just to run your laptop; it's all over

Ethernet! The 4800 works best when connected to a tablet-style PC for its ability to manipulate functions via touch screen. Ringing out a room can now be done with just a stroke or two of the pen. One of my later complaints about the 480 was it was very hard to get in and around the unit without having the remote. On the 4800 you can access everything from the front panel and nothing is more than a one or two button push. I mentioned earlier about the labeling of all your ins and outs per pro-

gram. This is a great time saver especially for a company like ours that has upwards of 40 different programs for all combinations of speakers and applications. For the old 480 we had a full sheet charted out per program of the input and corresponding output that you would have to reference every time you used it. Now it is all on the front panel display in full color.

Now, it is great to talk about the control and functional ease of the 4800; but how does it sound? Well, in one word great. It is just as good and better than the 480 due to the 96 kHz upgrade and greater "tuneablity" of the pre and post EQ. If you have used any product from the DriveRack family you will be doubly impressed by the 4800.

SUMMARY

Over the years I have noticed what seems to be an evolutionary/de-evolutionary ebb and flow to mankind's pursuit towards perfection. We never achieve perfection, but we always seem to get closer, and I think dbx has its greatest shot at it yet with the 4800. They have beat all the shortcomings from its predecessor and forged ahead with future thought, to not only how this unit will be used today, but also next year and years after.

*David Rittenhouse is a senior live sound engineer at RCI Sound Systems and a regular contributor to **Pro Audio Review**.*

Product Points

Plus:
+ 96 kHz DSP
+ Full color QVGA display
+ HiQnet
+ Gain/mute knobs on the front panel with lighted rings

Minus:
- People stuck in the archaic 1990s might feel left out
- The overwhelming urge to get a tablet PC

The Score:
More than just an upgrade and totally new and much better product!

Review Setup:

Yamaha PM5D, Yamaha 3500-40 consoles; EAW KF760, KF761, KF730, KF300 speakers; laptop PC.

Contracting

BY ANDREW ROBERTS

Lab.gruppen C68:4 Power Amplifier

I will admit that, as recently as a few years ago, I was one of those archaic souls who thought of power amplifiers as relatively simplistic devices. To me, it seemed that either an amp worked or it didn't. Turn on the switch, and juice comes out the back. In the years since my personal "dim-bulb era", amplifiers have gone from lumbering beasts to lean, thrifty, reliable, dynamos. I will also admit to being amazed at how something that you can lift with two fingers can move so much air. Nowadays, as if power-to-weight ratios weren't enough, amp manufacturers are creating amps that have on-board diagnostics and extensive protection and remote control capabilities.

Here in the states, Lab.gruppen (a Swedish amplifier manufacturer) existed in relative obscurity until the late 1990s. Today, their products appear on many high profile tours and their fP series amps are known as a benchmark for quality and competence. Recently, the folks at Lab.gruppen have released the C Series of amps - designed specifically for use in permanent installations.

FEATURES

There are four models in the C series line and they are all of the two-space, four-channel variety. The model I received for review was the C68:4, the flagship. The amp has a claimed power output of 1700 watts per channel (remember there's four) into 4 ohms (with the amp's Voltage Peak Limiter set to maximum - more on this in a moment), representing a massive power density. As

would be expected from any designated installation amp, the C68:4 is comfortable driving both low-impedance loads and 70V/100V, distributed systems.

The C68:4, like the fP series amps from Lab.gruppen, has a Class TD topology. Purportedly, the patented TD process combines the efficiency of a Class D amp with the sonics of a Class AB amp. The

amp features a regulated switch-mode power supply that, according to convention, should yield greater stability in the face of voltage drops or fluctuations.

The amp has an adjustable global input gain of +23 dB to +44 dB in 3 dB steps, meaning it should be possible to achieve a good signal-to-noise ratio regardless of incoming signal level (within reason of course). There are three dedicated DIP switches on the rear panel that, depending on the combination of their settings, can globally compensate for the level of incoming signal. After that, each channel has its own front panel attenuator that will dip from 0 dB to –Inf.. Appropriately, those volume controls are hidden behind a security panel on the front of the amp.

Other prominent features of the amp

include a Current Peak Limiter (CPL) and a Voltage Peak Limiter (VPL). The Current Peak Limiter is an internal limiter that ensures that the amp will not try to deliver more current to the outputs than what is physically possible for the transistors. It does this by comparing the output voltage and the output current - maintaining operation in what Lab.gruppen

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...you could drive both a distributed sound system (for paging perhaps) and a low impedance system (for performance), all from one C68:4.

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calls the Safe Operating Area (SOA). The Voltage Peak Limiter (VPL) is a process that matches the amplifier's output to the connected speaker load. This is achieved through another group of DIP switches on the back of the amp. There are eight separate settings for Voltage Peak Limiter ranging from 42 to 141. Fortunately, this function, unlike the previously mentioned CPL, can be individually applied to each of the amp's channels. That means that, with four channels, you could drive both a distributed sound system (for paging perhaps) and a low impedance system (for performance), all from one C68:4.

The C68:4 also has an extensive system of fault warnings and protections. First among them is the Very High Frequency Protection system. This sys-

Fast Facts

■ **Applications:**
Installation, live sound

■ **Key Features:**
Four-channel; Voltage Peak Limiter; Current Peak Limiter; Power Average Limiter; Class TD; compatible with NomadLink

■ **Price:**
$4,895

■ **Contact:**
Lab.gruppen/TC Electronic at 818-665-4900, www.labgruppen.com.

Contracting

tem detects continuous ultrasonic program material (starting at 10 kHz) at full power and mutes the output channel if the oscillation lasts for six seconds in an effort to protect high frequency speaker components. This feature is based on the

assumption that full-scale VHF signals do not appear in natural sources (like music or speech)- especially in a non-transient fashion. The VHF protection system has a corresponding LED on the front panel that indicates when it is functioning.

The amp also has a protection functions for very low-frequency DC output (also featuring muting and an LED indicator), "Hi" impedance to indicate an open load (LED indicator), and a "Lo" impedance protection system that indicates a short circuit or excessive load (muting the output). The amp also has a high temperature detection system that measures temps at the power supply and each amp output. When a moderate overage occurs, it will illuminate a front panel LED (flashing) and send info to the Nomad Link network. If the temperature reaches a dangerous level, the amp will mute outputs and the LED will be solidly illuminated.

One of my favorite protections on the C68:4 is the Power Average Limiter (PAL). This controls performance between the power supply and the mains inlet, limiting the maximum average power consumption according to the power supply capabilities. On larger models like the C68, it can be a show saving feature as it could prevent the amp from tripping the mains breaker. Whenever any protection circuit or limiters engage in the amplifier, front panel indication will notify the user with exact feedback on what limiter or protection circuit is engaged.

The C68:4, like so many modern amps, has a soft start sequence for power up to prevent tripping a mains breaker. Also, if many amps are coming on at the same time, you can use the Nomad Link network to establish a power up sequence.

IN USE

After installing the amp in my rack, I found the C68:4 to be an impressive design. For example, since it draws air from the front to the back, there are two large foam air intake filters that can be accessed for cleaning by removing the front panel cover (which also yields access to the channel volume controls).

NOMADLINK

The NomadLink Bridge 60E ($1,245) is the control counterpart to the C series amps. Bridging between the NomadLink daisy-chained network and standard Ethernet/LAN, it provides utilization of any standard Ethernet architecture when connecting a PC running the DeviceControl software. It has the ability to manage up to 60 units within the confines of cable length restrictions. It is a one-rack space unit that has a modest display and user interface. When used with a PC, it is a formidable tool for controlling a large group of amps. In addition to the previously mentioned power sequencing and monitoring capabilities, the system has some nice security features and I like how it has Ethernet connections in the front as well as the rear - facilitating easy hookup for periodic maintenance.

There are a slew of LEDs on the front of the amp- most of them I've already mentioned. There are also LED indicators for level (-4 dB, -10 dB) and Bridge mode. The front panel is also home to a power switch and a remote switch (to transfer power on/off commands to NomadLink network control). The rear panel has a large bank of DIP switches that address some of the previously mentioned functions as well as bridge mode (A+B and

C+D), a mode function for the VPL (hard or soft) and a switch labeled "Option Active" for future upgrades (thinking ahead, I like it!). Input signals are ushered in through Phoenix connectors and outputs are via screw terminals. Also, there are in and out connections for the network (Ethercon-housed RJ45) and a mains AC cord (30a).

At one facility I wired in the C68:4 to power stage monitors and a subwoofer. Like the fP series amps from Lab.gruppen, the C68:4 is powerful and very clean sounding. In my shop, I tested some of the protection features and it responded flawlessly- especially when presented with a dead short. The only feature missing is high-pass filtering. Yes, it protects against DC but it would be nice to be able to utilize some type of rolloff in the 30 Hz – 80 Hz range, further protecting loudspeakers.

SUMMARY

This amp is one of the, if not the most, comprehensive amplifier I have ever used. It has a remarkable set of features designed to protect itself and other components in a sound system (including the user, who will benefit from lower maintenance costs and improved system performance). It has a massive power density and it is incredibly lightweight considering the amp's power output capability. The C68:4 also has enormous potential for remote control when used with Lab.gruppen's NomadLink network. At $4,895.00 it is not cheap, but remember it delivers four channels of high output power. This is not a budget piece, it is intended for high quality systems where big power and excellent control are required.

Andrew Roberts, a regular contributor to **Pro Audio Review**, *is a sound reinforcement and recording engineer.*

Review Setup

Midas Venice 320, Allen & Heath GL2400 consoles; JBL SRX700 series cabinets; Audio-Technica, Shure, Audix and Sennheiser mics; Rane, TC, BSS, PreSonus processors.

NEW PRODUCTS

Carvin is celebrating 60 years in business. Congratulations!

A party thrown for Sting, after receiving an honorary doctorate, featured **Turbosound** Qlight speakers. Turbo TA and TFM speakers were running sound for David Gilmour's recent tour. Gilmour also used a Neumann KMS 105 on tour.

Soundcraft scored when SSE Hire, a rental house in the UK, bought six of the new Vi6 digital consoles.

For announcing its new fall lineup, Fox Broadcasting used **JBL** VerTec VT4889, VT4888, VT4887 speakers and VT4882 subwoofers (amongst other VerTec speakers). The operation was run by On Stage Audio. On the other end of glitz, VerTec VT4888 speakers and VT4882 subwoofers were also for a Shaolin Monks show in Dubai. Power was provided by **Crown** MA3600 and MA5002VZ amps.

Peavey's Versarray 112 line array modules and 218 subwoofers were used at the Dallas Guitar Festival. Power was provided by **Crest** Pro 5200 and Pro 9200 amps.

Creative Technology, a rental house in LA, has added 68 dV-DOSC modules, 22 dV-SUB subwoofers and 32 LS series amps to its **L-Acoustics** offerings.

FOH Engineer Gavin Pearce used a **Focusrite** Saffire interface for numerous recording and editing duties on the recent James Brown tour.

Sennheiser Evolution e 602 II Microphone

The latest from Sennheiser's Evolution line is the e 602 II. The e 602 II is a cardioid mic with an aluminum body making it significantly lighter than the original e 602. The e 602 II is designed for bass-heavy instruments such as kick drum tuba, bass guitar amps. It has an integrated standmount. Price: $289.
Contact: Sennheiser USA at 860-434-9190, www.sennheiserusa.com.

American Audio MCD-710 CD Player

For DJs (or wannabes) American Audio has a new CD/MP3 player out, the MCD-710. The MCD-710 is a dual-well/dual-drive rackmountable unit. It has complete controls for operating each well independently. On the CD side the 710 has the usual features and controls such as a jog/shuttle wheel, loop functions, BPM and a "Bop Effect." For MP3 playback it has added folder and organizational functions. Some functions such as pitch control work for both. Price: $399.
Contact: American Audio at 800-322-6337, www.adjaudio.com.

Gemini Sound XTR-500 Speaker System

For the sound pro on the go the Gemini Sound XTR-500 might be of use. The XTR-500 includes two passive satellite speakers with 10-inch woofers and a 15-inch powered subwoofer. The sub's amp outputs 230W to the sub and 85W to each satellite and it has a cooling fan. Wheels and cables are included. Price: $899.
Contact: Gemini Sound at 732-738-9003, www.geminidj.com.

PWS Helical Antenna Kit

Wireless antenna products are usually not a hot topic at pro audio chat rooms but when they are needed they ARE needed. Professional Wireless Systems – PWS, has a new helical antenna kit for sale. It consists of two HA-8089 helical antennas, two LPDA bat-wing-style antennas and a bunch of cables. It's all housed in a weather-resistant rolling case. Price: $1,850.
Contact: PWS at 407-240-2880, www.professionalwireless.com.

BY ANDREW ROBERTS

QSC PLX3602 Power Amplifier

A s most of us probably know, we audio professionals operate in a field where there is an accelerated evolution of the tools of our trade. Without some type of refinement or improvement, it is rare for any given product to have a life cycle beyond a few years. It has been nearly a decade since QSC introduced its PLX series of amplifiers. These less expensive brethren of the company's Powerlight series proved to be an instant success with musicians, institutions and even sound providers like myself. The source of their popularity was that they provided significant power and minimal weight at a very affordable price. My sound reinforcement business owns quite a few PLX amps and we have used them on hundreds of jobs in all manner of situations and operating conditions. I have seen them prevail through dust clouds, voltage drops and over-zealous band engineers. They have proven to be very competent and remarkably dependable. Therefore, it was with much anticipation that I received for review the PLX3602, a replacement for the venerable 3402, the former flagship of the PLX series.

FEATURES

To those familiar with the original PLX line, this new breed will seem like familiar territory, but with more refinements than overhauls. One change is that there are now six models in the line, including two models specifically designed for 4 ohm and 8 ohm loads (PLX1104 and PLX1804).

Like its predecessor, the 3602 is two

rack spaces tall. From there, the two amps have slight differences and, in some cases, more significant variations. For starters, the front panel looks remarkably different. QSC has eschewed the old gun metal gray for a new vibrant silver finish that looks like something from Mr. Scott's engine room on the Enterprise. Gone too are the

idiot-proof, making it easier to slip the amp into bridge mono mode or some other mode that may cause problems for inexperienced users. The new switches allow access to some previously seen functions (bridge/stereo/parallel, clip limiters, high-pass filters) and some new functions (low-pass filters).

Throughout the course of the band's boisterous performance, the 3602 delivered enough drum fill to dislodge tooth fillings!

old elephant ear style front panel handles. They have been replaced by discreet finger holds formed by flanges on the faceplate. As with the 3402, front panel controls are a power switch, volume attenuators and LED indicators (power, signal [-35 dB], -10 dB, clip, bridge mono, parallel).

The back panel is home to input connections (XLR and 1/4-inch balanced), speaker output connections (Speakon and binding post), and a host of configuration switches. Gone are the recessed DIP configuration switches from the first generation PLX units (which I spent many a moment trying to adjust - crouched over, Leatherman in hand, Maglite in mouth, trying to see the tiny toggles). The old DIP switches have been replaced by new switches that you can manipulate with your fingers. While I welcome this change, it could be argued that the amp is now less

As you may have gathered by the model number, the new amp is slightly more powerful than its predecessor. The 3602 is rated for 1,100 watts per channel at 4 ohms (20 Hz – 20 kHz, 0.05% THD) and 3600 watts in bridge mono (1kHz, 1.0% THD, 4 ohms). I would like to interject that, if the 3602 is as robust as its ancestor, it will be comfortable driving 2 ohm loads (*It is rated at 1,800W for 2 ohms – Ed.*). Continuing the PLX tradition, the 3602 is very thrifty too, consuming just 11.5 amps when powering a 4 ohm load at 1/8 power (the closest rating to actual use). It also weighs in at a remarkably light 21 pounds.

IN USE

The most fitting assignment for an amp with this level of power output is sub-

Fast Facts

■ **Applications:**
Live sound, sound reinforcement

■ **Key Features:**
Two-channel; bridge/stereo/parallel operation; clip limiters, high, low-pass filters

■ **Price:**
$1,549

■ **Contact:**
QSC Audio Products at 714-754-6175, www.qscaudio.com.

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woofer duty and, being the conscientious reviewer, I felt compelled to oblige. I used the 3602 to power a pair of 4 ohm cabinets (each loaded with four 15-inch woofers) and a pair of single 18-inch 8 ohm subs. The first event was a concert featuring a local jazz group in a lively, confined acoustic environment. The 3602 performed wonderfully - dealing with kick transients and delivering robust lows from the 4 ohm subs. While I had a high-pass filter set at 25 Hz on my speaker processor's sub output, it was nice to have the onboard rolloff at 33 Hz, had I needed it. Later, in bridge-mono mode, the amp was brilliant in powering the 8 ohm cabinets, even for more aggressive musical styles. It displayed lots of headroom and delivered a deep wonderful bass sound.

At another event, where I was doing sound for a group of Nigerian musicians, we were presented with a drummer who needed some more thump in his monitors. Since we were short on processing for such a request, I pressed the 3602 into service. I put the amp into stereo mode and applied the 100 Hz internal high-pass filter to one channel (for mid/high) and a low pass on the other channel (for a sub). The drummer had a big grin on his face during the sound check and, as I listened over his shoulder, I too was impressed with the clean thump of the kick drum sound. Throughout the course of the band's boisterous performance, the 3602 delivered enough drum fill sound to dislodge tooth fillings (and contaminate the drum sound for FOH).

At another event, I used the 3602 to power a string of 8 ohm cabinets for public address. The event was at the house of former President Bill Clinton and his wife, Senator Hillary Clinton. With more than 150 people gathered in their back yard, we were required to provide discrete distributed sound for several speeches. This included powering three cabinets per amp channel. While this is not a particularly stressful chore for an amp, it was a test of the amp's flexibility. I put the amp in full range mode and applied the internal 33 Hz high-pass filter. There was clean intelligible speech heard throughout the listening area.

SUMMARY

With increased power, more features, an easier to use configuration section, and a lower price ($1,549) than its predecessor, the PLX3602 is sure to carry the PLX popularity torch a long way. This amp is a wonderful choice for institutional or worship use but, at only 21 pounds, it is a blessing for portable sound reinforcement. If its level of reliability is equal to that of its PLX ancestors, the PLX3602 should be a staple in sound systems worldwide. If you want lots of power, good sound, low weight, and proven reliability at an affordable price- this is your amp!

Andrew Roberts, a regular contributor to **Pro Audio Review**, *is a sound reinforcement and recording engineer.*

Review Setup

Midas Venice 160, 320 mixers; Audio-Technica, Shure, Audix and Sennheiser mics; Rane, TC, BSS, PreSonus processors; JBL SRX and Community loudspeakers.
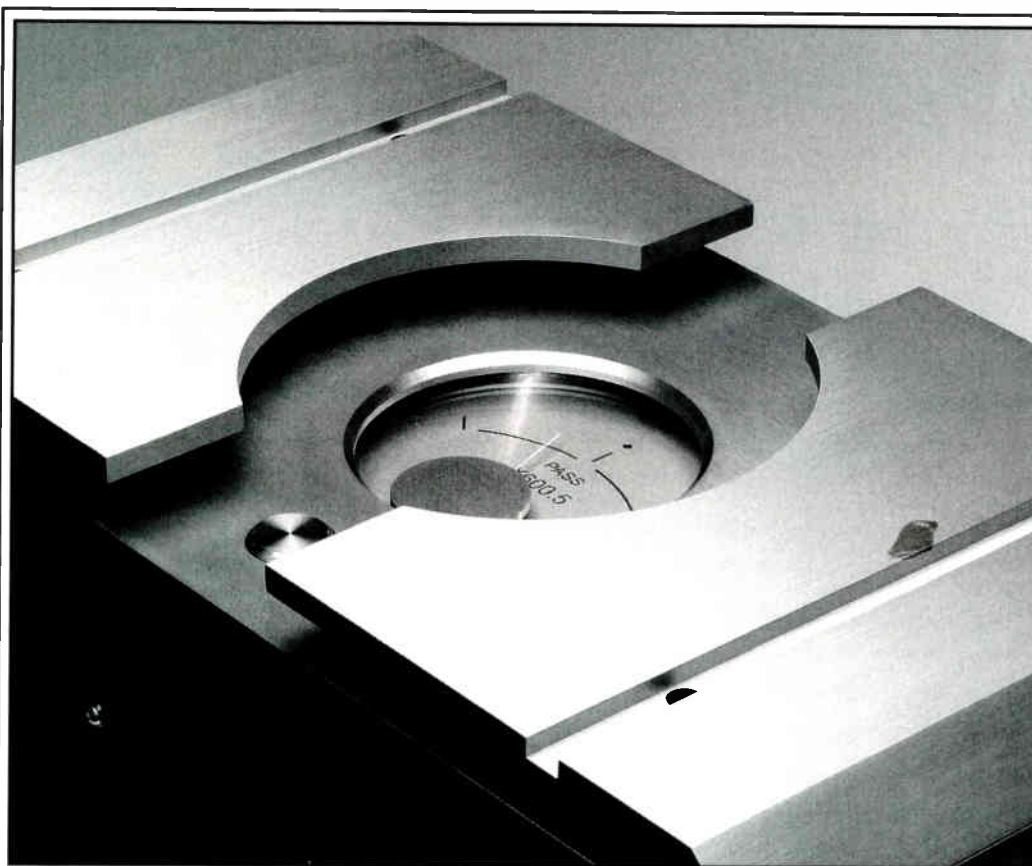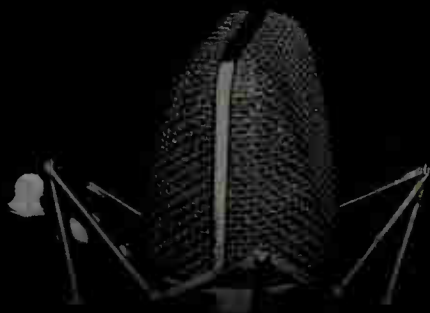
QSC PLX3602 Power Amplifier Bench Measurement Data

Maximum Power (1 kHz, 1% THD)

Stereo 8 ohm load	800 W, 29.0 dBW
Stereo 4 ohm load	1.2 kW, 30.8 dBW
Stereo 2 ohm load	1.6 kW, 32.0 dBW
Bridged 4 ohm load	3.2 kW, 35.0 dBW

Dynamic Output Power

Stereo 8 ohm load	977 W, 29.9 dBW
Stereo 4 ohm load	1.8 kW, 32.6 dBW
Stereo 2 ohm load	2.25 kW, 33.5 dBW
Bridged 4 ohm load	4.5 kW, 36.5 dBW

Dynamic Headroom

Stereo 8 ohm load	0.9 dB
Stereo 4 ohm load	1.8 dB
Stereo 4 ohm load	1.5 dB

THD+N at near rated power (20 Hz - 20 kHz)

Stereo 8 ohm load	<0.05% @ 600 W
Stereo 4 ohm load	<0.04% @ 1000 W

THD+N at 10W output (20 Hz - 20 kHz)

Stereo 8 ohm load	< 0.05%
Stereo 4 ohm load	< 0.04%
Damping Factor, 50 Hz re 8 ohm	645

Output Impedance magnitude at:

50 Hz	12.4 milliohm
1 kHz	17.5 milliohm
5 kHz	61.8 milliohm
10 kHz	117.9 milliohm
20 kHz	226.5 milliohm

Frequency Response

20 Hz - 20 kHz	+0.00, - 0.25 dB
	- 3 dB at < 10 Hz, 65 kHz

Voltage Gain, 8-ohm load
62.3X, 35.9 dB

Input Impedance
20 kilohm

Sensitivity for 0 dBW
45.4 mV, -24.6 dBu

Sensitivity for rated output, 775W, 8-ohm
1.26V, 4.2 dBu

A - Wtd noise
250 uV, 81.1 dBW

A - Wtd S/N re: rated power
110 dB

Channel Separation, 20 Hz - 20 kHz

Ch 1 > 2	> 64 dB
Ch 2 >1	> 70 dB

Notes: Unless otherwise noted or implied, all measurements are made with 8-ohm loads from the balanced inputs, stereo mode with the volume set to maximum. Both channels driven for all distortion and power measurements. Measurements made on one channel are made on channel one.

-Bascom King

QSC PLX3602 Power Amplifier Bench Measurement

The PLX3602 is the currently most powerful amplifier in the PLX series. A switching power supply is utilized resulting in low weight relative to the amplifier's output power. The output stage uses a three-tier Class H topology resulting in higher efficiency over the full output power range.

Frequency response as a function of open circuit, 8, and 4 ohm loading is plotted in **Figure** 1. As can be seen, the
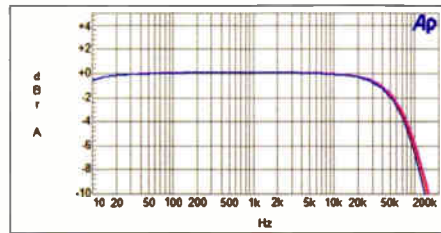
Figure 1: Frequency response at the 1W/8 ohm level as a function of output loading. Magenta = open-circuit, blue = 4 ohm.

response is very independent of load suggestive of a high damping factor. Frequency response of the 33 Hz and 100 Hz low-cut and the 100 Hz high-cut filters are shown plotted in **Figure 2**.
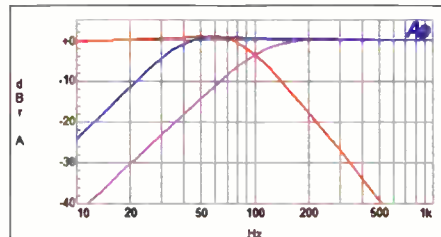
Figure 2: Frequency response of the low and high-cut filters. Red= 100 Hz high-cut, magenta = 100 Hz low-cut, blue = 33 Hz low-cut.

THD+N for a 1 kHz test signal and for 8 and 2 ohm loading in the stereo mode are shown in **Figure** 3 and 4 respectively.
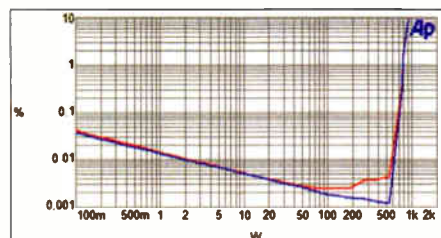
Figure 3: THD+N of 1 kHz signal vs. power output for 8 ohm loading in stereo mode. Red = Ch 1, blue = Ch 2.

Distortion is quite low in this design for this testing condition. THD+N as a function of frequency and power for 4 ohm loading is plotted in **Figure** 5. The frequently
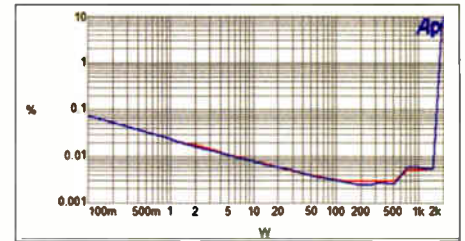
Figure 4: THD+N of 1 kHz signal vs. power output for 2 ohm loading in stereo mode. Red = Ch 1, blue = Ch 2.
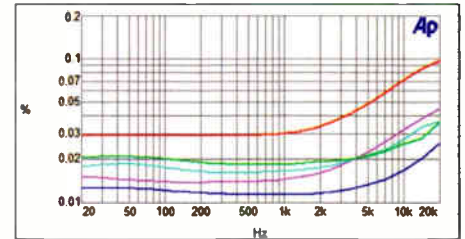
Figure 5: THD+N vs. frequency at different power levels for 4 ohm loading. Red = 1W, magenta = 10W, blue = 100W, cyan = 500W, green = 1 kW.

observed rise in high frequency distortion in many amplifier designs is in evidence here. A spectrum of the distortion and noise residue of a 1 kHz test signal at10W into a 4 ohm load is shown in **Figure** 6. Amazingly low is the amount of AC line harmonics. The signal frequency harmonics are dominantly even indicating good half-cycle symmetry at this power level.
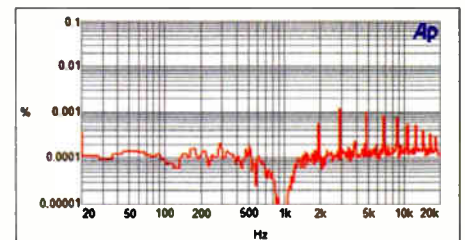
Figure 6: THD+N spectrum of 1 kHz 10W signal into 4 ohm.

Channel separation is plotted in **Figure** 7 for both testing directions. Unusual is the more or less flat nature of the separation vs. frequency. Much more usual is a characteristic that rises with frequency due to capacitive coupling in the signal circuitry.

Figure 7: Channel separation vs. frequency. Red = Ch 2 > Ch1, magenta = Ch2 > Ch1.

BY WILL JAMES

Crest Audio CC4000 Power Amplifier

I'm not sure how long you have been around the pro audio business, but odds are, if you've been in the business any length of time, you have used Crest power amps at some time in your career.

The 1980s saw the industry standard of power amps with models like the Crest Audio 8001, 9001 and the mighty 10001. They may well be setting a new standard with the new CC series.

FEATURES

The model we tested was the CC4000, so called 4000 because the amp is rated at 4,000 watts RMS in bridged mono mode at a 4 ohm load. But the other specs are equally impressive, with Crest rating this power amp at 2,000 watts @ 2 ohms, 1,350 watts @ 4 ohms and 800 watts @ 8 ohms, all ratings with both channels driven equally at 1 kHz.

Right out of the box, the CC4000 is a pretty good looking piece of gear, with a stout black steel case and a slick looking brushed silver control face with black expanded metal air intake. The control face contains two rotary input sensitivity dials, labeled Channel A and B (as opposed to L and R). Directly to the center of the two rotaries, are the LED input level stacks, displaying signal and amplifier status. The only additional item on the front panel is the Power On/Off rocker switch. And in the Crest tradition, it's a combination magnetic circuit breaker and On/Off switch.

The rear panel is equally efficient, with the IEC power connector residing alongside the output connectors. The outputs consist of a Neutrik NL4 and binding post (female banana) connector per channel,
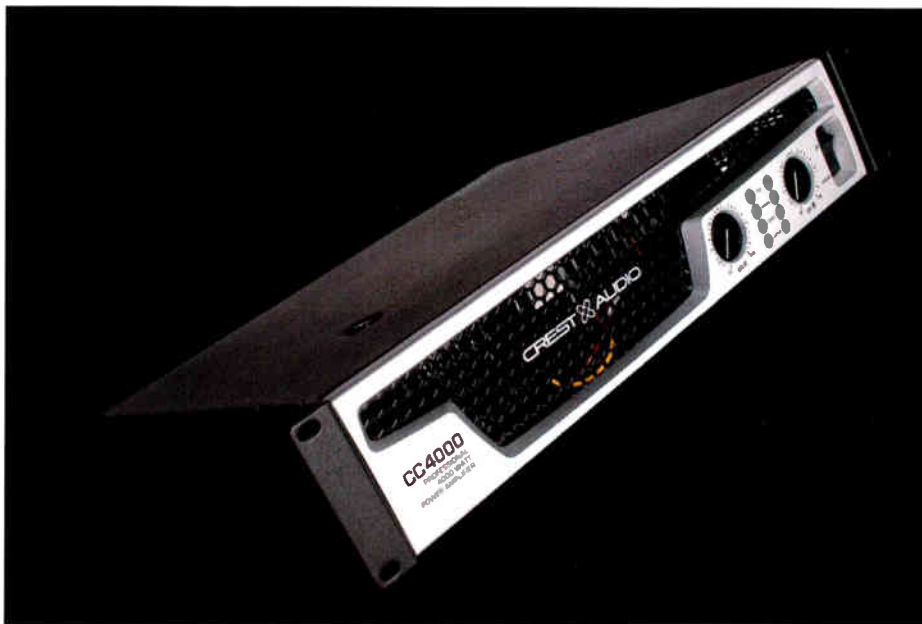
as well as a legend depicting the proper connector alignment for stereo and bridged use. At the other side of the rear panel, are the input connectors which are Neutrik combi 1/4-inch and XLR style connectors. This location also contains a three-position switch, allowing for input connector assignment to stereo (separate Channel A and B information), parallel (identical Channel A and B information), and bridged mono. If you have never done it, bridging mono is the somewhat unnatural act that combines the output of both channels into functioning as one power amp by using the positive swing of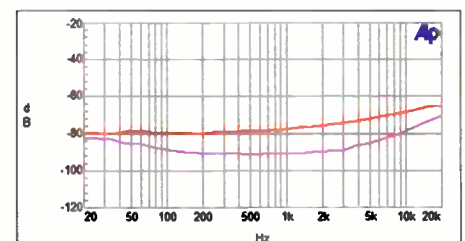 the wave in the Channel A amp and the negative swing of the wave in the Channel B amp. The output is then taken from the red banana output of one channel and the red banana output of the other channel. This particular Crest amp also assigns the bridged output to the Channel B Neutrik NL4 output connector.

IN USE

I had this Crest CC4000 for only a short time, so I decided to give it a brutal two-day workout. First off, I wanted to hear the transparency and response speed (the response speed is commonly referred to as the slew rate, which essentially describes the rate at which an amplifier can process

change in volume, and is measured in volts per microsecond). Although Crest rates this power amp very modestly at 15V per microsecond, it felt as though it responded much faster. The CC4000 was employed as a monitor amp, with Channel A in the lows character and Channel B in the highs character.

It acted out both parts with considerable aplomb and dexterity, responding to my old ears, with what I thought was much faster than rated by Crest. The vocal range was clean, concise and responsive, while the lower frequencies responded with equal speed and clarity. The CC4000 might be a little much for most monitor applications, but I felt that it would allow for the most critical listening environment in a loud, crowded casino showroom (Harrah's, to be specific).

We continued the show the second evening, which by the way, was a Doors tribute band called Peace Frog. The second night we decided to let it rip, and employed the CC4000 as a subwoofer amp. We connected the amp to a pair of A-Line Acoustics LS218 double 18-inch subs, and let the games begin. The CC4000 was even more at home as a sub amp, delivering incredibly smooth response at extended frequencies of 40 Hz with no difficulty at all. The kick drum had excellent

Fast Facts

■ **Applications:**
Live sound, installation

■ **Key Features:**
Two-channel; 2,000 watts @ 2 ohms, 1,350 watts @ 4 ohms and 800 watts @ 8 ohms; anti-clip limiting processor

■ **Price:**
$1,560

■ **Contact:**
Crest Audio at 201-909-8700, www.crestaudio.com.

BY DAVID RITTENHOUSE

Ultimate Ears UE-10 Pro In-Ear Monitors

n this modern age the thought of placing an element in your ear so that you can hear and perceive music or speech is not far fetched. Actually, it is done more and more by musicians everyday. Yet as I say this, I think back to my youth of 8-tracks and huge headphones and the thrill of listening to Pink Floyd' s *Dark Side of the Moon* for the first time and I realize that this concept started from an imagination into a reality only to be fully fulfilled at this point in time. With this grandiose statement I am here to tell you about the new in-ear monitors from Ultimate Ears, the triple driver UE-10 Pros.

FEATURES

This is a triple loaded in-ear driver from Ultimate Ears. It encompasses two low-end drivers and a single high-end driver with a proprietary surface mount crossover network. Sensitivity is rated at 119 dB @ 1mw with a frequency response of 20 Hz to 16 kHz ±3 dB. Impedance is rated at 13.3 ohms. The impressive point is The UE-10 Pro provides −26 dB of isolation. This is done passively unlike the more common "noise canceling" headphones which use an external mic and phase cancellation.

The UE-10 Pros are professional in-ear monitors unlike the more common "off the shelf" products. You are required to get impressions made from a qualified audiologist and send them off to Ultimate Ears for a custom fit. The impressions are turned into molds so that the monitors can be made to fit your exact ears. This precise fit is what gives you the isolation and the suburb response. It is interesting to

note that this is not an easy process, but more on that later.

Ultimate Ears provides you with some neat aspects such as initializing each ear piece and marking them blue for left and red for right. Each set of UE-10 Pros comes with its own set of cleaning tools and custom engraved metal carrying case. The connector is a 1/8-inch (3.5mm) mini available in 46 inches or 64 inches length cables. The cables are detachable and replaceable and available in clear, beige, brown and black. You can select the color of the ear pieces themselves from a wide variety including the most standard, clear (which I got) to the most eclectic.

IN USE

Before I even get to talking about the performance of the UE-10 Pros I had to get ear impressions made. This alone was an ordeal. Unfortunately, I must have run into a slew of incompetent audiologists. It took three tries to get a proper impression made. I think that the majority of audiologists I went to only had experience with hearing aids and not in-ear impressions. What I have learned is two fold. 1) If your audiologist has never heard of in-ear monitors...run out the

door and never look back. 2) Always go to an audiologist that the manufacturer recommends. Luckily, Ultimate Ears has a large base of recommended audiologists around the country on their website so you can find one close. Once those basic rules were followed the process was easy and I received the UE-10 Pros in about 14 days.

Like most people the first thing I did when I received my set of UE-10 Pros was to plug them into my iPod. It's the quickest and easiest way to test a new set of in-ears. Besides these will definitely sound better than those crappy white ear buds that you get with an iPod. The monitors slid right into my ears with a little

---

Fast Facts

■ **Applications:**
Live sound, sound reinforcement

■ **Key Features:**
Custom fit; triple drivers; swappable cables

■ **Price:**
$900

■ **Contact:**
Ultimate Ears at 800-589-6531, www.ultimateears.com.

Product Points

Applications:
+ Great clarity
+ Custom fit
+ Cool colors and case

Key Features:
- Can be difficult to obtain correct impressions

Price:
Overall, a dream realized!

World Radio History

BY ROGER WILLIAMS III

Crate Audio CPL2 Stereo Compressor/Limiter

St. Louis Music (now owned by Loud Technologies) has been a purveyor of musical instruments for more than 80 years, and their Crate division has been cranking out electronic instrument amplification for 25 of those years. After a brief flirtation with the Audio Centron line of professional sound reinforcement products, St. Louis Music decided to fold the Audio Centron line into Crate Audio.

Decidedly aimed at the MI market, they design products for the music stores and MI catalog houses of the world. However, I have recently noticed a renewed trend toward improved design and quality disciplines involving Crate's products, and was intrigued when the Soundscape CPL2 stereo compressor was dropped at my door.

FEATURES

The CPL2 is a dual-mono compressor, which can be linked for stereo applications. A look at the front panel reveals the prominent features. From left to right, you will find expander/gate threshold control with LED indicators, -40 dB to +20 dB compressor/limiter threshold control, 1:1 through infinity:1 ratio control, 0.1ms to 200ms attack time control, manual/auto mode switch, 0.05s to 4s release time control, -20 to +20dB output control, enable/disable button, and a "breath" control. Next is the true RMS detection stereo/mono mode switch, and then the second channel of the unit, which mirrors the aforementioned functions. The manual/auto, enable/disable, and mono/stereo mode switches each have LED function indicators, and each channel has

Fast Facts

■ **Applications:**
Sound reinforcement

■ **Key Features:**
Two-channel; compressor/limiter with expander; Interactive Ratio Control; Breath processor; auto function

■ **Price:**
$169

■ **Contact:**
Crate Pro Audio at 314-727-4512, www.crateproaudio.com.

12-segment 1 through 30dB LED gain reduction meters. The master power switch resides at far right.

Around to the rear jackfield, we find from left to right an IEC AC power socket with fuse housing, channel two's balanced 1/4-inch TRS phone and XLR output jacks, +4 dBu/-10 dBV input sensitivity range switch, balanced 1/4-inch TRS phone and XLR input jacks, and duplicate jacks/switches for channel one.

Two items I noticed straight away while going through the CPL2's features were the "breath" processor and the expander/gate function. The "breath" processor is claimed to add high-end clarity to the signal post-compression that will compensate for a perceived loss in this area, which they call dynamically controlled frequency correction. The expander/gate section features IRC (Interactive Ratio Control), which is claimed to continually adjust the expansion ratio and attack time to match incoming program material characteristics. I was anxious to try these accommodations in a real-world environment.

IN USE

I decided to try the CPL2 in my project studio, utilizing signals that typically need compression: electric bass and vocals. I plugged my '67 Fender Jazz bass into a Radial JDV direct box, from there into a Mackie 1604 VLZ mixer, and inserted the CLP2 into the channel strip. Monitoring the resulting signal through AKG K240 sealed headphones, I initially noticed that even with the expander dialed off and the output set at maximum that the noise floor was very low, good considering the use of single coil pickups. Dialing up the breath processor did introduce a slight increase, but not

too invasive. The expander section took some getting used to, however - a very sensitive control. When long sustained notes would fade in volume past the threshold dB setting it would cause the expander to chatter a bit. In soft passages or song ends this would have to be reckoned with. The compressor section did a nice job of controlling quick slap bass transients on a rendition of Sly Stone's "Thank You Fa Lettin' Me Be Myself," and the breath control gave a nice, wiry presence to the strings. The auto switch enables a good, program dependent setting for attack and release parameters, but if you want to go a little over the top you'll have to go manual.

On vocals, I used a Shure Beta 58 through the Mackie. The expander worked much bet-

Product Points

Plus:
+ Quiet Operation
+ "Breath" processor restores lost high frequency content
+ Solid construction
+ Price

Minus:
- Tricky expander threshold control

The Score:
A well-built low-noise stereo compressor/limiter/expander with some added bells and whistles at an ultra low price.

ter in this setting, shutting off background noise effortlessly between passages, although mic handling noise gave it some trouble. On a stand this wouldn't be a problem. The breath control added some nice air, except on sibilants, but judicious use produced a nice effect.

RAD-360

This wireless can go the distance!

FEATURING 50 MILLIWATTS OF POWER AND OVER 12 HOURS OF BATTERY LIFE

When you first use the RAD-360 UHF Wireless, you'll be tempted to see how far you can go before it actually drops out. Go ahead. Get it out of your system. It will most likely be somewhere between 500 and 1000 feet, depending on your location and line of sight.

The RAD360 is a complete solution for all of your wireless needs, whether it be Live Music, Fixed Installations, Presentations, or Houses of Worship.Handheld systems feature the OM series dynamic mics, critically acclaimed for their clarity, off-axis rejection, and ability to achieve high levels of gain before feedback without distortion. Bodypack systems feature a variety of lavalier and headset microphones, as well as instrument systems for guitar, flute, harmonica, and brass.

You can be sure of drop-out free performance whatever the venue. And best of all, the RAD-360 is simple to set up and use. You'll be up and running in minutes!

Main Features
- Frequency Agile – 193 channels available per system
- Interchangeable capsule assemblies for handheld transmitters
- Metal housing for receiver, bodypack, and handheld transmitters
- Battery power indicators
- RF level metering
- Noise squelch circuitry

Handheld and body pack use standard AA batteries

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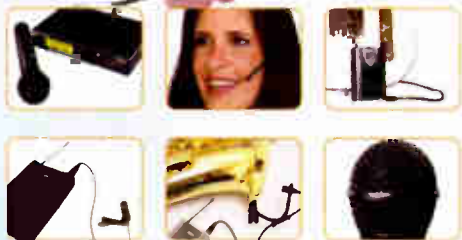
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TEL: 503-682-6933 FAX: 503-682-7114 www.audixusa.com

Audix Corporation. PO Box 4010. Wilsonville, OR 97070.

NEW PRODUCTS

Producer/engineer Ted Perlman uses **Pauly Superscreen** pop filters at his Buffalo Studios in Los Angeles.

Producer Anthony Resta used **Crowley & Tripp** ribbon mics on his latest project, Perry Farrell's new album at Resta's Boston-based Bopnique Studio.

Producer/programmer Andrew Philpott has added an **Eventide** Anthology II plug-in bundle to his collection of plug-ins.

Engineer Mike Roskelley chose **ATC** SCM 110A monitors and Sub/115 subwoofers for his surround monitoring setup at his new Salt Lake City studios. See picture of Mike and one of the SCM 110As.

Roving engineer John Agnello used his collection of mobile **API** gear (512B preamps, 560B ten-band EQs and 550B four-band EQs) as a partial front-end to a **Neve** 8088 console to work on the latest Andrew W.K. project at Water Music Studios in Jersey City, New Jersey. Down in Charleston composer/engineer Chris McLernon took his **API** Lunchbox rig (3124 four-channel mic preamp, 512C mic preamp) and visited singer London Weidberg to record a tune that was later used on the TV show, Vegas. McLernon recorded to a **Digidesign** Digi 002 DAW. See picture of McLernon and the Lunchbox.

Fostex PM-2 MKII Powered Monitor

The newest and largest member of Fostex's PM powered monitor family is the PM-2 MKII. The PM-2 is a two-way monitor designed for closefield and desktop uses. Its 8.75-inch woofer is made of olefin-coated Kevlar while the HF driver is a 1-inch soft dome UFLC (polyUrethane Film Laminated Cloth) tweeter. The monitor is biamped with dual 120W amplifiers. A "high-gloss" piano black finish rounds out the package. Price: $599 per pair.
Contact: **Fostex at 310-329-2960, www.fostex.com.**

TL Audio M4/40 and M4/48 Consoles

Definitely standing out these days in a world of digital and solid state consoles and mixers is the TL Audio M line of tube-based console – the newest being the M4. The M4 now comes in 40-channel and 48-channel versions. Besides tube preamp input sections, the M4 also has a four-band EQ and VU meters. It also offers an optional digital ADAT interface and a 24-bit/96 kHz digital output option. Prices: M4/40 - $22,450; M4/48 - $24,410.
Contact: **TL Audio/Independent Audio at 207-773-2424, www.independentaudio.com.**
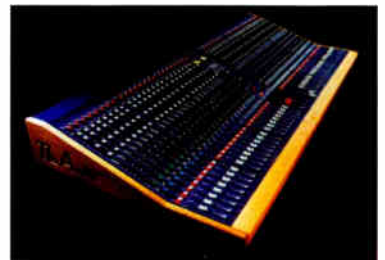
Universal Audio Neve 1073 and 1073SE Plug-ins

Designed for Universal Audio UAD-1 DSP card and Powered Plug-ins systems the Neve 1073 and 1073SE "channel module" are the result of co-operation between Universal Audio and AMS Neve. Not surprisingly, the 1073 is looks identical to its 40+ year-old analog namesake including three-band EQ (parametric mid), phase reverse and high-pass filter. Price: $249
Contact: **Universal Audio at 831-454-0630, www.uaudio.com.**

M-Audio Sputnik Microphone

Da! Hopefully it goes more than "Beep, beep!" The Sputnik is a lollipop-style mic with a gold-evaporated Mylar large diaphragm with a dual-metal backplate design. Patterns include omni, cardioid and figure 8. The heart is a mil-spec 6205M tube. The solid brass body has a nickel finish. The Sputnik ships (launches!) with power supply, mic cable, shockmount, bag and flight case (borscht-flavored Tang extra). Price: $699.
Contact: **M-Audio at 626-633-9050, www.m-audio.com.**

PRO AUDIO REVIEW

Gear & Software Reviews For The End-User

THE BUYERS SERIES:
STUDIO MONITORS
and HEADPHONES

Would you rather listen to your monitors...

or hear your music?

Advanced Dynamic Audio Monitors are state-of-the-art, no compromise speakers designed and built in Germany with one goal in mind: to accurately reproduce natural acoustic events. For audio engineers, this means the truest possible reproduction of their recordings that translate exceptionally well to the entire range of music playback systems on the market.

Based on a new approach to Dr. Oskar Heil's Air-Motion Transformer, ADAM monitors feature painstaking design, superior materials, and A.R.T. (Accelerated Ribbon Technology) for exceptional clarity and transient response in the high and midrange frequencies. ADAM monitors have been used in the most critical listening environments in audio recording, mixing, mastering, film, and broadcast. From Abbey Road to Jazz at Lincoln Center, Danny Elfman to David L. Newman, and Butch Vig to Eminem, ADAM monitors are trusted to reveal every detail.

New A7 closefield monitors shatter price barrier

ADAM Audio is now shipping the new A7 two-way closefield studio monitor, bringing the acclaimed ADAM sound to a lower price point.

The A7 combines ADAM's renowned A.R.T (Accelerated Ribbon Technology) folded ribbon tweeter with a state of the art 6.5" carbon fiber woofer, resulting in an extremely accurate monitor with all of the clarity, detail and spectacular imaging traditionally associated with the ADAM name.

Powered by two 50W amplifiers, the A7's front panel features a power switch and detented volume knob. The rear panel houses both balanced (XLR) and unbalanced (RCA) connectors, controls for tweeter level as well as two shelving filters for high and low frequencies.

At $999 per pair, ADAM innovation and quality is now within the reach of most anyone.

New Sub 8 compact subwoofer reaches new lows—in frequency and price

ADAM Audio is now shipping their new Sub 8, a brand new subwoofer aimed at the project studio market.

The Sub 8 is a small yet powerful subwoofer designed to extend the low end of any near field monitoring system. It houses an excellent 8" woofer with a large 50mm voice coil and is driven by a 160W ICE power amp.

The front baffle features two motorized knobs that allow the end user to tailor input level and crossover frequency settings. These knobs can be controlled with an included wireless remote control, enabling the user to optimize its settings without leaving the ideal listening position. The unit also provides an onboard 2.1 bass management system.

At $699, The Sub8 is an ideal match for ADAM monitors such as the A7, P11A, S1A and S2A. In addition to a traditional black model, ADAM will also offer a silver version that will be a perfect complement to the Artist desktop monitor. ∎

ADAM Audio produces a complete range of both passive and active monitoring systems to suit any given studio environment, from the A and P Series for home and project studios to the S Series for the most demanding professional environments and applications.
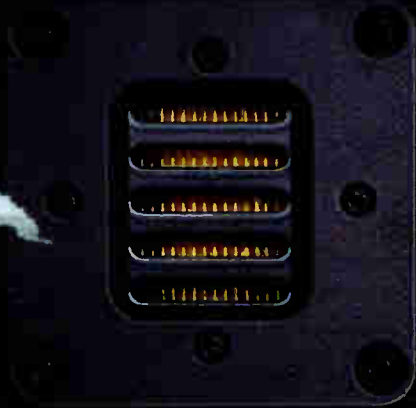
ATC Is Different

"All progress occurs because people dare to be different." – Harry Millner

It's been more than 30 years since Billy Woodman began pursuit of his singular vision of designing better loudspeakers for the most critical studio applications. Under his stewardship, ATC was formed and quickly made its mark with custom driver units capable of handling more power, producing less distortion and going louder than any other driver in the world.

Now, anyone with even a passing interest in professional audio knows that the sheer number of available loudspeakers is daunting, to say the least. Pick one, pick another – after all, what's the difference?

Woodman provides the clarity: "The aims of the forefathers of the industry seem to have been completely forgotten. Many loudspeakers of today are described as being 'musically involving' and 'having pace, rhythm and slam' or, as being 'a musical experience.' These words might have a subjective meaning to the originator, but they cause confusion and suspicion in the minds of the public, while providing the less scrupulous with a cynical cover for rather poorly engineered products."

Design and Engineering

Yes, it's in the design. It's in the engineering. The performance of a loudspeaker can be defined by its linear and non-linear behavior. That's the difference, and it's been the hallmark of ATC every step of the way.

Begin with Super Linear Magnet Material (SLMM) Technology at the heart of ATC SL woofers. The product of serious research and development, it addresses the inherently non-linear magnetic performance of soft steel and hysteresis distortion in driver magnet assemblies. Developing and implementing SLMM to replace soft steel around the driver completely eliminates the problem, resulting in new levels of revealed detail and previously unheard effects. It set a new standard in bass performance.

ATC long ago set the pace with Soft Dome Technology for mid-range drivers that have revolutionized studio monitoring. Again the result of intensive R&D, it achieves exceptional broad and even dispersion to produce a flat response every-

where in a room.

Also vital to constant dispersion, ATC understands that the relationship between direct and reverberant sound is very important in high-performance loudspeakers. Not only must the on-axis magnitude response be accurate and linear, but also that the behavior off-axis must be both broad and even, with frequency exhibiting no abrupt dips in amplitude.

Active Performance

Designed optimally, active loudspeaker systems can provide superior performance. ATC active systems offer three Class A/B monoblock power amplifiers, each isolated with its own power supply and optimized for its respective transducer. Six MOSFET amp blocks are also precisely matched to the needs of the drivers, dramatically increasing transient headroom and maximizing sound pressure levels.

These unique MOSFET blocks also incorporate a proprietary design that momentarily rounds off waveforms caused by clipping, instantaneously lowering the gain before the amplifier clips. The unbelievable speed of the process renders it inaudible while protecting the drive units and amplifier.

While ATC loudspeakers come in packages large and small, the consistency of each remains exactly the same. Put on a blindfold and listen to a smaller SCM100ASL and a larger SCM300ASL at the same level and try to distinguish the two. We're quite confident that it's a near-impossible task. The only difference will be that as size increases, so too does bass extension and dynamic range.

The same holds true whether the volume is loud, soft or anywhere in between. This is a true hallmark in proving the validity of adherence to the most advanced linear design and engineering principles.

As Billy Woodman plainly states: "There is no excuse for bad engineering." Yet this very thing proliferates in the loudspeaker market, seriously compromising your sound and your work. ATC chooses to take its own path, one of discernable difference when it comes to loudspeakers created to exceed the most essential professional needs. ∎

ACOUSTIC ENGINEERS

ATC Studio Monitors/TransAudio Group

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702-365-5155
www.transaudiogroup.com

Ultrasone Headphones — an absolute necessity
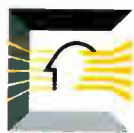
For the discerning professional or audiophile, Ultrasone headphones are an absolute necessity in the studio, during a live mix, for broadcast or when just enjoying music. Ultrasone has been winning awards and ears since their introduction in the states in 2004. Ultrasone has S-Logic™ technology. So only Ultrasone headphones offer such amazing clarity and detail of sound. And S-Logic™ technology delivers one other advantage – it helps protect your ears from any long-term damage. Combined with our superior customer service and industry-leading two-year warranty, it is the best **and safest** headphone on the market.

S-Logic™ Natural Surround Sound:

Invented by Florien Koenig of Germany, a musician with a doctorate in audio acoustical engineering, Ultrasone's S-Logic™ technology was a result of his dissatisfaction with the inaccurate, artificial sound of conventional headphones. His invention, S-logic™, replicates the ways we hear in everyday life. Other conventional headphones place the speaker in the center of the ear cup, driving an audio signal directly down the ear canal, slamming into the eardrum, totally bypassing the outer ear. S-Logic™ uses decentralized driver positions to direct the audio signal to the outer portion of your ear, the pinna, which is responsible for our perception of distance and direction. The audio signal is then reflected naturally down the ear canal, resulting in an incredible natural surround sound, with three dimensional qualities, as if the music is being played live, or from speakers, not headphones. This creates separate, distinct instrument placement in a much wider sound field. S-Logic™ is the only headphone system to utilize your entire sense of hearing. Mixing, tracking, even finding your pitch with absolute accuracy… it is all easier with an Ultrasone.

Safer Listening

Because the S-Logic™ technology does not aim the auditory signal directly down the ear canal, but at the pinna, pressure to the eardrum is decreased by up to 40% (3 dB - 4 dB) for the same perceived volume. This significantly reduces the risk of hearing damage – an absolute must if you spend a lot of time in headphones. DJ's and drummers have found that they can listen with less distortion and greater clarity at extremely high volumes while reducing the ringing and damage to their ears.

Reduced Fatigue

Engineers and broa[...] confirm that ear fatigue is greatly reduced w[...]rasones, due to the decreased pressure on the [...] Combine that with the superb isolation[...] Ultrasones are the most comfor[...] to wear for those long sessions.
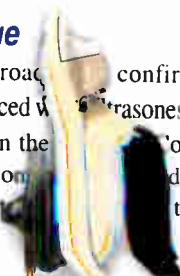
ULE technology

The ULE technology was developed by Ultrasone for users who spend many hours under headphones. Headphone drivers convert an electric signal into an acoustical signal, producing low-frequency magnetic fields. In response, Ultrasone developed a special shield inside the ear cup. Made of MU Metal, (ULE technology) this shielding is designed to reduce the radiation by up to 98% compared to ordinary headphones. The ULE technology has stood up to international review and is recommended by technical surveillance organizations.

Models

Ultrasone only makes headphones and the models encompass everything from the headphone with all the accessories or luxury you might desire to the durable model for heavy-duty studio use.

Ultrasone headphones are available in several models, open or closed, with heavy duty Mylar, sensitive gold, or extremely accurate titanium drivers. Frequency extension is as high as 8 Hz – 35 kHz. Our PROline series features added accessories, such as detachable cables, one straight and one coiled cable in every box, and extra ear pads, velour or leather depending on the type of headphones. Shorter cables for field or MP3 use are available. Our DJ series features a detachable cable with independent L/R volume controls and a stereo mono switch.

The iCans, released this year, are our portable headphone for the person who loves music, and wants to protect their hearing so they can keep on listening to music. No longer does the consumer have to compromise. They can have the same incredible audio experience and safe listening experience the professional had when creating the music

And we have the Edition series… Limited edition, numbered headphones for the audiophile who wants luxury with his listening. See our website www.ultrasoneusa.com for information on all of the models, awards we have won, and testimonials from professionals who love them. ■

ULTRASONE

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VL-X5

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VL-X5: TASCAM's new powered studio monitor.

- 90 watts total power: 60 watts LF (bass) + 30 watts HF (treble)
- 5" LF driver, 1" silk dome HF driver
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- Front-firing ports allow you to place them near a wall
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"...An incredible value. The low end is amazing..."
-EQ, May 2006

TASCAM

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attack response, and the bass keyboard (remember, it's The Doors!) had very rich extended synthesizer tones that purred at 40 Hz and 50 Hz.

SUMMARY

The Crest CC4000 rocks with some of the best power amps in its class. It reminded me how Crest was once king of the touring sound hill back in the day. They keep doing this kind of stuff, they just may regain that title. Power amps come and go, but this one is a keeper.

Will James, owner and chief engineer of Atlantis Audio and Lighting, is a contributor to **Pro Audio Review**.

Review Setup

Midas Siena/48-channel console; Yorkville TX2M monitors: Yorkville TX processor; A-Line Acoustics LS218 subwoofers; Audix D6 kick drum mic, Audix Micro-D snare mic, Audix OM7 vocal mic, Shure SM58 Beta vocal mic

Crest Audio CC4000 Power Amplifier Bench Test

(Unless otherwise specified, all tests reflect signals applied to the balanced inputs, both channels driving 8? loads, a measurement bandwidth of <10 Hz to 30 kHz, and the worst-case result.)

Rated Power
2 x 800 watts RMS into 8 ohms 1 kHz/0.05% THD; 2 x 1,350 @ 4 ohms, 2x 2,000 @ 2 ohms (0.1% THD), 1 x 4,000 @ 8 ohms (bridged, 0.1% THD)

Measured Performance
Power at clipping (1% THD+N); watts, 2 channels driven at 1 kHz

8 ohms	4 ohms	2 ohms	4 ohms, bridged
840*	1226*	>1650**	>2500**

*See text
** See text; limited by AC available powerline voltage, and/or safe available dummy-load resistance

THD+Noise at rated power at 1 kHz:

8 ohms	4 ohms
0.006%	0.008%

Freq. Response
(1w, 8 ohms) +0, -0.6 dB <10 Hz-20 kHz

Input sensitivity:
selectable: -25.3/-31.3 dBu for 1 watt into 8 ohms

Input impedance:
17k kohms (both legs)

S/N (A-wtd./8?, 10 Hz-22kHz msrmt. ban-width, Gain switch set to "20")
84 dB re: 1w; 112.6 dB re: rated power (800 watts)

Damping Factor (re: 8 ohms):

30 Hz	1 kHz	20 kHz
580	608	>650

This is an amplifier that make a lot of power; my bench tests were limited by sag in available powerline voltage — despite a 20 amp circuit, the heavy, supplied line-cord, and a 135-volt Variac — to the extent that 2 ohm and bridged-mode testing both reflect substantial reductions. Another handicap was my inability to jumper up more than 2 kilowatts worth of load resistance in 8 ohm or 4 ohm configurations, (and my reluctance to risk melting, at more than $50 a pop, any of the ten 250w 4 ohm and 8 ohm resistors I have…). Suffice to say that Crest's CC4000 met its specs, and then some, in every test I ran, falling short only where the above-named inadequacies limited performance. I did note that at low power levels the amp produces substantial mechanical buzzing (presumably from magnetostriction in its power transformer), and that at high powers on pure tones its switching circuitry (again, presumably), sings quite audibly. But where loudspeakers instead of dummy resistors are the loads, at these levels you are very unlikely to hear this, or much of anything else, other than the program! The amplifier got comfortably warm but never alarmingly hot, even on extended high-power runs; its dual fans seemed very effective, if hardly silent.

For the sake of clarity, all plots show one channel only; the two channels were effectively identical throughout.

Figure 1 shows frequency response into 8 ohms loads at, from top to bottom and offset to -1 and -3 dB for clarity, 1 watt, 100 watts, and 250 watts. The slight dip at 1 watt is well within Crest's specifications, and the increasing high-frequency rolloffs at higher power would seem exceedingly unlikely to have any audible impact.
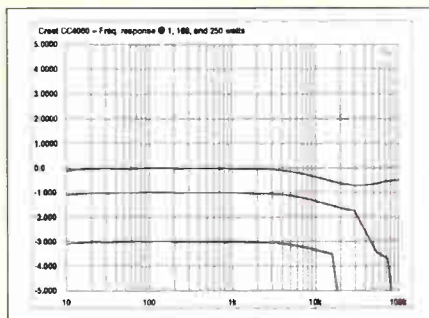
Figure 1: Frequency response into 8 ohms at 1 watt, 100 watts and 250 watts.

Figure 2 graphs THD+noise versus frequency into 8 ohms at three power levels: 1 watt, and 250 and 600 watts, ranging from bottom to top along the 7 kHz log line. (Both channels were driven in all cases, but single channels shown for clarity.) Distortion at other power levels and with lower-impedance loads was along the same lines, never exceeding 0.5% at high frequencies with 2-kilowatt, 2 ohm r
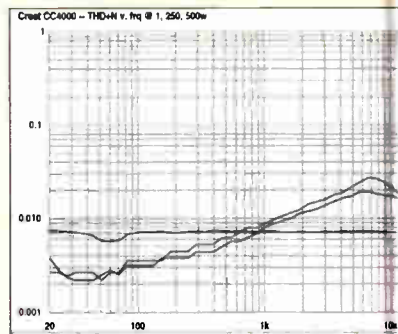
Figure 2: THD+noise v. frequency into 8 (at 1 watt, 250 watts and 600 watts.

Figure 3 plots power against distortior 8 ohm and 4 ohm loads (left and right, alo. the 1-watt line), in watts (horizontal) vs.
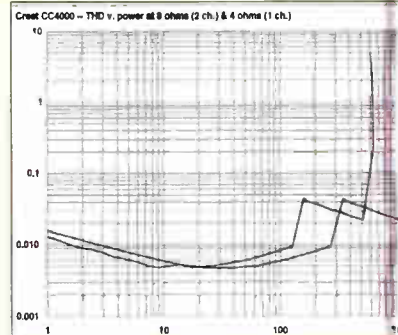
Figure 3: Watts (hor.) v. distortion (vert.) kHz for 8 ohm and 4 ohm loads.

THD+noise in percent (vertical). The 4 ohm represents only 1 channel driven due to a s age of suitable dummy-load resistors; the ping point with both channels driving 8 ohr short of the spec, likely not thru any fault the amplifier, but due to inadequacies of p line voltage, which sagged to about 110 volt this point, despite a maxed-out 135-volt V in the circuit. A 40-amp 120Vac circuit, st wired, would be required to really get the fu potential from this impressive power amp.

-D.

ence listening system, which includes, a Legacy/Coda High Current preamp, Esoteric DV-50 Universal DVD-A/SACD player with upsampling PCM converter, modified Vacuum State Electronics Sony SCD-777ES SACD player, Benchmark ADC-1 A/D converter and TASCAM DV-RA1000 high resolution recorder.

Primary speaker cables included Alpha Core solid silver, spade-terminated interconnects and Westlake Low PE Distortion Balanced cables. Amps on hand for comparison included the first generation Pass X250 and the Bryston 14BSST — one of my favorite studio amplifiers.

I played a number of my own high resolution 24-bit/192 kHz guitar recordings and auditioned a number of commercial and DMP label SACDs to get a feel for the X350.5's sonic signature.

Product Points

Plus:
+ Exquisite, accurate reproduction
+ Improved bass and transient response
+ Perfect for high resolution monitoring

Minus:
- Ouch, my hernia!
- Maxes out most credit card limits

The Score:
It's heavy, expensive and runs very warm, but can an amp sound any better?

On the DMP SACD release, Steve Davis *The Quality of Silence*, I immediately noticed the difference comparing the original X250. The transient response with recorded drum cymbals, and the upper register piano was more clearly delineated than the older version. You did not have to A/B the amps to hear the difference in presentation, and the more realistic reproduction of those instruments via the .5 version.

On my Martin D-35 acoustic recordings, the strums and delicate pick attack was much more out front in the stereo image with the X350.5. The details were much easier to pick out in the mix. I always thought that X250 always had a good width and depth to the image, but the extra detail on the X350.5 makes it starkly more present. The extra information con-

tained in high resolution recordings made the amp's improvements much more audible. The 24/192 kHz playback's improvement over 44.1 kHz was obvious using the old Pass amp, but the new amp brought it to an even higher degree of realism.

The other big difference was the tightness of the bass. With the big improvement in dampening factor, the X.5 series has lost almost all the warm bloom of the older version, making it just about as tight and fast as the bipolar-output Bryston 14BSST.

Speaking of the Bryston, the Pass now also nearly equals or perhaps surpasses it in transient response. I always thought that bipolar amps would always be better at the upper-frequency reproduction with more clarity and speed (and sometimes a slight harshness), but the X350.5 shows that the JFET design is up to the competition.

I also auditioned the Lipinski L-505 closefield monitors with the Pass X350.5. Lipinski recommends Pass Labs amps to

Studio

power its speakers. Those same qualities I heard through the Legacys were also delivered in closefield range with the Lipinskis. The L-505s do not deliver the bottom end of the Legacys, which have three woofers to extend bass to under 20 Hz, but the transient response and the mid bass were quite impressive up close.

My nitpicks with the X350.5 are few. It weighs twice as much as the Bryston, even though the Bryston produce about 600 wpc (8 ohms), and it does not accept banana plugs. I have several sets of high-quality cables that terminate with banana plugs, but I could not use them with the Pass.

I know there will be complaints about a $9,500 price tag, but a Dodge Viper does not cost $20,000. Low-volume, made-in USA high-end components are not cheap. Workers who assemble Pass amps don't make 50 cents an hour, and you don't see a power supply like the 350.5s in a $400 amp.

SUMMARY

If your speakers are passive, and you want to hear the most accurate two-channel playback possible, I strongly recommend the Pass X350.5. Its new design has resulted in a clearly audible improvement in sound over the previous generation of X Series. The sonic enhancements are more clearly magnified when listening to high resolution sources.

*John Gatski is publisher and executive editor of **Pro Audio Review**.*

TECHNOLOGY HIGHLIGHT

The X350.5 is the accumulation of the improvements made during the product life of the model X350. This is true of the ".5" series, in general, and at some point the products evolved enough to merit an upgraded designation and a new faceplate.

The original X amplifiers used MOSFETs as the input devices in single differential pairs. The .5 series uses cascoded, complementary JFETs that have lower noise and greater linearity, delivering better performance while allowing lower feedback and greater stability. As before, the circuit still consists of the Pass Super Symmetry design — a simple voltage-gain stage driving a simple current-gain output stage, with carefully matched parts balanced in a patented cross-coupled topology.

The output stage has been increased from 40 to 48 power MOSFETs, used as complementary followers in the symmetrical balanced circuit. The frequency compensation of these devices has been improved so that there is much less rise in distortion at high frequencies, and by enclosing the output stage in the feedback loop of the amplifier we have increased the damping factor from about 30 to about 250.

According to Pass, the decision with each product to use feedback on the output stage or not has always depended on the results of listening tests, and in the earlier designs it could went either way.

In the present product .5 the preference was the output stage in the loop.

The other major improvement in the .5 series is the application of a small amount of single-ended, Class A bias to what would otherwise be a highly-biased Class AB design. Current sources draw about one-half amp from the output to the negative supply rail, so that for the first watt or so the amp operates single-ended. Above that the amplifier operates push-pull Class A to about 30 watts. The idle dissipation of both output stages is about 400 watts.

Improvements were made to the power supply by working with manufacturer Plitron to produce a power transformer that was electrically and mechanically quieter, particularly under noisy AC conditions. This also improved the power factor (the spread of energy draw from the wall) to about 80%. The AC primary circuits also have improved filtering to reduce high frequency noise, both coming into and emitted by the power supply, and new massive high-speed/soft-recovery rectifiers have been added on the secondary side of the transformer for the same purpose. The bulk capacitance of the supply has remained the same, but the first of the two stages has improved filtering and decoupling.

Pass says that between these changes and improved circuit layout, all forms of noise have been improved by a factor of about two.

BY RUSS LONG

Digidesign Pro Tools HD 7

Digidesign, creators of the immensely popular Pro Tools DAW system, released Pro Tools HD 7 software, their latest and greatest version at last year's AES show in New York. Pro Tools is now up to version 7.1 and, as the user base continues to grow, Digi shows no sign of stopping until DAW world domination is complete. That's perhaps a bit of an exaggeration but not far from the truth.

FEATURES

The release of Digidesign Pro Tools HD 7 (Mac OS X and Windows XP) adds a multitude of new features and improvements that deliver considerably expanded audio and MIDI recording and editing capabilities, improved efficiency, increased mixing flexibility and power, and superior ease of use. Pro Tools HD 7 software is a paid upgrade ($245/CD or $175/software download). With regard to space (entire books have been written on Pro Tools and its feature set), this review is going to focus on the features that are new to Pro Tools HD 7 software. To check Pro Tools HD 7 system requirements please visit: www.digidesign.com/compato.

COMPOSITIONAL TOOLS

Pro Tools now supports REX and ACID files which are two of the most popular formats for loops and samples. Digi has finally added Instrument Tracks (something we've all been requesting for years), which combine MIDI and audio capabilities in a single channel strip. This simplifies routing for virtual instruments and MIDI sound modules. With Region Groups, any combination of MIDI and audio regions can be grouped to quickly and easily build arrangements. The new region looping feature allows you to assemble grooves or fill backgrounds behind video cues. Applying the groove input quantization to MIDI tracks provides the ability to adjust the feel of the track. It is also now possible to drag and drop audio, MIDI, REX, and ACID files directly from the desktop to the timeline.

Pro Tools 7 also adds real time MIDI processing capabilities, which enable non-destructive control of MIDI quantization, note duration and velocity, transposition, and timing. The Mirrored MIDI editing mode allows adjustments made to one MIDI file to affect all existing copies of that region. The Zoom Toggle allows users to quickly switch between two different user-defined track view settings in the Edit window, making it easier to work with both MIDI and audio. In addition to these features, nearly every MIDI operation window includes enhancements that make work with MIDI faster than before. Enhanced groove quantization features include the ability to apply groove template quantization across multiple tracks, add randomization to further humanize the feel of a groove, and apply groove template quantization to incoming MIDI signals. Sample-based MIDI tracks allow MIDI events to stay locked to time code regardless of session tempo changes.

More advanced MIDI users will enjoy the fact that an enhanced Select/Split Notes window enables users to split a MIDI performance into multiple tracks by MIDI note, velocity, duration, or position. The Remove Duplicate Notes command allows users to quickly clean up recorded or merged MIDI tracks. The Change Duration window has been revamped and now offers legato, overlap correction, and transform sustain pedal features and the Transpose window now allows users to transpose all notes in octaves and semitones.

EDITING AND ARRANGING

The Pro Tools menu structure has been reorganized to provide more logical and streamlined menu navigation. This took a little bit of getting used to at first but now it is the biggest reason I hate to work on a pre-7.0 rig. To make the adjustment easier, the new Tool Tips feature adds "rollover" descriptions of objects in the Pro Tools software interface.

Pro Tools 7 has added Separate on Grid and Separate at Transient functions that allow the editing of multitrack audio regions simultaneously based on a grid value or the transients in an audio file. Audio regions can now be quantized according to a grid or groove template. The Reverse Strip Silence functionality allows for extracting louder portions of audio tracks. The Link Track and Edit Selection feature enables the simultaneous quick application of track-level com-

mands globally across multiple tracks. Dragging multiple regions onto a single track is also possible via a new Region List drop order pref.

The Duplicate Track command now allows the specification of the number of tracks and which track parameters are to be duplicated. The Import Session Data feature now supports importing Markers and Mix/Edit Groups. The new DigiBase improvements include support for working with MIDI files and an added Tempo column to organize and view audio files by tempo. Red Book audio CDs can now be imported from the DigiBase Workspace on both Macintosh and Windows XP machines. The environment now supports up to 999 Marker/Memory Locations and the resizable I/O Setup and Disk Allocation dialogs allows simultaneous access to more parameters.

Fast Facts

■ **Applications:**
Studio, broadcast, post production

■ **Key Features:**
Mac, Windows, up to 192 tracks; up to 192 kHz; Instrument Tracks; Real-time MIDI processing; Region Groups and Region Looping; Supports REX and ACID files; increased RTAS performance; 10 sends per track; RTAS plug-ins supported on aux and master tracks

■ **Price:**
Upgrade: CD: $245, Download: $175

■ **Contact:**
Digidesign at 650-731-6300, www.digidesign.com.

MIXING

Digi has improved the support for multi-processor computers and the RTAS environment, resulting in a plug-in increase up to 150 per cent on dual-processor computers. A computer running Pro Tools 7 instead of an older version can typically run more than twice as many RTAS plug-ins and virtual instruments. Pro Tools|HD now supports RTAS plug-ins on aux inputs and master faders, which improves the compatibility between Pro Tools|HD, Pro Tools LE, and Pro Tools M-Powered systems. More

RTAS effects and instrument plug-ins are able to run simultaneously due to improved host processing efficiency.

Pro Tools 7 now supports up to 10 aux sends per track and up to 160 simultaneous channels of I/O. Pro Tools|HD 1 provides up to 96 tracks at 44.1 kHz or 48 kHz, up to 48 tracks at 88.2 kHz or 96 kHz or up to 12 tracks at 176.4 kHz or 192 kHz. Pro Tools|HD 2 Accel and HD 3 Accel provides up to 192 tracks at 44.1 kHz or 48 kHz, up to 96 tracks at 88.2 kHz or 96 kHz

or up to 36 tracks at 176.4 kHz or 192 kHz. The ability to use RTAS plug-ins on Aux Inputs and Master Faders is another nice feature included with Pro Tools 7. Not being able to use an RTAS reverb on an Aux channel always drove me crazy with old Pro Tools.

Send assignments can now be copied or dragged and dropped across tracks. The New Automate All command allows the easy arming automation of all plug-in

parameters at once. An all-new, free Dynamics III DigiRack plug-in (released concurrently with Pro Tools 7 software) allows easy and powerful control of mix dynamics. EQ III now includes a new filter band pass solo mode, making it easy to isolate and remove unwanted signals or noise in an audio track. The new Cut, Copy, Clear Special commands allow more refined editing of automation data.

In Use

For this review I installed Pro Tools HD 7.1 into a Mac G5 dual 2GHz processor with 2 GB of RAM running OS 10.4 The initial program launch opens a reorganized set of drop-down menus immediately signifying an improved user interface. I've been using the software for several months now and the more I use it, the more I love it. It is easy to take for granted from time to time but every time I work at a studio that hasn't upgraded I'm reminded over and over the value of this update.

My only big complaint with Pro Tools is with the edit window. I envy Nuendo users for being able to simultaneously view different kinds of automation data for a single track. I would love to be able to look at my vocal track and simultaneously see the volume automation, aux send level automation and panning automation but it's not possible with Pro Tools. Another nice feature would be the ability to "load user preferences." I work at different studios all of the time and only bring my own Pro Tools rig with me about half of the time. Instead of spending 5 - 10 minutes of every session trying to get all of the preferences (e.g. auto scroll, color coding, fade-in, out and crossfade type, dither, etc.) set to my liking,

I would love it if I could just plug in my thumb drive and have the rig immediately configured to my way of working. It also still seems awkward to me that the master fader inserts are post-fader. This means you never want to insert a compressor/limiter on your master fader because if you have a fade on your mix, the compression will gradually go away as the mix fades and the fade will sound like dung. When I mix in the box I run all of the channels to a stereo aux and insert by stereo buss compressing there. Then I

assign the aux to Master Fader where I write my fade.

All of this aside, I have never seen another DAW able to compete with the speed or flexibility of Pro Tools. The multitool makes the interface lightning fast and the added features make the environment more powerful then ever.

> **I have never seen another DAW able to compete with the speed and flexibility of Pro Tools. The multitool makes the interface lightning fast...**

Even with all of the additions and improvements included with Pro Tools HD 7 software, I have surprisingly found the software to be amazingly easy and intuitive to use. Menus have been stream-lined and organized more logically, though key commands remain the same. The Tool Tips feature provides descriptions of objects within the interface when the cursor is placed over them so you can get better acquainted with the interface.

SUMMARY

The new changes and updates to Pro Tools once again prove that it is a leader, if not the leader, in the DAW marketplace.

*Russ Long, a Nashville-based producer/engineer, owns The Carport recording studio. He is a regular contributor to **Pro Audio Review**.*

Review Setup:

Apple 2 GHz Dual Processor G5 w/2 GB RAM; Lucid Gen-X-96 clock.

World Radio History

BY JOHN GATSKI

Pass Labs X350.5 Stereo Amplifier

I have been listening to Nelson Pass-built amplifiers since 1994. From the Class A simplicity of the Aleph line to today's X and XA series, I have always found Pass' dedication to improving an amplifier's audio reproduction admirable.

Steps up in amplifier design usually mean subtle audible improvements. However, Pass' latest upgrade to his successful X series has resulted in a not-so-subtle improvement to the sound — especially with high resolution playback. In fact the Pass X350.5 auditioned here, just may have the most inner midrange and treble detail of any amp I have ever heard. High-end mastering and engineers and those studios with high-end reference listening rooms are the intended ears for this class of amp.

FEATURES

Price at a whopping $9,500 (Yeah, that is more than your Pro Tools rig, including the G5), the Pass X350.5 is an example of premium, made-in-USA build and parts quality. This heavy brute, weighs in at more than 150 pounds, but all that pain of moving disappears when you hear it.

Amplifiers are always minimalist electronics components, and the Pass is no different. On the front panel is the power button, a really cool looking backlit current meter and separate front panel handles. The back panel sports balanced XLR and unbalanced RCA inputs, a set of 12V triggers for remote off/on switching and a large wing nut-based binding post that works with spade or bare wire termination (To make a product line that meets

European and U.S. safety standards, Pass does not offer banana plug-capable posts).

The rear also contains the main power supply on/off switch, and a massive set of carrying handles, but it takes a stout man to lift 150 pounds contained in such a compact chassis. It took two to transport and place this bad boy in my studio. The sides of the amps are occupied by large heat sinks that would chop off a foot if you dropped the amp. The amp runs very warm at low volume and hotter at louder levels. Make sure there is plenty of ventilation.

Design-wise, the X350.5 has the same basic SuperSymmetry design that Nelson Pass and crew developed in the late 1990s, which utilized carefully matched output devices in a low feedback, fewer-stage configuration. The new tweaks are employed in the .5 series, which includes the X150.5, the X250.5, 350.5 and 600.5 monobloc. Among the changes: JFET input devices have replaced MOSFETS (more linearity, Pass says), high-bias Class A on the input stage with limited feedback loop, (a feature seen previously on the Pass' flagship XA Series), better filtering on the power supply and more power supply capacitance. The refinements are claimed to net better bass performance and transient response at lower distortion than the last version.

Relevant specs for the Pass include 350 watts- per-channel at 1 percent distortion or less; dampening factor is listed

> **The transient response with recorded drum cymbals, and the upper register piano was more clearly delineated than the older version.**

at 250 ref 8 ohm nominal load (hence, the audibly tighter, faster bass presentation). Frequency response is listed from DC to 100 kHz at minus 3 dB. Gain is 30 dB. The X350.5 idles at 600 watts and consumes 1,800 watts at maximum output. All this power is still possible through the 15 amp 120V power supply.

IN USE

I connected the Pass X350.5 to my refer-

BY RICHARD ALAN SALZ

Yamaha HS50M Studio Monitor and HS10W Subwoofer

Yamaha NS10 aficionados have been searching for a replacement for the out of production studio monitor for some years now. Judging by the mildly insane prices that some people are willing to pay on eBay (a recent pair in nice shape went for $599) there still exists healthy demand for NS10s.

On the other hand, is the venerable NS10 representative of today's consumer-level speakers in the way it was of the speakers of 20 years ago? Not in my opinion! End users' speakers have improved enormously in the intervening years, today's studio monitors need to be much more linear in response than the NS10s could ever hope to be. With that in mind, let's take a look at a 21st century Yamaha studio monitoring system.

FEATURES

The Yamaha HS50M is the smaller monitor in a series (with a larger model, the HS80M also available) that is designed for utilization with the HS10W subwoofer. When it comes to build quality Yamaha does not disappoint, fit and finish is excellent. Both the HS50M monitor and the HS10W subwoofer feel very solid for their compact sizes, which isn't surprising given their MDF construction and onboard amplification. Speaking of amplification, the HS50M supplies 45 watts of power to the 5-inch woofers and 25 watts of power to the 3/4-inch dome tweeters. The HS10W subwoofer's 8-inch driver is supplied with 150 watts.

Surprisingly, no distortion figures are furnished for the amplifiers, the power output is specified as "dynamic output," and the 55

Hz to 20 kHz frequency response of the HS50M monitor is only specified to within −10 dB! Inputs to the HS50M monitor are on separate balanced XLR and 1/4-inch balanced/unbalanced TRS/TS jacks. The subwoofer offers balanced XLR and balanced/unbalanced TRS/TS inputs, but provides only XLR outputs for use in passing through audio to the satellites. Both the monitors and the sub have a snazzy white LED lit Yamaha tuning fork log that lights up when they are powered up.

A continuously variable input potentiometer (with a center detent at +4 dB gain) allows for matching to a wide variety of systems, but I really wish both the speakers and the sub had a stepped attenuator. It would be comforting to know that settings (other than the one detent) were truly equal between the speakers, rather than just eyeballed. The HS50M monitor offers high-cut, mid EQ, room control, and high trim switches in order to match the speakers to their environment. Grilles are not included, but the tweeter is protected behind a mesh screen. The subwoofer offers a phase reverse switch as well as switchable (and sweepable) low-pass and high-pass filters. Both systems are magnetically shielded, made in China, and carry a one-year parts and labor warranty. The excellent manual offers useful information geared towards using the monitors for stereo, stereo with subwoofer, and 5.1 surround sound operation.

IN USE

I auditioned the HS50M monitors with and without the HS10W subwoofer. The first thing that I noticed about the monitors is that they play much more loudly than you might expect given their compact size and small drivers. Tonally these Yamahas are much more extended both at the top and bottom of their frequency response than the venerable NS10s, but there is a slightly phasey quality which hinders both the imaging and the transient attack preventing instruments such as tambourines from sounding "there" in the way truly high end speakers do. It's not drastic, but it is there nonetheless.

Taken as a sub/sat system, there is a slightly ripe sound in the upper mid bass, which seems to be centered around the low C of a four string bass. I also noticed that some kick drums sounded a little bit more tubby than I remembered. There is a little bit of overhang overlaid onto some signals, and if you really push the sub, the tube that couples the port to the inside of the subwoofer tends to vibrate and add a bit of a rattle to the output. Perhaps that's its way of letting you know that you should turn things down a bit? Other than that, the HS10W sounds authoritative and ballsy especially for an inexpensive sub. I'd submit that its response is similar to a good quality consumer sub.

None of these tonal and spatial aberra-

Fast Facts

■ **Applications:**
Studio, post production

■ **Key Features:**
HS50M – two-way; 5-inch woofer; 3/4-inch tweeter; EQ controls; 45W LF amp; 25W HF amp. HS10W – 8-inch woofer; 150W amp; magnetic shielding

■ **Price:**
HS50M $499 per pair
HS10W $599

■ **Contact:**
Yamaha at 714-522-9011, www.yamahaproaudio.com.

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tions are "deal killers" but it is important to point out the differences you might expect to observe between these monitors and others of the same basic size and construction which cost far more.

Unlike some other budget monitors, the Yamahas handle high levels gracefully. I would be omitting the truth if I said that the sound didn't become a bit hard and abrasive at extremely high levels but at any sane listening levels the Yamahas are generally smooth sounding, lacking the ragged sound of lesser designs.

One of the review speakers (which evidently had already made the rounds as it was marked "C Stock" had a defect in that the XLR input was markedly attenuated compared to the TRS input. The other speaker did not share that defect, and a quick trip to the cable bin turned up a female XLR to TRS cable that I connected to the XLR-only speaker output of the subwoofer.

As with the recently reviewed Wharfedale Diamond monitors I ran some mixes (both 'in the box' using Nuendo, and through the superb Audio Developments AD146 console) whilst monitoring through the Yamaha monitors. I found that mixes translated well to systems ranging the gamut from a clock radio with a CD player to the honky Bose system in my A4 Avant (please don't get me started on Bose...) as well as a friend's Boston Acoustics home theater system.

On more revealing systems, I found the high end to be a bit crispy combined with a mid bass hole (corresponding neatly with the slightly tubby character I noted earlier). In all fairness, a little more time served with the Yamahas would probably go a long way towards producing perfectly balanced mixes.

I found the Yamaha monitors to throw a fairly well defined soundstage, wider than deep. As with other inexpensive monitors, low level detail (such as reverb tails) was harder to resolve than with higher-end models.

Without the subwoofer in use, the HS50Ms still generate enough low frequency output (other than the really low notes, of course) that one could feel confident using them to set the relative levels of bass drum and four string bass, for example.

SUMMARY

Keep in mind that the original NS10s were never meant to be used as primary monitors, (that function was delegated to UREI 813s, Tannoy Golds, and Altec Big Reds among others) they were instead used a reality check to make sure that finished mixes would sound correctly balanced on consumers systems. One might look at them as a bridge between the professional and consumer worlds. As with the original NS10s, the new Yamaha system accurately mimics the more extended yet still somewhat colored home theater, domestic listening, and mobile systems of today. Bottom line? Yamaha has succeeded in developing a worthy heir to the NS10s.

Richard Alan Salz heads Vermont Audio Labs (www.vtaudio.com) a multifaceted audio consultancy.

Review Setup

Audio Developments AD164 console; FBB Fretless Bass; Audio-Technica 4060, Audix D6 microphones; Steinberg Nuendo 3.2.

World Radio History

BY STEPHEN MURPHY

Lavry Engineering
LavryBlue Mic Pre Module

O ver the last two decades, engineer Dan Lavry earned a reputation as a *converter designer extraordinaire.* He has designed converters for well-known pro audio manufacturers including Apogee Electronics, Otari, New England Digital, as well as for his own company Lavry Engineering (changed from dB Technologies in 2002).

While many top mastering and recording engineers revere Lavry's high-end Gold series converters, the introduction of the modular and more-affordable LavryBlue series makes Lavry's innovative converter technology available to a much larger market. Last year's introduction of a two-channel microphone preamplifier module ($968) greatly expands the potential market and applications of the LavryBlue series.

FEATURES

The LavryBlue series is a user-configurable modular system based around a single-rack space chassis (model LE 4496). The chassis has five single-width module bays; modules are powered by the LE 4496's internal power supply, which interfaces with the outside world via a standard IEC power cable.

The LavryBlue series modules include the M•AD 824 two-channel A/D converter, M•DA 824 two-channel D/A converter, M•BY2 synchronous up/down sample rate converter, M•SYNC master clock sync, and the M•PREAMP two-channel microphone preamplifier. This review primarily focuses on the LavryBlue two-channel mic preamp module. The overall LavryBlue system and con-

Fast Facts

■ **Applications:**
Studio

■ **Key Features:**
Two-channel microphone preamp module; gain settings via ±1 dB gain selector toggle switch and two-digit LED readout; optimized input impedances for ribbon, dynamic and condenser microphones; 10 dB pad; phantom power supply per channel

■ **Price:**
$968 (preamp module); $3,498 (as tested)

■ **Contact:**
Lavry Engineering at 206-842-3552, www.lavryengineering.com.

verter modules were covered in depth by mastering engineer Alan Silverman in the April 2004 issue of **Pro Audio Review** (the review can be found at *www.proaudioreview.com*).

The LE 4496 chassis can accommodate two of the double-width, two-channel mic preamp modules, yielding a total of four channels per rack unit. Lavry essentially treats the two channels within each module as a stereo pair – each channel has its own input gain setting but all other controls are global, affecting both channels in the module.

The M•PREAMP module employs a novel (and highly accurate) gain-setting scheme: each channel is equipped with a momentary three-position mini toggle switch that adjusts the input gain up or down in 1 dB increments, while a two-digit LED display indicates the channel's current gain value. The normal gain range of the preamp pair is 21 dB – 70 dB; a toggle switch on the left side of the module activates a 10 dB pad on both channels, resulting in a gain range of 11 dB – 60 dB.

The final control on the front panel of the module is a three-position toggle switch that selects optimized input impedance for ribbon, dynamic or condenser microphone pairs. Selecting "condenser" automatically activates 48V phantom power for the input pair. In a move typical of Lavry's technical excellence, a double-regulated phantom power supply is implemented on each channel to eliminate any cross-channel interaction that may affect separation.

Additional user-configured settings (via internal jumpers) include signal phase reverse, balanced/unbalanced output choice and a phantom-power defeat for mics that utilize external power supplies.

IN USE

As I mentioned earlier, a single LavryBlue LE 4496 rackmount chassis can accommodate two of the M•PREAMP stereo mic preamp modules, but many other configurations

are possible using the other modules in the LavryBlue range. For this review, I was sent a LE 4496 outfitted in a manner that exemplifies the flexibility of this modular system. The configuration included the M•PREAMP two-channel mic preamp module, an M•SYNC master clock sync module, an M•AD 824 two-channel A/D converter, and an M•DA 824 two-channel D/A converter.

This configuration provides everything one would need for a top-notch two-channel recording/monitoring system that would be ideal for critical location recording or as the centerpiece of a high-resolution digital production studio.

So how did it perform? Let's cut to the chase: I was tremendously impressed by the LavryBlue mic preamps, and the LavryBlue converter system in general. The preamps were extremely quiet and clean, and provided a gain range about as ideal as I could wish for. The preamps ably accommodated everything I threw at it, from very quiet ribbons to super hot condensers – I'd be hard pressed to name another preamp that matched so well in every instance.

I had the opportunity to use the LavryBlue on a variety of sessions and sources over the course of several weeks. Sources included male and female vocals, classical and jazz piano, a string quartet (stereo room pair), electric guitars and bass, and acoustic guitar. In all cases, the preamps felt true to the sources and mic choices, and yielded a wide and dynamic representation of the live performance. The piano, strings and acoustic guitar recordings really sold me on these preamps – crystal clear with plenty of dynamic depth and excellent channel separation.

To quantify the Lavry sound (or lack thereof): among the solid state crowd, the Lavry preamps definitely fall on the critical recording, super-clean (Millennia-ish) side of the fence, and away from the distinctive character set (API etc.). The Lavry preamps

BY TOM JUNG

Violet Audio ADP61 Studio AV Preamp

Violet Audio is a small audio electronics company based in Melbourne, Australia catering to both the commercial and pro audio markets.

The ADP61 is a multichannel preamp/tuner with built in 24-bit, 192kHz A/D and D/A converters with a wide variety of decoding options.

The ADP61 is very easy to set up and use without even cracking the manual. Dedicated push buttons on the front panel and a simple infrared remote control make operation straightforward without struggling through pages of menus.

THE GOODIES

Dolby and DTS digital decoders are supported along with their entire inventory of matrix surround formats. A six-channel stereo mode can take a stereo input and send it to all six outputs without any additional processing. Dolby matrix Pro Logic II Movie and Music modes are supported as well as Dolby Digital 5.1 and Dolby Digital Surround EX 6.1 the later employing a center surround channel. Likewise DTS matrix Neo 6 Movie and Music modes are designed to take a stereo source encoded or not and turn it into surround however not discrete. Additionally DTS-ES matrix provides for a center surround while DTS and DTS ES are discrete modes and sound slightly better than their Dolby counterparts due mainly to the higher bit rate. The DTS 96/24 playback mode is also supported as long as you have the software for this format. All formats are automatically recognized and are displayed on the front panel of the ADP61.

Seven (6.1) XLR balanced outputs make the ADP61 a good choice for interfacing to a professional multichannel active monitor set up. The balanced analog multichannel inputs appear on a 25-pin D-sub connector; unfortunately a breakout cable is not supplied with the unit, a simple six-way XLRF to 25-pin would have been nice. Six multichannel unbalanced inputs are available on RCA jacks.

One of the six stereo analog inputs is balanced on XLR and has an input sensitivity switch for –10 dB, 0 dB and +4 dB why you say? More on that later. Four of

the remaining five stereo unbalanced analog inputs have both video and S-video input switching as well. Stereo unbalanced along with video outputs S-video outputs pretty much covers the analog ins and outs. Digital inputs include AES/EBU on XLR as well as coax on RCA and optical TOSlink.

THE TEST

My first listening set up consisted of a Philips SACD 1000 transport feeding the

TOSlink input of the ADP61 whose balanced XLR outputs are feeding three SLS PS8R active ribbon monitors for the front and three channels of the new Bel Canto S300/M300 amplifiers powering three SLS S8R passive ribbon monitors for the surrounds.

Source material consisted of two DTS 5.1 Surround recordings I made in the mid 1990s. The first was a project of big band arrangements played by what we called the DMP Big Band, originally recorded in 20-bit PCM then encoded to DTS, which

was at the time the most Hi-Res audio release format available. The second project was the Sacred Feast choral recording originally mastered in DSD and transferred to 20-bit PCM then encoded to DTS. The ADP61 did a fine job of decoding the DTS encoded signal and sending it to the balanced XLR outputs. Both the brass in the big band project and the voices in the choral project are very telling musical content, if anything is wrong in either the digital or analog world you'll

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I also played several familiar CDs into both the AES/EBU and coax inputs with similar results - clean, clear, full-bodied sound with low distortion.

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know it instantly. Both records exhibited good tonal balance, very little harshness and accurate placement of voices and instruments. I also played several familiar CDs into both the AES/EBU and coax inputs with similar results - clean, clear, full-bodied sound with low distortion.

Next I wanted to check out the analog inputs so I fed the output of my EMM Labs DAC8 D/A converter into the unbalanced analog inputs (since I had no breakout cable for balanced) and played a few SACDs. Wow! What a disappointment.

[*Violet Audio responds: 'From what we know the EMM Labs DAC8's outputs are +4 dB balanced but the ADP61's RCA inputs are unbalanced -10 dB level designed for CDs and DVDs so a mismatch is highly likely in such situations.*]

Apparently the analog signal is converted to digital inside the ADP61 and not so well at that. The image collapsed, the detail went away and the whole presentation fell apart.

[*Violet Audio responds: 'This is correct. All analog inputs other than the 25-pin D-sub are converted into digital signals to allow for onboard DSP'.*]

I also tried the balanced analog stereo input feeding it with a +4 dB signal and it was distorted until I reset the input sensitivity switch to −10 dB, apparently I was overdriving the A/D, with standard +4 dB? Not cool.

[*Violet Audio responds:' The input sensitivity switched was incorrectly marked on many early units. It seems that -10 dB is where +4 dB should be and vice-versa. We have posted this warning on our website, www.violetaudio.com.'*]

If your monitoring needs are limited to digital sources be they Dolby, DTS or PCM and you need a tuner the ADP61 might work just fine for you, but if you are looking for an all around digital and analog solution I'd keep looking.

[*Violet Audio responds: 'The primary reason for making the ADP61 was to give the ability of 5.1/6.1 decoding for active speaker systems, AES/EBU, optical and coax inputs and provide balanced outputs on XLR connectors as well downmix facilities. We feel we achieved that with the ADP61. Certainly the revised model, with*

added features such as a subwoofer crossover control, input volume memories and better interior electronics, is a much improved model. The configuration that Tom reviewed is no longer available. We hope he will take a look at the new version.']

The MSRP for the ADP61 is $1,495 and includes a rackmount kit. If it was $600 or so I might not be so critical.

Contact: Violet Audio/Kaysound at 514-633-8877, www.violetaudio.com.

Tom Jung, founder of DMP Records, is **Pro Audio Review**'s *technical consultant and a regular contributor.*

Studio

RME ADI-192 DD
Universal Format Converter

BY DR. FREDERICK J. BASHOUR

As I keep acquiring new digital recording equipment—without necessarily getting rid of the older gear—I continually run into interfacing and compatibility issues. In fact, I think I've reviewed at least three digital format converters in this magazine over the past 10 years! First, I needed to convert from the now long obsolete Yamaha MEL-2 format to AES/EBU, ADAT ODI and TDIF gear, so the Spectral Translator Plus of the mid-1990s did that for me. I still have three of them. A few years later, I wrote about the Z-Systems 32-32r system (PAR, 4/98) which enabled me to keep 32 stereo channels of AES/EBU gear continually

Fast Facts

■ **Applications:**
Studio, post production

■ **Key Features:**
Eight-channel triple, 24 bit/192 kHz format conversion between AES/EBU, S/PDIF, ADAT optical, and TDIF in any combination, and simultaneously; eight-channel sample rate conversion

■ **Price:**
$1,495

■ **Contact:**
RME/Synthax at 330-259-0308, www.synthax.com, www.rme-audio.com.

hooked up, and route them at will.

But then, I got into higher sample rate recordings, so the number of channels I needed effectively doubled (at least using the "double-wide" protocol — which came first), so I added another Z-Systems unit, various small M-Audio "necessary boxes" such as the CO3 and their S/PDIF and ADAT ODI 1x4 distribution units. And, of course, I can't forget the TASCAM IF-AE8HR (*PAR*, 9/01), which enabled me to record four channels of 88.2 kHz material on one TASCAM DA-78HR. I still have many tapes recorded in that format (which is still my "B Machine" backup system when recording sessions directly to hard disk) — often used in pairs, since playback of all eight tracks necessitates syncing up two DA-78s with two format converters in order to record eight tracks.

Finally, we cannot forget the extremely useful Apogee PSX-100 (*PAR*, 9/01) and Trak2 (*PAR*, 5/02) converters, because they both manage to do format conversion as well as ADC duties. In fact, it was because I had to commandeer my Trak2 a few months ago — in order to convert four 88.2 kHz channels coming from my venerable TASCAM recorder/converter system into the ADAT S/MUX format necessary to send some French organ music into my FW-1814 M-Powered Pro Tools system, that I finally said, "Enough already with multiple back-to-back-to-back format conversion systems! Surely someone must make a single box now which can do all these conversions at once."

Yes, there is! The answer is the RME ADI-192 DD, a single rack-space unit that

does more slicing and dicing than any piece of digital equipment I've ever seen. I can simultaneously connect digital gear with in three formats (ADAT ODI, TDIF, and AES/EBU). and output whichever of the three inputs I select to all three outputs at once. It operates with PCM sample rates up to 192 kHz, and observes the double and quad-wide protocols of S/MUX and S/MUX 4. Not only that (speaking of slicing and dicing), but one can also change sample rates along the way if desired. since it also features an eight-channel SRC circuit — and a fancy high-tech clock generating system to keep everything locked together and free of clicks and glitches.

FEATURES

Basically. the RME ADI-192 DD consists of three separate eight-channel digital format converters housed in a single rackspace unit. It also contains switchable SRC circuits that allow for both sample rate conversion and clock decoupling at a very high level of accuracy. The format conversion between AES/EBU, ADAT ODI and TDIF operates in both directions at the same time. RME's so-called Intelligent Clock Control permits extreme flexibility; the internal clock can operate up to 192 kHz, or the unit can lock to external word clock, or any channel pair of the digital input signals. LEDs of different colors show the present state of incoming and outgoing signals, SRC processing, and the lock and/or sync states.

The unit's front panel is laid out very logically, divided into seven sections with 46 LEDs and nine pushbuttons. (Over on the far right, next to the power switch, is a pair of TOSlink S/PDIF I/O jacks, which can substitute for AES/EBU channels 1-2.) It's really easier to just look at the front panel, and the flow chart in the manual, than it is to describe the elegant switching and routing setup.

The three input formats monitored on the left are sent to the three output source selectors on the right. Their selector pushbuttons give them access to any of the three inputs. Since the three selectors operate independently, one can either send a single input source to all three together, or send each input source to a different output format destination. And, as I said, along the signal pathway, one can select the SRC circuitry to act on one of the three inputs.

whose output can then be sent to all three outputs at the new sample rate.

If only one AES input is used, the ADI-192 DD automatically activates "distribution mode," in which the single stereo input will be copied to all AES/EBU outputs. This special mode — active for the low "single speed" sample rates only — is also available when using the front S/PDIF input, and also allows SRC operation.

The rear panel is simplicity itself. Four sets of AES/EBU inputs and outputs, two pairs of ADAT ODI TOSlink-type inputs and outputs, and two TDIF DB25 connectors are arranged logically. as well as the word clock BNC I/O ports, with their associated 75 ohm termination switch (with confirmation LED).

IN USE

I unpacked the unit and — having salivated over my downloaded user's manual for what seemed like an eternity while waiting for it to arrive — I was entirely ready to attempt the four channel 88.2 kHz setup I had been doing previously with my tricked-out Trak2. I mounted the

ADI-192 DD in the rack out in the studio — a setup I use for tracking when I'm playing piano at the same time. In that rack, the analog-to-digital converters are my Genex GXA8 (*PAR*, September, 2002), and that unit outputs AES/EBU at all sample rates up to 192 kHz. I hooked four short Mogami AES/EBU cables between the Genex's outputs and the RME's four XLR inputs.

To get into Pro Tools M-Powered (running on my 1.5GHz 12-inch PowerBook), I now use my new M-Audio ProjectMix I/O control surface, so I simply connected a short plastic fiber optic cable between the first of the RME's pair of ADAT ODI outputs (which, in S/MUX mode, transmits channels 1-4) and the PM I/O. I powered everything up, ran the M-Audio control panel and selected S/MUX and fiber optic input, and opened an old four-channel, 88.2 kHz session template in Pro Tools 7.1. The mics were already plugged into my Manley Mic-EQ500 and D.W. Fearn VT-2 mic preamps and powered up, so as soon as Pro Tools booted up the ses-

sion, and I put it into record on the appropriate four channels, the proper meters started reading, and I knew that RME had gotten everything right. A quick listening test confirmed this; I had the two stereo pairs I was expected. I know from bitter experience that, when bit-splitting isn't working, sometimes one gets mono, sometimes one gets only one channel, or sometimes nothing at all. But if you get good-sounding stereo, you know you're home free.

When using both TDIF ports (main and aux) on the RME unit, two DTRS machines can be connected; receiving Channels 1 - 4 and 5 - 8, respectively, at 88.2 kHz/96 kHz, or Channels 1 - 2 and 3 - 4, respectively, at 176.4 kHz/192 kHz. This mode, previously available to me only on the Digital Audio Denmark 2408 converter (and its successors, built for Merging Technology) is, in fact, the way I recorded several sessions in Europe a couple of years ago, including the 38 DTRS tapes I made in Paris for my Langlais organ music project. This was the material that caused me to seek out the ADI-192 DD in the first place, as detailed in this article's introduction. So, yes, I can now take these tapes directly into Pro Tools with only one device!

If only one AES input is used, the ADI-192 DD automatically activates "distribution mode," in which the single stereo input will be copied to all AES/EBU outputs. This special mode — active for the low "single speed" sample rates only — is also available when using the front S/PDIF input, and also allows SRC operation.

RME's implementation of Bitclock PLL (as opposed to simple

LIPINSKI
S O U N D

Powered Stand

word clock PLL) in its "SteadyClock" technology is said to combine the advantages of up-to-date digital technology with analog filter techniques. In fact, RME states that it finally accomplishes the result Digidesign's so-called SuperClock (256 times the WC frequency) was supposed to do, but often failed, due to the limits of high frequency transmission technology, and that regaining a low jitter clock signal of 22 MHz from a slow word clock of 44.1 kHz is no longer a problem.

Here in my studio, I have two very high quality word clock generating and distribution systems — Apogee's Big Ben (*PAR*, 3/04) and an updated version of the Rosendahl Nanosyncs unit reviewed for *PAR* in March, 2002 — necessary since I have elaborate recording installations in both my control room, and in the large studio room itself. Both units improve the sound of any source locked to them, and I can now picture a setup in which each of *them* is locked to the RME ADI-192 DD. Only one of them can be in the same room with it, so I'll just have to experiment which method (AES or WC) works better over the 40 feet or so (as the wiring within the walls goes) distance between my control room and studio equipment rack which contains the Nanosyncs.
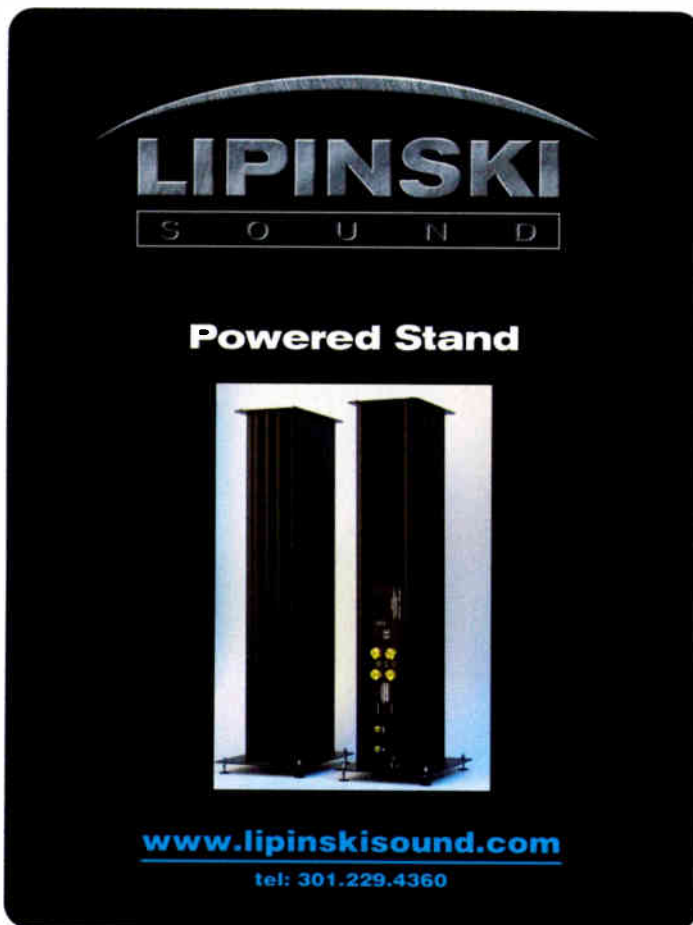
RME claims that its eight channel SRC circuit exhibits exceptional conversion quality, formerly available only from synchronous SRC devices. I tested two such devices for *PAR* (from Weiss Engineering and dCS) back in June, 2000, but no longer have either one available for comparison. I do, however, have a Lucid SRC 9624 (*PAR*, 5/01), a Kurzweil DMTi (*PAR*, 6/97) as well as the SRC circuitry built into my various MOTU hardware interfaces, and I can state that the RME ADI-192 DD's asynchronous conversion process sounds much better than anything else I have here in Studio Dufay.

While monitoring the effect of the conversions through my Weiss DAC 1, Mk. II (*PAR*, 12/05), an extremely accurate high end digital-to-analog converter, I was pretty hard put to tell any difference at all in when upsampling, and when reducing sample rate, I heard only the typical sound of the lower sample rates themselves. Amazing! RME suggests that, since its SRC process is so transparent, it be left on all the time so that its clock decoupling and signal conditioning features would be available on all digital audio bitstreams coming in and out of the ADI-192 DD. I'm tempted to do this, for it virtually guarantees the lowest amount of jitter and a complete lack of clock sync issues.

SUMMARY

PAR's editor doesn't allow any of its writers to use the term "Swiss Army Knife," but if I *could*, the RME ADI-192 DD would certainly qualify. It is, in fact, the "missing link" (if he'll let me say that) I've been searching for to *finally* ensure that all my digital audio equipment will play nicely in my studio sandbox. I'm sure it would do the same for any other studio that needs to interface as many (past and present) digital audio formats as I do. Very highly recommended.

Dr. Fred Bashour holds a Yale Ph.D. in Music Theory, and currently performs as a jazz pianist and church organist. During the past 25 years, he has received credits on hundreds of recordings released on over a dozen labels. He has also been a regular contributor to Pro Audio Review since its second issue.

AKG Perception 100/200 Microphones

BY RICHARD ALAN SALZ

It seems like at least half of the microphones on the market today are from companies that didn't even exist 15 years ago, much less in the "golden age" of microphones back in the 1950's. On the other hand, studio grade condenser microphones are now available for less than the cost of a nice dinner for two. The AKG Perception 100 and 200 answer the question about what would happen if an "old world" manufacturer developed a "new world" priced microphone.

Like more and more products (and not just in the audio field) the Perception microphones are made in China rather than in Austria with AKG's other products, but let's put that information off to the side for a moment.

FEATURES

Both the Perception 100 and 200 are transformer-coupled condenser cardioid microphones, with externally polarized one-inch diaphragms. They differ from each other in that the 200 adds a low-cut filter (12 dB per octave at 300 Hz) and 10 dB pad switches, a metal spider-style shockmount and a rather nice hard shell box.

Both microphones seem exceedingly well made, feel solid in the hand, and the fit and finish would not be out of place on a microphone costing 5 - 10 times the

Fast Facts

■ **Applications:**
Studio, sound reinforcement

■ **Key Features:**
Condenser; cardioid pattern; 1-inch diaphragm, low frequency cut filter, 10 dB pad (Perception 200)

■ **Price:**
Perception 100 - $199
Perception 200 - $319

■ **Contact:**
AKG at 615-620-3800, www.akg.com.

price. Without a doubt they are the nicest budget microphones I've seen in terms of quality of design, styling and execution.

The manufacturer's stated specifications for the Perception 100/200 include a frequency response from 20 Hz to 20 kHz (with the deviation appearing to be +3 dB/-6 dB according the frequency response chart) and a sensitivity of 18mV/Pa. Signal-to-noise ratio is 78 dB, maximum SPL is rated at 135 dB for the Perception 100 and 145 dB for the Perception 200 with the pad engaged at less than 0.5% THD. The Perceptions' impedance is specified at nominal 200 ohms, and the self-noise is a respectable 16 dBA, which is lower than other microphones in its price class. The Perception microphones carry a two-year warranty.

IN USE

For this review, I received a single microphone of each model. Although they differ a little bit in terms of their feature set, they do use the same diaphragm and basic electronics, so I compared them to each other, and used them as a stereo pair. Both microphones exhibited similar tonal characteristics and were matched closely enough in order to use them as a stereo pair in non-critical applications.

The overall sound of these microphones

is probably best described as modern in terms of having the expected rising high frequency range typical of newer condenser microphones. Unlike some other low-buck condensers they manage to avoid shrillness while still sounding extended in the highs.

The Perception microphones had a compact and reasonably focused sound, with male vocalists easily cutting through dense mixes. Compared to higher-dollar microphones, a hazy and electronic veil could be noted on the Perceptions. At their price point, however, I have yet to hear any microphones that sounded better.

On a FBB Custom fretless bass played through a Marshall bass cabinet the Perception mics portrayed the extended and taut sound of this combination. Beware that the cardioid pattern is fairly tight, and as such bass response will fall off sharply if the microphone is significantly off axis. Incidentally, many of the Chinese microphones seem to improve in low frequency extension the more they are used. PAR editor John Gatski recommends positioning microphones in front of a subwoofer playing 80 Hz test tone at a healthy level as a good method to "loosen up" recalcitrant diaphragms.

The Perception 200 was a great choice miking up a Tom Anderson Strat-style guitar played thorough a Marshall JCM 800 2210 half stack, yielding an edgy tone which was much ballsier than the typical Shure 57 sound. Make sure to turn the gain way down if you plan to change the settings of the low cut or pad switches, as they insert a painfully loud pop when moved whilst powered up.

The Perception microphones will suffice as room microphones but they don't have the resolution of low-level detail and spatial positioning that is the hallmark of a really great microphone. Then again, you can purchase a Perception 100 for less than $100! As with the low frequency extension, the resolution of low-level detail also may improve as the diaphragm becomes more compliant through usage.

Using the pair of Perception microphones on a Rainsong acoustic (the 100 pointed towards the bridge, and the 200 pointed towards the neck joint with the low cut engaged) I was more than pleased with the frequency balance and transient speed when monitored through an Audio Developments 146 console. A small cut in the midrange was all that was needed for a keeper track.

SUMMARY

You're not going to be selling your vintage C12s in order to buy a bunch of Perception 100/200 microphones, but considering the asking price, I don't think there is any question that the AKG Perception microphones are a terrific value. Live sound companies in search of ruggedly built, good-sounding and inexpensive large-diaphragm condensers need search no longer. Highly recommended!

Richard Alan Salz heads Vermont Audio Labs (www.vtaudio.com) a multifaceted audio consultancy.

Review Setup

Audio Developments AD146 console, FBB Fretless Bass, Tom Anderson guitar, Steinberg Nuendo 3.2.

BY RUSS LONG

SPL Transient Designer 4 Model 9842 Processor

Jeremy Ramsey from Rack & Roll Audio in Nashville turned me on to the SPL Transient Designer 4 Model 9842 ($1,349) a couple of years ago and I've been hooked ever since. I bought the four-channel version and I routinely compliment it with a second four-channel unit during tracking sessions. The Transient Designer provides the ability to increase or decrease the attack of a sound source and to extend or shorten its sustain. The box actually makes it possible to shape the dynamic path of a signal practically giving you the ability to change the microphone distance and position after the recording has been made. It accomplishes this using Differential Envelope Technology (DET) which offers a new way to provide the independent level shaping of the dynamic response of a sound.

FEATURES

The 1U Transient Designer 4 measures 19 inches x 1.7 inches x 9.3 inches and weighs 7.5 pounds. The box has four female and four male XLR connectors for input and output. The input impedance is 22 kohm and the output impedance is less than 600 ohms. The unit's nominal input level is +6 dB with a maximum input level of +24 dBu and a maximum output level of +22.4 dBu. The box has a frequency response of 20 Hz-100 kHz with a maximum load of 600 ohms and a total harmonic distortion of 0.01% @ 1 kHz. The unit accepts a standard IEC power cable and is switchable

between 115 and 230 volts.

DET is the first analog solution for the level-independent shaping of envelopes allowing transients to be accelerated or slowed down and sustain prolonged or shortened. The Transient Designer has only two controls, Attack and Sustain, essentially deeming traditional dynamics controls such as threshold, ratio and gain

trolled by the same side chain voltage so as to maintain a coherent and stable stereo image. When operating in link mode, the control elements of the odd channel (Channel 1 or 3) control the pair.

To assure impeccable signal quality, SPL developed a hybrid-component balanced input/output stage using all laser-trimmed resistors with a tolerance of

> The Transient Designer Model 9842 has earned its place on the short list of modern day classics. The box provides simple control of the attack and sustain of a signal in a very easy way.

unnecessary. The Transient Designer's internal processing is highly developed, so that while the processing going on inside the box may be very complex, the user only has to deal with these two intuitive controls. The degree of dynamic processing required to do this couldn't be duplicated even using a chain of several conventional compressors, yet only two controls per channel are required to allow the user to completely reshape the attack and sustain characteristics of a sound. Attack can be amplified or attenuated by up to 15 dB while Sustain can be amplified or attenuated by up to 24 dB. All of the necessary time constants (attack, decay and release) are set automatically in a musical fashion according to the input signal's characteristics providing quick and natural sounding results.

For stereo operation the link function connects channel pairs in an odd/even configuration (Channels 1 and 2 and/or 3 and 4). The linked channels are con-

0.01%. The result is a CCMR (common mode rejection) that is better than -80dB at 1 kHz.

IN USE

Although the Transient Designer is a particularly powerful device, I found it to be exceptionally intuitive to use. Turning the Attack knob increases or decreases the level of the signal's transient and turning the Sustain knob increases or decreases the sustained portion of the sound. That's pretty much all there is to it. Pressing the channel's On button activates the processor and illuminates a red status LED inset in the button. When the On button is switched out, a hard-bypass relay circuit is engaged. This circuit engages automatically if the unit loses power making the box perfectly suited for sound reinforcement situations. Each channel also has a "Sig" (signal present) LED that illuminates when the input signal exceeds -40 dBu.

World Radio History

The Transient Designer's forte is no doubt drums and percussion. I have spent hours with the box effortlessly shortening or lengthening the attack of all kinds of percussive signals with no negative sonic artifacts whatsoever. It works well with kick drum, snare drum, toms, congas, bongos, etc. I found that it is almost like being able to change the amount of drum damping after the recording has taken place. I had good results emphasizing the attack of a loop to increase the energy and aggression of the loop in the mix. I found that using the box on snare and toms when mixing, is like being able to return to the tracking session and change the amount of drum damping. If the drums were recorded in a room that was too ambient, the Transient Designer can shorten the room's decay time in a very musical way or during tracking the snare, toms, or overheads can be shortened without being physically dampened.

I was able to get the same results with practically every other instrument I used it on including acoustic guitar to get a mellower, rounder, more relaxed sound, on piano to decrease the ambience and on upright bass to add attack and in every case had positive results.

I've found that by increasing the attack and decreasing the sustain, instruments can be moved more to the front of the mix without being turned up. This works exceptionally well for percussion sounds that need increased clarity but need to remain low in the mix. I've also had some good results using it on acoustic guitar with the same approach.

Using the box to add attack to a bed of highly distorted electric guitars was amazingly able to bring out some clarity in the individual notes. While mixing a track that included a stereo choir (that was recorded in an overly ambient hall) I found that turning down the sustain provided an intelligibility that I wasn't able to attain through any other means.

SUMMARY

The Transient Designer Model 9842 has earned its place on the short list of modern day classics. The box provides the simple control of the attack and sustain of a signal in a very easy way. Unlike traditional compressors, the Transient Designer's processing is not governed by the signal level but rather by its dynamic characteristics, so all signals
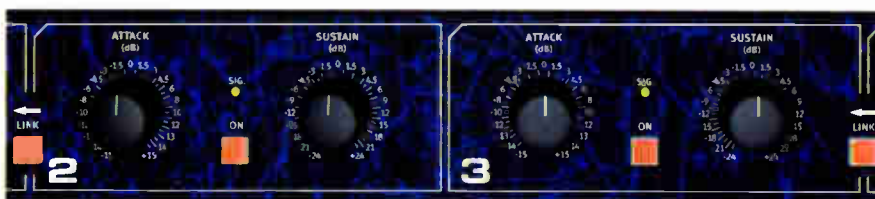
(loud and soft) are processed equally.

Russ Long, a Nashville-based producer/engineer, owns The Carport recording studio. He is a regular contributor to **Pro Audio Review**.

Review Setup:

Apple 2 GHz Dual Processor G5 w/2 GB RAM; Digidesign Pro Tools 7.1; Lucid Gen-X-96 clock; PMC AML-1 monitors.

Grammy-Winning D.I.Y.

BY STROTHER BULLINS

Singer/songwriter Will Owsley — a multitalented producer, engineer, session guitarist and solo recording artist — is almost always in the recording studio. To be exact, he's almost always in *his* recording studio, a well-stocked home facility located in the Nashville area, nearby to many of his musical collaborators and creative cohorts. As a longstanding industry professional — from collaborating with the likes of Shania Twain, Amy Grant, and the Neville Brothers to gaining a Grammy nomination for the engineering of *Owsley*, his inaugural and dynamic solo release — Owsley has had countless opportunities to use many of the world's best recording equipment.

HOME SWEET HOME

However, while all the great places Owsley has recorded are nice, home is where his heart is. "It's my own little home, and I stay down there," he says of his basement studio. "Except when I do sessions, I rarely venture out. A lot of the people that I work for just give me hard drives and say, 'Hey Will, would you play on my record?' I'll take it down to my studio and I have all my guitars, instruments, and stuff down there. I record my parts right there."

For his home sessions, Owsley often uses his vintage Neve mic preamps, UREI 1176 and 1178 limiters, and Focusrite Red 3 compressor in conjunction with his a Soundelux ifet7, a new microphone favorite. The Soundelux ifet7 is a cardioid, phantom-powered, two-in-one re-creation of the Neumann U47 and Neumann U87 vintage FET microphones.

"For my console, I have an old, fully-discrete Soundcraft 1600," Owsley enthusiastically explains of his beloved analog mixer. "Its EQ and mic preamps sound unbelievable. I often use the Neve and 1176 for a front end to my new Soundelux ifet7, going into either Pro Tools or RADAR. If I go into RADAR, I usually transfer over to Pro Tools, just because it has the automation, all the editing capabilities that you could ever want. Plus, the session's probably going to end up on Pro Tools anyway!"

Because Owsley possesses a quite impressive and distinctive voice, he has learned all about great large diaphragm microphones, becoming "a vintage microphone adorer," he explains. "Long ago when I was a writer for Sony Music, they had a Neumann FET 47, which I used to sing on several things," Owsley recalls. "Also, way back in the day, I was in a band called The Semantics on Geffen Records. For the record, I sang through a FET 47. In both instances, I remember how much I liked my voice through it. When I saw that Soundelux came out with the first-ever quote/unquote 'copy' of the FET 47, I was immediately interested. I've also used a Neumann U87 a lot for voice, and since the ifet7 is a combination of a FET 47 and a U87 — it has a switch that selects between two settings — I had to hear it just for the concept alone."

FLEXIBLE MICROPHONE

A flexible microphone is a must for a professional self-recordist such as Owsley and, thus far, he has recorded a wide variety of instruments and vocal sources via his beloved ifet7. "I really love the way it sounds," he declares. "I've recorded upright bass with it, and we all know that FET 47s sound instantly unbelievable on kick drum. On the 'instrument' mode, it sounds very much like a FET 47, but it has a bit more gain, and it really captures bass transients well. Also in the instrument mode, I've cut bass drum, bass, and guitars. On guitars, the ifet7 has a very open sound, and it's a great combination with an SM57. I've also cut mandolin and acoustic guitar with it; any kind of stringed instrument records fantastically with it in the instrument mode, too."

Owsley feels that the ifet7 especially shines in vocal miking applications. In the vocal or 'V' setting, I've used it on my voice and lots of backgrounds," he explains. "It sounds huge and — to me — even better than a U87. The top end is really open, and it also has great lows. I was instantly a fan of it in that mode. I feel that the vocal setting is similar to the instrument setting, but it has more of a defined top-end."

While he's eager to talk about his new microphone, he insists that it is largely because of the benefits it offers others like him: self-recording musicians at varying levels of success. I'm just really impressed with it," he concludes. "A lot of microphones do one thing really well, but not two things really well. I think that Soundelux has really broken ground with a microphone that has two great sounds for the price of one."

Currently, the super-busy Owsley is promoting a new full-length release entitled *The Hard Way* — while staying busy with continual production, engineering, and session gigs. His groundbreaking 'digital record deal' with Universal Records — a great example of his progressive approach to the recording industry — is helping promote his new music to many new fans in an exciting new way.

"It's the first-ever all-digital record deal," describes Owsley. "It means that Universal has the rights to my music in the digital domain. They market and promote my music on the Internet, and I'm free to do other deals as far as the hard copy of my record is concerned; I have placed pressings of my own at www.owsleymusic.com. It's a new way of thinking for breaking artists, and Universal is tied in with iTunes and AOL. A couple of weeks ago, I was the free download of the week on the front page of iTunes — on the home page — and we're having great results with it."

Contact: www.owsleymusic.com.

Strother Bullins is a North Carolina-based freelance writer specializing in the professional audio, music and entertainment industries.

twist (the twist is key to proper insertion). The standard Steely Dan was dialed up and… "Wow!" First off the monitors were extremely comfortable and the quality was amazing. Every nuance was detailed in great accuracy with unbelievable separation. After playing around with the iPod EQ a little to boost the bass to test the response I was thrilled. I was disappointed that the low end had to be boosted significantly to hear the low-end thump as my other headphones didn't need any boost. So a real world test was in order.

There was a show coming up where with the Annapolis Symphony Orchestra would be playing the songs of The Eagles, which screamed "test these out". Since the band that was performing with the ASO was all on in-ears (mainly to keep volumes down as not to disturb the orchestra) using the UE-10 Pros as my "cue ears" would be perfect. Hearing a properly EQed kick drum for the first time through the UE-10 Pros was great. Any fears I had that the low-end response wasn't up to par were tossed out the window. Though I will say, if I was a drummer I would want a sub or shaker under my throne so that I could feel the low end as well as it can be a little deceptive to hear it, but not be moved by it.

SUMMARY

In all, even after the hassle of the incompetent audiologists, I could not be happier with the UE-10 Pros. The precise fit and clarity are inconceivable — the custom metal carrying case is pretty cool too. I have already recommended several of my fellow engineers to get a set. A dimly lit room and *Dark Side of the Moon* will be a whole new experience (mind altering drugs not included).

*David Rittenhouse is a senior live sound engineer at RCI Sound Systems and a contributor to **Pro Audio Review**.*

Review Setup:

Apple iPod; Yamaha PM5D console; Shure PSM700.

board, which as has been related was the simplest and poorest clock that Phillips suggested at the start of the CD era.

The clock to me is the most important part of the player with regards to it conveying the ambience in a recording. Some of the best transfers to CD from original recording have been made by engineers who borrowed a rubidium clock for some of their CDs.

Also, the audio chips used in electronic devices can also affect the ambience and transients. Only Steely Dan's engineer in his column has, for example, ever mentioned the dreadful 5532 Signetics chip which engineers still use. It actually eliminates ambience and has horrible transients on real music, not test benches.

Besides the clock there are the nebulous CD tweaks which make a big difference. Both the Statmat (by Ringmat) and the latest Marigo Signature III disk above the CD, lend great clarity to items such as the fingers sliding on a guitar string, which would improve

In a live setting with a local rock band I patched the CPL2 into the buss insert for the drums with good results, taming an otherwise skull shattering 6.5-inch x 14-inch chrome Ludwig snare and 26-inch bass drum (the guy likes Bonham).

From a fit and finish standpoint, the CPL2 is solid, with nice easy to read graphics on a silver powdercoat chassis. I also appreciate the IEC power cord, as opposed to those infernal wallwarts. This also means the unit had to pass NRTL (Nationally Recognized Test Lab) testing with an internal power supply, as evidenced by the ETL logo. The unit has a high 50 kohm input impedance and low 60 ohm output impedance, which should easily enable good impedance matching with most other brands of pro audio equipment.

I liked the Crate CPL2 stereo compressor; it has all the basic functions needed in a dynamics processor, with a couple of useful additions. And while it probably won't be found in a mastering lab, at a list price under $170 dollars, it is a good means to entice the MI crowd into embracing dynamics control.

Roger Williams III is a long-time sound engineer and guitarist.

some of the problems that John found in his listening session.

In many years of listening, my noncomprehensive survey of CD players at many years of CES, has found only one CD player, in a 2005 demo, which was really impressive - a Gryphon model for $12K. It is one of only two players I have ever seen in their specs that has a heated crystal clock - this one having two separate clocks.

It is amazing that 20 years later, we can only make a handful of CD players that properly convey the information on the disc.

Norm Relich
Lansing, IL
Purdue University Calumet Physics

Correction
*It should be corrected that Media Numerics' RockNet 300 Network (**PAR** June, 2006) is not Ethernet-based as stated. It is a proprietary system that uses CAT-5 cabling.*

weren't always the best choice for every application in which they were tried – sometimes color and distortion were just the ticket. But if I were forced to make a desert-island choice, I would easily opt for the accuracy of Lavry.

SUMMARY

From a utilitarian standpoint, the only criticism I can level at the Lavry stereo mic preamp is its inherent stereo (and not dual-mono) nature – the fact that input impedance and pad settings affect both channels. "Stereo" is right there in the name and I'm sure the global design helps keep it affordable, so I won't quibble too much. Although this review wasn't about the converters, I have to say that they are absolutely top notch and worth every penny. The LavryBlue series A/D and D/A converters in combination with the stereo mic preamp module makes a powerfully attractive and flexible high-resolution two-channel recording and monitoring system.

PAR Studio Editor Stephen Murphy has over 20 years production and engineering experience, including Grammy-winning and Gold/Platinum credits. His website is www.smurphco.com.

BY STEPHEN MURPHY

Building the Perfect Beast (Part 3)

Series Recap: One no longer needs to be manically obsessed or semi-masochistic to conquer the configuration of a cutting-edge Windows-based workstation capable of handling a multitude of high-resolution audio tracks and/or uncompressed HD video. By virtue of a stable OS, advances in data processing technology and easy access to online resources, nearly everyone can configure a computer from the ground up. With the continuing goal of keeping neophyte do-it-yourself computer projects from turning into SIY (screw-it-yourself) projects, this series endeavors to provide a general roadmap to the configuration of a Windows XP-based audio workstation.

ASSEMBLY REQUIRED

Last month we looked at the process of choosing the components that form the core computer system. To summarize, the simplest and safest way to ensure success is to pick the processor (or multiple processors) you wish to employ, and then use the CPU manufacturer's lists to choose a compatible motherboard and CPU heat sink/fan cooling system. Use the motherboard manufacturer's compatibility lists to choose an appropriate case (based on mobo size and other features such as number of drive bays), power supply, RAM, and video card (or at least video card slot type).

Be sure to use your target audio application's knowledge base and forums for hardware requirements, recommendations and compatibility issues. The forums can be a particularly good resource for audio interface and video card recommendations, and even complete, battle-tested system configurations.

For those who have asked, the system I built during the writing of this series is based around two of AMD's latest-generation dual-core 64-bit Opteron 280 CPUs (the equivalent of four 2.4 GHz processors). The rest of the core system consists of a Tyan Thunder K8WE motherboard, a PC Power & Cooling Turbo-Cool 850 SSI power supply and Wintec DDR 400 PC3200 RAM (4 GB).

For a bit of flexibility and future proofing, I chose a Matrox Parhelia AVPe 128MB PCIe x16 video card because it supports up to three displays and a variety of output standards (DVI, VGA, HD analog component and composite/S-video). For the system drive I am using a basic Maxtor 100 GB ATA/133 drive partitioned in two for a XP Professional and XP x64-bit dual-boot configuration. To be able to handle uncompressed HD video (one of the fundamental reasons for building this beast), my "media drive" is a RAID comprised of four Maxtor SATA II 250GB drives (16 MB cache/7200RPM) running in RAID 0 mode on the mobo's NVIDIA SATA II RAID controller.

Assembly of the components takes patience but is relatively straight forward provided you follow the manufacturers' instructions. There are a number of good sites that detail basic computer assembly, though most of the information you need is covered in the mobo manual (the Tyan manual was refreshingly detailed and clear – another sign of the improving times).

THE SOFT SIDE

Unlike Windows operating systems of the past, XP requires very few tweaks for use with audio/video applications. Before we get to the tweaks, here are some tips to make your XP installation go a little smoother:

1. Have a computer available and connected to the Internet for product reference, troubleshooting and driver downloads.

2. Before you start, download the latest drivers for your motherboard and video card – burn these to a CD.

3. When installing XP, you are given the opportunity to install third-party (i.e. motherboard-specific) disk and Ethernet controller drivers – these are usually supplied on a floppy with the mobo. If you plan to install a RAID, be sure not to skip this step.

4. Once XP is installed, use Norton Ghost to save a mirror image of the system drive. Ghost essentially allows you to restore your system to a clean install (or any other point at which you ran a Ghost image).

5. Go through the online Windows Update process, which should ultimately upgrade your OS to Service Pack 2 (SP2) – be aware that this may require at least two update processes. Ghost again.

6. Also save a Ghost image after applying system tweaks and after installing key software applications. You can save a half-day of your life and several tufts of hair by having good restore points.

The only essential change in Windows XP for use with audio applications is to set processor scheduling to Background Services: Start > Settings > Control Panel > System > Advanced > Performance Settings > Advanced Tab > select Background Services. While you are there, you can also set visual effects to a minimum by clicking on the Visual Effects Tab > select Adjust for

Sound Reinforcement Amplifiers

Lab.gruppen fP6400

Features: Two-channel; level controls; MLS Switch load matching; limiters; stereo/parallel/bridge switch; heat sinks; 650W @ 16 ohms, 1,300W @ 8 ohms, 2,300W @ 4 ohms, 2,900W @ 2 ohms per channel; 2,500W @ 16 ohms, bridged, 4,400W @ 8 ohms, bridged, 5,000W @ 4 ohms, bridged. Price: $2,395.
Contact: Lab.gruppen/TC Electronic at 818-665-4900, www.labgruppen.com.

Peavey CS 4080 HZ

Features: Two-channel; level controls; DDT limiter; bridgeable; 2,450 W @ 4 ohms per channel; 4,080 @ 4 ohms bridged. Price: $1,599.
Contact: Peavey Electronics at 866-443-2333, www.peavey.com.

Crest Audio Pro 9200

Features: Two-channel; level controls; impedance sensing; Automatic Clip Limiting; AutoRamp soft start; variable speed cooling fan; 2,100 W @ 4 ohms per channel, 6,500 W @ 4 ohms bridged. Price: $3,392.
Contact: Crest at 866-812-7378, www.crestaudio.com.

Yamaha PC-1N Series

Features: Two-channel; networkable; level controls; high-pass filter; limiter; Ethernet; dual heat sinks; stereo/parallel/bridge switch; 200W – 925W @ 8 ohms per channel. Price: starts at $1,049.
Contact: Yamaha Corporation of America at 714-522-9011, www.yamaha.com/proaudio.

Electro-Voice CP4000S

Features: Two-channel; level controls; Class H; limiter; three-speed cooling fan; 900W @ 8 ohms, 1,500 W @ 4 ohms, 2,100W @ 2 ohms per channel; 3,000W @ 8 ohms, bridged, 4,200W @ 4 ohms, bridged. Price: $2,390.
Contact: Electro-Voice at 800-392-3497, www.electrovoice.com.

Avlex PAHS-3820B

Features: Two-channel; low-cut filter; twin two-speed cooling fans; Class G; 800W @ 8 ohms, 1,200W @ 4 ohms, 1,900W @ 2 ohms per channel, 2,400W @ 8 ohms, bridged, 3,200W @ 4 ohms, bridged. Price: $750.
Contact: Avlex at 877-447-9216, www.avlex.com.

FBT Symbol 12000

Features: Two-channel; level controls; limiters; Ultra Linear Cascode Circuitry design; soft start; bridgeable; twin cooling fans; 600W @ 8 ohms, 1,000W @ 4 ohms, 1,500W @ 2 ohms per channel, 3,000W @ 4 ohms, bridged. Price: $2,199.
Contact: FBT at 800-333-9383, www.fbt.it.

Behringer EP2500

Features: Two-channel; level controls; low-cut filter; limiters; variable speed cooling fan; stereo/parallel/bridge switch; 500W @ 8 ohms, 750W @ 4 ohms, 1,200W @ 2 ohms per channel, 2,400W @ 4 ohms, bridged. Price: $489.
Contact: Behringer at 425-672-0816, www.behringer.com.

Phonic America XP5000

Features: Two-channel; limiter; high-pass filter; variable speed cooling fan; short circuit, subsonic, DC, thermal, RF protection; 2,500W @ 2 ohms per channel; 5,000W @ 4 ohms, bridged. Price: $1,119.
Contact: Phonic America at 813-890-8872, www.phonic.com.

American Audio VLP1500

Features: Two-channel; level controls; twin cooling fans; 500W @ 8 ohms, 750W @ 4 ohms per channel; 1,500W @ 8 ohms, bridged. Price: $399.
Contact: American Audio at 800-322-6337, www.adjaudio.com.

Mackie M Series

Features: Two-channel; variable low-cut filters; sweepable high-pass filter; subwoofer function; CD horn EQ; T-Design Constant Gradient cooling; Class H. Price: starts at $899.
Contact: Mackie at 800-258-6883, www.mackie.com.

Dynacord H 5000

Features: Two-channel; internal CPU controller; level controls; LCD screen; Class H; networkable; IRIS-Net module slot; 1,500W @ 8 ohms, 2,500W @ 4 ohms, 3,500W @ 2 ohms per channel. Price: $4,250.
Contact: Dynacord/Telex at 800-392-3497, www.dynacord.com.

Apogee Sound CA Series

Features: Two-channel; level controls; variable speed cooling fan; air filter; 180W – 550W @ 8 ohms, 275W - 940W @ 4 ohms per channel. Price: starts at $780.
Contact: Apogee Sound at 800-443-3979, www.apogeesound.com.

Ashly Audio PE Series Model 3800

Features: Two-channel; limiter; subsonic filter; Ethernet; optional DSP cards; 1,900W @ 4 ohms per channel. Price: $2,970.
Contact: Ashly Audio at 800-828-6308, www.ashly.com.

New BGW Systems GT5600

Features: Two-channel; level controls; Super-H circuit design; soft start; DC, thermal protection; variable speed cooling fan. Price: $2,635.
Contact: New BGW Systems at 323-278-0001, www.bgw.com.

QSC Audio PL 6.0 II

Features: Two-channel; level controls; PowerWave Power Factor Correction technology; Class H; data port for QSControl; variable speed cooling fan; soft start; bridgeable channel pairs; 1,300W @ 8 ohms, 2,200W @ 4 ohms, 3,500W @ 2 ohms per channel. Price: $3,599.
Contact: QSC Audio at 800-854-4079, www.qscaudio.com.

Carvin DCM2000

Features: Two-channel; level controls; soft start; mute; opto-isolator limiter; short circuit, no load, thermal protection; variable speed cooling fan; heat sink; bridgeable; steel chassis; 425W @ 8 ohms, 700W @ 4 ohms, 1,000W @ 2 ohms per channel, 2,000W @ 4 ohms bridged. Price: $519.

Contact: Carvin at 800-854-2235, www.carvin.com.

Camco Vortex 6

Features: Two-channel; digital control; limiter; remote control; 2,300W @ 4 ohms per channel. Price: $4,905.

Contact: Camco/Ashly Audio at 800-828-6308, www.ashly.com.

Inter-M R-500 Plus

Features: Two-channel; level controls; limiter; thermal, short circuit protection; convection cooling; 250 W @ 4 ohms per channel. Price: $492.

Contact: Inter-M Americas at 866-636-5795, www.inter-m.net.

Amplified Design International 600.6

Features: Six-channel; detented level controls; three independent power supplies; twin cooling fans; bridgeable; 400W @ 8 ohms, 600W @ 4 ohms, 750W @ 2 ohms per channel. Price: $3,629.

Contact: Amplified Design International at 770-433-9434, www.abamps.com.

MC2 E45

Features: Two-channel; Power Reduction Circuit; limiters; 2,250W @ 4 ohms per channel. Price: $4,595.

Contact: MC2/Group One at 516-249-1399, www.mc2-audio.co.uk.

Nady Systems SPA-2400

Features: Two-channel; clip limiter; low-cut filter (30 Hz); soft start; variable speed cooling fan; bridgeable; 750W @ 4 ohms, 1,200W @ 2 ohms per channel; 1,500W @ 8 ohms, bridged, 2,400W @ 4 ohms, bridged. Price: $539.

Contact: Nady Systems at 510-652-2411, www.nady.com.

Gemini GXA-1600

Features: Two-channel; two-speed cooling fan; aluminum heat sinks; protection circuitry; bridgeable. Price: $559.

Contact: Gemini at 732-738-9003, www.geminidj.com.

Crown I-Tech 8000

Features: Two-channel; Class I; Power Factor Correction; analog and 24-bit/96 kHz digital inputs; Crown IQ processor/control card slot; Crown IQ-compatible; 2,100 W @ 8 ohms, 4,000 W @ 4 ohms, 3,500 W @ 2 ohms per channel.

Contact: Crown International at 574-294-8200, www.crownaudio.com.

Stewart Audio DA-70-4PFC

Features: Four-channel; level controls; switch mode power supply; convection cooling; 35W @ 8 ohms, 70W @ 4 ohms per channel. Price: $792.

Contact: Stewart Audio at 209-588-8111, www.stewartaudio.com.

Lory Electronics A21

Features: Two-channel; gain trims; FLPT-FET technology; blown fuse indicator; separate headphone driver; automatic standby; VU meters; 140 W @ 8 ohms per channel. Price: $675.

Contact: Lory Electronics at 213-925-0575, www.loryco.com.

Alesis RA500

Features: Two-channel; level controls; convection cooling; bridgeable; 150W @ 8 ohms, 250W @ 4 ohms per channel, 500W @ 8 ohms, bridged. Price: $459.

Contact: Alesis at 401-658-5760, www.alesis.com.

Crate Audio SPA2000

Features: Two-channel; level controls; protection circuitry; twin two-speed cooling fans; 750 W @ 8 ohms, 1,100 W @ 4 ohms per channel, 2,000W @ 8 ohms, bridged. Price: $749.

Contact: Crate Audio at 800-727-4512, www.crateaudio.com.

Yorkville AP6020

Features: Two-channel; limiter; twin cooling fans; 800W @ 8 ohms, 1,300W @ 4 ohms, 2,000W @ 2 ohms; 4,000W @ 4 ohms, bridged. Price: $1,999.

Contact: Yorkville at 716-297-2920, www.yorkville.com.

Best Performance.

There are a number of other tweaks that can slightly enhance performance and/or prevent unintended interruption of audio processes. To get you started, here are the three most rudimentary and benign tweaks:

1. Disable the screen saver: Right click on desktop > Properties > Screen Saver > None.

2. Turn off power schemes: Start > Settings > Control Panel > Power Options > set monitor, hard discs and system standby to "Never" and save as your own preset. While you are there, click on the "Hibernate" tab and make sure it is not enabled.

3. Disable System Sounds: Start > Settings > Control Panel > Sounds & Audio Devices > Sounds Tab > set Sound Scheme to "None."

Next month we'll look at some of the more advanced tweaks for improved performance, reliability and protection from unwanted intrusion from the internet. We'll also look at the state of 64-bit and some of my favorite audio computer peripherals.

PAR Studio Editor Stephen Murphy has over 20 years production and engineering experience, including Grammy-winning and Gold/Platinum credits. His website is www.smurphco.com.

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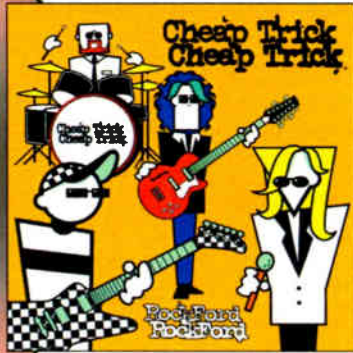
Single: "Perfect Stranger"

Album: *Rockford* (Cheap Trick Unlimited/Big3 Records)

Date Mixed: Spring 2006 at Long View Farm Recording Studio in North Brookfield, Mass.

Single Producer: Linda Perry

Single Engineers: Linda Perry, Jim "Pinky" Beeman, Steve Thompson, and Roger Moutenot at Kung Fu Garden in Los Angeles; Big3 Studio in St. Petersburg, FL; Long View Farm Recording Studio; and Studio 419 in Nashville

Single Mixer: Steve Thompson

Mastering: Ted Jensen at Sterling Sound in New York City

Other Projects: Thompson has worked with more than a few illustrious and varied artists, including Guns 'N Roses, Madonna, Metallica, Blues Traveler, Korn, Blondie, Wu Tang Clan, Simple Plan, Brian McKnight, among many others.

Single Songwriters: Linda Perry, Robin Zander, Rick Nielsen, Tom Petersson and Bun E. Carlos

Mix console: Neve 8068

Recorders: Pro Tools|HD

Monitors: Yamaha NS-10, KRK E8, UREI 813

Processing: Universal Audio Pultec EQP-1A, "various plug-ins," "little" compression or reverb, and "sometimes a nice, tight, ping-pong delay" on lead vocal tracks.

Somewhere in the prayers of reverent, melodic, and pop-dependent rock musicians, engineers, and producers, "Thank you for Cheap Trick" is surely muttered on a regular basis. Those who don't neglect the bestowers of the musical blueprints they regularly depend upon. (If you're a wonderful, well-written, yet defiantly ragged rock musician and you don't understand where I'm coming from, then, I'm sorry, but you clearly don't know the history of the very musical style you embrace.)

Producer, engineer, and mixer Steve Thompson — who fully recognizes the awesomeness of Rockford, Illinois' finest — recently jumped at the opportunity to mix the band's 16th studio release named after its hometown. *Rockford* — an energetic, varied, and expectedly fresh tour of Cheap Trick's many attributes — features a Linda Perry (ex 4 Non-Blondes)-produced and cowritten single entitled "Perfect Stranger," upon which vocalist Robin Zander, guitarist/main writer Rick Nielsen, bassist (and 12-string bass inventor!) Tom Petersson and drummer Bun E. Carlos also share writing credits.

"I was very excited and very honored to have been asked to work with them," Thompson explains. "Cheap Trick has always been hungry, **Steve Thompson** always puts on a great show, writes great songs, and never sat on their laurels — they know what a great song is. I don't even put them into an age group, either. A lot of the bands today were influenced by what Cheap Trick has done."
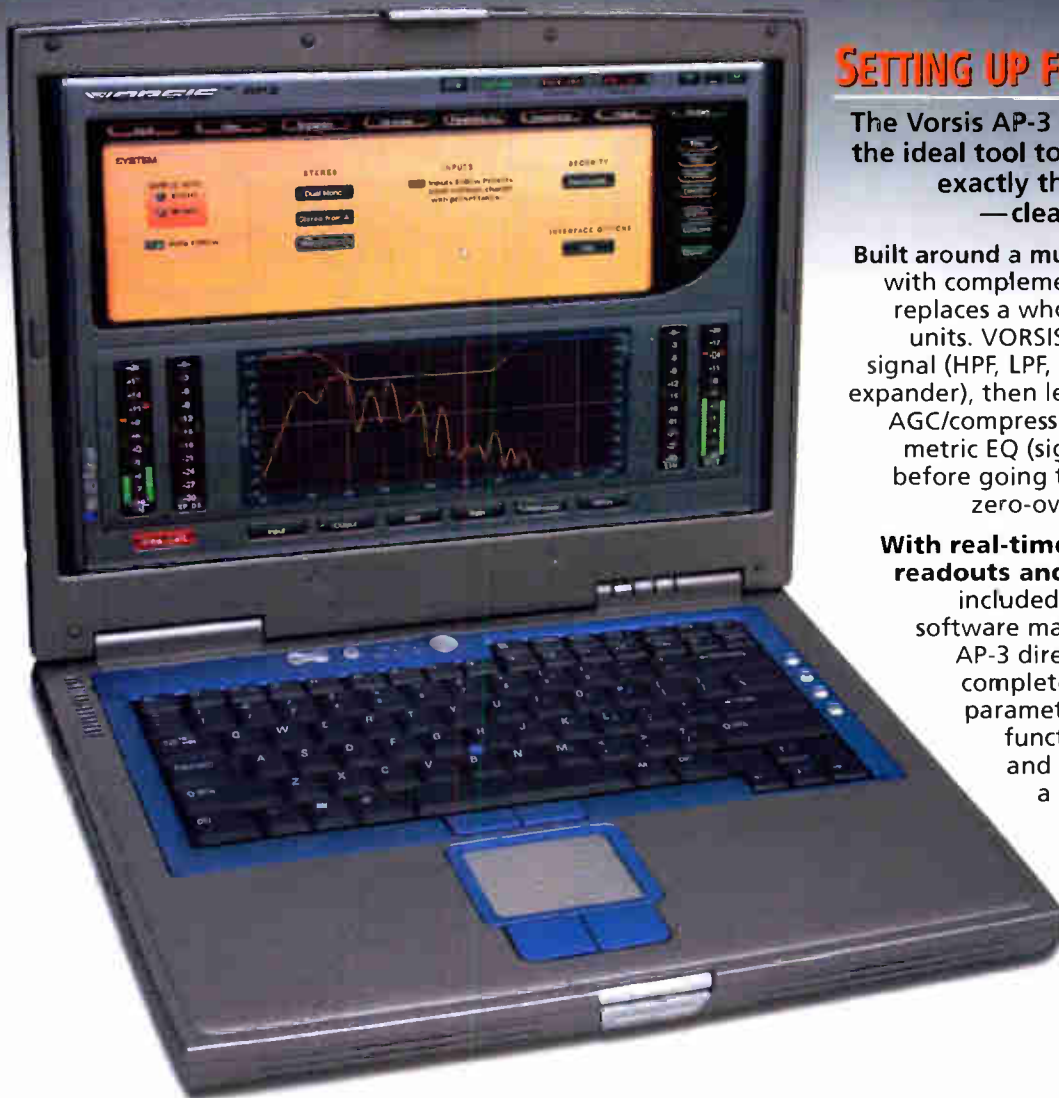
Like the rest of *Rockford*, "Perfect Stranger" came to Thompson as Pro Tools|HD files, and Thompson regularly forwarded completed mixes to the band, who toured on and off throughout the completion of the album. "The whole band was involved in the mixing of it," credits Thompson. Following a few suggested guitar overdubs ('I thought the main rhythm parts could be a bit heftier,' he explains of his suggestion to Nielsen), "Perfect Stranger" was mixed using little board compression and whatever else the mixer deemed worthy and that was within reach of Long View Farm's Neve 8068. "I'll use what I can — Pultecs, modern stuff, plug-ins, or whatever. I'm a guy who has worked on so many different types of projects that I don't like doing one certain thing. I try not to follow a rule."

Strother Bullins is a North Carolina-based freelance writer specializing in the professional audio, music and entertainment industries.

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