Producto Review

The Review Resource for Sound Professionals







■ N ======= J.gruppen, Yamaha Live Sound Amps

Benchmark Media New DAC1 High-End Goes USB!

A-T M50 Headphones Great Accuracy, Small Price!









40 SERIES: UNMASKED

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. Listen in at www.audio-technica.com.



AT4040

Technically Advanced Large Diaphragm

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



AT4033/CL

A Studio Classic

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.

AT4047/SV

Vintage F.E.T. Sound

The rich, warm sound takes you back to the classic F.E.T. studio microphones. But there's nothing remotely old-fashioned about the technology behind that vintage sound.

AT4050

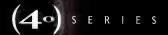
Multi-pattern Condenser

With three switchable polar patterns, the AT4050 exhibits a remarkable combination of qualities: high-SPL capability with transparent and airy uppers/mids complemented by rich low-end qualities.

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.



www.audio-technica.com

(A) audio-technica always listening.

www.yamahaca.com And ready to go. The Tn Series of amplifiers from Yamaha have the power needed to drive your speakers the distance. With a 2 Ohm UL approved rating, these amps are set to get to work using a NEXO system or any other line array. With Yamaha's own EEEngine technology, power consumption is reduced by up to 50%. Additional features include removable handles, easy air-filter maintenance and robust construction. Network control capability wraps up the package and superior sound quality ships it out the door. Yamaha's Tn Series Amplifiers. Sound Intensified. When you need help, time zones shouldn't matter. Yamaha provides coast-to-coast **AHAMAY** 24/7 technical support. With dedicated staff and regional service centers, assistance is around the corner. If we can't fix it over the phone, we'll put a part or a person on the next plane out. It's that simple.

Yamaha Commercial Audio Systems, Inc. • P. O. Box 6600, Buena Park, CA 90620-6600

commercial audio

©2007 Yamaha Commercial Audio Systems, Inc.

INSIDE THIS ISSUE



July 2007 | Vol. 13 Issue 07



Evaluating audio products for professionals in commercial recording, broadcast production, audio for video/film, project studios, live sound, contracting and multimedia.

64

STUDIO

- 8 Live News and New Products
- 10 Benchmark DAC1 USB Converter by John Gatski
 - **BENCH TEST**
- 14 | THE BIG CPU UPGRADE Eight Ways to Make Your In-The-Box Mix Sound Great! by Russ Long
- **18** Moving to Apple Mac Pro by Rob Tavaglione
- 20 Moving to Microsoft Vista by Stephen Murphy
- 22 PC Man to Mac Guy: Buying My MacBook Pro
 by Dan Wothke
- 24 Digital vs. Analog Summing by Alex Oana
- 32 | STUDIO SENSE Lectrosonics and the Lavs of Doom

by Stephen Murphy

- 34 | X/AUDIO
 Bits and Pieces: My New DAW
 by Strother Bullins
- 38 Future Sonics Atrio Series Ear Monitors

by Rob Tavaglione

39 | THE BUYERS SERIES: LIVE SOUND CONSOLES AND SPEAKERS

Cover photos courtesy of Nickelodeon and Apple

POST

- 48 Post News and New Products
- 50 Sound for Animated Feature Films by Mel Lambert
- 54 Polarity Post: Heroes for Higglytown

by Strother Bullins

56 The Wonder Pets Rescue TV Sound

by Frank Beacham

LIVE

- 60 Live News and New Products
- **62** Yamaha Tn Series Amplifier by Dave Rittenhouse
- 64 Lab.gruppen FP+ Series Amplifiers

by Dave Rittenhouse

CONTRACTING

68 Contracting News and New Products

DEPARTMENTS

- 6 | PUBLISHER'S PAGE
 New Owner, Same
 Great Reviews!
 by John Gatski
- 70 | BUYERS GUIDE Live Sound Amplifiers
- 74 | SINGLE SLICE
 Mitch Easter "Sights Set
 On Heaven"

by Strother Bullins

the ART & SCIENCE of MOVING AUDIO

New AV-M8 Mic Input Module

Fast, simple audio conferencing over Cat-5

- · 6 mic inputs plus 2 mic/line inputs
- · ADAT® In and Out for cascading two modules
- · Per-channel gain, phantom power, low-cut filter
- Reversible mounting flanges or rack ears



New AV-P2 Output Module

Audio source selection, press feed, and translation applications

- · Secure set-and-forget channel configuration
- Independently selectable output levels for left and right channels
- Simple plug-and-play installation

AT AVIOM, WE THINK OF AUDIO AS BOTH ART AND SCIENCE.

Aviom's team of world-class engineers has created the kind of groundbreaking technology and robust product design that requires serious hardcore science. But true innovation also requires a measure of "outside-the-box" thinking that can only be described as artistry.

That unique combination produced A-Net®, the world's premier audio transport technology and the power behind Aviom's award-winning family of audio distribution, networking, and digital mixing products. With A-Net, moving audio is fast, easy, and cost-effective.

Aviom solutions have been specifically designed to save contractors time, money, and labor-without jeopardizing the integrity of the audio. Our strength is moving audio-whether it's one channel or hundreds—from point to point to point, or throughout an entire network.

So when moving audio is a key part of the project, choose Aviom to do the job faster and easier, for less.



A-NET THE POWER OF DIGITAL, THE SIMPLICITY OF ANALOG

Connectivity is essential in any systems environment. All Aviom products are activated by A-Net*, the world's premier audio transport technology, designed specifically for the unique demands of streaming audio.



For more information visit www.Aviom.com/AV

FROM THE PUBLISHER

John Gatski

New Owner, Same Great Reviews!

Astute readers may notice a new logo affixed to the front of this month's

> issue: NewBay Media. Pro Audio Review - along with its sibling titles Audio Media, TV Technology, Radio World Broadcast & Production Italy — was purchased by NewBay Media in July. NewBay Media formed

> > last year and purchased the

U.S. division of CMP, includ-

ing Pro Sound News, Systems Contractor News, Guitar Player, Bass Player, EQ and Residential Contractor News, as well as a number of

The new publishing structure is great news for PAR in that it gives the magazine more resources to cover the industry's products, as

other titles

evidenced by the amount of creativity and resources that go into cartoon audio production. In this issue, PAR takes a behind-thescenes look at the audio side of two popular TV animated series, The Wonder Pets and Higglytown Heroes, as well as a macro look at animation audio from several big film studios.

The available digital tools that enable pristine music production also allow the makers of today's hottest cartoons to produce the best sound - whether in stereo or in multichannel. And a few old tried and true audio techniques continue to be implemented. For example, check out Frank Beacham's The Wonder Pets profile, where the classical music and singers are all scored at the same time.

| PRODUCT UPDATE

Those who emailed me about my review of the Trident 8T-16 board, asking about smaller configurations, listen up. John Oram has told



well as sets up logical synergies with Pro Sound News — the industry's best news and information source for many years.

It also means that the staff gets to work again with our old pal Frank Wells, the NewBay pro audio division editorial director and editor of Pro Sound News. Frank got his foot in the pro audio magazine world in 1995 as one of Pro Audio Review's lead bench testers. In 1997, he became the editor of Audio Media U.S., and then joined PSN in 2000. It will be great to work with Frank and all those in the new organization.

| ANIMATED ABOUT AUDIO

As a father of an almost four-year-old, I can tell you that cartoons are pretty serious stuff. The TV and movie studios know that as well, as

me that, spurred by demand, an eight-channel version is ready for those who need fewer channels for their location or studio sound needs. It will contain the same features and be offered with an optional meter bridge. Price is estimated to be under \$2,500, with the meter bridge about \$600. It has the same EQ, direct outputs and channel routing as the 16, and it comes in a rack or tabletop configuration.

We are still waiting to get our hands on a Toft Audio ATB analog board that came out many months ago to see how it stacks up against the competition. I hope to have that review and a lab bench test ready by AES time.

John Gatski is publisher/founding editor of Pro Audio Review.

Pro4udio Review

Vol. 13 Issue 07 | July 2007

Evaluating audio products for professionals in commercial recording, broadcast production, audio for video/film, project studios, live sound, contracting and multimedia.

Publisher & Executive Editor: John Gatski ext. 119 igatski@aol.com Publisher, Pro Sound News: Margaret Sekelsky Frank Wells Managing Editor Tony Ware ext. 126 e-mail: tware@imaspub.com Reviews/Features Editor Strother Bullins e-mail: strotherPAR@earthlink.net Studio Editor Steve Murphy e-mail: smurphy@imaspub.com Technical Consultant Tom Juna

Contributors: Tony Angelini, Bruce Bartlett, Dr. Frederick Bashour, Frank Beacham, Wayne Becker, Carlos Garza, Will James Bascom H. King, Daniel Kumin, Russ Long, Rich Rarey, David Rittenhouse, Andrew Roberts, Alan Silverman, Rob Tavagllone, Mark Ulano, Roger Williams, Tom Young

Production Director Davis White Ad Traffic Manager Lori Behr Publication Manager Michelle Norman Graphic Designer Lourdes Lilly National Sales Manager Matt Rubenstein Ad Coordinator Caroline Freeland ext. 153 Circulation Manager Kwentin Keenan ext. 108

Extensions refer to office number 703-998-7600

CORPORATE -- NewBay Media, LLC

President and CEO Steve Palm Chief Financial Officer Paul Mastronardi Controller Michael Ruber **Marketing Director** Nancy Gregson **Group Circulation Director** Denise Robbins Vice President of Web Development Joe Ferrick

Market Groups **Executive Vice President.**

Carmel King Video/Broadcast Division Vice President, Group Publishing Director

Pro Audio & Systems Division Vice President, Group Publishing Director

Music Player Network John Pledger Vice President, Group Publishing Director

Video/Broadcast Division Publisher, Technology & Learning Jo-Ann McDevitt

Pro Audio Review Founded by Stevan B. Dana



PRO AUDIO REVIEW (ISSN 1083-6241) is published monthly by NewBay Medla LLC, 5827 Columbia Pike, 3rd Floor, Falls Church, Va. 22041. Subsciption information can be found at www.proaudioreview.com, by calling 703-998-7600 or writing to the above address. Letters to the editor are welcomed at the above address

Phone: 703-998-7600 | Fax: 703-998-2966

Periodicals postage paid at Falls Church, VA 22046 and additional mailing offices. POSTMASTER: Send address changes to Pro Audio Review, P.O. Box 1214, Falls Church, VA 22041.

©2007 by NewBay Media LLC. All rights reserved.

REPRINTS: Reprints of all articles in this issue are available. Contact Emmily Wilson (ext. 110) at Pro Audio Review.

Printed in the U.S.A.







Adam Goldstein



TRIDENT IS LISTENING

Thank you for publishing the review of our Trident Series 8T console [PAR April 2007, pg. 16].

Your comments have been well received and improvements made. Firstly, in regards to the instruction manual, we commissioned a professional copywriter who is also a recording engineer and we have the manual included with every mixer, as well as available for your readers if they mail to us here: 8Tmanual@tridentaudio.co.uk.

The remarks about the choice of control pot have been addressed, and although we have kept the "flexi-mount" product we have buffered the side-to-side movement of the control knob.

We also have a metal knob option (avail-

reedback

We want to hear from you. Send your comments to letters@proaudioreview.com. Please include name, city, state and job title and firm in the email. For product submissions, contact Tony Ware at tware@imaspub.com.

able for any 8T console) that is incorporated in a super deluxe model of the console recently built for mega record producer Roy Thomas Baker.

Called the Trident Series 8T-RTB, this product features 32 channels with aluminium (that's how we spell it in the UK!) control knobs, full meterbridge, and the original chipset as used in my original Series 80 design to ensure the EQ is faithfully reproduced.

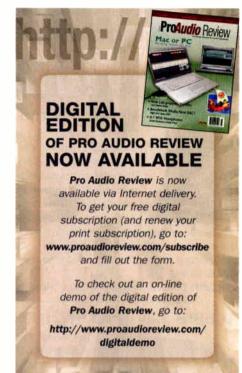
We also have an addition to the range with an eight-channel version (Series 8T-8) that is available in desk mount, like the rest of the range, and in a 19-inch rack mounting option, with meterbridge for both configurations. To complement this model we have designed a sidecar that can be seen at our website at www.tridentaudio.co.uk.

We have units available if you, and your readers, are interested in following our progress with this Series.

Keep up the high standards of audio reviewing.

John Oram

Chairman & Managing Director TRIDENT AUDIO Ltd, U.K.



Get Your Optical Definition



\$649.99 MSRP



STUDIO

The latest news and products

NEW PRODUCTS

TASCAM X-48 Hybrid Hard Disk Workstation



When it comes to memorable recordings, the most talented producers and musicians capture an X factor. Now let TASCAM's latest make it an X-48 factor. The long-incubating X-48 is the world's first standalone 48-track Hybrid Hard Disk Workstation. Combining the ease-of-use of a hard disk recorder with the fluid features of a computer-based digital audio workstation, it's perfect for a multitude of uses, including as music tracking, studio recording and even live recording. And since it records Broadcast WAVE files, you can upload your files to your favourite

DAW for easy editing or mixing. Or do it "in the box," as the X-48 comes with its own editing capabilities, as well as recording at up to 96 kHz/24 bits, a VGA display output, a built-in DVD+RW and a 80 GB hard drive, plus much more. The X-48 will allow your signature to mark any spot.

PRICE: \$4,999

CONTACT: TASCAM | # 323-726-0303 > www.tascam.com

ALESIS Trigger iO Percussion Interface



This should drum up some interest from percussionists: Alesis is now shipping this USB/MIDI, pad trigger interface. Using electronic triggers placed on traditional acoustic percussion sources or electronic drums, this I/O converts them to MIDI note messages. Then USB 1.1 and 10 inputs allow performance options to be expanded to incorporate

computers, samplers, synthesizers, etc. into recording sessions. Dual zones can be set on the inputs, and sensitivity levels/velocity curves/crosstalk/thresholds/re-trigger times manipulated. Up to 20 programmable presets can be stored and recalled, and a foot-switch helps keep things rollin'. BFD Lite software comes bundled, offering professional-quality drum libraries.

PRICE: \$299

TC ELECTRONIC Digital Konnekt x32



No matter the stress of recording, TC Electronic's Digital Konnekt x32 helps you keep your sound

together, offering an all-in-one digital patchbay, format converter and 16x16-channel FireWire audio interface with Jitter Elimination Technology. This unit helps engineers and producers integrate digital outboard effects and keyboards with modern workstations for a cohesive setup that's second-to-none. It's easy with eight channels of conversion between AES/EBU, ADAT (up to 96 kHz SMUX), S/PDIF and TOS, plus Word Clock I/O, and aligned to any sample rate ranging from 44.1 to 192 kHz. It allows sample rate conversion on up to four separate stereo inputs at once. Plus, its Integrator plug-in helps manage effects sends, routing, levels and latency compensation for seamless integration of hardware effects processors. The Digital Konnekt x32 can't do all the work putting together what's in your head, but it can make sure what comes out heads in the right direction.

PRICE: \$1,745

A-DESIGNS 500HR Rack Frame with Power Supply



Do you find yourself tracking and mixing large sounds in limited-space locations? Then the 500HR is perfect for your project studios. Designed

for portability and durability, this single rack space, dual-slot regulated power supply with 48 V phantom power and two balanced XLR inputs will accommodate 500 Series compatible modules. At just six pounds when empty, this steel chassis is great for taking "your sound" across town. And it's versatile – just switch out modules to convert it from a channel strip to a stereo mic preamp, or mix-and-match EQ, etc. The potential uses for this affordable unit are practically limitless, even if your space and funds aren't infinite.

PRICE: \$335

CONTACT: A-Designs Audio | ≈ 818-716-4153 ⊃ www.adesignsaudio.com

You know we at **PAR** are studious about the studio, so here's the latest and greatest on what's making waves, and capturing them. Sound off, one two three four!

EW

z

NEWS

NEWS

NEWS

NEWS

S

NEW

EWS

EWS

EWS

MEWS

EWS

Memphis, Tenn. Indie label Archer Records has invested in a 48-channel API Vision console, chosen to take advantage of all-discrete circuitry and automation (plus the studio's previously owned outboard API processors) when recording live Americana, Jazz, Folk, Rhythm & Blues and Rock in stereo and surround

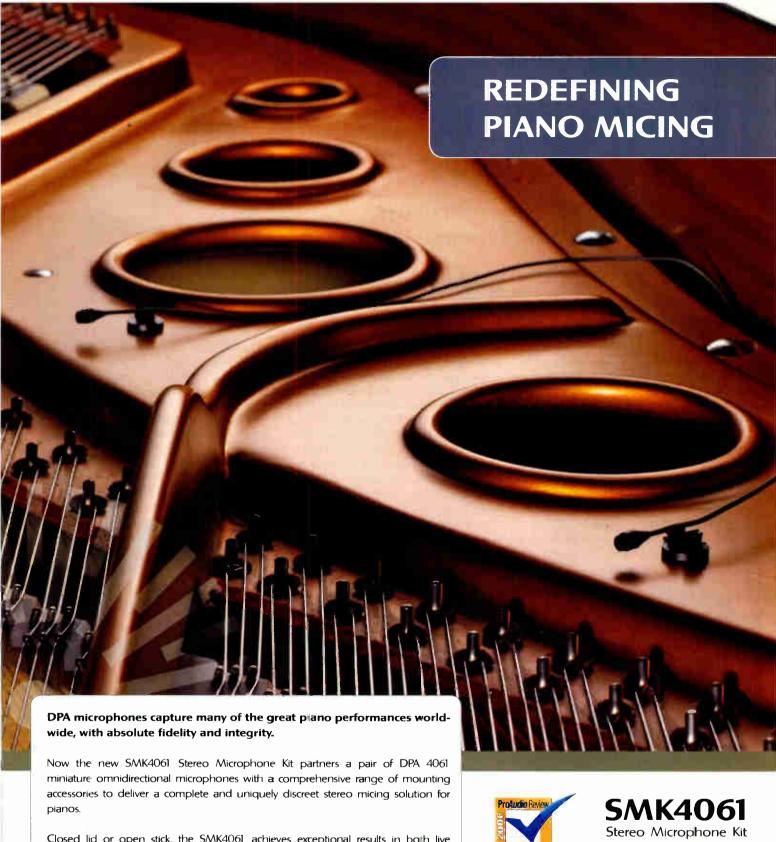
Engineer Michael Haynes opted for **Crowley and Tripp's** Studio Vocalist and Naked Eye ribbon microphones while recording Danya River's "very rich, even low-end and captivating breathiness" at the Hideout Studio in Austin, Tex. A **Sage Electronics** preamp, **API** 5500 two-channel equalizer, **Neve** tape drive processor and Neve Portico compressor complete the chain.

In honor of the 20th anniversary of the stellar Trinity
Session recording, the Cowboy
Junkies reprised the original
location of the Church of the
Holy Trinity in Toronto (pictured)
to recapture the spirit of the
original recording, but for HD
DVD. Making sure none of the
depth of hauntingly memorable
acoustics were lost, almost 50
Shure microphones augmented
a Holophone H2-PRO to flesh
out a mobile studio.



Veteran country, blues and bluegrass engineer Kim Person has acquired a **TRUE Systems** Precision 8 mic pre for recording drums, nylon string guitar, ubright bass and piano at her studio in coastal Virginia, in Nashville and on remote.

| Pro4udio Review | July 2007



Closed lid or open stick, the SMK4061 achieves exceptional results in both live sound and recording applications.









The microphones for sound professionals with uncompromising demands for musical accuracy



STUDIO | REVIEW

by John Gatski

Benchmark DAC1 USB Converter

A top-rate DAC combined with USB is A-OK for virtually any computer.

Benchmark Media introduced the company's under-\$1,000 reference stereo DAC a few years ago, and we threw accolades at it. The master reference sound quality, excellent utility with plenty of I/O options and exemplary measured performance, made this DAC a favorite among discriminating engineers.

Fast forward to 2007. With more and more computers functioning as record/playback, editing and mixing machines, Benchmark has added USB connectivity to the DAC1 and made some subtle component changes to improve its compatibility with outboard gear.

FEATURES

The \$1,275, made-in-USA DAC1 USB is a 24-bit/192-kHz converter that operates at a 110 kHz sample frequency; Any sample rate

BENCH TEST | Page 12

two channels of high-res audio, and offers a connection option that is common to almost any modern computer.

Spec-wise, the Benchmark offers the same excellent, accuracy, signal-to-noise and jitter reduction performance of the original DAC1. [Check out Bascom King's bench test. – Ed.] The analog output line output has a bit more oomph for driving difficult preamps than the original. And the headphone outputs feature a 10 dB gain reduction jumper that restructures the circuit to better suit high-sensitivy headphones; one output also features a deaftable mute-on-insert function to simplify switching between headphone/loudspeaker monitoring.



audio played through the DAC1 is either downsampled (from 176 kHz or 192 kHz) or upsampled (from 32 kHz through 96 kHz). Like the non-USB DAC1, the USB brethren offers the outstanding HPA2 headphone circuit, AES/EBU (XLR), TOSlink and coaxial SPDIF (BNC) inputs, as well as fixed and variable controlled RCA and XLR analog outputs.

The big difference, of course, is the addition of USB input to allow flexible routing for computer audio use. According to Benchmark, the designers settled on USB 1.1 for two-channel 24-bit/96-kHz computer compatibility because it requires no special drivers to use with the various software programs. The 12 Mb/s USB 1.1 conduit has plenty of bandwidth and data transfer rate for

IN USE

I first listened to the DAC1 with a variety of sources, including the TASCAM DV-RA1000 an Alesis Masterlink and my G5 Apple desktop system via AES/EBU output. With 96 kHz and 44.1 kHz sources it sounded like the original DAC1, though I thought it sounded a bit more open when listening with the AKG K 701 headphones. The newer version sounded slightly more present than the original. Maybe my original DAC1 needs a tune-up.

I also connected the the DAC1 USB into my reference listening system with Pass X350.5, Coda preamp and the DV-RA1000 as the source. The Benchmark characteristics of very accurate playback, a great sense of space and detail of well-recorded PCM audio was obvious. Drum cymbals, guitar and other intense transients were in abundance.

After several days of listening via the AES/EBU to get an audio reference, I then connected a MacBook Pro to the DAC1 via the laptop's USB and optical digital output in order to switch back forth between the two conduits. I heard no difference in the audio quality and no disruptions, spitting or sputtering. The USB stream worked flawlessly.

I did discover that you need to pay close attention to the internal sample rate settings of the Apple computer; you have to make sure that when you play 96 kHz audio, the Audio/MIDI system set up is set to 96 kHz. The Apple system sample rate does not always follow the source audio. I played a Toast-authored DVD Video disc with 24-bit/96-kHz stereo guitar tracks, and monitored via the DAC1.

BENCHMARK continues on page 13 ➤

| TECHNICAL INSIGHT: BENCHMARK DAC1 USB

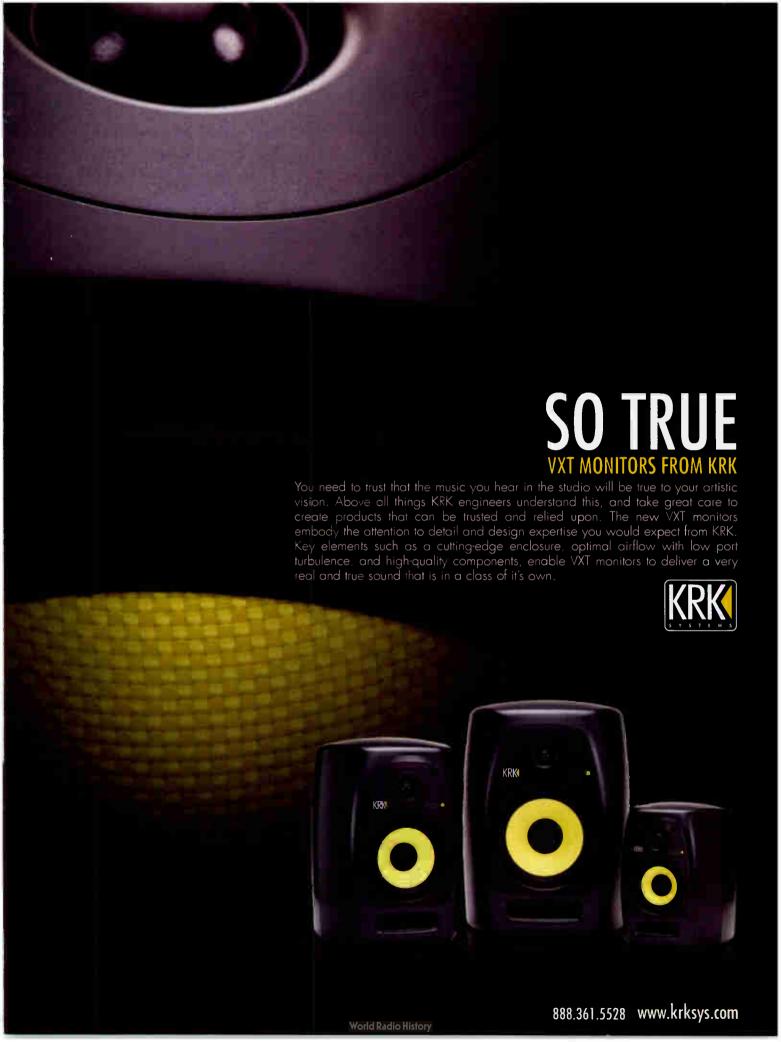
According to Benchmark Media Systems Designer John Sau, the DAC1 USB uses an AKM4114 AES digital audio receiver that was selected for its ability to accurately recover data in adverse conditions that include high-jitter, high-noise and low digital signal levels. The coaxial, XLR, and optical digital inputs operate at sample rates up to 192 kHz. The DAC1 USB inputs at sample rates up to 96 kHz and 24 bits are handled by a TAS1020B running Benchmark's native USB firmware.

The DAC1 USB does not use the AK4114 or the TAS1020B for conversion clock recovery. The interface clocks are isolated from the D/A conversion clock.

At the core of the DAC1 USB, an AD1853 D/A converter operates at a fixed frequency derived from an ultra low-jitter crystal oscillator. This D/A circuit was selected because of its low distortion. The transition band of the digital filter in the AD1853 is frequency-shifted by upsampling to 110.633 kHz. The upsampling process eliminates alias effects caused by near-Nyquist audio content.

The upsampling is asynchronous and is accomplished with an AD1896. This device was chosen for its low spurious tones, low distortion, and exceptionally low PLL corner frequency. The PLL has a corner frequency of 2 Hz and provides 100 dB of jitter attenuation at 1 kHz.

- John Gatski



BENCH TEST

Benchmark DAC1 USB D/A Converter

BENCH MEASUREMENT	DATA	22 Khz BW	113.0 dB	
		A weighte	d 115.5 dB	
MAIN OUTPUT LEVELS @ 0 dBFS		96.0, 192.0 kHz F	s	
Output switch in calibrated position		Wideband	89.0 dB	
Unbalanced outputs	2.001 V	22 Khz BW	113.dB	
Balanced outputs	1.269 V	A weighte	d 115.5 dB	
Output switch in variable position, front panel control at maximum, Lch/Rch		USB input		
paner control at maximum	i, Leil/ Reil	44.1 kHz 16 bit		
Unbalanced outputs	2.917/2.818 V	Wideband	91.0 dB	
Balanced outputs	1.850/1.788 V	22 Khz BW	113.0 dB	
		A weighte	d 115.0 dB	
HEADPHONE OUTPUT LEVEL @ 0 DBFS		44.1 kHz Fs		
Front panel level at maximum		Wideband	91.0 dB	
High impedance loading Lch/Rch		22 Khz BW	112.0 dB	
	2.912/2.814 V	A weighte	d 112.0 dB	
Both channels loaded	with 50 ohm	96.0, 192.0 kHz F	s	
Lch/Rch		Wideband	88.0 dB	
2.824/2.746 V, 159/150 mW		22 Khz BW	113.0 dB	
		A weighte	d 115.0 dB	
OUTPUT IMPEDANCE				
Line Outputs		DYNAMIC RANGE		
Unbalanced outputs	30.5 ohm	44.1, 96.0, 192.0	kHz 116.0 dB	
Balanced outputs	133 ohm	USB input		
Headphone outputs	0.16 ohm	44.1, 96.0, 192.0	kHz 116.0 dB	
FREQUENCY RESPONSE		QUANTIZATION NO	NEE	
44.1 kHz Fs		-		
	20 Hz 24 4 kHz	44.1, 96.0, 192.0	kHz 111.0 dB	
+0.0, -3.0 dB 20 Hz – 21.1 kHz		CHANNEL SEPARA	TION	
96.0 kHz Fs		44.1, 96.0, 192.0 kHz Fs		
+0.0, -3.0 dB 20 Hz – 45.6 kHz 192.0 kHz Fs		Ch 1 > Cl		
+0.0, -3.0 dB 20 Hz – 51.3 kHz		011 7 01	120 dB 20 Hz – 1.6 kHz	
10.0, 0.0 ub	20112 01.0 KHZ		100 dB @ 20.0 kHz	
TOTAL HARMONIC DISTORTION		Ch 2 > Cl		
22 kHz measurement filter			110 dB 20 Hz – 1.8 kHz	
44.1, 96.0, 192.0 kHz Fs			90 dB @ 20 kHz	
< 0.001%, 20 Hz – 20 kHz		USB Input @ 10 k	Hz	

Note: Measurements were made at 44.1, 96.0 and 192.0 kHz sample frequencies and with bit depth of 24 bits triangular dithered at the unbalanced outputs unless otherwise noted. Digital input was via AES/EBU input unless otherwise noted. The adjustable output levels were set for 2V at the unbalanced outputs.

104.5 dB

95.3 dB

+/-

LINEARITY ERROR

96.0, 192.0 kHz Fs

44.1, 96.0, 192.0 kHz Fs

80 kHz measurement filter

+/- 1.5 dB 0 to -120 dBFS < +10 dB @ -140 dBFS

< 0.0014%, 20 Hz – 20 kHz

< 0.003%, 20 Hz - 38 kHz

SIGNAL TO NOISE RATIO

44.1 kHz 16 bit	
Wideband	88.0 dB
22 Khz BW	93.8 dB
A weighted	95.8 dB
44.1 kHz Fs	
Wideband	89.0 dB

44.1, 96.0, 192.0 kHz Fs

BENCH MEASUREMENT DATA

44.1, 96.0, 192.0 kHz Fs

Ch 2 > Ch 1

Ch 1 > Ch 2

The Benchmark DAC1 USB is one of the latest entries in the new category of audio D/A converters that accept USB input from a

computer. Testing the behavior of a D/A converter using its USB input is not simple at this point. Currently, I don't know of a device that converts AES/EBU or SP/DIF data to USB in real time, which would permit straightforward real time testing of most parameters of interest. One has to record test wave files on some recording device and then play them back via a computer USB output into the D/A converter under test. Audio Precision has a handy utility, WfmGeneration, for generating typical test files on the HDD on one's AP host computer. Now the problem is to play them back out of the USB output of the computer with a suitable software program that is bit transparent.

Fortunately, the Foobar 2000 program does this nicely. Some other programs tried either added distortion and/or noise when played back via USB into the DAC1 USB (the distortion is present with other USB devices and non-USB sound cards as well, not just the DAC1 USB; distortion is due to poor volume controls or sample-rate/bit-depth handling in the media player software) compared to the digital test signal input from the Audio Precision. Benchmark has found that the Tascam HD-P2 solid-state recorder is bit transparent in the digital domain. I was able to borrow theirs for a short while to make up some of my own files, which were then transferred by FireWire to my AP (Audio Precision) computer and then played back with Foobar.

Frequency response for the sampling frequencies of 44.1, 96.0 and 192 kHz is shown plotted in **Figure 1**. With frequencies

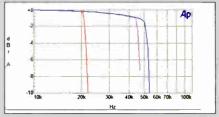


Fig. 1 Frequency response at 44.1,48.0, & 96.0 kHz Fs. Red = 44.1 kHz, Magenta = 96.0 kHz, and Blue = 192.0 kHz. Lch shown

extended down to 10 Hz, there was no rolloff at the low-frequency end. One major surprise in the frequency response measurements was the reduced high frequency bandwidth at the 192 kHz sample frequencies. Checking with Benchmark on this revealed

BENCHTEST continues on page 30 ➤

STUDIO

REVIEW

BENCHMARK Continued From Page 10

PRODUCTPOINTS



- · Glitch-free USB 1 audio input
- · Very accurate sounding PCM converter
- · Audiophile-quality headphone amp
- · Balanced and unbalanced outputs



- No power switch
- · No bit/word status display

SCORE

One of the best sounding DACs available gets a computer interface

It did not sound high-res, so having been tipped off about the Apple audio sample rate setting not matching the source, I opened the Audio/MIDI Setup menu and the system was sending out 44.1 kHz internal downsampled audio. I manually switched to 96 kHz, and the

guitar track DVD-V sounded much better.

The Benchmark's USB DAC actually creates digital output redundancy for Apple computers since the high-end Macs already allow digital input and output via optical at up to 96 kHz. Many Windows machines, however, don't have any built in digital audio I/O. Thus, the Benchmark USB DAC1 is a perfect mate for those computers with no other digital connection. And its sound quality is better than most of Firewire or USB interface box converters that I have heard.

I have only minor quibbles with the DAC1. It still does not have a power switch (The Mytek has a nice rocker switch mounted on the front). [Benchmark responds: Automatic power management turns off portions of internal circuitry/the LED display with no digital input present, and resumes, making a power switch unnessecary.] And no small high-end converters I have evaluated have any incoming status display of word length and sample rate. On the plus side, the manual is as good as there

is, with detailed, informative and easy-toread info and factory benchtest measurements. The manual also contains some interesting Apple vs. PC computer observations relating to implementation of digital audio.

SUMMARY

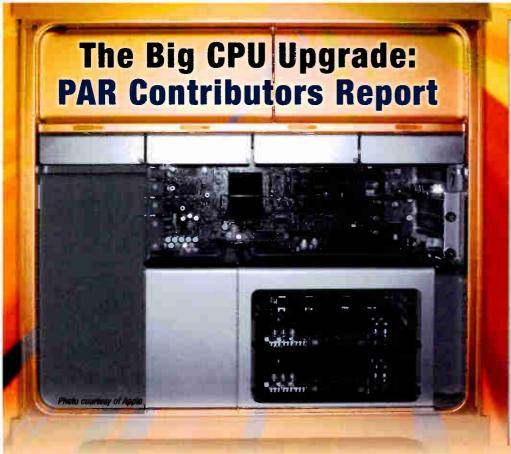
Benchmark has taken the top-performing, low-jitter DAC1 and given it a USB connection that makes it compatible with virtually any computer. For simple two-track monitoring, the DAC1 USB also eliminates the need for extra sound cards and breakout boxes — USB or Firewire. The feature adds quite a bit to the price, but you are getting a DAC and a major Class A headphone amp in one chassis — with one more input option.

John Gatski is publisher and founding editor of Pro Audio Review.

| EVALUATION SETUP

Apple MacBook Pro 17-inch, Benchmark DAC1, TASCAM DVRA-1000 DVD recorder, Alesis Masterlink, Pass Labs X350.5 power amplifier, Legacy Codas preamplifier, AKG K 701 headphones





Not too long ago, the overwhelming majority of audio engineers considered a tactile mixing desk as the "heart" of their workspace. Digital or analog, the console was their studio centerpiece: the creative nexus where source audio converged and evolved before its final transformation to a stereo or multichannel mix.

Although not exactly obsolete, the physical mixer now shares its spotlight with (and has increasingly been replaced by) the digital audio workstation, with its own internal mixer, EQ, effects, routing and editing capabilities, musical performance features and, of course, a high-resolution multitrack digital recorder - all at a comparatively low, low price point. Simply stated, the heart of the modern music production studio is essentially a "home" computer.

With that in mind for this special feature section, Pro Audio Review details a variety of major issues that affect the modern, DAW-dependent pro studio engineer: purchasing and upgrading a CPU for studio use, adapting to the latest Mac and PC hardware and OS updates, and maximizing the ever-growing potential of ITB (in-thebox) music production, while weighing the pros and cons of doing so. Here, your guides are the best in the business: PAR Contributors who - just like you - find a CPU as the heart of their modern music production rig.

- Strother Bullins

Eight Ways to Make Your In-The-Box Mix **Sound Great!**

by Russ Long

have been routinely mixing in the box for several years now, and I love it. As with an analog mix, there are a lot of ways you can screw up an in-the-box mix. However, if you do things right you can end up with something that sounds fantastic. Some of the following ITB mix tips are directed more towards recording than mixing, which emphasizes the importance that a recording makes in attaining a great mix.

TIP 1: DO SOME GREAT ANALOG PROCESSING BEFORE YOU CON-**VERT TO DIGITAL**

As close as so many plug-ins come to the real things, they still aren't; and they typically don't sound as good as the real things, either. Once basic tracking has been completed, most people record one instrument at a time. Start with a great mic and pre and then use a real Neve, GML, Pultec or Daking EQ, for instance, when you are recording. Compliment it with a great sounding compressor like the Tube-Tech CL-1B, Teletronics LA-2A or the Empirical Labs Distressor. Just don't wait to slap on a modeled EQ and compressor plug-in during mix.

TIP 2: USE GREAT CONVERTERS

Unfortunately, cheap converters are generally cheap for a good reason. I've had great luck tracking to Pro Tools through Apogee AD16X, Lynx Aurora,

TIP 3: PRINT YOUR ANALOG INSERTS

One of the strengths of mixing in the box is the ability to instantly recall your mix. If you end up using some analog processing on a track during a mix, print it to another track. This keeps you from having to recall the analog device during a recall.

TIP 4: USE A GREAT ANALOG STEREO-BUS INSERT

I find that using a great analog signal path on my stereo buss makes my in-thebox mixes resemble the work I do on an



Pendulum 6386

iZ RADAR and Prism Sound ADA-8XR converters. During the mix, better converters allow you to hear more resolution and detail, which in turn improves your mix. When it comes time to print, if you are mixing to another medium (e.g. analog tape), good converters again improve your mix.

analog console. I use three analog pieces in this insert: the Universal Audio 2-610, the Pendulum 6386 and the Pendulum PL2. I patch line-in on the 2-610 and use a slight amount of EQ (usually a 1.5 dB boost at 10 kHz and a 1.5 dB boost at 70 Hz). I add this EQ before I start to add

MIX TIPS continues on page 16 ➤



USB.006



It's that easy to get stal ted with the MXL USB.006. No menu's, no drivers to install, Just plug it in, set your preferences and you're recording! Ease of

operation is just the beginning. Inside this great looking body you get a true Gold Diaphragm Studio Condenser Microphone - not a cheap dynamic or noisy electret. Its got real analog volume adjustments to ensure precise gain staging and sounds just like a studio mic should. The MXL.006 comes with a 10' USB cable, a table stand, a travel case and is MAC and PC compatible.

Audition one today at most fine music retailers.

(800) 800-6608 / www.mxl-usb.com





plug-ins, because I find that it makes me use less equalization on individual tracks (see tip 5).

Don't forget: If you are mixing in Pro Tools, you want your stereo buss insert to actually be on a stereo aux rather than on the stereo buss. The stereo buss inserts are post-fader, so the dynamic processing will change as you begin to fade or make any other stereo buss adjustments. I like to assign everything to a stereo aux and then assign the stereo aux to the stereo buss.

TIP 5: DON'T OVERUSE PLUG-INS

Plug-ins are great, but they can start to have a cumulative negative effect when used in great number. The mythical "Pro Tools sound" that so many people like to talk about hating is really about people mixing while using way too many plugins and completely screwing up their gain-structure. This point leads us to tip 6 ...

TIP 6: GET YOUR GAIN STRUCTURE RIGHT

Try to keep everything running as close to unity as possible. Red "over" indicators within the application indicate that something needs to change. Distortion is often pretty in the analog world, but never in the digital realm.



Empirical Labs Distressor

TIP 7: TAKE ADVANTAGE OF BEING ABLE TO TWEAK OVER A LONG-PERIOD OF TIME

When I mix an entire album in the box I like to schedule a few days where the artist, the producer and myself can all live with the mixes before coming back to the studio and finalizing the project. Often the changes we end up with are somewhat minor (i.e., things that we could never justify doing a recall for if we were mixing analog, but they ultimately make it a better album). When working on an analog console, it's difficult to do more than two recalls a day, but when mixing in-the-box it's possible to make tweaks

on every song on an entire

album in a day.

TIP 8: RECORD AT THE RIGHT SAMPLE-RATE

If you're mixing in the box and your project is going to end up on a CD (44.1 kHz) you should record at 44.1, 88.2 or 176.4. The conversion from 48 kHz, 96 kHz or 192 kHz to 44.1 kHz isn't pretty and will ultimately have a negative effect on the sound of your project. Yes, I know, 48 kHz does sound slightly better than 44.1 but it's not worth the trade-off of the ugly conversion.

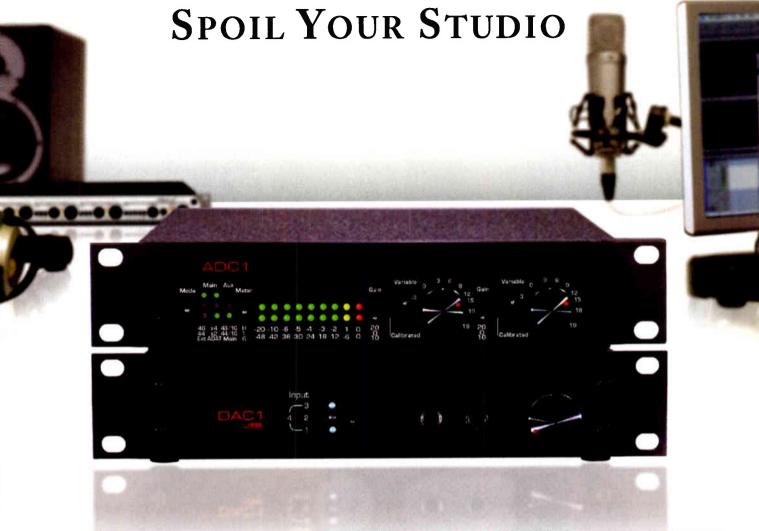
SUMMARY

Hopefully this has provided some digital fodder for thought for all of the in-the-box mixers out there. Keep it real and happy mixing!

Russ Long has done 5.1 DVD mixes for Allison Moorer and Mercy Me and is an indemand engineer for live sound recordings. He has multitracked live performances for Chris Tomlin, Lisa Loeb, Salvador and Nichole Nordeman.

Lynx Aurora 16

INDULGE YOUR EARS



After investing time and money into your gear, room, and setup, you expect your tracks and mixes to shine through with all of their potential tonal brilliance and clarity. Don't let inferior digital audio converters limit your results - your studio deserves the best!

Inferior A-to-D converters mask the musically harmonic colors you work so hard to create, and your audio content is lost - forever. Inadequate D-to-A conversion impairs your listening ability by adding artificial overtones and distortion. This sabotages your mixes because they won't translate from the control room, even with high quality monitors.

Benchmark's exclusive UltraLock™ technology frees your tracks from non-musical, jitter-induced distortion to reproduce your sounds transparently and faithfully. The award-winning DAC1 and ADC1 converters are world-renown for solving the devastating effects of jitter and distortion. We're proud of the many testimonies from successful engineers who credit the DAC1 and ADC1 as essential equipment in their studios.

Learn how your recordings can finally reach their full potential.

Call us today: 800-262-4675, or visit our website: www.benchmarkmedia.com/par



INDULGE YOUR STUDIO WITH THESE SPECIFICATIONS

ADC1

2-Channel 24-bit 192-kHz A/D Converter

- Ultra-low distortion (0.0003%) for transparent recording
- Wide Dynamic Range > 120 dB A-Weighted
- UltraLock™ for jitter-free slave operation
- Multiple outputs and configurations

DAC1 USB

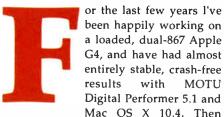
2-Channel 24-bit 192-kHz D/A Converter

- Breakthrough USB technology for transparent & driverless 96/24 playback
- No drivers, No configuration, No kidding! Windows Vista/XP/2000 & Mac OS X
- HPA2™ high-current, 0-Ohm headphone amplifier
- UltraLock™ for jitter-free playback

Sevenuark ...the measure of excellence!

Moving to **Apple Mac Pro**

by Rob Tavaglione



along comes the new Intel-powered Mac Pro, which sounds enticing, but I am in no

MOTU

position to sacrifice stability or needed peripherals with a busy music studio that has to stay productive.

While faster speed, more plug-ins and higher track counts are worthy benefits, the process (and cost) of switching to the new Mac is not to be taken lightly. Even so, I finally took the plunge, though with careful consideration.

I will share insight on my move to Mac Pro in the text below, what problems arose during the process, and what results can be expected. While this is not intended to be a Mac Pro review, per se, let us nonetheless take a closer look at the machine behind all the fuss.

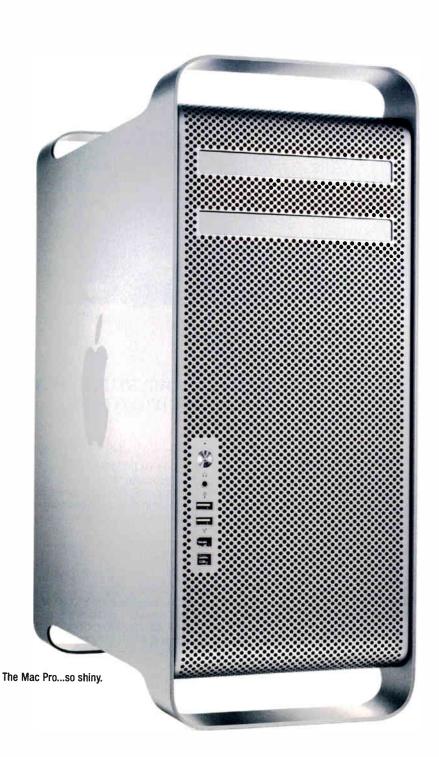
OUT OF THE BOX

I bought the Mac Pro with dual 2.0 GHz processors, 1 GB of RAM, a 160 GB hard drive and the DVD-RW drive (compatible with dual layer discs). Out of the box, this computer is visually striking with its all brushed aluminum and gray plastic look - just like the previous G5. I opened the Pro to find an audio engineer's delight: an extremely sleek interior, a logical layout and no visible wires!

I immediately found it interesting that, even with all the changes inside the box, my experience as an audio user remained nearly unchanged.

There are now two PC boards for mounting RAM (667 MHz, PC-2 5300, DDR2, ECC DIMM memory, fully buffered), each with four slots, each board neatly sliding out and requiring no tools for memory installation. This arrangement allows for a total of 16 GB of RAM, an obscenely high amount that will greatly benefit video editors and those who can afford the \$2,500 (street) price tag.

Inside, storage abilities are nicely implemented with slots for four SATA drives allowing up to four terabytes of storage. Whereas the G4 used ATA drives, the G5 and Mac Pro use SATA (Serial ATA) drives that have different mounting holes, pin configurations and are a little smaller. The drives are housed behind brushed aluminum, side-by-





The Mac Pro's dual PC boards for mounting RAM

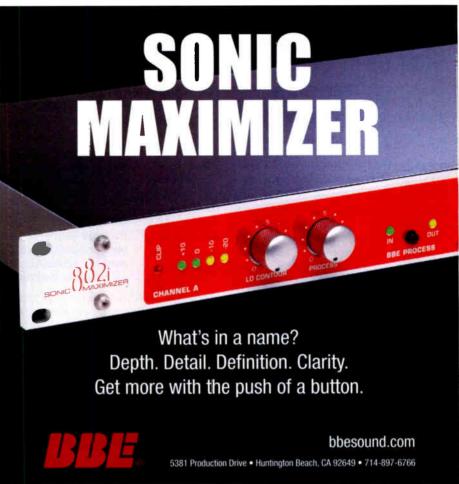
side, in a setup that is more ergonomic and much easier to install than any previous Mac model. You'll definitely want to add an additional drive or two, and will be pleased to find them no more expensive than the already affordable ATA drives.

UP AND RUNNING

Mac Pro ships with OSX 10.4.8, which showed no major operational differences when compared to my previous 10.4 OS including Core Audio and the Audio Units plug-in protocol. The biggest change for me was the new and advanced mouse with its scroll wheel and new zoom features. I immediately found it interesting that, even with all the changes inside the box, my experience as an audio user remained nearly unchanged; such continuity has always been a strong point of Apple's OS progressions.

As I installed disc-burning software (Toast 8, a Mac standard that now includes access to all your Audio Unit plug-ins) I realized a big difference compared to my G4: much quieter operation. Although I use an iso-box for housing my noisy hardware, this newfound quiet will be a welcome change for many desktop users.

I installed a second 160 GB hard drive (stock drive for apps, secondary for audio files) that needed formatting. Using Disc Utility, I partitioned, formatted and initialized the drive. As I transferred a session to this new drive I enjoyed a significantly faster process: only one minute for a 4 GB transfer and three minutes for 10 GB. I then brought the rest of my workflow with me on



a FireWire drive, which was quick and effective although I had been given the option to network my two Macs for direct transfers when I first booted up the Mac Pro.

The new RAM cards are very easy to work with and require no tools; I installed an additional gigabyte of RAM in mere seconds. Or maybe should I say "attempted to" as the machine's hardware profile did not recognize the additional memory. Some deeper digging in the manual revealed that added RAM must be in pairs. Adding a second one GB module did the trick at about \$300 street for 3 GBs total. I have noticed faster operation, particularly when pushing the limit with rapid editing, enjoying faster drawing of fades and edge trimming. My DAW includes a powerful pitch corrector; it loves this new environment, calculating new pitch curves nearly instantly and crunching this heavy math much quicker than before.

Track counts are somewhat higher, but exact improvements are hard to quantify. Previously on the G4 it was not unusual for a session with 40 tracks or more (at 48 kHz) to get jumpy, unresponsive and buggy, sometimes even crashing. Now a similar session, comparably loaded with many plug-ins

MAC PRO continues on page 28 ➤



www.proaudioreview.com

July 2007 | Produdio Review | 19

Moving to Microsoft Vista

by Stephen Murphy

t has been a little over six months since the official release of Microsoft's longawaited Vista operating system - time enough to gain perspective from the professional audio standpoint. Since I have amassed a book-load of notes on Vista and have very limited space for this survey, I'm going to jump right in and zip

through the major points without much ado. To quote one of my favorite lines from the film Repo Man: "Since time is short and you may lie, I'm going to have to torture you. But I want you to know, it isn't personal."

THE BIG PICTURE

As with previous major OS releases from Microsoft - and any good Hitchcock movie - the Vista story arc is full of suspense, surprise and a healthy dose of plot twists. On the subject of pro audio and Vista, the story has several satisfying resolutions, but it is also rife with cliffhangers and far too many pages are downright missing. Also like previous Windows versions, the Vista release has resulted in a prodigious amount of technical and anecdotal information, plenty of healthy discussions and, as always, lots of chaff from rumor mill - much of which has thankfully been put to rest at this point.

Now that the dust has somewhat settled, a prevailing opinion - which I share - has

emerged: Vista provides a good number of worthwhile general-use improvements and new features, falters in a few others, and ultimately doesn't quite live up to expectations for a major release. Microsoft does deserve credit

Search T

for its proactive dialog with the pro audio community of Vista's approach audio handling, which will in turn benefit all multimedia applications.

For the general public, switching to Vista is generally a smooth experience, with few attention-grabbing impediments. The most immediately noticeable aspects of Vista are its new translucent "Aero Glass" comprehensive visual design, and its infamous User Account Control warnings (which pop up with annoy-

ing frequency but serve the higher purpose of

protecting your installation).

The Aero Glass design is a significant aesthetic improvement over the XP GUI, and the new Desktop Composition display engine (which makes Aero Glass possible) provides a number of other welcome GUI enhancements, such as the hover-over active thumbnail renderings of objects minimized to the task bar and objects in the Alt-Tab switching dialog. Display performance, in theory, is improved through Windows' management of per-application display rendering and scaling via off-screen display buffers and offloading of an increased percentage of rendering duties

Call 1498 Cali 1500 Jenny's Cale III Month Calenda The update-as-you-type Start Bar search and Instant Search box are excellent. Unfortunately, they can't help you find non-existent Vista drivers...

to the graphic display card.

I put "in theory" in the above sentence because pro audio users will want to disable many of the animations and effects (as with XP), and not-insignificant numbers of users have experienced reduced graphic performance or, worse, reduced performance of an application as a result of graphic processing. However, certain hardware configurations and respective driver implementations are probably to blame in most of these cases. As always, troll your DAW's forums and FAQs for known working configurations.

THE LONG AND SHORT **OF VISTA AUDIO**

For audio professionals, the most significant aspects of Vista's new approach to audio are its low-latency Wave RT (Real Time) audio driver, per-application volume control (instead of XP's global control) and an impressive amount of user control of application process prioritization via the Multimedia Class Scheduler Service (which can finally elevate audio streaming beyond "Background Services").

There are a number of other audio features and "effects" found in the Sounds section of Vista's control panel that many home/office users might find helpful. Included here are Microsoft's take on speaker protection (highpass filter), inter-application loudness control and even a form of RTA speaker equalization using a microphone and test signals - all of which should be disabled by pro audio users.

The factor having the biggest bearing on the use of Vista for pro audio applications is still, six months after its introduction, the haphazard availability of compatible software and hardware drivers.

Here's the over-generalized skinny: most pro audio developers have announced intended Vista compatibility (seems a no-brainer, but there are still a few that haven't officially announced), more than half of audio hardware manufacturers have released Vista driv-



Unlike this picture from the MS website, live thumbnails on the task bar is most useful when you want to see what's happening on a window that's not on the top; truly great for progress bars etc.

ers (some of which are still "Beta" and a good percentage do not offer 64-bit drivers), and a much smaller number of software developers

have released Vista 32/64-bit versions.

As this list is changing with somewhat encouraging frequency, I'm not going to attempt to provide a list that will be outdated by the time

you read this. I did want to praise RME and MOTU on the hardware side and Cakewalk on the software side for their early and fullon adoption of Vista across their product lines. The two major copy-

protection ven-



Some of the spiffy new audio features...that DAW users should avoid.

dors, Syncrosoft and Pace, are also up to speed, with Syncrosoft as the early adopter.

FUTURE VIEW

If you are considering the Vista upgrade here are some important points to keep in mind:

Version: There is a confusing array of versions and sub-versions of Vista. For most pro audio users, the two to consider are full-install Home Premium and Ultimate. Note that the Vista upgrade requires the removal of XP from the system — a major change in policy that, given the state of compatibility, is simply unacceptable. That said, I cannot confirm or deny the existence of a workaround floating around the web ...

Audio Components: Keep in mind that, even if your major components are Vista-compatible, many ancillary objects such as plug-ins, authoring applications, codecs, macros, etc. may not be. Do a comprehensive survey, preferably over several days or weeks of pro use, and write down everything you use and check with the manufacturers' web sites for compatibility info.

Don't forget about the hardware in your computer, including graphics cards, additional FireWire or USB cards, audio DSP cards (for example, at this date UAD is compatible, PowerCore is not), high-end video I/O, etc.

Performance: This is one heated subject on audio web forums. From my own experience (wholly unscientific, since much of what I use in XP is not available in Vista), Vista performed on par with, if not better than, XP in all

BBE Sonic Maximizer and 3-way Crossover...
Two processors, one great unit.

bbesound.com

5381 Production Drive • Huntington Beach, CA 92649 • 714-897-6766

cases. Others have reported the opposite. It seems that certain configurations and CPU loads will yield widely varying results. Again, check the app and hardware forums for known good setups. There are some true comparisons of performance on the Internet, with my favorite being www.rainrecording.com/vista



Vista's Flip3D desktop and an example of sidebar gadgets — très cool.

- definitely worth checking out.

In addition to manufacturers' websites and the one mentioned above, here are a few more reference sites that cover the state of Vista compatibility and other useful reference information: www.iexbeta.com (comprehensive hardware and software compatibility lists, plus blogging on a number of related subjects); www.dawbench.com has excellent benchmark test results; also check out Sweetwater's Vista section in the forums for some ad hoc compatibility lists and comments.

It's six months since the release of Vista and unfortunately we're still playing a game of "Chicken or Egg." So, what is the "way forward" for audio professionals, you ask? Well, how soon you can upgrade is contingent on your specific configuration and respective compatibility. If, like me, you have a broad variety of essential A/V apps, plug-ins and hardware, you're going to have to wait or try out a dual-boot config to get your feet wet. In general, Vista will prove to be desirable if for no other reason than the eventual elimination of XP support.

Stay tuned to my Studio Sense column in *Pro Audio Review* for ongoing Vista coverage and configuration tips.

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

www.proaudioreview.com July 2007 1 Produdio Review 1 21

PC Man to Mac Guy: **Buying My** MacBook Pro

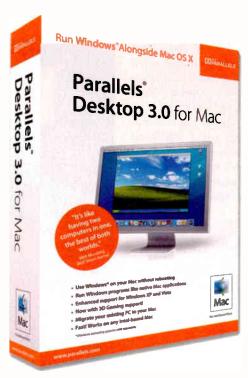
by Dan Wothke

have dabbled with Macs for the past 12 years, but for various reasons Microsoft Windows landed as my operating system of choice. Yet, once the MacBook Pro hit the market I took notice. After checking in with various professionals using the MacBook Pro (and subsequently wearing out message boards while lauding it), I decided to check it out for myself. I originally began looking at the new MacBook based on the Intel chipset; I liked the smaller size. I finally decided on the 15-inch Pro (Intel Core 2 2.16 GHz and 1 GB of RAM) version mainly for the dedicated video RAM (128 MB on my model) and FireWire 800 option.

One thing that I can't understand is why the MacBook Pro still does not have a hardware-based right click. The right click is no longer just a PC feature; it is very useful on both PC and Mac platforms. The MacBook Pro does offer an option to use two fingers on the track pad while simultaneously clicking on the button to act as a right click (which also works Windows). So why not go all the way and install a right click?

I would be surprised if they don't eventually integrate a right click directly on the surface. After all, it is okay to admit that

PCs do have a few useful options that would benefit a Mac; I, like many others, have become proficient in the two-finger technique and will use a wireless USB mouse when the situation is right.



Parallels Desktop virtualization software

The built-in Bluetooth option is great for a mouse, but has been reported to cause problems Windows side. especially when working with audio applications. So I steered away from that option and disabled the Bluetooth altogether.

The other missing keypad option that I often reach for is Delete. The Mac Delete functions like a backspace, not a true Windows delete. In order to actually delete a file in Windows using the

keypad I have to hold down Function and press Delete. The problem is that the keys are on opposite sides of the keypad and cannot be accomplished with one hand, so I would either choose to delete from the



For immediate service, fax card to 703-820-3310



NO POSTAGE
NECESSARY
IF MAILED
IN THE
UNITED STATES

BUSINESS REPLY MAIL

FIRST-CLASS MAIL PERMIT NO 3068 FALLS CHURCH VA

POSTAGE WILL BE PAID BY ADDRESSEE



Get your own copy of Pro Audio Review!

rill out the form below and fax to Incomplete forms will not be pro		ubscription. Or, for faster service, visit http://www.proaudiorev	iew.com/subsc	ribe/
•		to Pro Audio Review! No, thank yo	ou.	
-		Digital Edition. (Email address required.)		k voi
	_	e the fax number provided to communicate	_ no, man	· you
		products or services:		
	_	selected businesses and organizations allied to the	6/07	BZ
professional audio/video indust		,		
1. TYPE OF AUDIO BUSINESS (only one):		Signature (required)		
☐ A. Commercial Recording Studio	☐ F. Mastering Facility	orginature (required)		
□ B. Broadcast Station/Studio	R. Duplication	Date		
C. Video/Film Production	☐ K. Project Studio	Date		_
Q. Video/Film Post-Production	L. Music Studio	N - 1 - 1 - 1 - 1 - 1		
 J. Indep. Audio/Production Facility 	☐ T. Record Company	Name (print)		
☐ E. Multimedia Production Facility	D. Educational Facility			
G. Venues/Auditoriums	□ N. Dealer/Distributor/Retailer	Job Title		
☐ H. Contracting/Installation	M. Manufacturer			
S. Facility Design/Maintenance	 P. Manufacturers Sales Rep 	Company		
□ I. Audio Consulting	V. Sound Reinforcement			
□ W. Rental Firm	☐ O. Other	Address		
2. JOB FUNCTION (/ only one):		Addicas		
□ P. Engineering/Technical Mgt.	 Q. Engineering/Technical Staff 	City/State/Zip		
☐ A. Corporate Mgt/Owner	☐ F. Sales/Marketing	City/State/2ip		
□ B. General Management	☐ G. Education/Training			
C. Production	 J. Purchasing 	Phone		
☐ K. Consulting	□ E. Musician/Performing			
☐ L. Independent Engineer	☐ M. Producer	Fax		
☐ N. Sound Design/Editor/Mixer	☐ H. Other			
3. PURCHASING AUTHORITY (only one):		· Email		
☐ A. Authorize ☐ B. Evaluat	e C. None	(required for the Digital Edition)		

menu or continue developing my two-finger-on-the-pad with click combination. After doing a little searching I did find a free utility from www.olofsson.info to map the Mac keyboard to my referred Windows XP style. There are many out there, and surely there is one that will help any user when swapping between operating systems.

Aside from some keyboard and mouse adjustments, the MacBook Pro has been running both platforms without any problems ... and running them quite well.

I decided to setup my Mac to dual boot with Windows Boot Camp, beta version 1.3. Setup was a breeze, as all that was required was my Windows XP CD (I installed XP PRO with SP2) and a blank CDR so the Mac could burn all of the drivers for the Windows install. One should note that Mac cannot directly write to NTFS partitions, only FAT32. Due to the efficiency and security provided with NTFS, I went that

route with my Windows partition. I can read and copy from the Windows hard drive when booted in Mac but cannot write.

On the Windows side, be prepared to spend an extra \$50 to purchase Mac Drive (www.mediafour.com). This handy utility will mount Mac-formatted hard disks (currently using HFS) when booted into Windows to either read or write. The only adjustment for Mac Drive was to turn off the feature so as to always read dual disks as a Mac format. This is nec-



essary when using CDs set up for both platforms, such as VST or audio installa-

tion programs.

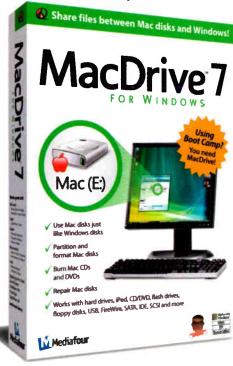
Another great utility when running both Mac and Windows OS is Parallels. This will run a true version of Windows in a virtual type of when machine booted into the Mac OS. My end goal is to have Parallels run my current Windows partition while booted into the Mac OS. This is now available with Parallels 3.0, but I have read that it will cause conflicts with Boot Camp; I decided to wait for the bugs to get worked out as I don't need any

additional complications in my transition to Mac world.

Aside from some keyboard and mouse adjustments, the MacBook Pro has been running both platforms without any problems ... and running them quite well. In order for Apple to further establish the MacBook Pro as a formidable computer in the world of Windows laptops, some compromises could be made on the keypad and mouse pad to smooth the transition for loyal Windows enthusiasts everywhere. Yet, all in all, for PC Windows users Apple's dual-platform capability is a great tool. Having both operating systems at my disposal has already proven worth the adjustments despite any interface shortcomings.

Dan Wothke began his journey as a musician and naturally progressed to exploring audio engineering. He has since accrued over 10 years of studio, live sound and technology experience. Dan currently runs the gauntlet of all things media in his role as Media Director at Belmont Church in Nashville.

MacBook Pro photo courtesy of Apple.



Mediafour's MacDrive 7

www.proaudioreview.com

July 2007 ! Proludio Review | 23

Digital vs. Analog Summing

by Alex Qana

'd venture to say that I'm not the only mixer who grew up in an analog world who unequivocally loves the convenience of the DAW. Should I care about analog summing? What about all the anecdotal evidence? Am I selling my projects short when I mix in the box?

A year-and-a-half ago I bought what is arguably the most transparent and neutral analog summing, or ASum, system available: 32 channels of passive Folcrom feeding a pair recent and old mix sessions unusable.

In pondering these issues, I decided to do a shootout to answer the following question: Is analog summing worth the trouble or should I just sell the equipment?

A REDUCTIVE COMPARISON

I had recently cut some inspiring live tracks in the studio to Pro Tools | HD with Division Day — an analog-savvy band who had previously only tracked to tape and mixed on consoles to tape. It seemed appropriate I should make an attempt to use my analog summing gear. I printed one mix three ways: digitally summed, 'ASum' via Digidesign 192 D/A converters, and ASum via Apogee DA16x D/A converters.

In each case, the D/As fed 14 channels of

ent. The digital mix had a noticeably harsher top end but a nice bottom. The ASum mixes had something distinct in common: both had smoother top end above 2k. I felt that the DA16x was slightly smoother, with more body and tonal clarity for each sound, in comparison with the Digi 192.

I gave these three identical mixes to a trusted colleague for his own blind listening test. LA-based engineer Bob DeMaa has some great ears, and his tastes aren't always the same as mine. In this case he reported hearing identical qualities to the ones I observed.

To anyone who can afford it, the \$5,000-plus required to get into this summing game is absolutely worth the improvement. However, most listeners of an ITB mix wouldn't know there was anything inherently lacking (as we

> truly are deep in the era of the MP3). Regardless, I am relieved by how similar digital summing is to top-quality analog summing... especially because I've been taking the easy digital route lately.

A THOROUGH COMPARISON

If making an informed decision regarding analog summing were just about a simple A-B test through a transparent signal path we could look at the score and the costs and you could turn the page.

Pro Tools "bounce to disk" = good Analog summing via Digidesign 192 = better Analog summing via Apogee DA16 = best

But there are two reasons to dig a little deeper for outcomes this test could not reveal.



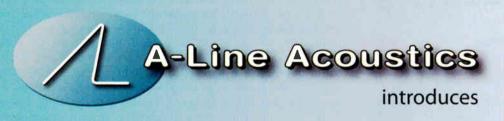
AMS Neve 8816

of Millennia HV3s (inside TD-1s) for makeup gain. The frequency response of this setup does not fluctuate more than 0.1 dB from 20 Hz - 20 kHz, for whatever that's worth. So why haven't I used it in over a year?

One answer is it's easier not to use it. Mixing entirely inside the DAW is totally recallable and repeatable, except for when software updates really screw you and render

Folcrom sparing one pair of outputs for monitoring. The D/As chopped to their own clocks, were gain-matched, and each analog bounce was re-digitized through the Digi 192's A/D. (At this point it is important to note that the mix in question was performed as an in-the-box mix then split out analog for the test.)

Small differences were immediately appar-

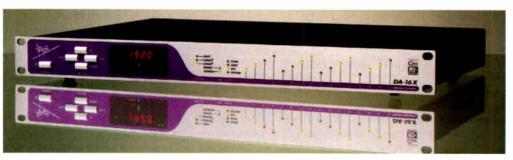


EMMA EVERY MAN'S MODULAR ARRAY

EMMA SOLUTIONS - affordable, portable line array speakers for quality uniform sound with quick, easy set-up. Powered DSP controlled modules may be stacked or flown. 6.5" LF drivers and 6.5" HF ribbons, with 100° wide dispersion, match 4x10" subwoofers to deliver smooth, accurate audio performance. USA made EMMA enclosures are available in custom finishes.

www.a-lineacoustics.com





Apogee DA-16x

One, my gain structure was set up to work for internal digital summing — levels much lower than I could get away with hitting the analog summing - and two, my analog summing equipment is about as uncolored and undistorted as it gets, which might be missing a range of tonal opportunities available exclusively in the analog domain.

If your DAW uses fixed-point math (like Pro Tools) it will blossom when you get those faders closer to zero. I took my test mix and increased the output gain of all the faders (or output levels of the last plug-in on each channel) by eight dB over the gain structure appropriate for the digital bounce. This put most of the faders near unity and resulted in output levels at the D/A converters up near the top with a few dB of headroom to spare. Subsequently, I decreased the analog makeup gain at the preamp by 8 dB. The summed mix was at the same level; but, boy, did it sound different!

THE IMPORTANCE OF DISTORTION

At a recent recording engineer's conference in Tucson, Arizona, one of the seminars was devoted to distortion in recording. It was one of the most spirited panels, with over the top adjectives flying left and right about every-

one's favorite ways to mangle and blow up tracks. I had been thinking a lot about distortion recording and **Crane Song Phoenix** and Massey **Tape Head**

was getting back around to the idea that the digital age trend of higher fidelity through lower distortion wasn't making better music.

I work really hard every day to find ways to inject distortion into my mixes using tools like Amp Farm, Tape Head, Phoenix, Analog Channel, Sans Amp and random, cheesy VST plug-ins. More germane to the summing discussion is the subtle, naturally occurring distortion missing in digital and inherent in the analog domain. Without getting into a lengthy technical discussion, transformers, tape, discrete circuits and tubes found in highquality signal paths create harmonics that increase the tonal density of sounds, making them thicker, warmer, more vibrant or all of the above. The proliferation of vintage equipment modeled or cloned as plug-ins underscores the importance of distortion in making recordings sound better.

REAL WORLD PROCESS

For an album I was hired to mix a year-anda-half ago, the producer and I had talked about taking the finished mixes to an SSL room here in LA in order to split out the tracks and sum through the desk. I reasoned if I bought analog summing gear I could theoretically do a version of the same, reaping the benefits of analog summing at my own studio, saving the client thousands ... and then I'd own the gear. The client agreed.

There are many great analog summing options out there, such as the Neve 8816 reviewed by Russ Long in the December 2006 issue of PAR; it offers an affordable version of a time-proven mixing system. I was hesitant to commit to one summing flavor, so I invested in the flexibility of the Folcrom. Then I faced the tough choice of what type of makeup gain I would want for raising the output

the fully passive Folcrom to line level.

There's virtually nothing inside the Folcrom, which makes it devoid of coloration. What might at first seem like a pain in the ass is truly where the genius of the Folcrom lies: determine the tone color of the mix with the mic pre you choose for the make up gain. In a studio chock full of mic pres - which would otherwise lie dormant during the mix phase of a project one might have a lexicon of classic tones to impart to a mix. Hook up a pair of Neve,

SUMMING continues on page 26 ➤

LOWEST COST PER PRINT



Introducing the Nexis 100AP the premier automated disc burning and printing solution! Using low cost non proprietary ink cartridges. the Nexis 100AP offers the lowest print cost in the industry.

Featuring Natural Pick™ technology, no mispicks or double picks, assuring the user a 100% duty cycle.

Using our patented AutoCal™ the Nexis line can self calibrate providing longer and more trouble free service life than the competi-

XLNT Idea brings the same level of innovation and design to their entire product line of DVD/CD disc creation solutions.

Contact us for a dealer near you 1-800-479-1689



www.proaudioreview.com July 2007 | Produdio Review | 25



API, Helios, Telefunken, Trident or whatever pres that you have available, then think of how radically different the mix would sound through each of those.

My studio happens to be a dedicated mix room: so, unlike lions at a zoo. I don't have a pride of bored mic preamps licking their

dB. Then you want to make some rides, or you realize the kick isn't loud enough.

Just as you finesse that final five percent of magic into the song, you have to put your engineer's cap back on and bring each individual track down by 2 - 3 dB without messing up your compressed busses, effects returns and automation moves. What a perfect time to get out the old fluoroscope and watch the neurons light up the left hemisphere of your skull while the right side goes dim. Honor your mix as a musical performand still maintain a minimum 3 dB of headroom on each at the converter output. This may be an obvious reason of why analog summing sounds better. Not because it's analog but because it saves each DAW track from the abusive process of gain reduction the D.A.E. (Death to Audio Effervescence) engine. [Go ahead, make up your own acronym — it's fun!]

Today, my faders sit a lot closer to unity and I like it. Remember how people would say, "Oh, the higher your track count the worse your mix sounds in the box?" That concern is moot with analog summing.

Universal Audio 2-610

chops from across the hardwood moat. I decided that in my first venture I wanted most to hear my mix as-is, but with the benefits of summing happening in the analog domain. This meant I needed a clean pre a Millennia.

I purchased a pair of super transparent TD-1 pre-amps, knowing they would also work well as a DI and mic pre for the occasional mid-mix overdub. I'm glad I did. It turns out that not only do they spit out exactly what you put in, but the built in two-band parametric EQ is unbelievable: the best high frequency analog EQ I've ever heard ... simultaneously sweet and exciting. The other mic pre I have available for make up gain is a Universal Audio 2-610, which supplies delicious tone coloration from the opposite end of the spectrum.

The first time I broke out a mix into the Folcrom the producer and I heard the difference immediately. Track-by-track things seemed to get simultaneously clearer and wider as I routed incrementally away from Master to each of 16 analog outputs. I distinctly remember my producer friend Sandy Chila grabbing my arm and exclaiming, "I hear that!"

At the time, to be honest, the difference was apparent but didn't seem to merit all the trouble. Then I started revising those same mixes. My process was to get the mix 95-percent of the way there, then break it out to do the fine-

tuning. You know that you have your mix pretty much dialed in when you're patting yourself on the back that it is peaking at -0.1

ance and create an environment to enable your flow.

Don't bring all your faders down, push 'em up! The first time I felt this way, I had

To anyone who can afford it, the \$5,000-plus required to get into this summing game is absolutely worth the improvement. However, most listeners of an ITB mix wouldn't know there was anything inherently lacking (as we truly are deep in the era of the MP3).

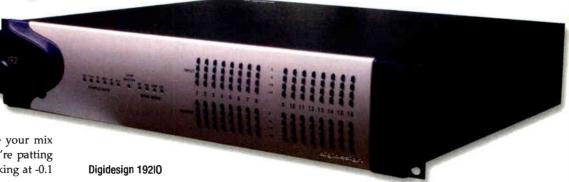
been mixing ITB for so long that I forgot that analog is headroom heaven. When I did my initial shootout discussed at the beginning of this article, I was able to bring each channel up by more than 100 percent

SUMMARY

It's not just about the sound when it comes down to determining the value of analog summing; it is also about how the sound affects your mixing process and where that might lead you. Digital summing can be great If productivity and having frequent recalls are key and you're not mixing lots of dense arrangements with high track counts.

With analog summing, more choices become available, such as the type of D/A converters and the ability to incorporate other outboard gear. Striving for audio bliss involves compromises such as parting with lots of money. Another compromise can be process. To some, it may be more valuable to have quick and easy mix recall. I can now safely release a portion of the guilt I've been feeling over mixing in the box lately because the current state of fidelity Pro Tools offers isn't bad. But I've been reminded that worthwhile sonic improvements are within my grasp via analog summing and the process might inspire me to make better mixes that transcend esoteric audio quality debates.

Alex Oana is an 11-time Minnesota Music Award winner, including three for Producer of the Year. So he got too big for his britches and moved to LA, where they slapped some sense into him. Contact him at www.alexoana.com.



Virtual. Perfection.



StroboSofit

StroboSoft Software Strobe Tuner for Mac / PC

Perfect tuning for instruments or samples can be made with razor-sharp precision using StroboSoft's 0.1 cent accuracy borrowed from our legendary hardware tuner line.

Take advantage of over 30 preset Sweetened Tunings™ that compensate for certain instruments' design flaws and optimize their sound. Use the integrated Buzz Feiten Tuning System® presets for your BF-equipped instruments.

Standard chromatic tuning mode and innovative 'Instrument Mode' provide professional tools for quick tuning or complete instrument set-ups.

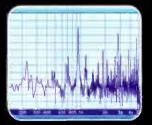
With more than 50 preset alternate tunings and unlimited capability to store your own presets, quickly dial in the tuning you need or compile play lists for studio sessions or your live shows.



Note/Octave window offers real-time response and multi-window (below) provides cent offset, Hertz value, and MIOI note number.



Store unlimited presets for all your instruments or create a preset for each song in your set to compile a set list for



Use the spectrum analyzer to view the fundamental note and its harmonics, view noise floors, or to help you isolate tuning issues. Oscilloscope also included

Legendary, Peterson 0.1 cent accuracy / 52 preset alternate tunings / 26 Instrument presets / Over 30 exclusive Sweetened™ Tunings / 12 selectable temperaments / Buzz Feiten open and intonation presets / Intonate Mode / 96kHz support / 20Hz-5kHz range

VST plug-in available July 2007.

Free 1.22 update to registered customers.

Check it out today at www.strobosoft.com

peterson STROBE TUNERS

- Industry Standard for 59 Years













(spread out across all channels), will operate freely without delays. Your performance will vary based upon innumerable factors such as sample rate, virtual instruments, buffer sizes, plug-ins, etc. Powered plug-ins will certainly help, but watch for incompatibilities and critical updates.

A RATHER SIMPLE TRANSITION

All things considered, the transition from G4 to Mac Pro was really quite simple, not a source of frustration. Although some software and peripherals weren't so easy to deal with (see the accompanying sidebar on peripherals), added hardware was a pleasure to install and was reasonably priced. All aspects of performance are stellar, the best I've ever worked with, and significantly but not dramatically better than before.

Reports of blazingly fast operation way beyond previous levels may be a bit exaggerated in most cases, but the possibility of all that RAM does beg the question: Just how fast will a maxed out Mac Pro work with RAM-intensive DAW software? Mac's OS remains simple, stable and versatile (due in part to the ease of the Audio Units protocol), but the long awaited release of the next OS — Leopard, in Fall 2007 — will be closely watched by stability-demanding pro users.

Rob Tavaglione is owner of Catalyst Recording in Charlotte, NC, and a mixer for the NBA Bobcats.

Photos courtesy of Apple.

ON MAC PRO PERIPHERALS AND SOFTWARE

It was time to install my audio interface and software and get back to work. In order to use my MOTU interfaces I had to get a new PCI card, as Mac Pro uses the new PCI Express (PCI-e) standard. I foolishly fumbled with the single retaining bar that now clamps down the cards and made another attempt before I got the card to seat right and recognize both interfaces. Next I installed Digital Performer 5.1 and predictably faced no problems; DP has always installed perfectly for me. The 5.1.1 updater was also required, but I was up and running in less than 10 minutes. I had already been using DP 5.1.1 (it is Universal Binary), so I could notice the subtle differences with Mac Pro. Basically, responsiveness improved with quicker command response, faster fader draws and improved functioning of the multi-tool. Although the audio quality remains unchanged, "bounce to disk" functions are now much faster and quick re-positioning of playback on the timeline is greatly improved.

Universal Audio UAD-1 UltraPak-powered plug-ins also required some new hardware in the form of UA's PCI-e card. After installing the new card and reauthorizing my previously owned plugs (via a web-based command/response code), I was almost functional. The UAD-1e processors operate via the Audio Units protocol; the Mac Pro's Core Audio must re-inspect the plugs before use. No errors were found and I was once again enjoying powered plug-ins, including detailed automation.

A Focusrite Liquid Mix required a new

Intel Mac installer, as well as updated snapshots and emulations (about 390 MB total), which were available free of charge from the Focusrite website. After Core Audio's inspection, the Liquid Mix showed significant improvement in this new environment compared to its operation with my G4 (please see *PAR* October 2006 for the full review). Tip: Please remember to adjust buffer sizes, plug-in delay compensation and "always use in real time" parameters (a setting unique to Digital Performer) for best results.



Focusrite Liquid Mix

Unfortunately, my Waves Masters mastering bundle didn't make the transition. It is not unusual for plug-in development to lag behind such major changes in hardware and pro users should plan for at least some incompatibility in the short term. Waves plans to release this bundle with Intel Mac and MAS (MOTU Audio System, MOTU's plug-in format) compatibility in August '07, but it falls too late for the press time of this evaluation. In the meantime, Waves tells me that Masters is compatible with DP versions no higher than 5.01.

Rob Tavaglione



Performance is Everything.



The VX-5 handheld condenser from Audix brings studio quality sound to the stage.

The flexibility, resistance to plosives, off axis rejection, and ability of the VX5 to handle high levels of sound pressure will satisfy the most demanding performers and engineers.

In addition, the VX-5 will capture acoustic instruments such as guitar, woodwinds, brass, percussion toys, and drum overheads with outstanding clarity and accuracy.

For award winning performance, make the VX-5 your mic of choice.



For more information go to www.audixusa.com or call 1-800-966-8261

Audix Corporation. PO Box 4010. Wilsonville, OR 97070. Audix Corp 2007. All rights reserved. Audix and the Audix logo are trademarks of Audix Corporation.

BENCH TEST

Benchmark DAC1 USB D/A Converter

BENCHTEST Continued From Page 12

that they up or down sample everything to 110 kHz, as they feel that the chipsets and digital filters don't work as well at 192 kHz as they do at 96 kHz. 110 kHz was chosen as it was judged to be optimum for the process. This is the reason for the reduced high-frequency bandwidth at the 192 kHz (and 176 kHz) sample frequencies. As an observation on output regulation with a 600-ohm load, the output dropped about 0.4 db for the unbalanced outputs and 1.7 dB for the balanced outputs.

Total harmonic distortion plus noise in a 22 kHz measuring bandwidth as a function of signal frequency and sampling frequency was virtually identical for all sampling frequencies. A plot of distortion vs. frequency at a sampling frequency of 44.1 kHz in a 22 kHz BW is shown in **Figure 2** for both channels.

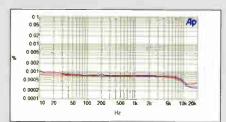


Fig. 2 Total harmonic distortion plus noise as a function of frequency at 44.1 kHz Fs. Lch = Red, Rch = Blue.

Plotted in **Figure 3** is distortion vs. frequency with the measurement bandwidth opened up to 80 kHz with the 96.0 kHz sampling fre-

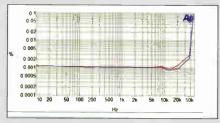


Fig. 3 Total harmonic distortion plus noise as a function of Frequency at 96.0 & 192 kHz Fs. Lch = red, Rch = Blue.

quency. THD+N of a 1 kHz test signal as a function of level down from 0 dBFS full scale is shown plotted in **Figure 4** for a 44.1 kHz sampling frequency. Results are essentially the same for all three sampling frequencies.

As has been the case with most of the D/A converters I have tested, the deviation

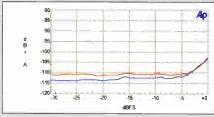


Fig. 4 Total harmonic distortion plus noise as a function of signal level at 44.1 kHz Fs. Lch = Red, Rch = Blue.

from linearity is pretty much the same as a function of sampling frequency. Deviation with linearity for sample frequencies of 44.1, 96.0 and 192.0 kHz is plotted in **Figure 5**.

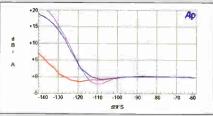


Fig. 5 Deviation from linearity of a 1 kHz test signal as a function of sampling frequency. Blue = 44.1 kHz, Cyan = 96.0 kHz, Green = 192 kHz. Rch shown.

As is usual for most D/A converters tested, channel separation was essentially independent of sampling frequency but not direction. A plot of separation for the two testing directions, L > R & R > L is plotted in **Figure** 6 for a 44.1 kHz sampling frequency.

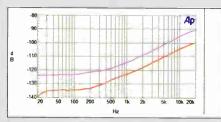


Fig. 6 Typical channel separation vs. frequency at analog output. Sampling frequency 44.1 kHz. Analog output. Red = L > R, Magenta = R > L.

The above has described the usual measurements that I make into the AES/EBU digital inputs. I was able to make a number of measurements with the USB inputs, some of which are mentioned in the measurement text. One aspect of the USB input that was of interest was its susceptibility to jitter in the interface. I had devised a simple test with my AP that introduced a 1 UI 500 Hz sine wave

jitter signal on its digital output along with a signal frequency of 1 kHz at 0 dBFS. This test was at 44.1 kHz with 24-bit resolution. This signal when applied to various D/A converters would introduce distortion sidebands of 500 and 1500 Hz in varying amounts. A typical result of such a test is shown in **Figure 7** for a D/A converter that was quite affected

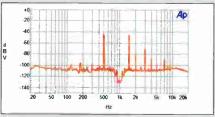


Fig. 7 Example of effect of 500 Hz 1 UI jitter on digital input signal on THD+N spectrum on a particular D/A converter.

by the added jitter – these sidebands disappeared when the jitter was turned off. When I ran this test into the AES input of the DAC 1 USB, the results were very good with no 500 or 1500 Hz sidebands showing. When played back via the USB input of the DAC 1 USB, the results were virtually identical to AES input data. These results are shown in

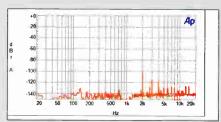


Fig. 8 Spectrum of THD+N spectrum of a 1 kHz 44.1/24 signal under various conditions. See comments.

Figure 8.

One anomaly that occurred in the measurement data was that the played back 44.1/16 silent track made by the WfmGeneration utility might have suggested that the supposedly dithered signal didn't come out that way via the USB input. A more likely cause is that the generated wave for some reason wasn't 16-bit dithered.

From what I have seen here, I would conclude that the performance of the DAC1 USB using its USB input is comparable to using its regular digital inputs.

-Bascom King

new age



Telex Communications: 12000 Portland Avenue South, Burnsville, MN 55337 phone: 1.800.392.3497 fax: 1.800.955.6831 World Radio History

RCM-26

Dual Channel Controller Remote Module

DYNACORD®

by Stephen Murphy

Lectrosonics and the Lavs of Doom

No.

date three panel members, though one of the expected 12 dropped shortly before the shoot, leaving one of the risers with just two panel members.

I slated a Sony ECM-88 wired lavalier for each of the panel members, plus a backup 88 for each riser. Being mobile, the host

would require a top-notch wireless microphone solution: one that can suffer the potential slings and arrows of outrageous interference without a glitch. The DC area is a tri-state confluence of broadcast TV stations, secretive government agencies (and all that that implies), and millions of Blackberry/TDMA devices, most of which seem to find their way to the press club on a regular basis. This was definitely a job for Lectro!

I arranged for a redundant set of two Lectrosonics LM Digital Hybrid Wireless transmitters, two Lectro M152 omni lavs, and my new favorite receiver, the Venue Digital Hybrid modular system.

Next came the issue of sound reinforcement, an aspect of audio production not oft called for in my typical studio and postproduction environs. The live audience, fanning out from the backside of each riser, would need to hear the host, the panel, two handheld wireless Q&A mics and the video playback segments. The panel members on each riser needed to hear the host, the panel minus their own mics, the Q&A and the video playback. The host needed to hear everything and everybody (as did I on my local monitors). The broadcast mix needed all live mics returned to the fourth floor control room, minus the video playback audio they were sending up to me.

he nation's capitol, earlier this year.

Since we last left our hero, MurphMan had resumed his guise as hard-working, modest and ruggedly handsome freelance

engineer/PAR Technical Editor Steve Murphy. Toiling away without complaint in the dank catacombs deep below the National Press Club, Murphy is once again accosted by his Draconian boss, J. Jonah Rothman.

As the boss exits a memo dislodgesfrom the impossibly tall stack just deposited on the desk and drifts to Murphy's feet, where he has just enough time to scan its contents before it is spirited off by an oversized rodent.

"Holy TPS-Report!" Murphy exclaims.
"Arch-villain Open Mike and his 15 henchmen
— The Lavs of Doom — are planning to attack
the club!"

Slumping back onto what passes for his desk chair (an empty spool of multi-core cable and some bubble wrap), mild-mannered Murphy comes to the realization that protecting the viewing public from Open Mike and the Lavs of Doom is too much for his superpowered alter ego MurphMan to handle on his own. The time has come to use the LastResort-O signaling device to call in help — help that can only come from the all-powerful Lectro!

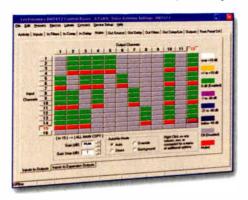
THE BACK STORY

The above is completely true. Well, for the most part. At times...

OK, so I exaggerated a bit: the deep, dank catacomb is really the spiffy NPC Broadcast Operations Center, which is not underground at all, but rather on the fourth floor of the National Press Building. Oh, and my boss is not named J. Jonah Rothman — it's Howard and he's nowhere near as Draconian as depicted.

But Open Mic and the 15 Lavs of Doom did come to the press club, albeit in the form of an hour-long national public television broadcast called *A Public Voice*. The forum-style show, now in its 16th year, is taped in the National Press Club's storied ballroom on the 13th floor — where over 40 years ago Nikita Khrushchev said he would "bury us" and years later Louis Armstrong made his last live recording.

Emmy-winning journalist Frank Sesno hosted this year's *A Public Voice*, which explored the topics of energy sources and consumption. The forum included an 11-member panel made up of nationally known politicians, journalists, policy analysts and policy makers, plus a small live audience. The broadcast taping is the culmination of several public National Issues Forums (see *www.nifi.org*) held across the country. Instead of the usual combative banter, the panel members are instead asked to watch numerous video taped excerpts



Lectrosonics LecNet2 software

from the earlier public forums and respond to the challenging and often-poignant questions and comments from the local attendees.

THE SET UP

The guest panel was set up in an "in the round" style, with four low risers arranged in a large diamond for the panel members, and a large open space (approximately 35 x 35 feet) in the middle where the host could stalk about. The live audience sat in rows fanning out from the panel's risers. Speaking of cameras, the event was taped from a switched six-camera feed, with a studio camera in each of the four corners of the large diamond, another studio camera for audience shots, and an ENG camera in the east balcony (which was also home to my temporary "control room" setup).

Once I retrieved the production outline from the, uh, large rodent, I went to work assessing the audio requirements of the event. Each of the four risers were set up to accommo-

THE ENDEAVOR...

Let's review a basic audio formula: OM(n) + SR(SpE) = UFB(F/B!), where OM(n) is a large number of Open Microphones, where SR(SpE) denotes "Sound Reinforcement Speakers Everywhere", and where the sum of the equation is...well, let's just say the part in the parentheses stands for "Feedback".

For the small live audience, I opted for a powered JBL SRX712s placed wedge style on the outside of each riser facing outwards. Monitoring for the members of the panel was provided via a single Galaxy Hot Spot placed on the back of each riser, with its back to the JBL pointing at the back of the chairs. Together with fellow engineer David "One Fader" Sless (who, during the show, potted up the tape roll feed to my mixer from the comfort of the fourth floor control room), I experimented

with front firing placement of the Hot Spots, but found the plush chairs and warm bodies between the mics and the Hot Spot provided good monitoring with better iso.

The tape playback, Q&A mics and redundant host lavs would pose no above-the-ordinary trouble. The challenge I was facing was the 11 active and four backup wired lavs. Unlike other high lav count broadcasts on which I work — such as the Kalb Report, where the host addresses each guest by name, giving me time to pot up — each member of this panel was free to chime in at any time, and in any combination. In other words, there was no predicting and no way I could get away with 11 mics worth of room noise, handling noise and other noises emanating from the panel.

...AND THE RESOLUTION

Coming to my rescue, with superhuman aplomb, was Lectrosonics and its DM1612 digital audio processor. Some may be tempted to call the DM an automixer, but it is more than that — much more. But let me take a step back.

I'll be honest. Never in my long(ish) career have I had occasion to use an automixer. It's not that I have it in for them; they've just never

part of the world in which I've operated (studio music production and video post production). It was a review of a Lectro DM processor in this very magazine a few years back that led me to revisit the device for this application.

Much to my delight, the DM1612 was ideally suited for the job. Without getting bogged down in review-like detail (there is a perfectly functional review available in *PAR* December 2006) it turned out that this was exactly the type of use for which the 16-input, 12-output device was produced. The principal reason the DM processors work so well is that, in each mix, the output is never greater than the equivalent of one mic's signal or noise floor. Though impossibly complex in its internal processing, for the end user the result is simple, smart and incredibly effective.

Through the miracle of the 1612, a laptop, a USB cable and LecNet2 software I was able to design a system that took in all 11 active and four spare lavs — level-adjusted and lightly compressed for each talent — and produce numerous discrete automixed outputs that formed the basis of riser monitor mixes (columns 1-4 in the LecNet2 matrix graphic), four iso mixes for my main mix per-riser con-

trol (columns 5-8) and two full-lav mixes (main and safety, columns 10 and 12).

All of the DM1612 outputs in use were routed to the line inputs of an Allen & Heath GL-2400 console, where I folded in tape roll audio, Q&A mics and the host's Lectro lavs to create all of the respective mixes required of the broadcast taping and the house monitoring.

Though I created the four per-riser iso mixes for fall-back control at the analog mixer, I am happy to report that those mixes went unused, and with the exception of the mix-minus subs sent to the respective Hot Spots, all other mixes (house, monitor, mults, broadcast) used the main "all lavs" single-fader mix. Additional credit needs go to Lectro for the DM's per-input feedback suppression super powers, where anti-feedback adjustments affect only the problem channel's signal, not the overall mix EQ!

So thank you Lectro, from this mild-mannered but very picky engineer. Open Mic and the 15 Lavs of Doom were relegated to Singlefader Land, where they could do no more harm than a single mic. And the public was once again safe in The Nation's Capitol, and beyonnnnddddd...!!!



www.proaudioreview.com

July 2007 | ProAudio Review | 33

by Strother Bullins

Bits and Pieces: My New DAW



Photo condit Doug Blog/Allung

et me tell you about my new DAW.
With it, I consistently record great using the stuff you have?

Sounding 24-bit/96-kHz basic tracks

Four: How many years have you wanted the stuff you have?

Five: How many years have you loved music?

Okay, let's work through these (back to front):

Five and Four: I guarantee answer five is greater than answer four. (If not, you've known that you're a commercial artist for years.)

Three: The answer is yes? Congratulations. You are a story of success.

Two: That much, huh? You could use some more? That's what I thought. Maybe you should take a second look at the stuff you've purchased most recently. Are you now working to pay for it? If so, I hope it's all worth it.

One: It's easy? Cool. How's the "art" these days? Sounding good?

It's difficult? How's it sounding for you? Good? Well, at least you've got that.

In an ideal world, we all answered the same for questions five and four. We're supposed to love this stuff, folks. We all know that there are much easier ways to make money. Those who unteer in the music program at a local school — any grade K-12. They will love you for it.

Can't do it? I'll bet that you can work in the joy of music somewhere. I have living proof: my mom is a great piano player and a retired (career) 5th grade

teacher. She worked her musical prowess into her classroom studies and kids loved it. She's the pianist for her church; she plays and sings with the kids there. And most important to me, she plays for my kid on a regular basis.

The results? Everyone wins. An appreciative audience, right there in front of you, joins you in enjoying your talents: real happiness in pro audio.

STORM THE GATES

Here at *PAR*, I've had a unique position to witness how professional music producers, recording engineers and recording/performing musicians use the gear we review. I also am intimately aware of their world; I've worked for studios long enough to learn what ultimately matters is convenient quality.

And talk about today's convenient quality! Nearly every decent recording device today does 96k. Cheaper mics sound better than ever. You don't need a big mixing desk anymore. (And mixing live with a mouse is just a generation away, my friends.) I could go on and on.

The gates of recording have burst open; the playing field is officially leveled. So pros, please take heed (and stop slouching in your Aeron): nearly everyone who really wants to can now step into this business. What really separates the ever-shrinking number of million-dollar recording facilities from the talented ITB recording engineer? An unnecessary tactile mixer and an environment that, in most cases, is ideal for accurate listening, as well as creative and conducive acoustics. Sure, a world-class acoustic environment is a great place to work (believe me, I regularly miss my time spent at Masterfonics' The Tracking Room). But an ideal environment is not a requirement for producing ideal audio. Sample some of our industry's best new DSPenabled powered studio monitors, get creative with new acoustic treatment products, and grab that mouse. You, too, can mix for excellence and possibly even make the industry's best mastering engineers smile.

In other words: recording establishment, it is game on. Head's up.

Strother Bullins is the Reviews and Features Editor for Pro Audio Review.

With it, I consistently record great sounding 24-bit/96-kHz basic tracks both in my home studio and at live sound gigs. It is incredibly mobile, robust and rock-solid, has yet to freeze, fits neatly in my rack, comes nannels of very good AD/DA convert-

crash or freeze, fits neatly in my rack, comes with 24 channels of very good AD/DA converters, and interfaces incredibly well with my other (formerly my main) DAW, which is currently regulated to editing and effects duty. I no longer need my Mac nearby to instantly capture the musical magic in the room.

The DAW? It's the Alesis ADAT HD24 XR hard disk recorder (\$2k street). I track to dirtcheap IDE hard drives then edit, tune and effect (less) before mixing (and then by necessity, not whim). For tracking purposes, it's easier and more enjoyable to use than my other DAW, and the musicians I work with seem to agree. Finally, my Mac is now used solely for what it does best: not capturing, but processing and designing ... oh, and receiving e-mail. (And I love my Mac, FYI.)

Sure, you may consider my new ADAT HD24 as old technology. You may say that

ALESIS INCOME OF THE PROPERTY OF THE PROPERTY

there's clearly a better, more efficient way, etc., etc., etc. But I challenge you to try (or revisit) recording with a Mac-less or PC-less hard disk recorder like the Alesis. You may find yourself in a personal recording revival ... just as I did.

| HAPPINESS IN PRO AUDIO

Attention: Professional Engineers, Producers and/or Musicians. Answer the following questions for a free career evaluation.

One: How difficult is your job?

Two: How much free time do you have?

Three: For those of you who are making enough money to live like you want with the

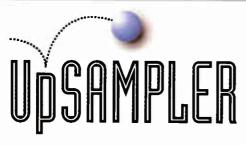
are now doing it "for the money" may have lost their muse and/or their desire to perform.

How to get it back? Share your art for free, preferably in person with your audience. Relive (or experience for the first time ever) the fun of what it's like to share your talents. One great demographic to share with? Kids.

Have a kid? Share your artistic abilities with them. If you don't want to play an instrument with them, at least play one of your great sounding recordings in the house. Be there, talk about it, enjoy it.

No kids? No problem. Volunteer to show some kids what it's like to record music or vol-





Capsule Reviews and Product Review Updates

Tape Head Plug-in for Pro Tools

TDM and RTAS only.

Contact: http://www.masseyplugins.com/

We mixers have our tricks to achieve the sounds we hear in our heads, and Tape Head is a short cut to get there.



Tape Head by Massey Plug-ins is the rare audio device that does one thing very simply and very, very well. It distorts things - just like tape or a stomp box. The unremarkable drive knob on the left belies the sophisticated programming and great ears behind it.

When I turn it up I hear tiny bubbles and, just like Don Ho sang, "Tiny bubbles make me warm all over." But I swear it's not from his song or from champagne that I get that feeling. When I hear a certain kind of

Anyone who has heard it knows... VT-2 Vacuum Tube Microphone Preamplifier (VT-1 Single channel version also available) D.W. FEA www.dwfearn.com 610-793-2526 West Chester • PA • U.S.A.

pleasing distortion I feel the sensation of thousands of clearly defined, smooth, rounded edges in the sound.

Unlike a refined tape machine modeler such as Crane Song's Phoenix, Tape Head does not want to behave. The default setting colorfully saturates, but Tape Head begs to be cranked up to a perfectly balanced overdrive. But, dude, odd harmonic distortion from analog tape is so totally not the same as, like, even harmonic distortion from tubes. Totally, I'm often searching for a cool way to distort stuff in Pro Tools. Tape Head is a very cool way.

Tape Head can be subtle and help any track sit better in a mix, or it can make sounds leap out and snarl. Luckily the trim knob is there to contain all this and keep you under digital clipping. The most complicated control on Tape Head is its tone switch - a little silver, DIY-looking toggle. It's like a bias/EQ control on your tape deck: it changes the tone from dark to bright. [According to Steven Massey, in the latest update of Tapehead (out since this review) the brightness control has received an extra "Mid-Bright" setting that makes the plugin even more flexible. — Ed.] "Simple" and "DIY" also defines the price: \$69. You can't spend money better unless you give it to charity.

Tape Head is one of the few plug-ins I can think of that would have a chance against its double in a parallel analog universe. When a tool gives you an immediate, positive, visceral feeling, it ceases to be mechanical and becomes musical. Tiny bubbles in the wine, make me happy, make me feel fine.

Alex Oana

Audio-Technica ATH-M50 Closed Back Headphones

Contact: http://www.audio-technica.com/

Audio Technica's new studio headphone, the ATH-M50 (\$199) is its latest generation of studio reference headphones. The headphone features 45 mm neodymium magnet drivers. Rated response is 15 Hz - 29 kHz, no tolerance specified. Specs include 99 dB sensitivity and 38 ohm impedance. The headphones weigh 10 ounces and offer well-padded headband and ear cups. The cups swivel for quick one-ear sessions and the headphone/cords can be folded neatly into the included carrying pouch.

I found the ATH-M50 very comfortable during auditioning with a high-end Benchmark HP2 headphone amp fed by an Apple G5 and Lynx card, plus a number of source components including highresolution flash recorders such as the TASCAM HDP2. The headphones' closed-back design with thick earpads also meant very good isolation in noisy environments. At first listen, the mids and highs were pretty close to my reference (and more expensive) AKG K 701 open phones, but the mid-bass seemed slightly emphasized. After a couple of weeks of continued use, however, much of the mid-bass prominence disappeared, though not completely.

After the long break-in, I also noted the excellent imaging and transient response of the headphone. High-resolution source audio is quite revealing through the ATH-M50's, with subtle reverb decay and other transient sounds coming through clearly but not over-hyped or edgy. The impedance makes them quite easy to drive, much easier than the high impedance phones such as the AKG K 701, which often cannot be driven to louder levels by the new crop of portable recorders.

Overall, the ATH-M50 is a highly accurate sounding and comfortable closed headphone.

- John Gatski



PRODUCTION

THE PROFESSIONAL'S SOURCE









real world solutions from industry professionals!



800-947-1182 | 420 Ninth Ave, New York 10001 | We Ship Worldwide

by Rob Tavaglione

Future Sonics Atrio Series Ear Monitors

"Ear monitors" trademark holder Future Sonics offers the "great" (and affordable) Atrio Series.

Personal monitoring is everywhere you look: on TV, on stage and in the studio. Once dominated by expensive custom-fit systems, these professional quality monitors have seen a rapidly falling price point due to a growing number of universal-fit systems. Now IEM innovator Future Sonics ("ear monitors" trademark holder) has entered the affordable fray.

| FEATURES

The Atrio Ear Monitor (\$199) is a single driver design that promises wide frequency response, excellent dynamics and fits into the ear canal with either compressible foam or silicon earpieces. There are two silicon types provided: hard or soft (unflanged and flanged, respectively) with both in three sizes. The ear monitors hold in place via a semi-flexible wire fitting that can be bent to conform to the outer ear's contours. From there, a single 1-meter black cable (claiming reduced handling noise) connects the Atrios to either your wireless beltpack or headphone amplifier with an eighthinch stereo mini-plug.

FAST FACTS

APPLICATIONS

Studio, live, broadcast

KEY FEATURES

Three sizes of compressible foam or hard silicon or soft silicon earpieces: 1-meter black cable; eighth-inch stereo mini-plug; 20 Hz - 20 kHz frequency response, 32 ohms, 112 dB @ 1 mW and 26 dB of ambient isolation

PRICE

\$199

CONTACT

Future Sonics | # 877-FSI-EARS ⊃ www.FutureSonics.com

Included is a handy vinyl carrying case with room for the monitors, two sets of foam earpieces, the silicone earpieces and the all important earwax remover tool. Specs are 20 Hz $20\,\mathrm{kHz}$ frequency response, $32\,\mathrm{ohms}$, $112\,\mathrm{dB}$ @ 1 mW and 26 dB of ambient isolation.

IN USE

The three most important factors in successful personal monitoring are fit, fit and fit. A tight seal that goes deep into our often asymmetrical ear canals is absolutely necessary - for isolation and adequate bass response.

On my first test I tried the hard rubber earpieces, only somewhat comfortably, and found the Atrio pair to have a smooth, yet articulated top-end response, but a bit lacking in bass. Suspecting a poor fit in my troublesome left ear, I switched to the slightly cumbersome foam earpieces and found the missing bottom end. In this particular instance I heard an additional 6-9 dB of bass response, critical to say the least. I then tried the flanged, soft rubber pieces with much more comfort and satisfaction. After a quick lick (a little unsanitary, yes, but experienced users know just how well this works) I got the monitors deeply seated, with nary a bit of discomfort or leakage, an impressive amount of bass and the best performance yet.

Compared to the older, single-driver Shure E1 pair that I had on hand, the Atrios had slightly smoother highs, more accurate dynamic response and healthier bass, with much deeper extension. The Shures were slightly more efficient, about three dB louder in informal testing.

Although designated for specific left or right use, the Atrios used as marked didn't conform to the ear too closely. In fact they seemed to fit best with the wire hanging below the ear. I got a neater look, with less protrusion, after I installed them backwards, switching left and right. For on-camera use, the monitors extend visibly outside the ear a bit. Not such a problem with custom fits or a "flesh" coloring, except the pair for review was navy blue (black is also available) and too visible for some applications.

In the studio, I had a drummer do a few songs in "surround the ear" headphones and then the Atrios. His first comment was, "Wow, these have a lot of bass." He was also fond of the isolation, which allowed him to hear his drums via the microphones, but not via leakage. Our click track was deftly handled; it was clearly "followable" at reasonable levels with zero leakage into nearby mics.

I also had a vocalist try the Atrios with positive success. He first commented on how nice it was that these wouldn't fall off (like headphones) and then expressed how much he liked the clarity and deep bass response. Our only



problem was the occasional loosening of the fit during vocal notes that had the singer widely opening his mouth. I have found this to be a problem on stage, where a sudden increase in leakage can be very distracting. Custom ear molds are often the best solution for such vocal applications, and Future Sonics offers its SofterWear custom fit sleeves for \$149 (plus a fitting fee with an audiologist, approx. \$125).

I SUMMARY

The Atrio Ear Monitors sound great and will definitely satisfy any quality and performance expectations. Even though their mounting scheme requires some effort, I did achieve a very good fit and expect most users to do the same. These may not compare to Future Sonics custom MG4 (\$798), but the Atrio Series will easily meet typical needs on stage or in the studio ... especially for this price.

Rob Tavaglione is owner of Catalyst Recording in Charlotte, NC, teacher of college level audio courses and mixer for the Charlotte Bobcats (NBA) and Sting (WNBA). Contact him at rob@catalystrecording.com.



YAMAHA COMMERCIAL AUDIO SYSTEMS, INC.

Yamaha Celebrates 20 Years of Digital Console Manufacture with Over 2500 PM5Ds Sold Worldwide

Celebrating 20 years of digital console manufacturing, Yamaha Commercial Audio Systems, Inc. continues on the fast track to providing complete sound system solutions with its latest digital audio products for the commercial installed sound and touring sound markets. Yamaha PM5D Version 2 software is available as a free download and adds over 30 new features which have been based upon

input from the company's vast customer base of engineers, sound companies and contractors. New features include Virtual Soundcheck that allows for an easy switch between live inputs and inputs from a multi-track recorder. Yamaha PM5DV2 includes extended Effects Library Programs: De-Essor, add-on **VCM** (Virtual Circuitry Modeling) effects now built in such as analog

modeled compressors, EQs (Comp 276, Comp260, EQ601), and OpenDeck tape simulation. Effects can be used as additional 31-band GEQs and all GEQs can be switched to PEQs. The new Channel Move feature allows channels to be moved for layout reorganization.

Yamaha PM5DV2 also adds new security features such as Read-Only Scenes which cannot be erroneously stored-over, edited or

erased, by loading from a card or Studio Manager — a great feature for protecting templates and basic scenes. Load Lock prevents internal data from being overwritten from a memory card or Studio Manager, and Output Isolation protects output channels and parameters when the console is used at festivals or performing arts centers with multiple engineers who might have their own Scene Memory setups.

The new Yamaha DSP5D is a stand-alone unit that functions as a DSP-expander for the PM5D digital console increasing the PM5D to 96 mono plus 16 stereo input channels, ideal for tour sound and installation applications where more DSP processing power, inputs, card slots (2 provided), effects and dynamics processing are required. The unit can also be used stand-alone with a PC and Yamaha Studio Manager software.

When a PM5D is used in conjunction with the new Yamaha DCU5D Ethernet Audio Cascade Unit and a DSP5D, the DSP5D can be placed on stage or set in a remote location and controlled from the PM5D up to 100 meters using a CAT5. Unique to Yamaha, the new DCU5D can connect 32-bus cascade ports of a PM5D digital console to a DSP5D, expanding the mixing power of the console's input channels, effects processors and other console connectivity features. A second DSP5D unit can also be added for further expansion of up to 144 mono plus 24 stereo input channels. The unit is controlled via the master PM5D console and DSP5D units can be placed onstage, in an orchestra pit, or anywhere distance requires the flexibility of CAT5 cabling. A DCU5D can be used with existing customers' PM5D consoles, and uses a single Ethernet cable for connectivity.

When using ADK's new LYVE Tracker with a Yamaha PM5D, the 4U rack-mounted unit will provide an excellent recording solution capable of up to 192 simultaneous tracks. The LYVE Tracker is custom designed for use with Yamaha digital consoles and built to withstand the rigors of life on the road in a concert tour environment and the perfect solution for houses of worship, studio or broadcast use.

LYVE Tracker is rock solid, loaded with powerful features, yet easy to set up and use. Just Plug and Play! The system features Intel's new Core 2 Duo and RME MADI technologies and is equipped with removable hard drive bays and a DVD-RW for backup and archiving. The unit is offered in three models with capabilities of recording 64, 128 or 192 tracks respectively. Available with a choice of

Steinberg Nuendo 3 or the newly-released Cubase 4 engines, the package includes a complete plug and play, one-cable MADI connectivity package. Cubase 4 natively records in the industry standard broadcast wave format and can import and export OMF 1, 2 and AIFF which are compatible with Pro Tools,

AVID, RADAR, Nuendo and

most other DAW and computer recording programs.

For more information, contact Yamaha Commercial Audio Systems, Inc., 6600 Orangethorpe Avenue, Buena Park, CA 90620; telephone 714-522-9011; e-mail casales@yamaha.com; or visit www.yamahaca.com.

Yamaha Commercial Audio Systems, Inc. offers a complete line of integrated professional audio products for the commercial recording, production, broadcast, live sound, and sound reinforcement markets. The company has a dedicated dealer network, intensive field training, and 24-7 tech support in the U.S. and Canada.

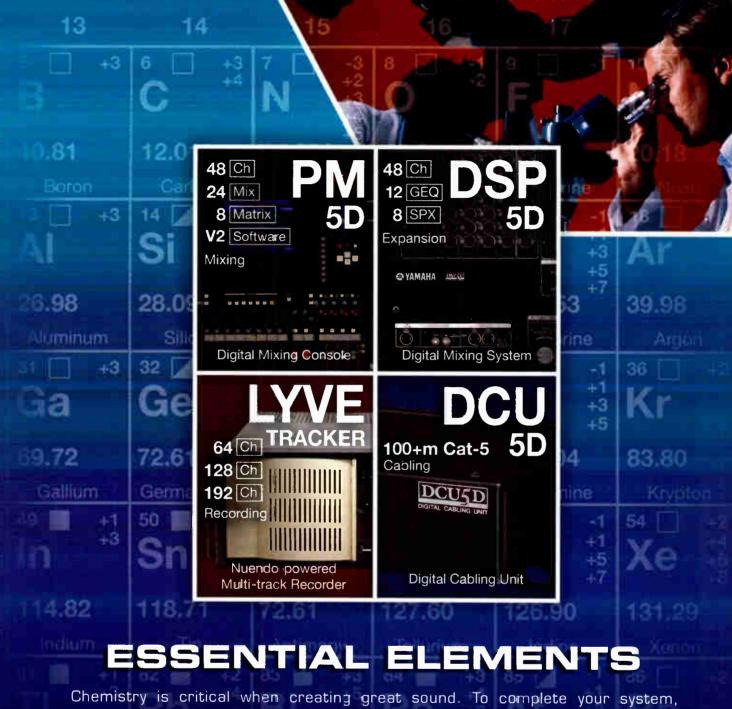




Yamaha Commercial Audio Systems, Inc.

6600 Orangethorpe Ave Buena Park, CA 90620

Tel: 714-522-9011 www.yamahaca.com casales@yamaha.com



Chemistry is critical when creating great sound. To complete your system, 4 compatible components are vital. The foundation for any solid mix begins with the PM5D digital console. Add in three fundamentals: an expander box, cabling and a multi-track recorder, and the reaction is sure to be positive.

For more information on the multifaceted PM5D system, visit www.yamahaca.com



Yamaha Commercial Audio Systems, P D. Box 6600, Buena Park, CA 90620-6 ©2007 Yamaha Commercial Audio Systems

QSC AUDIO PRODUCTS INC.

Object Will Sound Larger Than It Appears



Designed for the most demanding concert and installation audio professionals, the WideLine-8 system sets new standards for astonishing acoustical performance and compact size.

Line arrays are increasingly employed in a wider range of venues and applications. Frequently these venues have height or sightline restrictions that limit the placement and size of the array. Rising fuel costs are putting pressure on touring system providers to downsize their transportation requirements. At the same time, client and audience expectations for audio performance have never been higher. To meet these needs, QSC embarked on a program to develop the most compact, highest output line array speaker system available. QSC's WideLine-8 packs full size line array performance into an ultra compact package measuring less than 20" (508 mm) wide and 9" (229 mm) high.

The classic hot rod formula – putting the most horsepower available into the smallest chassis possible.

Each tri-amplified WideLine-8 element uses a pair of high-power, neodymium magnet, 8" low-frequency drivers. Both cover low frequencies while only one extends into the midrange, thus maintaining horizontal dispersion control at crossover. High frequencies are handled by a premium, 3" diaphragm, neodymium compression

driver mounted to QSC's patented, 140° multiple aperture diffraction waveguide. WideLine-8 signature wide coverage delivers a broader stereo image, reduces the need for supplemental fill speakers and allows more placement options.

The waveguide exit extends nearly to the top and bottom of the enclosure to create a continuous acoustical source with minimal discontinuities between adjacent elements resulting in greatly reduced destructive interaction.

The companion WideLine-8 sub sets a new benchmark for extraordinary acoustic output from a miniscule enclosure. At first glance, the enclosure seems too small to even house its pair of 4" voice-coil, long-excursion 12" woofers, yet it produces 135 dB (peak) SPL and has LF extension to 32 Hz, with the punch needed for popular music. The extremely low profile of the WL212-sw sub

(less than 15") also makes it a great choice for installed and portable use under a stage or in other tight quarters. The 4th order bandpass design matches the width of the WideLine-8 elements allowing it to be flown inline with or behind a WideLine-8 array.

No corners cut — No sacrifices made — QSC quality throughout.

Premium materials are used throughout the system. The enclosure is constructed of Baltic birch plywood and coated in an environmentally friendly, waterborne polymer finish that is field repairable. To keep the weight to a minimum and prevent rust, suspension fittings are made of aluminum.

An elegantly simple 4-point suspension system combined with lightweight, compact size and excellent handling ergonomics mean that a one person crew can easily deploy an array. A single AF3082-L array frame suspends up to twelve line array elements or up to twelve line array elements plus four subwoofers. The WideLine-8 system is equally at home in the air or ground stacked.

Special attention is given to ground-stacked applications. The WideLine-8 sub suspension system includes a unique provision that allows the lowest line array element stacked over a sub to be tilted up or down by as much as 10°, offering better coverage of listeners near the stage. WideLine-8 subs may be used as a base for stacking up to six WideLine-8 elements.

There's more to the WideLine-8 than just loudspeaker technology. The WideLine-8 is part of a complete system solution including the new QSC PL3 amplifiers. Signal processing is also part of the WideLine-8 story. Choose the networked audio, control and flexibility of QSControl.net version 3 or the simplicity of the new SC28. The SC28 is the first loudspeaker processor to implement Intrinsic CorrectionTM, a holistic approach to loudspeaker tunings that produces amazing results while dramatically reducing the time and effort required to set up a system.

The WideLine-8 offers installers and touring pros an ultra-compact, concert-capable line-array system unlike anything else that's available.

QSC, the QSC logo and WideLine are registered trademarks of QSC Audio Products, Inc. in the U.S. Patent and Trademark office and other countries. Intrinsic Correction is a trademark of QSC Audio Products, Inc.



QSC Audio Products Inc.

1665 MacArthur Blvd. Costa Mesa, CA. 92626

Tel: 714-754-6175
Fax: 714-754-6174
www.qscaudio.com
info@qscaudio.com



42



Caution: Object Will Sound Larger Than it Appears

NEW WideLine-8



Just when you think compact performance has been pushed to the limit, along comes the WideLine-8. At 20" (508 mm) the WL3082 is just a little wider than the amps in your racks. And it's packed with the same DNA as its slightly bigger brother, including the patented* multiple aperture diffraction slot waveguide. The WideLine-8 applies to audio the classic hot rod, high performance formula — put the most horsepower available into the smallest chassis possible. The companion WL212-sw sub applies the same formula and delivers 135 dB SPL

from an enclosure that's at home in a flying array or as a base for ground-deployment.



And there's more to WideLine™ than superlative performance

WL212-sw

and compact size. WideLine is part of a complete system including the PowerLight™ 3 ultimate analog amplifiers and your choice of signal processing solutions from the drag and drop flexibility of BASIS™ to the pre-programmed simplicity of the SC28 featuring QSC Intrinsic Correction™.

Let QSC show you how a complete, compact QSC WideLine rig fits your needs. For more information, call 1-800-854-4079 or click online at qscaudio.com

*U.S. Patent No. 7,177,437

Hear the Power of Technology

L-ACOUSTICS

Sleek, Stylish and Integrated



offerings include upgraded co-axial speakers, a new amplified controller range and an ultra-compact line-source loudspeaker system.

The model 8XT is a compact, high-power 8"

Renaissance - L-ACOUSTICS new product

The model 8XT is a compact, high-power 8" coaxial loudspeaker perfect for many point source applications. The passively crossed two-way system can function as a full range speaker, for vocal or music reinforcement or as a fill speaker on either a stage lip, under balcony or break-out room or as a very discrete floor monitor. The 8XT may also be combined with our SB118 subwoofer for Dance club, DJ or other full-range+sub applications. These different conditions result in different preset parameter settings exclusively found in the new LA4 amplified controller. The 8XT features a variety of mounting options including an conical pattern with a peak SPL of 123 dB 1 watt at 1 meter in the "FILL" mode.

The model 12XT is a full-size, 2-way 12" coaxial loudspeaker that fulfills the higher SPL requirements of a stage monitor and other high SPL and full frequency applications, but still maintaining a compact and low profile footprint. Many other FOH and fill applications can be addressed with the 12XT as well. Mounting option include a fly-point, pole cup and mounting yoke. The 12XT produces 129 dB (peak at 1W@1M) utilizing the fill setting and its pattern is 90 degree axi-symmetric. As with the other point source products, the 12XT features extensive preset configurations exclusively within the LA4 amplified controller.

KIVA is our newest offering in the category of line-source loudspeaker products. KIVA is an ultra-compact enclosure containing two 6 1/2" low frequency drivers and a single 1/5"

diaphragm compression driver. The KIVA system attributes are enhanced with a very discrete and fully captive rigging system. Both flown and ground stack operation are supported.

KILO is the low frequency extension companion enclosure for the KIVA system. Sporting a single 12" transducer, the KILO enclosure adds extended range to the 50-100 Hz region of the a KIVA system. The enclosure is capable of producing 125 dB (peak) in its operating bandwidth and features the same integrated flying hardware as the KIVA enclosure.

The LA4 and LA8 amplified controllers are the next generation of processing and power for the full range of L-ACOUSTICS products. Both controllers retain a four channel amplifier format with a 2 input, 4 output DSP front end processor. The LA4 is 1000 per channel; the LA8 is 2000 watts per channel (both at 4 ohms). Simple preset recall of the extensive L-ACOUSTICS library is done via the front panel. Support of legacy products is included in the library. Additional control can be obtained through a standard Ethernet connection to a PC computer using the Network Manager software. Many man hours of development went in to the DSP section which features the use of cascaded A/D converters which achieve a 130 dB dynamic range. Additional features of the DSP section include the incorporation of linear phase filters and a superior dual-mode speaker protection limiter which protects from both over excursion peak operation as well as thermal protection for longduration operation.



L-ACOUSTICS US

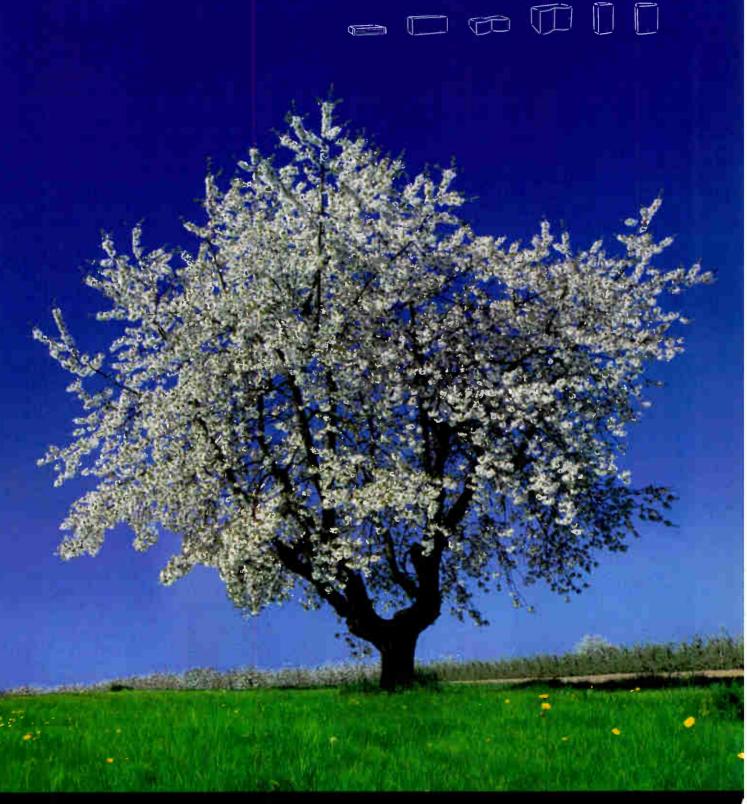
2201 Celsius Ave, Unit E Oxnard, CA 93030

Tel: 805-604-0577
Fax: 805-604-0858
www.l-acoustics.com
info@l-acoustics.com





RENAISSANCE



L-ACOUSTICS is recognized the world over as a key innovator. Our reference systems backed by an uncompromised dedication to training and customer support have all contributed to our success. But we cannot rest there. Innovation drives us forward and once again our R&D roots are regenerating. The L-ACOUSTICS family tree is now blossoming with an incomparable new lineage including amplified controllers, new XT coaxials, the KIVA Line Source Array, and a revolutionary new SB subwoofer. Prepare to share in our renaissance and pick up a taste of the new fruits by visiting us at www.l-acoustics.com



A-LINE ACOUSTICS

Hybrid-Active Solutions for a Greener World, Greener Wallet

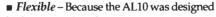
The old phrase "work smarter, not harder" is as crucial today as ever – not only to succeed, but also to survive in the audio production business. A-Line helps sound professionals meet the challenge of the ever-changing, competitive market with line array systems designed to be more efficient, cost-effective, convenient and flexible than ever before. In addition, A-Line helps to conserve our environment by building products locally in the U.S.A., reducing the costs and energy needed to import systems from overseas. Drawing upon extensive research with audio professionals, A-Line

has developed innovative solutions for more compact, better performing and easier to use speaker systems. The result is lighter, more productive and energy efficient hybrid-active line arrays that take up less cargo space for a greener world and a greener wallet.

The "Swiss Army Knife" of line arrays:

- AL10 (2x10" B&C neodym., 1.4" B&C neodym.) Hybrid-Active with EZAL
- ■LS218A (2x18") Powered Subwoofer
- AS215A (2x15") Powered subARRAY with EZAL

■ Efficient – Line arrays have become a popular alternative to point source speakers because they project better to the back of a room while providing acceptable listening volume toward the front. However, bigger is not always better. Obviously, top-quality components are necessary for high definition audio, but other factors help to determine coverage and pattern control. Frequency crossover points must be considered for optimum driver efficiency and clarity. Typically, the length of the line determines the projection and lower frequency pattern control more than the size of the enclosures. Therefore, the sum of a quantity of smaller enclosures can be greater than fewer larger ones. Powered systems can be more efficient utilizing DSP to optimally match amplifiers and drivers in a controlled environment. Reliable 1.5kW dual voltage Bang & Olufsen ICEPower modules deliver premium audiophile sound. The powered AL10A and unpowered AL10X in Baltic birch with metal grille and steel rigging hardware weigh just 80 and 72 lbs. respectively, which means that eight cabinets may be flown on 650 lbs. rated Genie Lifts.



in both powered and unpowered versions, A-Line offers its customers the best of both with the world's first Hybrid-Active system. The AL10X unpowered extension is optimized to be powered by the AL10A. Enclosures, available from 90 to 140-degree wide dispersion, can be combined for maximum

coverage from back to front. Factory-supplied DSP programs match enclosures with subs and other applications. Compact design means effective arrays up to six units high can be ground-stacked on specially designed wheelboards with adjustable stabilizers. Up to eight can be flown from Genie lifts and up to twenty-four boxes can be flown in a single line with

pattern control down to 50 Hz! The AS215A subARRAY may be added for larger format applications where four-way operation is desired.

- Convenient Believe it or not, there are actually some audio professionals who run sixteen box line array shows by themselves using A-Line arrays. That's how easy it is to set-up and operate an AL10 system! Wheelboards up to six units high roll-in, attach quickly to hoists or Genie forks and lift right off the ground into place. The patented Easy Alignment System (EZAL™) allows the user to individually aim and focus boxes up to 5 degrees under load, without decoupling, making A-Line the fastest, easiest and safest line array system setup avail-Stacked three high, 1kW powered LS218A subwoofers, easily roll into place and operate on the ground from specially designed wheelboards. Computer networking is also available for remote control/monitoring, along with Cobranet digital audio network connectivity with analog backup.
- Cost-effective Designed and built in the U.S.A., an AL10 system costs less than comparable products from the competition. A-Line's unique Hybrid-Active system makes it even more affordable. All woodworking and metal fabrication is done in-house. Proprietary drivers and components are selected and sourced from the best manufacturers in the world to meet demanding performance standards, not because we have a division that makes them. Also, A-Line puts its money into its products, rather than expensive hospitality suites at trade shows. All aspects of field transportation, operation and servicing are carefully considered. An AL10 system is VERY TRUCK-PACK FRIENDLY. While LS218A subs include tiltback wheels, they may also be transported

three-high and loaded six across inside the width of a truck. Four wheelboards, four to six high, of AL10s also fit inside the width of a standard truck interior. Therefore, a twenty-four box AL10 line array with eighteen LS218 subs takes up less than ten linear feet inside the trailer AND you

Truck interior. Therefore, a twenty-four box AL10 line array with eighteen LS218 subs takes up less than ten linear feet inside the trailer AND you ory-supplied can leave the amp racks behind. An A-Line with subs and Affiliate reported substantial savings in fuel,

equipment and labor because events previously requiring five trucks now only need four – a 20% savings.

All things considered, A-Line speaker sys-

tems save labor, time, space and energy...which add up to a lot of green.





A-Line Acoustics

510 East Washington St Corry, PA 16407

Tel: 814-663-0600 Fax: 814-664-7429 www.A-LineAcoustic

www.A-LineAcoustics.com
BMayer@A-LineAcoustics.com



"It's a no brainer!"

Phil Papotnik is serious about his sound. As owner of Raven Sound, a major pro audio provider in Western PA, he needs serious gear that will cover everything from corporate events and theatre, to festivals and concerts. We talked to Phil about why he chose A-Line Acoustics for all his line array systems.

"First, we found that A-Line's AL10 powered system uses top-shelf components like B&C drivers and built in ICEpower® amps complete with DSP. Alternating powered and unpowered AL10's in a line array gives us a hybrid active system eliminating the need for power amp racks.

Next, we found these AL10's lighter and more compact than competing designs. We can set up 6 on the ground, 7 on a Genie Lift or up to 24 in the air. With the EZAL levers on the sides, we can focus them under load without dismantling the array at all. So we can set up in less time with fewer people!

The AL10's also come in 90° and 150° wide dispersion models so we can combine them to achieve better throw to the back, while providing more even coverage up front.

Then there's the sound — the definition, clarity and vocal transparency — FANTASTIC!
Finally, they actually COST LESS!
When you add it all up, it's a NO BRAINER!"

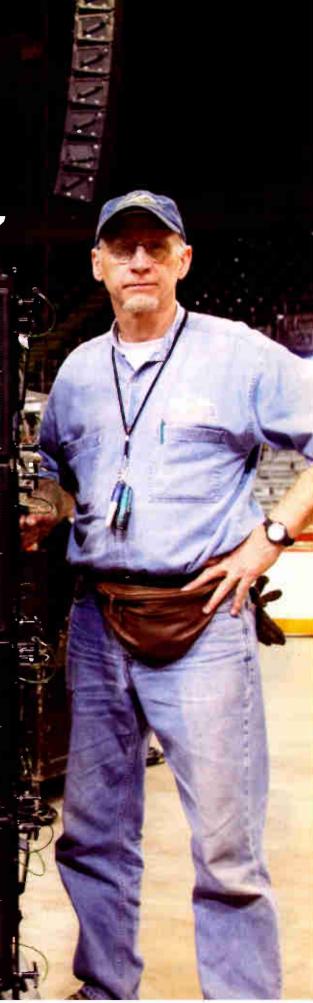
Thanks Phil, may we quote you?

Get the whole story at www.a-lineacoustics.com

A-Line Acoustics Corry, PA, USA 814-663-0600



World Radio History



POST

The latest news and products

NEW PRODUCTS

CEDAR DNS2000 Dialogue Noise Suppressor



Say goodbye to unwanted noise, and say hello to a new PC interface for CEDAR's popular

DNS2000. Actually, say anything you want in any situation you want, as this dialogue noise suppressor was previously only available to Mac users of Pro Tools but now allows any Mac or PC Pro Tools box superior noise reduction in film, live TV and radio broadcasts. It's even been used by forensic investigators to enhance speech caught on tape. What's more, the unit offers intuitive control for speed and simplicity and, as the 1RU rackmount processor itself is platform generic, it is completely flexible – use it with PC software one day, and a Mac (Intel or PowerPC) the next. Key features include timecode automation, less than 0.2 ms latency (so there's no loss of lip sync), and 24-bit I/O and dual 40-bit f/p DSPs for better audio quality. control and selectivity without hogging your host computer CPU. It couldn't be any clearer - this is the noise suppressor for the uncompromising engineer.

PRICE: \$7.972

CONTACT: CEDAR Audio Limited | ≈ 207-828-0024 ⊃ www.cedaraudio.com

SONIFEX RB-SC2 Dual Sample Rate Converter



There's no reason why the warmth of an analog recording can't be carried to its digital conversion

process. And you'll warm up to the RB-SC2 dual sample rate converter quickly, as this 1U rack-mount produces AES/EBU, S/PDIF and TOSlink optical level digital audio outputs from balanced optical level digital inputs at sampling rates that can be set by internal clock or various external synchronizer sources. You'll enjoy being able to sample at frequency rates as high as 192 kHz. And its special X-Lock mode lets the unit function as a full bi-directional sample rate converter by synching converter 1's output with converter 2's input, and vice versa. Optional video synchronizing boards are available, and a serial RS232 port is included so that the RB-SC2 settings can be controlled remotely. In this dual both sides win.

PRICE: TRA

RYCOTE S-Series Windshield System



For those engineers recording on location, you'll want to catch wind of this Gloucestershire-based company's innovative new mid-priced entry in wind suppression and enhanced usability. The S-Series design involves a one-piece web suspension made of durable, flexible, lightweight plastic, and able to accommodate microphones from 19 to 25 mm in diameter. While the windshield uses a twistlock system to attach 100 mm diameter pods (of different lengths) to the chassis ring, adding a faux fur covering and >30 dB wind suppression. An XLR tail with Neutrik NC3, gold-finished connectors and pistol grip handle are standard.

PRICE: \$349 - \$379

CONTACT: Rycote | **☎** +44 (0) 1453 759338 **⊃** www.rycote.com

XLNT IDEA Nexis 100AP CD/DVD Publisher



A rockin' demo deserves a rockin' look, and the Nexis 100AP CD/DVD Publisher can offer you just that. Print and burn up to 100 discs in one session with easy, hands-free, unattended operation, featuring the AutoCal calibration and Natural Pick guarantee of a 100-percent duty cycle. Package includes integrated oneclick software, high-speed DVD/CD burner, four-color 4800 dpi inkjet technology and up to 100-disc capacity. Nothing rocks like easy of use, and the Nexis 100AP uses non-proprietary inkjet cartridges available at virtually any office

products store. DiscWorks software means you can easily setup multiple burn-and-print jobs, or rip music CDs with your favorite MP3 software such as iTunes or Windows Media Player — all in a compact, attractive desktop unit. Self-publishing never looked so professional.

PRICE: \$1,595

CONTACT: XLNT Idea | ± 800-479-1689 ⊃ www.xIntidea.com

Film composer John Ottman (pictured), no stranger to scoring stages, has installed a pair of Klein + Hummel O 300 D speakers and a K+H O 800 subwoofer in his private facility, which he used while scoring this summer's nigh-energy Fantastic Four: Rise of the Silver Surfer.

MEV

NEWS.

NEWS

EWS

NEW

S



The Merging VCube and New York's Sound One have found themselves in sync. Needing a computer-based video playback system that could handle almost any modern file format during dubbing and re-recording, Sound One chose Merging Technologies for high-resolution materials.

AMS Neve was recently called in to Hollywood's Widget Post studios to perform an upgrade on the DFC console. It has been upgraded with Gemini audio processing, Encore Plus automation and a console-wide TFT meter-

Production studio owner Andy Prinz has incorporated Audient's SUMO Summing Amplifier in his Zurich facility. After tracks are edited in Nuendo the SUMO's internal compressor/limiter and eight fully balanced stereo inputs/outputs (plus AES/SPDIF up to 192 kHz) are used for the final mix for Prinz' record label, Offshore Music, as well as mastering for the Swiss arm of EMI Music. London-based producer Gareth Johnson also used a SUMO whilst mixing tracks for the Teenage Cancer Trust concerts' DVD, Virgin Radio broadcast and iTunes downloads.

The second installment of Carlos Garza's "Sampling of Sound Design Tools" (teased in PAR June 2007) is being postponed to a future issue. Plese follow this very "Post" section regularly to ensure you don't miss the other half of this very exciting round-up.

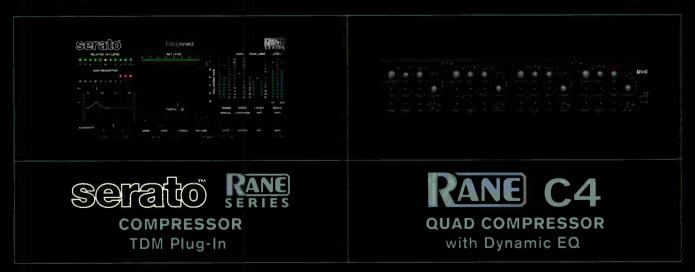
www.proaudioreview.com

"The Serato Rane Series Dynamic EQ is fantastic. This is one tool I want to take with me everywhere."



:: GREG NELSON, FOH: Pearl Jam and Incubus

AVAILABLE IN YOUR CHOICE OF FLAVORS - SOFTWARE OR HARDWARE



www.serato.com

www.rane.com

IN THE CONSOLE OR IN THE RACK

by Mel Lambert

Sound for Animated Feature Films

he world of animated features has changed dramatically during the past decade. Developments in computer-aided imaging and motion-capture techniques provide movie directors with the ability to create realistic environments and human characteristics. All of this helps the audience suspend disbelief and be cast firmly into the action — any self-respecting director's aim. But this paradigm shift also affects the multichannel soundtrack that sound designers, supervising sound editors and rerecording mixers are being asked to prepare.

"In the past, during the days of Disney and early pioneers of full-length cartoons, sound-tracks were larger than life with heightened sound textures that drew people into this stylized cartoon world," recalls Technicolor Sound Service's Richard Anderson. "But now we are being asked to re-create, from scratch, a highly realistic sound world that the animated character will inhabit. In other words, our sound-tracks need to sound more like live-action films, with particular attention to fully enveloping environments through which the CGI characters are moving. It's a wonderful challenge to achieve that reality with sound."

Randy Thom, director of sound design at Skywalker Sound in Marin County, Calif., agrees. "Directors are looking for an enhanced sense of realism in soundtracks, which means that we have to start from the basic voice track

[used to provide sync references for CGI elements] and generate everything else." Thom worked on *The Iron Giant* and *The Invisibles*, for which he garnered an Oscar nomination. He is currently enjoying the success of *Ratatouille*, a hit featuring his work as supervising sound designer and mixer.

Tom Myers also works at Skywalker Sound, where he handled sound design and co-mixing functions (with Gary Summers) on *Cars*, last year's blockbuster ani-



Remy, cooking rat in Ratatouille

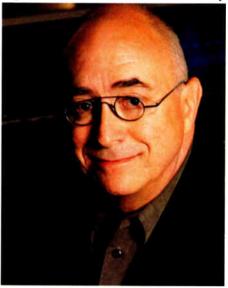
mated feature from Pixar/Disney, co-directed by John Lasseter and Joe Ranft. Previous animated features include *Toy Story, Toy Story II* and *Monsters, Inc.*

"For Cars, beyond the overall sounds of these automobiles, we needed to generate individual personalities for each of the characters," the sound designer stresses. "John Lasseter was adamant that each car would sound realistic. We carefully studied the artwork for each character with John and developed a list of each

one's parameters. If a character was portrayed by a '59 Chevrolet Impala, for example [Ramone's persona, voiced by Cheech Marin], then the engine and transmission had to sound right; the same for Doc Hudson's Hudson Hawk [voiced by Paul Newmanl Lightning McQueen [voiced by Owen Wilson]. We recorded a vast number of [four- and six-cylinder] cars for literal accuracy, while never forgetting that they needed to be dramatically accurate, as well."

The production also called for a number of Foley recordings, Myers recalls. "John Lasseter came to the stage and worked closely with the Foley Walkers. He would explain each character to them and help the [artists] understand how to differentiate their mannerisms in sound.

"We strived to maintain logic for the life inside *Cars*," the sound designer continues. "In reality, we developed our own arbitrary Law of Physics. For example, we decided that the engines would not run all the time — they



Skywalker Sound's Randy Thom

would only run if the car moved five feet or more. That meant that we could maintain reasonable dialog levels without them being drowned out. We also developed incidental noises for car rolls and suspension noises as they moved about."

NEARING FAR FAR AWAY: AUDIO AGAINST A CHANGING PICTURE

Supervising Sound Editor Richard Anderson's latest project is DreamWorks Animation's *Shrek The Third*, co-directed by Chris Miller and Raman Hui; previously, the seasoned editor also worked on *Over the Hedge, Madagascar* and *Shark Tale*.

"I asked a lot of questions during the initial spotting sessions [on the picture editor's Avid with partially completed animated scenes]," Anderson says. "I needed to know what the surfaces would be — hard floors and carpet, rock or sand, for example — as they weren't rendered yet in many scenes. Co-supervisor Tom Jones and I developed our laundry list of sound effects needed for each scene. Then, using our database of stored sound effects [at Technicolor Sound Services] we transferred them to Pro Tools [DAWs] for editing and synchronizing against picture." Sound assistant Mark Coffey



Tom Myers, Cars sound designer

also recorded real-world effects for the production, including wagon-wheel rolls.

"Foley elements, which we try to record once the picture is locked, included swords and other props," Anderson continues. "These were put into premixes with similar cut library sound effects so that the re-recording mixer can deal with all of the 'flavors' of one item at the same time. We then recorded footsteps sounds and pre-dubbed them as a separate pass. Our Loop Group recording session for miscellaneous voices was held back until the last possible moment [because of waiting for shots to be animated].

"The biggest challenge for an animated feature like Shrek The Third is logistics," the editor concludes. "How to organize the project against a changing picture and where some of the physical surfaces in the picture are is still under development. For example, the animator may decide to change one element in a shot, such as a sword hit, but leave rest of the shot and its length the same. We then need to resync that sound for that shot. The change note from the picture editor's Avid might not list that internal change since the shot length is the same. DAWs make life a lot easier, but it still takes careful attention to hundreds of details."

AN AUDIO WAVE: DESIGNING WITH STORYBOARDS AND RENDERINGS

Steve Ticknor was supervising sound editor and sound mixer on Summer 2007's Surf's Up," co-directed by Ash Brannon (Toy Story and Toy Story II) and Chris Buck (Tarzan, Pocahontas and Little Mermaid), while working at Sony Pictures Studios in Culver City, Calif. Ticknor joined the production in October 2006. The animated feature comprises a behind-the-scenes documentary at the annual Penguin World Surfing Championship, focusing on up-and-comer Cody Maverick (voiced by Shai LaBeouf).

"Most scenes were still in storyboard format," the sound designer recalls, "with maybe 20 percent in rough-rendered format. But I could get a sense of what the story involved, and what was going to be needed in the final soundtrack."

Pre-dubs were performed on a dual-operator Digidesign ICON D-Control digital console on Sony's Stage 7, with finals scheduled to move into the facility's Anthony Quinn stage, which sports a Harrison MP3 Digital Production Console, working with re-recording mixer Tateum Kohut. Stage 7's ICON accommodates some 400 mixdown channels and is powered by Pro Tools | HD7 Accel systems per side.

Ticknor recalls the Surf's Up soundtrack's biggest challenge was developing realistic water and ice sounds. "There are a lot of scenes where water plays an important part in establishing the environment. So Martin Lopez, the sound designer on Surf's Up, captured a bunch of sounds while on vacation in Hawaii's North Shore. During three full days he made a number of stereo recordings on a portable DAT of the huge water that is native to Hawaii, as well as open ocean those sounds you find in surf - he went out in a two-seat kayak as far as possible from the beach. We also gathered watersplash sounds from

a local pool, plus everything we thought we might need from surf boards, churning water, sand and the rest."

To further add realism to these and other tracks, Ticknor decided to further process the sound using a "worldizing" technique. "We

Left-to-right: Ulrika Akander/dialog editor: Mike Benavente/ADR supervisor; Matt Hanson/sound assistant: Steve Ticknor/supervising sound editor and re-recording mixer: Sarah Bourgeois/sound assistant; Martin J. Lopez/sound designer; Jason King, Foley supervisor at Fox Studios' Howard Hawks Theater, in Century City, Calif.

took the tracks on a portable Pro Tools | LE system and DV Toolkit to El Mirage in the Mojave Desert and replayed them via an Mbox Pro and Genelec high-end studio monitors," the supervising sound editor continues. "Then we **ANIMATED FEATURES** continues on page 52 ➤

THE PERFECT MIX

hear back.

- · Unlimited system size
- · Local control of eight channels (stereo mix and six "niore-me's")
- · ADAT, analog, HearBus inputs are switch-selectable from front panel
- · Built-in DSP Limiter with threshold adjustment

· Master Volume

- · Stereo AUX In
- . Balanced Line-Outputs
- . Built-in Mic Stand Mount
- · Standard CAT5e power

mix back

- Affordable, flexible 16x12x2x2 monitor mixer
- · 16 Mic/Line inputs each with passive split and two stereo and twelve mono level controls
- · Pin 1 lift and Mic power switches per channel
- Dual switch-selectable effects sends
- Stereo Aux inputs
- · Channel Inserts
- . Four-band EQ with dual sweepable mids
- · Talkback with intercom and optional remote
- 16 master outputs each available as analog, ADAT, and HearBus

www.HearTechnologies.com

1-256-922-1200



July 2007 | Produdio Review | 51 www.proaudioreview.com

POST | Feature

ANIMATED FEATURES Continued From Page 51

recorded the sounds using three sets of stereopair microphones located 20 feet, 40 feet and 60 feet from the source, plus two additional mics, onto a Fostex eight-track hard-disk recorder." El Mirage comprises a dry lakebed, a basin, Shadow Mountains and Twin Hills.

"Because of inevitable aircraft fly-bys, we needed to visit the location twice," Ticknor

says, "but we secured some wonderful natural ambiences and sound slaps against the nearby mountains. To further extend our pallet of sounds we also worldized the tracks on the Sony Scoring Stage using the same mics at 20/40/60-foot distances which, because of the more damped environment of the enclosed sound stage, gave us a softer, warmer feel with

a natural acoustic. Combined with the original sounds onto our pre-dubs, these extra sounds — mixed at reasonably low levels — added a sense of distance to scenes where the characters are out in the surf looking back to the beach, for example, or to form a more dense sound without resorting to artificial reverb."

Ticknor recalls having some 250 tracks for the pre-dub, including music. "For the finals we will probably have 300 - 400 effects tracks running from our pair of Pro Tools DAWs.

"We held off recording Foley as long as possible since we didn't receive a locked picture until late in the production process," the editor offers. "Gary Hecker — a real artist — used 60 pounds of snow that we brought into Sony's Foley Stage to create realistic sounds along with surf board noises and plenty of waves."

AN OGRE OF A MIXING SESSION

Turning to mixing and blending together these various dialog, music and effects elements to create the soundtrack, Andy Nelson, re-recording mixer on *Shrek the Third*, confirms that directors of animated features are looking for a highly realistic soundtrack and one that does not sound like familiar cartoons. "The ground rules were that it should sound like a fairy tale come to life with very realistic sonic landscapes," he offers. Nelson also worked on the first two *Shrek* films.

For Shrek The Third, Nelson was working with his long-time co-mixer Anna Behlmer, who oversees effects mixing while Nelson handles dialog and music. "The dialog track [used to synchronize the animation] is always recorded dry on an ADR stage; my job is to bring that to life and make it sound more realistic," Nelson concedes. "I use subtle high-frequency EQ and level changes to suggest that the character is moving away or towards the audience. It's all done very gently using the [Howard Hawk Stage's AMS Nevel DFC digital console so that it doesn't distract from the action and sound artificial. I also use touches of reverb to place the voice in a correct perspective. My favorite tools are the Junger Audio De-esser, Lexicon 960 reverb for dialog and TC Electronic Reverb 6000 for music tracks."

The stage's AMS Neve DFC Digital Film Console connects via a Nexus AES-format and MADI digital router to a quartet of 64-channel Pro Tools | HD2 Accel workstations, plus additional Pro Tools DAWs from dialog, effects and music editors. The router handles approximately 500 AES-format inputs, 200 AES-format outputs, 120 analog inputs and 104 analog outputs. Recording is handled by a single 48-channel and two 16-channel Pro Tools | HD2 Accel systems connected via fiber optics to a 6.5 terabyte storage area network. All Pro Tools playback



| Produdio Review | July 2007 www.proaudioreview.com

systems, recorders and edit stations are networked via Gigabit Ethernet or GIGE Fiber. Picture source is from a Fairlight Pyxis random-access system offering SDI- and HDSDI-format sources from a fiber-channel SAN.

Nelson prefers to keep the dialog track primarily in the center channel. "I don't favor panning dialog tracks, unless a character is maybe moving off-screen and I will follow the movement into the left or right screen channel. But it's always with gentle pans — nothing dramatic. As a personal preference, I use reverb and ambience rather that panning to add distance to the character and place them in a more appropriate environment."



Anna Behlmer/effects re-recording mixer; Richard Anderson/supervising sound editor; Andy Nelson/dialog and music re-recording mixer at Sony Pictures Studios' Stage 7, in Culver City, Calif.

The re-recording mixer also avoids excessive use of the subwoofer. "Too many films rely on low-frequency tracks," Nelson considers. "But because theaters do not offer consistent LF performance, I prefer to not rely on the subs, and use the full-range loudspeaker channels instead."

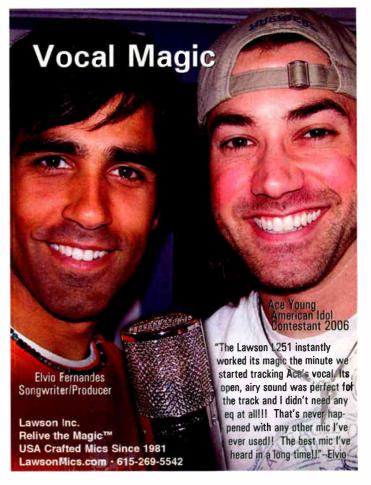
Nelson concedes that the majority of 5.1-channel ambience comes from the effects elements, handled by Anna Behlmer. "Good effects mixing places the voices into a realistic environment rather than just existing as talking heads," Behlmer offers. "I received hundreds of tracks from [supervising sound editor] Richard [Anderson], and divided these into hard and background effects. The latter I blended into five background pre-dubs, grouping together 'Winds,' 'Water' and other similar elements. That gave me the flexibility to add or subtract these groups into the 5.1-channel soundtrack. From working on *Shrek* and *Shrek* 2, I knew that we would have several scenes in the swamps so I built several very dense, very intimate backgrounds for those sections to retain a sense of continuity from the previous films. I built everything as 5.1-channel ambiences, using stereo background elements that can be moved into the front or rear soundstages, or to favor the left- or right-hand sides."

Behlmer considers Foley to be particularly important for animated features such as *Shrek The Third*. "We need to put back all of the detailing that would normally be captured with production dialog," she stresses. "We need to hear the clothing movements, leather vests, boots and the rest – otherwise you lose that added level of reality that makes the on-screen action come to life."

As will be readily appreciated, animated feature films such as *Shrek the Third*, *Surf's Up*, *Cars* and the latest animated blockbuster, *Ratatouille*, come to life on the screen through care and attention from a team of talented supervising sound editors, sound designers and re-recording mixers. It is a make-believe world that benefits from a great deal of sonic seasoning to ensure that it plays like a live-action production.

Mel Lambert has been intimately involved with production and broadcast industries on both sides of the Atlantic for many, many years. He is principal of Media&Marketing, a Los Angeles-based consulting service.





www.proaudioreview.com

July 2007 | ProAudio Review | 53

World Radio History

by Stroher Bullins

Polarity Post: Heroes for Higglytown

he fictional world of Higglytown is literally teeming with heroes waiting to be discovered. Thus, Patrick Fitzgerald — a senior sound designer, mixer and (with Chief Engineer/President Roger Wiersema) co-owner at Polarity Post in San Francisco — could be considered quite a hero himself. For each episode of the Disney-distributed animated hit series Higglytown Heroes, Fitzgerald wrangles, creates and mixes all audio elements to entertainingly reveal just how each citizen of this town can be, in just the right moment, a hero.

Higglytown Heroes balances fantastical animation (all people are CGI Russian nesting dolls) with organic sounds, as well as some traditional cartoon-style aural anchors. Together, these elements capture the attentions of late-preschool-aged children while allowing each story line to be easily understood and relatable.

For four years now Fitzgerald and his team at Polarity Post have handled both sound design and final mix duties for *Higglytown Heroes*. Each episode breaks down into two "elevenish minute" stories per half-hour. All dialogue is first recorded and edited in Los Angeles and sent to Wildbrain for animation. Polarity Post then receives edited dialogue tracks from the animators, along with three elements from three different composers per

story: an underscore for the entire show and two songs. One — the "dilemma song" — and two — the "hero song" — explain the task at hand, and which Higglytown Hero will render the situation, respectively.

CREATIVE FOLEY FOR REALISM

Fitzgerald had a desire from the very beginning to make the Higglytown Heroes

world a very real place to its viewers. After all, he explains, its look is very much "3D CGI," so "the world wanted to be real. It's not a cartoon in the sense of zany, weird characters — other than no one in Higglytown has legs — so my approach was to make it feel like a real place."

This required lots of Foley for each episode, organic background ambiences and a realism that resembles a live action drama. Then, purely cartoonish sounds would be used to garnish the "realistic" audio base. "To me, real-life sounds grounded it in reality," Fitzgerald offers. "On top of that realistic layer, we could do more cartoony type things — zips, slide whistles and weird, kooky

things. I would occasionally do callbacks to early Hanna-Barbara and early Warner-Bros. kind of sounds, and hopefully the younger generation of viewers will be introduced to

that classic cartoon audio. There is such a palette of established sounds in the cartoon world that it's hard to actually break away from that."

AUDIO FOR CHARACTER MOVEMENT

Fitzgerald cites one of the most interesting aspects of sound design for Higglytown was in its character movements. After all, when the main characters don't have legs but must still move, you have to get a bit creative. "That was probably the most unique

thing that we did creatively and systematically," he offers. "Characters in Higglytown don't walk — they hop. The sounds alone weren't terribly involved; it was an approach akin to doing Foley footsteps. When it's right you'll know it and it doesn't call a ton of attention to the movement."

AUDIO TOOLS FOR HIGGLYTOWN

Fitzgerald works exclusively in Pro Tools | HD and the entire Polarity Post facility is either HD or TDM systems. "There's three of us here that work on the show, so we all use a combination of HD systems and TDM systems," Fitzgerald explains. "Edwardo Mendoza mainly focuses on dialogue pre-mixing, helps out with some sound effects, and will do most of the 2D effects. Brad Seminoff handles everything that carries from episode to episode, such as character movements, vehi-



MOTU MachFive

cles and thematic effects. He lays those things out and he'll create his stuff in MIDI tracks, firing off to the sampler. He also does Foley. I'll do whatever else isn't covered and mix."

Effects are laid out in various templates using the MOTU MachFive sampler. "It integrates quite nicely with Pro Tools."

While Polarity Post offers both a Digidesign ICON mixer and a Euphonix CS2000 console, a dedicated work surface is not in the mix for *Higglytown Heroes*—Fitzgerald works in the box. As a matter of fact, he insists that doing so saves him quite a bit of time and effort. "For instance, I may say, 'At this frame, I want this sound to be 6 dB hotter than at the last frame and 50-percent left.' It's a lot faster using a DAW when doing this kind of thing than working with a control surface."

Strother Bullins is the Features and Reviews Editor for Pro Audio Review.



Reignited NEW DIRECTIONS. NEW DEFINITIONS. NEW DEVICES.

The NAB Radio Show[©] promises to deliver a one-of-a-kind networking opportunity for station professionals representing all formats and market sizes. Discover what's hot, what's not, and how to boost your station's success. This year's focus is on:

- New Directions Find out how new technologies are changing your markets and learn what works (and what doesn't).
- New Definitions Gain insight into how PPM will revolutionize the way you define and track your audience.
- New Devices Go Digital. Discover multicasting and wireless data text options.

Register today for the most explosive week in radio!

"The NAB Radio Show technical sessions are very valuable for engineers. This year's RF Safety Course alone covers issues of extreme importance to all practicing engineers."



MY MX

Talmage Ball
Vice President of Engineering
Bonneville International



September 26–28, 2007 Charlotte Convention Center Charlotte, North Carolina

www.nabradioshow.com



by Frank Beacham

The Wonder Pets Rescue TV Sound

ows of glowing computers line the 18th century brick walls of a former gunpowder store, located a few yards from the site of the historic Fulton Fish Market at New York City's

South Street Seaport. The physical setting — a high-tech workspace for new media creation housed in a restored Greek Revival commercial building constructed in 1790 — serves as a metaphor for *The Wonder Pets*, a digitally-created television show that tips its hat to a musical authenticity digital technology has yet to emulate.

Currently the top-rated preschool program on commercial television, *The Wonder Pets* is part of Nickelodeon's preschool programming block called "Nick Jr." However, to categorize the series as interesting only to preschool children is about as foolish as claiming that ice cream has no appeal to adults.

Using a technique called "photo puppetry animation," a team incorporates live action video and digital photographs of miniature sets into an integrated animation workflow using Apple's Final Cut and Adobe's Photoshop and After Effects.

However, it's the sound — specifically the music — that jumps out on the first viewing of an episode of *The Wonder Pets*. While most modern television productions are now created by a single composer/musician overseeing a MIDI synth system, *The Wonder Pets* uses a live orchestra with human musicians playing real instruments. The difference is stunning.

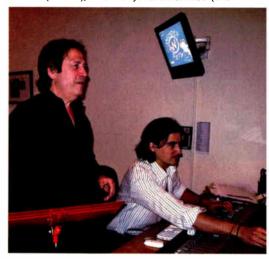
AN ANIMATED OPERETTA

Each special 11.5-minute animated story is actually an operetta with deep roots in the musical theatre. Because the show's animal characters travel the globe on a series of adventures, each story showcases the traditional music of the locale, and, in turn, the work of some of the finest international composers.

The Wonder Pets is the brainchild of Josh Selig, a Sesame Street alumnus (from child actor to international producer) who clearly has a deep love for both children's television and music. Selig's Little Airplane Productions formed a team for The Wonder Pets that no doubt would have even impressed the pioneering animators at Walt Disney Studios.

The show is about three classroom pets — Linny the Guinea Pig, Ming-Ming Duckling and Turtle Tuck — who travel the world saving other animals in distress. As is common in musical theatre, the characters often sing their parts as the story unfolds.

Animation is headed by Jennifer Oxley, the company's creative director, and music production is overseen by veteran producer/engineer Jeffrey Lesser, who has worked on the series with composers that include Robert Lopez (Avenue Q), Jason Robert Brown (Parade), Michael John LaChiusa (The



Jeffrey Lesser and Steve Robellido lead a fastpaced orchestral recording session

Wild Party) and Larry Hochman, who received recent Tony nominations for best orchestrations of Monty Python's Spamalot and the 2004 Broadway revival of Fiddler on the Roof.

COMPOSING PET(S) SOUNDS

Lesser — with extensive credits in feature film, television and music recording — brings a versatile background to *The Wonder Pets*. In film sound, he has worked with directors including Brian De Palma and Michael Wadleigh. In television, he produced and directed *Candid Camera* with Allen Funt.

Working behind a Control | 24 console with audio engineer Steven Rebollido, Lesser clearly enjoyed his interactions during a recent three-hour session with lead composer Larry Hochman and 12 orchestra members. This team — with Matt Henning as MIDI synthesist and Fred Barton as orchestrator — is one of three units that Lesser supervises during a fast-paced season that will create 40 complete story segments.

A key to Lesser's sound is to showcase the live musicians (from 12 to 17 per story) who are mixed with some virtual instruments in



the background to boost the orchestra's "size" to about 35 instruments.

"MIDI synth in general has a buzzy, bright quality that by definition interferes with the voices," Lesser said. "We use warm acoustic instruments and approach the mixing and recording as a combination of a Broadway show and a feature film. Because this is a mini operetta, we've elevated the music to a very important role in the series."

Prior to the orchestral recording session, weeks of preliminary work — from script creation and music composition to character voice and sound effects recording — has been completed. Lead composer Hochman, who has worked on almost all of the episodes including the pilot, noted that the show is unique among television projects because he is asked to write a complete score after the script is finished.

"Usually you write only the portions that are to be sung and then animated, and then you'd connect it with some minimal underscoring," Hochman said. "But we actually write a complete 10-minute piece of music. So the show is music driven."

After Hochman composes a piece he records it, playing piano and performing both the singing and acting parts. That performance is used for place makers. The actors learn their lines by listening to his rendition of their parts. After the actual dialogue and singing parts are recorded, Hochman's voice is replaced, leaving only the piano; the full orchestra replaces the latter.

Using MIDI, Matt Henning creates a barby-bar real-time structural map of all elements of the episode from Fred Barton's orchestration. This structure becomes a template for every line of dialog, sound effects, and all musical elements as audio is tied to picture. During the recording session, Henning sits with the orchestra as Barton conducts to make sure any changes conform to the MIDI map. "I make sure whatever I put in place agrees with what they do," Henning explained. "Sometimes live players may put in something that we miss."

Hochman works in the composition process with MOTU's Digital Performer, an integrated MIDI and audio sequencer. "I convert my work to a standard MIDI clock and email to Jeffrey a file that has all the tempo and click information," Hochman said. "He imports that file into Pro Tools."

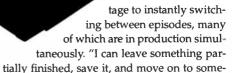
LITTLE AIRPLANE'S TOOLS FOR AUDIO FLIGHTS (OF FANCY)

Little Airplane owns two Digidesign Pro Tools | HD systems. The 16 Focusrite Class A mic preamps in the Control | 24 system are used for the orchestra's close and room microphones. Lesser chooses from a selection of Neumann, AKG and Rode microphones, preferring AKG's C1000s for strings (room), Neumann KM184 for violins (close), Rode NT5 and NT1000 for woodwinds, and the Audio-Technica AT3035 on brass.

Hochman, who sits among the orchestra

members during recording sessions, converses over a \$20 Radio Shack PRO-302 mic with Lesser, who stands behind Control | 24 console. "I grew up with boards in the analog world and to me it's very comfortable to be working on a board rather than just a computer screen," Lesser explained.

The Pro Tools system also offers Lesser the advan-



Digidesign

Control i 24

thing else. When I return, things are exactly as they were." Lesser normally avoids the use of exotic plug-ins for the production, staying mostly

Lesser normally avoids the use of exotic plug-ins for the production, staying mostly with reverb and compression tools from Digidesign and Waves. Some episodes, however, are an exception. For example, there's an upcoming episode where *The Wonder Pets* travel to Liverpool to save a band of baby beetles.

Hochman, an avid Beatles fan who also served as composer of that episode, tapped into his deep knowledge of Beatles technical lore for the show. "We used a Rickenbacker six string (Model 325) with the bright pickup just like the one owned by John Lennon," Hochman recalled excitedly. "We used amp simulators to get the right sound and (Beatles producer) George Martin's original compression. We did A-B comparisons with the original Beatles recordings to make sure we had it right."

OVERCOMING LIMITATIONS OF BROADCAST AUDIO

As with all television productions that seek sonic excellence, *The Wonder Pets* must deal with the compression limitations of broadcast sound. Lesser got his first dose of reality last year in the opening scene of the show's first episode.

"We tried to mix certain vocals in the background with sound effects," Lesser recalled. "When we heard it on TV the broadcaster's 'vocal lifter' changed the character of the scene. So we got the message that we must compensate for TV sound."

One way to avoid trouble in TV land is to avoid high impact sounds. "Big thunder or timpani drums on the low end seem to grab those compressors and pull everything else

WONDER PETS continues on page 58 ➤



Left to right: Josh Selig - Wonder Pets creator, Sofie Zamchick - Linny the Guinea Pig, Teala Dunn - Turtle Tuck and Danica Lee - Ming Ming Duckling

www.proaudioreview.com

July 2007 | ProAudio Review | 57

POST | Feature

WONDER PETS Continued From Page 57

down," observed Lesser. "We try to find a comfortable balance between the vocal and the music. And we learn ways to fool the compressor. The goal is to make the show sound more like a feature film than television."

As word of mouth about *The Wonder Pets* spreads through New York City's community of composers and musicians, Selig and his team are finding it easier to attract major talent to the show in its second season.

During a recent recording session, Jerry Bock — music composer (with lyricist Sheldon Harnick) for *Fiddler on the Roof*, the classic Broadway musical that opened in 1964 — observed the production from the con-



trol room. The producers are preparing an episode about a fiddler crab that insists on

playing his fiddle on a roof. They want Bock to compose the music.

"Every show is fresh and we seek composers who have connections to the locale or subject in each story," Lesser said. "So far we've been lucky, attracting many top composers and musicians."

There's one big fish, so to speak, that *The Wonder Pets* still dream of catching: musical theatre's living legend, Stephen Sondheim. He has an open invitation to compose music for the show. Perhaps he'll be interested if they decide to do an episode called "Sweeney Cod" or "Gypsy Moth."

Frank Beacham is an independent writer/producer living in New York City.





DESKTOP AUDIO

THE PROFESSIONAL'S SOURCE



real world solutions from industry professionals!



LIVE

The latest news and products

NEW PRODUCTS

AKG Wireless C 5 WL 1 and D 5 WL 1 Mics

Short wires giving you a short fuse? Soothe that short temper wih the musical and spatial freedom of new wireless versions of the C 5 and D 5 vocal microphones condenser or dynamic microphones — featuring dent-resistant spring steel elastic wire mesh caps for safe protection of the capsule element, as well as integrated shock absorber systems for compensation of handling noise (and smooth response, highest gain before feedback Laminate Varimotion diaphragm on the D 5). Working with the WMS 4000 wireless system/HT 4000 handheld transmitter these mics are feature-packed with backlit display, battery life indicator, silent mode, environment scan, auto setup frequency search, rehearsal functions and full compatibility with HiQnet. It's sorta like that

PRICE: TBA

CONTACT: AKG USA | = 818-920-3212 > www.akgusa.com

A-LINE EMMA Solutions-1

Bruce Willis movie: Live (wire) free, diehard.



When you want to bring audio and video to life, EMMA Solutions-1 brings the power to you. This compact, affordable, fully powered, true line array system can be easily set up and operated by just two people, yet can cover large ballroom-sized venues and theatres. Its wide 100-degree dispersion coverage and efficiency reduces the need for satellite point source speakers in most A/V applications. Built-in Bang & Olufsen ICEpower

modules with DSP/networking options eliminate the need for heavy amp racks. The system features a two-way 806A module with ribbon drivers — delivering highly intelligible coverage for theatres, houses-of-worship, clubs and social halls — and a matching S410A subwoofer. The low-profile columns can easily fit in a minivan, yet be stacked, wall mounted or flown to blend in with stage sets for more effective presentations. **PRICE:** \$8,799.

CONTACT: A-Line Acoustics | = 716-524-4084 ⊃ www.a-lineacoustics.com

QSC PowerLight3 Amplifiers



Want to shun the disruptive apparitions of bad audio? Well, make like the movie Poltergeist and "go into the light!" with QSC's new PowerLight 3 range. Offering full compatibility with advanced digital processing and QSControl, the PL325, PL340 and PL380 models range in power from 1250 watts to 4000 watts per channel at two ohms. And all this in two-rack space chassis that are compact

and light enough to carry around with a touring rig. No matter how high or low the power you need, you'll be wielding PowerLight switchmode power supply – offering highest efficiency and improved audio performance. Plus, you'll be the king of the soundboard with DataPort supports for remote computer/DSP control, detented gain controls for precise calibration, selectable input gains, clip limiters and low-frequency filters, removable knobs to prevent tampering, front and read LED status indicators and much, much more.

PRICES: \$1,649 (PL325), \$2,149 (PL340), \$3,599 (PL380) CONTACT: QSC Audio | ≠ 714-754-6175 ⊃ www.qscaudio.com

SOUNDCRAFT FX Mixer Range



Mixing is a little of this, and a little of that. Now Soundcraft offers a lot of options in a little footprint with the new FX16ii, EFX and MFX multipurpose mixers, each featuring no less than 32 legendary Lexicon 24-bit digital effects. The FX16ii is a refresh offering Lexicon effects, 26 inputs, 16 direct outputs, four stereo returns and rotating connector pod, which allows cables to be connected conveniently behind the

rack to save space. The compact EFX, available in eight- and 12-channel versions, updates the EPM with Lexicon effects, Soundcraft's highly regarded GB30 mic preamp, one aux send and one FX send on each channel. The MFX is the most versatile, with two group busses, two aux sends, one FX send on each channel and two stereo input channels, all offered in three frame sizes — eight, 12 and 20 inputs. Whether in a small theatre, music venues or house-of-worship, don't just mix it up, mix it up right with Soundcraft. **PRICES:** \$539 (EFX8), \$649 (EFX12), \$749 (MFX8), \$899 (MFX12), \$1,199 (MFX20), \$1,079 (FX16ii)

CONTACT: Soundcraft/Harman Pro | **=** 818-920-3212 **⊃** www.soundcraft.com

On July 7, 2007, the international Live Earth concerts set out to broadcast a shout heard round the world in support of environmental charities, and various microphone manufacturers helped make sure the message was loud and clear. Shure Beta 58s and wireless in London helped a reunited Spinal Tap "reportedly set off car alarms as far away as the Netherlands." Meanwhile. Sennheiser supported artists such as Madonna, the Foo Fighters, Bloc Party, Jos Stone and the Beastie Boys, as well as hosts including Chris Rock, Ricky Gervais and Eddie Izzard.

The Shure SM58 was also spotted on stage with Wilco at June's Bonnaroo Festival in Tennessee, and Jordin Sparks closed out the latest season of American Idol using Shure UHF-R wireless. While Sennheiser SKM 5000 microphones with SK 5212 beltpacks assured a performance by the cast of "Spring Awakening" perked up the 61st Tony Award ceremonies on June 10. And specially made white and gold SKM 935 evolution microphones helped Nelly Furtado (pictured) keep "Loose" and energetic on tour across Europe, the USA and Canada.

S

ΕV

MEWS NEWS NEW



Furtado's buddies Justin
Timberlake and Timbaland brought
"Sexyback" across the world with
a **Digldesign** VENUE D-Show system; rigs including versions of the
D-Show and associated racks were
also seen recently on tours by
Placebo and Sara Evans.

At KROQ's annual "Weenie Roast"

Digidesign, Shure and other manufacturers performed alongside

Korn, Linkin Park, Social Distortion
and Interpol, but the main act was
the new Martin Audio W8L

Longbow line array, which made its
West Coast debut powered by MA
4.2 amps and Dolby Lake

Processors.

60 | ProAudio Review | July 2007 www.proaudioreview.com

We worked hard on the PL380.

Now let's go out and play.



If owning the ultimate 8000 watt, Class D amp is your idea of fun, you're going to love the PL380.

Introducing the efficient, lightweight and well-protected PL380. It's powerful enough to handle even the largest drivers while delivering well balanced high and low frequency performance. The PL380 replaces existing amps without affecting signal delay and converts up to 85% of AC power into audio power—that's half the losses, half the heat of competitive Class D amplifiers.

Most importantly, the PowerLight™ 3 offers something no other Class D can: QSC quality, performance and value.



A full selection of input and output connectors. Switched LEDs show your settings at a glance.



LIVE | REVIEW

by Dave Rittenhouse

Yamaha Tn Series Amplifier

Commercial Audio enters the world of large touring amps.

Everyone seems to want power, power and more power these days. Shockingly, I am not talking about politicians; I am talking about engineers and their large touring amplifiers.

In today's market of high-power applications, engineers like line arrays and subs that can produce low end powerful enough to rattle your very digestive tract. That fact lends itself to an ever-increasing variety of manufacturers seeking to fill that never-endThe big questions are these: Will Yamaha stand out amid all the others with something "new?" Does a world that is moving to be more "green" need more power hungry amps? [Of course it does. — Ed.]

| FEATURES

Yamaha Commercial Audio has released three new amps for its Tn line: the T5n, T4n and the T3n — "T" is for touring, "n" is for networkable, and the power handling is 5k

EEEngine is its use in the conversion aspect of the power processing. The Tn Series of amplifiers realizes efficiency that matches Class D without compromising the sound quality of Class AB, accomplished by utilizing the power consumption and heat generation to make the amp more efficient with a greater topography and sound.

The networking aspects of these amps utilize two RJ45 connectors and a set of amplifier ID dipswitches for capability. The Yamaha ACU-16C amp control unit is required to network this amp with other Yamaha amps. It then becomes possible to network via CobraNet with the additional purchase of a Yamaha NHB32-C network hub/bridge; you can also send the audio inputs to your network over said Cobra-Net using the 16-channel DA converter housed in the ACU-126C. Yamaha backs the Tn Series with a five-year warranty.

IN USE

I was sent two T5n amps (\$3,799 each) to work with for a few months. First thing I



ing power void. Now, Yamaha — with 30 years of experience in amplifier technology — has stepped up with its first offering for large touring amp business: the Tn Series.



watts, 4.4k watts and 3.8k watts for the 5, 4, 3 models respectively at 2 ohms. Each Tn is a 2U dual channel amp with a frequency response of 20 Hz - 20 kHz and total harmonic distortion and noise of <0.1 percent. Damping factor is rated at 800 at 8 ohms at 1 kHz. Input gain is selectable from 32dB or 26dB. The back panel features XLR and EuroBlock inputs, as well as binding posts and Neutrik NL4 Speakons for the outputs. The front panel features a power switch, channel gain knobs and indicator lights for temp, protection, remote access, clip, signal and mute.

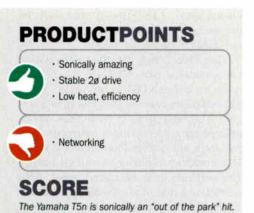
The Tn Series, like most other amps, can be configured for stereo, parallel or bridged operation. Also, the Tn Series is certified at 2 ohms by UL. With the new EEEngine (Energy Efficient Engine) amp drive technology, Yamaha claims a 50-percent reduction in power consumption and heat compared to conventional amplifiers; this amp does come with and require a L5-30 30-amp twist lock for the AC power. The true shining star of the

noticed when I pulled the amps from their boxes was the 30-amp power connector. I thought this was going to be a less power hungry monster, as that is still a burden imposed.

The next thing was the weight. The T5n weighs in at 14 kg., about 31 lbs. — not real heavy but not the lightest amp, either.

With that aside, I sent them out for my engineers to fire things up and see what happens.

A common use for these amps would be in line array and sub applications, so I sent them out with some EAW KF730s and SB850 subs. The T5n performed wonderfully and sounded amazing. I was very impressed with the tonal quality of the amps and the efficiency in the reproduction. The amps also remained cool on hot outdoor shows and the fans were quiet in their operation. The latter helps a lot in this town due to the large number of "talking head" events we do; it can be extremely embarrassing when the audience can hear the amp fans running over the mics



on the podium.

We also worked on a demo/shootout of some different speaker manufacturers in our shop for a couple of prospective clients. Our senior engineer Brian Bednar and I felt that the T5n would be the perfect match for some NEXO PS10 and PS15 cabinets. The amps performed very well under the stresses we put them through and the Yamaha/NEXO amp/cabinet combination sounded wonderful (and personally, I think the best); howev-

Overall, a well respected amp with great possibility

er, the client's decision was still unknown to me at the time of this review.

Most of the engineers who the used the amps seemed to like them as a standalone sub amp rack due to their clean sound and "beef" in comparison to some other higher powered amps. The comments that I received from my staff were all favorable.

In my personal usage I thought the T5n had decent power and good sonic qualities yet really missed when it came to the networking factor. You have to buy two different external pieces of gear (from Yamaha) to network everything together. I was not sent the networking components so I was not able to test these features. I do think that some amplifier manufacturer needs to step up and allow their amps to be networked for management and control via standard IP addressing — use a DHCP hub/router and that is all. I mean, if I can do this with the new RF microphones systems why can't I do it with my amps?

SUMMARY

Yamaha has made its mark by always being on top of the market and producing

quality products that have become industry standards. The T5n is an amp that is right on the cutting edge and at a price point that will bring quality to the club/touring market. I don't know that I will start to see it on

Our senior engineer Brian Bednar and I felt that the Tn5 would be the perfect match for some NEXO PS10 and PS15 cabinets.

upcoming riders, but I do know I won't be shocked if I do. As far as the Tn being a newer, "greener," less power hungry and more efficient amp is concerned, only time will tell. But if it has spurred a new way of thinking, then hopefully others will step up and follow suit.

David Rittenhouse is the senior account executive and A1 engineer at RCI Sound Systems in Beltsville, Maryland, and a regular contributor to PAR.



LIVE | REVIEW

by Dave Rittenhouse

Lab.gruppen **FP+ Series Amplifiers**

The FP+ Series offers big power and "amazing sound" in a small, lightweight and cool-running chassis.

Swedish amplifier manufacturer Lab.gruppen has built on its reputation with the new FP+ Series of touring amplifiers. These are monsters in power but not in size. But the questions still loom large that we must answer: whether the touring world needs more power and less bulk, or is this just another highly touted, fast-sell upgrade we all could have lived without.

| FEATURES

The FP+ Series comes in four different offerings: the two-channel 13000 (pictured) and 7000 models, 6500 W and 3500 W per channel at two ohms, respectively; and the four-channel 10000Q and 6000Q models, 2500 W and 1500 W per channel at two ohms, respectively. Off the top, the FP+ Series' most amazing feature is that all models are only two rack spaces small, including the four-channel ones. All amps are bridgeable: the four channel amps are bridgeable A+B and C+D channels with automatic -6 dB summing compensation. All amps are equipped with full LED meters per channel, VHF (very high frequency) monitoring filters and LED indicators, as well as open load and high temperature indicators on the front panel. The FP+ Series amps all weigh less than 27 lbs.

As you may assume from the amps' light weight, FP+ Series models are switch-mode power supply amps maintaining a consis-

tent output down to 90V (US). They come with a 30-amp twist-lock AC connector, and there are separate 230V or 115V (US) versions. The 7000 and 6000Q can be changed to a 20-amp plug, if required.

The FP+ Series are Class TD output stage amps; Class TD is a patented class that combines the high output of a digital Class D with the sonic signature of a Class B. This helps reduce the amount of voltage that is unused, which ends up in the output stage as heat and helps create a great sounding amp that doesn't weigh a ton. All FP+ Series amps come with XLR inputs and binding post output connectors; the 10000Q, 7000 and 6000Q ar also available with Speakon outputs.

Each amp in the series is equipped with a new VPL (voltage peak limiter). This will monitor and limit the output voltage of any channel to maximum peak output in selectable increments of 195V down to 38V depending on each amplifier's output rating. This is an additional piece of protection for your high-dollar cabinetry. Frequency response is listed as 6.8 Hz - 34 kHz ±3 dB with a signal to noise ratio of 112 dBA from 20 Hz - 20 kHz. All amplifiers are input gain selectable via dip-switches at the rear panel in dB steps of 23, 26, 29, 32, 35, 38, 41 and 44.

As with most — or at least any decent modern day amplifiers, the FP+ Series is

networkable. Lab.gruppen designed its proprietary NomadLink network to run on standard CAT-5 cables. Yeah!

The literal hub of this network is Lab.gruppen's new NLB-60E single rack space unit. One NLB-60E can control up to 960 amplifiers over the aforementioned single CAT-5, and up to 16 NLB-60E's can be utilized per system. That is control of up to 960 amps and 12.5 million watts (using all 13000 models) of power on a single laptop! The network does not carry signal, only control and real time monitoring of every amp within a closed loop chain. This means you can still use your existing snake (analog or digital) and your existing DSP processing without having to change your entire rig around your new amps ... a systems tech's dream. Every FP+ amp comes



standard with the ability already built in with both CAT-5 ins and outs - with no need to buy additional cards, only the hub (NLB 60E) and some CAT-5.

The FP+ Series is also RoHS compliant. The EU (European Union) has set new standards in the manufacturing of electronics,

LAB.GRUPPEN continues on page 66 ➤



SCILLS Get Serious

You've got a problem. You can't get the PA you need at a price you can afford.

It's got to get loud, and it has to sound smooth. It needs to handle the abuse of the road. Just like you.

You need power, you need detail, and you need clarity.

Your problem just got solved.

Ferrofluid Cooled Drivers

Community's Exclusive Carbon Ring Cone Technology*

Linear Phase Crossovers

Hand-laid Fiberglass Horns

Switchable Bi-amp / Passive Modes

DYNA-TECH™ Active Driver Protection

Music / Voice Presence Switch

Switchable High-pass
Output on Subwoofers

Community

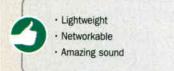
communitypro.com

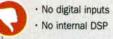
LIVE | REVIEW

LAB.GRUPPEN Continued From Page 64

requiring them lead-free. In today's push to make products more "green" this is great for the environment. Lab.gruppen also stands behind its products with an excellent six-year warranty.

PRODUCTPOINTS





SCORE

Truly a series of amps for the modern day

IN USE

I was able to test and use three FP+ Series amps — the 13000, 10000Q and 6000Q — over several months. The first thing to note, of course, is the power-to-weight advantage; no more heavy and bulky amp racks to run my line arrays or multiple bi-amp monitor mixes. When I can take out just three amps and run an entire EAW KF730 line array with SB850 subs or get four bi-amp mixes out of just two amps — something that used to take up four amps and eight rack spaces — I am beyond thrilled. Plus, the FP+ Series is so much lighter than the older amps we have now. This can save the back of an old soundman like me.

This is all good, but the

proof is not in

weight but in how they sound ... and that is amazing. Running the 13000 on 18-inch subs at four ohms is nothing but pure power (4400 W per channel). It had the sonic quality I used to get from the old, heavy, non-switching power supply

amps from the early '80s. I was disappointed

that Lab.gruppen did not list the damping factor in its specifications, but, that aside, I was very impressed.

Also, the amps did not get "cooking" hot like those older amps, and maintained a reasonable operating temperature throughout their use. I didn't have a large enough compliment of 13000s and 7000s to run RCI Sound Systems' larger EAW KF760 line array, but would still like to try that out in a rock concert-type setting. I am sure it would make a difference specifically in the overall sonic quality and sound of the rig.

I did have the ability to run the 10000Q and 6000Q in a larger variety of configurations. The first was with the aforementioned KF730 rig, and the 10000Q proved itself with ease. I can run a bi-amped rig in stereo off of one amp and make it sound better than it did with our other amps. Now, I know that this is not an amp "shootout," but we did do a head-to-head comparison in RCI's shop against a US-made amp we shall not name. Everyone we pulled in to hear this did not know which amp they were listening to and everyone chose the Lab.gruppen. This does not prove anything scientifically, but it does make an impression. I was simply overwhelmed with the reactions.

Furthermore, using the 10000Q (low) and the 6000Q (high) models for a set of four monitor mixes was thrilling. I had never heard RCI's EAW SM500 and SM200 sound so good. If you

right. Coming from RCI Sound — which does not already have a structure for remote amplifier monitoring or control but does not want to leap into a manufacturer's infrastructure that is confining all the way down to the signal path — this is the perfect fit. RCI is still analog and has yet to see a "solve all" digital solution at a good price. Speaking of digital, digital inputs of some sort would be a handy addition to the line, much like models from other manu-

...we did a head-to-head comparison in RCI's shop against a US-made amp we shall not name ... everyone chose the Lab.gruppen.

facturers that incorporate full DSPs in their amps. I have heard rumors of Lab.gruppen's work with Dolby Lake, but there is no evidence of that in this model. Maybe the future will hold even more for this series ...

SUMMARY

In the last few days and months, I have come to rely on these amps. As a company,

RCI already owns some
FP+ Series predecessors (fP 6400 amplifiers). They have been
rock solid workhorses over the last few
years. I would
love, and am
pushing, to
update the rest
of our old amp
racks with a variety of
amps from the FP+ Series.

It is a new day and age and truly a time to look to the future of less weight with a bigger punch. I feel that Lab.gruppen has hit the nail on the proverbial head. Let's hope this has come in time for these amps to replace the old heavy ones and save all of our backs!

David Rittenhouse is the senior account executive and A1 engineer at RCI Sound Systems in Beltsville, Maryland, and a regular contributor to PAR.

| REVIEW SETUP:

EAW KF730, KF300, SB850, SM500, SM200 Dbx 480

think that amps are not as important as good speakers, you have been mislead and are corrupting your signal chain in the process. Don't ever start thinking, "Yeah, our amps are old but they still work; we can still get by," because you will only hurt your client and your gear in the process.

Unfortunately, the NLB 60E hub was not able to be field-tested to network the amps together. This was a disappointment, as from my vantage point Lab.gruppen got the proprietary structure for these amps

66 | Pro4udio Review | July 2007



Is it a gamble each time you set up your powered loudspeakers in a difficult environment? Mids don't cut...bass too thin...amps overheat...audio shuts down? If so, you're obviously not using Yamaha MSR400s. Behind their subtle, professional appearance is a speaker capable of overcoming the most adverse circumstances. Put a pair in front of a good sized crowd and see for yourself. We're talking loud and clean.

MSR400...your best bet for great sound, reliability, and value.

- 12" long-throw woofer / 1.75" compression driver
- 400W combined max output power (300W + 100W)
- Usable as a short or long throw monitor...also flyable
- Add MSR800W powered subs for max punch and impact
- Under 50 pounds each



www.vamaha.com/livesound

CONTRACTING

The latest news and products

NEW PRODUCTS

AUDIO-TECHNICA SpectraPulse



Audio Technica has introduced its latest technological venture dubbed SpectraPulse, an innovative Ultra Wideband wireless microphone system specially designed for the installation market. According to A-T, SpectraPulse offers "secure wireless operation ... free from RF competition, frequency coordination and 'white space' issues."

In late June, A-T's Steve Savanyu explained and demonstrated SpectraPulse technology during a major rollout to the press and key

attendees at the kickoff of InfoComm 2007 in Anaheim, Calif. In his presentation, Savanyu hit all the high points of SpectraPulse, focusing on its unique application of patented Ultra Wideband (UWB) technology that allows it to bypass today's ever-bustling RF environment. Other highlights noted by Savanyu included its simplicity of use and security features.

According to the official press release, "SpectraPulse utilizes up to 14 simultaneous channels that operate flawlessly without RF competition, frequency hunting/coordination, white space issues [referring to the portions of the television broadcast frequency spectrum to be available after a US transition to digital TV in February 2009], or infringement from other wireless systems or radio sources."

For its application of UWB technology, A-T joined forces with Multispectral Solutions, Inc., a developer of Ultra Wideband systems for communications, radar and geo-positioning industries. By this, A-T's SpectraPulse is the first commercial sound implementation of UWB, a technology allowing "the wireless transmission of data in extremely short-duration pulses over a wide spectrum of frequencies," an instantaneous bandwidth of 500 MHz within the 6 GHz frequency spectrum. As a result, the signals travel in precisely timed sequences near noise-floor levels, as explained by A-T documentation. The decoding of the pulses require SpectraPulse technology, making it secure from signal interception and interference; further, A-T employs another level of security: an encryption package that meets the NIST-approved AES 128-bit standard developed by the good of US of A government for securing "sensitive material."

Product wise, Audio-Technica now offers the SpectraPulse Ultra Wideband wireless microphone system with the following components: the mtu101 microphone transmitter unit, the drm141 digital receiver module, the aci707 audio control interface, sep128 system encryption software and cei007 charger encryption interface, the later of which sounds a lot like a gadget built by Q for James Bond — no wonder this stuff has pro audio folks so excited!

 $\textbf{PRICES}: \ \mathsf{mtu101}, \ \$1,\!300; \ \mathsf{drm141}, \ \$8,\!600; \ \mathsf{aci707}, \ \$4,\!600; \ \mathsf{sep128}, \ \$2,\!000; \ \mathsf{cei007}, \ \$1,\!700.$

KALTMAN CREATIONS Spectran HF4040

RF Spectrum Analyzer



Nothing interferes with work more than interference. The new Spectran HF4040 RF Spectrum Analyzer can play your wingman in the narrowing wireless bands and help you avoid those meddling megahertz and gleaning gigahertz. This low-cost handheld analyzer is the world's first of its kind. It addresses Channel Selection, Interference Identification, RF System Monitoring, Testing and Exposure Limits. The HF4040 frequency range is from 10 MHz on up to 4 GHz, making it perfect for analyzing VHF, UHF and 2.4 GHz environments – finding peaks and zooming in on problems. It is a true RF analyzer, unlike scanners on the market that allow intermittent intrusion and slow scanning. The HF4040 is sold in a system package that includes an aluminum

carrying case, SMA full range antenna, batteries & charger (charger doubles as an AC adaptor), PC software for enhanced visual monitoring, a HyperLOG 700 MHz - 2.5 GHz precision directional antenna (other directional antennas available), and a mini tri-pod stand. Kaltman Creations has pre-configured the HF4040's 10 Hot Keys to address the more popular US RF bands: Wireless Microphone, Wi-Fi, Assisted Listening, Telco / Cellular, Intercom and Radio / TV Broadcast. The end user can also reprogram the Hot Keys as they see fit, plus manually set-up any frequency sweep range they wish. Trust the Spectran HF4040 to run interference on those important plays!

PRICE: \$1,400

Some athletes aim for the ring, others the cup, but the Chicago Cubs were recently enthused to be bestowed with the crown ... Crown CT Series amps (3000's and 600 models), that is, 11 of which were installed in Wrigley Field equipped with PIP-Lite cards, BSS Soundweb London, CobraNet and HiQnet.

MEN

EWS

NEWS

EWS

EWS

 \overline{z}

NEWS

NEWS

Sometimes folks say a concert is so loud or so clear it leaves them seeing God. And then sometimes audio is so crisp and spacious it helps you hear the word of the Lord, as is the case with recent Danley Sound Labs installations at Lakeview Baptist Church in Auburn, Ga. and Meadowbrook Baptist Church in Gadsden, Ala. Lakeview received two TH-115 subwoofers and two SH-50s hanging below a single Danley SH-100 in order to tame a 1,800-seat, 50 ft. tall sanctuary. QSC RMX Series amps power this center cluster, as well as additional SH-100's fed by a Yamaha M7 C148 console. The same models of loudspeaker and amp, tied to a Soundcraft Series 2 console. helped convert a gymnasium into Meadowbrook's new haven.

If loudspeakers could have a kid's TV show it might be called "Where in the World is D.A.S. Audio?" as the Spanish company has installed its Aero line arrays in Bolivia and Miami (that's sorta it's own country, ya know). At the Sonilum Convention Center in Santa Cruz de la Sierra, Bolivia, Aero 48 and 28A support live tours, banquets, exhibitions, trade shows and assorted audio/video extravaganzas. And in Miami's American Airlines Arena [pictured with Dave Vickery (Director of Broadcast Services), Jorge Arronte (Manager of Arena Sound and Matrix), and Ralph Avlarez (Owner, Interface Sound and Productions] eight CA-28A modules (two-way speakers with integrated amps) are optimized to deliver commentator dialogue to the Miami Heat's home court premium seats.



68 | ProAudio Review | July 2007

Standing the touring world on its ear...



FP+SERIES



The new FP+ Series from Lab.gruppen. From any perspective, it's the future of touring amplification.



Power @ 2 ohms. All models 2 U and 26.5 lbs

FP+SERIES Models





BUYER'S GUIDE | Live Sound Amplifiers

LAB.GRUPPEN FP10000Q

FEATURES: Four-channel; Class TD; 1,300 W per channel @ 8 ohms.

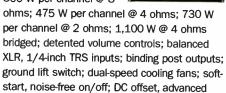
2,100 W per channel @

4 ohms, 2,500 W per channel @ 2 ohms; 5,000 W @ 4 ohms bridged; adjustable gain, voltage peak limiters; detented front controls; protection indicators; network monitoring. PRICE: \$6,945.

CONTACT: Lab.gruppen/TC Electronic at 818-468-8432, www.labgruppen.com.

NADY Systems Pro Stereo Power Amp XA-1100

FEATURES: Two-channel; 300 W per channel @ 8



protection circuitry; 2RU. PRICE: \$499.95.

CONTACT: Nady at 510-652-2411, nady.com.

PEAVEY Commercial Series CS 4080 HZ

FEATURES: Two-channel; 2,040 W per channel at 4 ohms; 1,250 W per

channel @ 8 ohms;

4,080 W @ 8 ohms bridged; two variable-speed

fans; 2RU; DDT speaker protection.

PRICE: \$1,599.99.

CONTACT: Peavey at 601-483-5365,

www.peavey.com.

CREST AUDIO CC 4000

FEATURES: Two-channel; 800 W per channel @ 8 ohms stereo; 1,350 W @ 4 ohms stereo; 2,000 W @ 2 ohms stereo; 4,000 W @ 4

ohms bridged; toroidal transformer; recessed.



stepped attenuators; variable speed fan.

PRICE: \$1,560.

CONTACT: Crest Audio at 601-486-2000,

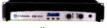
www.crestaudio.com.

CROWN CDi 4000

FEATURES: Two-channel; runs 2/4/8-ohm loads plus 70/140 outputs; 1,200 W per channel @ 4 ohms; 1,000 W per channel @ 70 V; onboard DSP; USB, HiQnet; advanced protection circuitry; LCD; 2RU; 19 lbs.

PRICE: \$1,800.

CONTACT: Crown Audio at 574-294-8200, www.crownaudio.com.



L-ACOUSTICS LA4

FEATURES: Four-channel: two inputs; 1,000 W per



channel @ 4 ohms; onboard DSP with preset library; sophisticated driver protection system; Ethemet I/O ports for remote

control/monitoring; lightweight 2RU; LCD.

PRICE: \$5,775.

CONTACT: L-Acoustics US at 805-604-0577. www.l-acoustics.com.

YAMAHA P Series II P7000S

FEATURES: Two-channel; 3,200 @ 4 ohms bridged; lightweight; Neutrik Speakon, 1/4-inch, 5-way binding post outputs; XLR, 1/4-inch TRS

inputs; high-efficiency EEEngine: sweepable high-, low-pass filters.

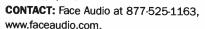
PRICE: \$969.

CONTACT: Yamaha Corporation of America at 714-522-9011, yamaha.com.



FEATURES: Two-channel; stereo, parallel mono and bridged modes; 650 W per channel @ 8 ohms; 1,200 W per channel @ 4 ohms; Class H; 2RU; advanced protection circuitry; balanced XLR, TRS inputs; Speakon NL4, binding post outputs; power up/down

muting; clip limiters. **PRICE: \$1,169.**



INTER-M Q-4300

FEATURES: Two-channel; 750 W per channel @ 8 ohms; 2,600 W @ 8 ohms bridged mono; variable speed fans (front-to-back); XLR, 1/4inch TRS input; 5-way binding post, Speakon output per channel;

on/off muting; advanced protection circuitry.

PRICE: \$2,360.

CONTACT: Inter-M Americas at 562-921-0313, inter-m.net.

D.A.S. SOUND PRODUCTS PowerPro PS-1400

FEATURES: Two-channel: 450 W per channel @ 8 ohms; 700 W per channel @ 4 ohms; 1,400

W @ 8 ohms bridged mono; steel 2RU; lownoise fans; low-weight



toroidal transformer; ground lift; voltage sensitivity; rear switches.

PRICE: \$850.

CONTACT: D.A.S. Sound Products at 305-436-0521, dasaudio.com.

YORKVILLE Audio Pro AP1020 **Power Amp**

FEATURES: Two-channel; 400 W per channel @ 4 or 2 ohms; 800 W @ 4 or 8 ohms bridged; toroidal transformer; steel 2RU; recessed knobs; front-panel status LEDs; balanced XLR.

1/4-inch TRS inputs;

Speakon, binding post outputs; advanced protection circuitry; userdefeatable peak limiter; "soft turn-on" circuit; variable-speed fans; rear-mounted ground lift, impedence switches.

PRICE: \$695.

CONTACT: Yorkville at 716-297-2920,

www.yorkville.com.

ELECTRO-VOICE TG-7

FEATURES: Two-channel; 2,500 W per channel @ 4 ohms; 3,500 W per channel @ 2 ohms: three-step Class H topology; microprocessorcontrolled management; optional RCM26 module for remote IRIS-Net supervision.

PRICE: \$4,450

CONTACT: Electro-Voice

at 952-884-4051, www.electrovoice.com.

FBT Kempton KA Series

FEATURES: Two-channel; 500 W per channel @ 8 ohms, 840 W per channel @ 4 ohms, 1,100 W per channel @ 2 ohms; 1,500 W @ 8 ohms bridged, 1,950 W @ 4 ohms bridged; Class H design; selectable 30 Hz low-pass filter; stereo, parallel, bridge switch.

PRICE: \$589.

CONTACT: FBT USA at

800-333-9383, www.fbtusa.com.

YAMAHA Tn Series

FEATURES: Two-channel; 2,300 W per channel @ 4 ohms, 5,000 W @ 4 ohms bridged (Tn5); 2,000 W per channel @ 4 ohms,

4,400 W @ 4 ohms bridged (Tn4); 1,400 W per channel @ 4 ohms, 3,800 W @ 4 ohms bridged (Tn3); 26 dB input gain; copper-plated transformer; EEEngine technology.

PRICE: \$2,399 (T3n), \$2,999 (T4n), \$3,799

CONTACT: Yamaha Pro Audio at 714-522-9011, www.yamahaca.com.

QSC AUDIO PowerLight 3 PL380

FEATURES: Two-channel; 4,000 W per channel @ 2 ohms; Class D; 85-percent plug-to-plug efficiency; remote DSP monitoring via BASIS; detented gain selection; removable knobs; Dataport;

BUYER'S GUIDE | Live Sound Amplifiers

M/F XLR; 2RU. PRICE: \$3,599.

CONTACT: OSC Audio at 800-854-4079.

www.qsc.com.

MARTIN MA Series 18K. 12K. 9.6K Digital Power Amplifiers

FEATURES: Two-channel; 1RU; 20,000 W @ 2 ohms bridged; Switch-Mode, PWM technology employs 95-percent input power; Power Factor Correction steady current draw: mains voltage variation immunity; DSP via front panel LCD, RS485 interface; optional Sharc card.

PRICE: TBA.

CONTACT: Martin Audio

at 519-747-5853, www.martin-audio.com.



FEATURES: Two-channel: 850 W per channel @ 8 ohms, 1,450 W per channel @ 4 ohms,

1,900 W per channel @ 2 ohms (Power H 2500); 1,500 W per channel @ 8 ohms, 2,500 W per channel @ 4 ohms, 3,500 W per channel @ 2 ohms (Power H 5000); three-stage Class H; auto-switching 100 - 250 V, 50 - 60 Hz;

remote DSP diagnostics via IRIS-Net V1.2; 2RU; 33 lbs.

PRICE: \$3,800 (Power H 2500), \$4,400

(Power H 5000).

CONTACT: Dynacord at 800-392-3497,

www.dynacord.com.

CARVIN DCM2500 Power Amp

FEATURES: Two-channel; 500 W per channel @ 8 ohms, 850 W per channel @ 4 ohms, 1,250 W per channel @ 2 ohms; 1,700 W @ 8 ohms bridged, 2,500 W @ 4 ohms bridged; continuous toroid transformers; hightransparent 50 V slew rate; M/F balanced XLRs; 5-way binding post, Speakon, 1/4-inch

outputs; limiter, ground lift, paralleling, bridge switches; recessed attenuators; LED indicators; soft-start; advanced protection relays; variable-

speed fan; steel 3RU. PRICE: \$599.99.

CONTACT: Carvin at

800-854-2235, www.carvin.com.

POWERSOFT K20 Digital Amp

FEATURES: Two-channel: 20,000 W @ 2 ohms bridged; 1RU; Switch-Mode, Power Factor Correction dissipates only 5-percent input as heat; 95 - 265 V, 50/60 Hz; front-panel

LCD/RS485 interface diagnostics; Smart Card functionality; optional

KDSP control/protection configurations.

PRICE: \$11,990 (K20 DSP).

CONTACT: Powersoft at 973-785-1105,

www.powersoft.it.

PRODUCTS & SERVICES

Pro Review





AUDIO EQUIPMENT EXCHANGE

ProAudio Review

ACOUSTICS



Full product line for sound control and noise elimination.

Web: http://www.acousticsfirst.com



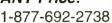


CASES

Case Specialists

www.NewYorkcasecompany.com

We Will Beat ANY Price!

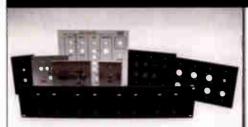






CONTROL PANELS

CUSTOMPANELS **METALWORK**



Gustom panels in Aluminum, Steel, Stainless & Brass. Wide choice of finishes. Silk screened or engraved graphics. Custom wood mounting systems & desktop consoles.

PANELAUTHORITY INC.

PHONE815,838.0488

FAXB15.838.7852

www.**panelauthority**.com

EQUIPMENT

ALAN SMART COMPRESSORS



- C1, C2 & Multi-Channel Versions
- Smart 2 Tools SSL to Pro Tools Interfaces
- Smart D.I. Boxes

All Exclusively Distributed by: 323-469-1186 • SunsetSound.com



Space Available

call Tina @ 970-728-8376

2007 CLASSIFIED **ADVERTISING RATES**

	1x	3x	6 x	12x
1-9 Colmn Inch	\$105	\$100	\$95	\$90
10-19 Colmn Inch	\$100	\$95	\$90	\$85
Distributor Directory	\$ 165	\$150	\$145	\$140
Professional Card	\$115	\$110	\$105	\$100

Advertising Sales Representatives

U.S. & Canada Matt Rubenstein

mrubenstein@imaspub.com P/ 914-524-5045 F/ 914-524-5046

Europe, Middle East, Australia & New Zealand John Gatski

jgatski@aol.com P/ 703-998-7600 x119 F/ 703-671-7409

Japan

Eiji Yoshikawa

callems@world.odn.ne.jp P/ +81-3-3327-5756 F/ +81-3-3322-7933

China, Hong Kong & Southern Asia/Pacific Wengong Wang

wwg@imaschina.com P/ +86-755-578-5161 F/ +86-755-578-5160

Product Showcases & Classifieds

Tina Tharp

tinatharp1@aol.com P/970-728-8376 F/ 970-728-8356

Pro4udio Review

ADVERTISER INDEX

This listing is provided for the convenience of our readers. Pro Audio Review assumes no liability for inaccuracy.

Page No.	Advertiser	Website
71	ACO Pacific Inc.	www.acopacific.com
47	A-Line Acoustics	www.a-lineacoustics.com
24	A-Line Acoustics	www.a-lineacoustics.com
2	Audio-Technica U.	S. Incwww.audio-technica.com
29	Audix Corp	www.audixusa.com
5	Aviom, Inc	www.aviom.com
37	B&H Pro Audio .	www.bhproaudio.com
59	B&H Pro Audio .	www.bhproaudio.com
19	BBE Sound	www.bbesound.com
21	BBE Sound	www.bbesound.com
23	BBE Sound	www.bbesound.com
17	Benchmark Media	Systemswww.benchmarkmedia.com
31	Bosch Communica	tions
52	Carvin Corporation	www.carvin.com
65	Community Profes	sional Loudspeaker
33	Conservatory Of F	ecording Arts & Scienceswww.cras.org/pro_audio.html
36	D.W. Fearn	www.dwfearn.com
9	DPA Microphones	Incwww.dpamicrophones.com
51	Hear Technologies	www.heartechnologies.com
		www.holophone.com
35	JBL Professional	www.jblpro.com
11	KRK Systems LLC	www.krksys.com
45	L-Acoustics US .	www.l-acoustics-us.com
53	Lawson, Inc	www.lawsonmicrophones.com
53	Linear Systems	www.linearsystems.com
15	Marshall Electronic	swww.mxl.usb.com
63	McCauley Sound,	nc
13	Oram Professional	Audiowww.oram.co.uk
27	Peterson Electro-M	usical Products www.strobosoft.com
7	PMI Audio Group	www.joemeek.com
61	QSC Audio Produc	s, Incwww.qscaudio.com
43	QSC Audio Produc	s, Incwww.qscaudio.com
49	Rane Corporation	www.rane.com
		www.sae.edu
69	TC/US	www.labgruppen.com
		www.true-systems.com
		www.xIntidea.com
		al Audio Systems, Inc www.yamaha.com
		al Audio Systems, Incwww.yamaha.com
67	Yamaha Corp Of A	nericaprojecticawww.yamaha.com/proaudio
75	Yorkville Group	www.yorkville.com

"Sights Set On Heaven" | Mitch Easter



SINGLE: "Sights Set On Heaven" ALBUM: Dynamico (Electric Devil)

DATES RECORDED AND MIXED: Summer 2004 and Summer 2006, respectively, at

the Fidelitorium in Winston-Salem, NC

SINGLE PRODUCER, ENGINEER, MIXER: Mitch Easter

PRO TOOLS OP/EDITING: Matt Boswell

MASTERING: Brent Lambert at The Kitchen Mastering in Chapel Hill, NC

OTHER PROJECTS: Easter has produced, recorded and mixed key projects for R.E.M., Don Dixon, Marshall Crenshaw, Suzanne Vega, Dinosaur Jr., Pavement, Wilco and his own band. Let's Active.

SINGLE SONGWRITER: Mitch Easter

CONSOLE: Neve VR-60

RECORDERS: Studer A800 at 15 IPS with IEC equalization and Dolby SR onto Quantegy GP9 tape and Pro Tools HD with Prism Sound ADA-XR AD/DA multichan-

nel converters

MONITORS: UREI 813 mains and Auratone closefields

SELECT MICROPHONES: STC (Coles) 4038, Neumann KM184, Groove Tubes GT-

1. AKG C451

SELECT OUTBOARD PROCESSING: Empirical Labs Distressor

(vocal), Delta Labs DL-5 harmonizer (drums), Dolby

740 Spectral Processor (mix buss)

ENGINEER'S DIARY

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

Mitch Easter, a noted aural architect of the American jangle-pop scene in the early 1980s, has finally released his first solo album. Easter — whose first big gig was producing the breakthrough R.E.M. album Murmur and its hit single "Radio Free Europe" bides his time in his hometown of Winston-Salem while producing, engineering and mixing in his personal analog heaven, the Fidelitorium,

Like the rest of the tracks on Easter's 2007 Dynamico full-length release, "Sights Set on Heaven" was recorded to a Studer A800 analog multitrack alongside a bevy of interesting and decidedly retro outboard processors, instruments and amplifiers. For Easter, one of the song's most interesting processors was the Delta Labs DL-5 harmonizer from the early '80s.

"The drums have loads and loads of it," he explains. "The DL-5 was an early harmonizer with tons of artifacts. The first record I ever heard that on was David Bowie's Low. On ["Heaven"] the snare sound disintegrates into this digital fizz that I just love. On the mix buss, I used a Dolby 740 Spectral Processor, a boost-only EQ that's dynamic: as the volume goes down it does more, and as the volume goes up it does less. It really brings out details in a nice way."



Mitch Easter

Although Easter happily lives in an analog recording world, his Dynamico — largely consisting of basic tracks recorded over years' past — was mixed from Pro Tools|HD for simplicity's sake. "Because I had so many different tape formats [for the album], I didn't want to constantly realign the tape machine; I had 16-track, 24-track, stuff with noise reduction and without, different speeds ... all this junk. So, with all the songs in front of me on a screen, I could just go for it. I copied the audio over at 96 kHz with some very nice converters: Prism ADA-XR multichannel AD/DAs. I always really, really liked Prism's two-channel converters that I would hear while mastering. To my ears, they have a superior sound. Although I figured that the new Digidesign ones were fine, I'm just contrary enough that I wanted something else."

Clearly Superior...

Ever wonder why high efficiency professional PA systems never sound as clear and concise as a studio monitor?

Well so did we... until now.

The new Unity™ Series U15 from Yorkville is a radical new loudspeaker design with ultra clear reproduction, exceptional linear frequency response, and extremely low distortion. We combined a single compression driver with three sealed back midrange drivers on a patented Unity™ horn. This single point source assembly combines well with a high efficiency Neodymium woofer to redefine the boundaries of affordable professional PA.





Hear the U15 today at your Yorkville dealer or find out more on the web at www.yorkville.com/unity

YORKVILLE SOUND 550 Granite Court, Pickering, Ontario LTW 378 • Phone: 935-837-8481 • Fax: 905-839-5776
YORKVILLE SOUND INC. 4625 Witmer Industrial Estate, Niagara Falls, NY 14305 • Phone: 716-297-2920 • Fax: 716-297-3689



Real Gear. Real People.

0

Surround. Simple.

Jack Says: "I love my H2-PRO.

Now I can't wait to get my new H3-D. You should buy one. You'll love it too.

(woof, woof! – Jack the Dawg)





















HO LOPHON

Surround Sound Microphone Systems www.holophone.com • 01.416.362.7790