New Products, P. 12, 38

NewBay Media

JULY/AUGUST 2014

ProAudio Review

The Review Resource for Sound Professionals

Why Gear Doesn't Matter

"Why All This Stuff, Dad?"

Sam Fuston, age 5, and his dad's stack of shootout preamps from 3D Audio's *Preamps in Paradise* DVD (2004).

reviews:

JBL 3 Series Studio Monitors

It's the same M50. The one praised by pro audio engineers, online reviewers, and cult followers worldwide for its sonic signature, long-wearing comfort, and excellent build quality. But now with refined earpads and three detachable cables. So you can take these iconic headphones from the studio to the street. And everywhere in between.

PURE. PROFESSIONAL. PERFORMANCE.

audio-technica.com

audio-technica.

Works seamlessly with:
GarageBand, Auna, Cubasis and all core audio iPad recording apps

iTrack Dock

The professional dock for recording music on iPad

iTrack Dock combines premium microphone pre-amps and audio quality with the simplicity of iPad to provide a comprehensive iPad recording interface that acts as the hub of your iPad recording studio. Connect and control your iPad, microphones, monitors, headphones and instruments from iTrack Dock and quickly capture your performance with the easy to use 'Tape by Focusrite' app - or any other core audio iPad recording apps.

- Compatible with all sizes of Lightning iPads - iPad 4th Gen, iPad Air, iPad mini & iPad mini with retina display
- Record and monitor at superior resolution
- Two award-winning Focusrite mic preamps
- Powers your iPad & class compliant USB MIDI device
- Class compliant USB MIDI controller input
- Dedicated monitor & headphone controls
- Free 'Tape by Focusrite' iPad recording app

Made for iPad

For more information visit
www.focusrite.com/itrackdock

Focusrite®
SOUND IS EVERYTHING

* iPad not included.

Studio

Covering Recording, Broadcast Production, and Post Production

34

Sound Reinforcement

Covering Live Sound,
Contracting, and Installed Sound

40

Departments

14

Cover Photo: Harry Butler

M2 Master Reference Monitors
The Village Studios Los Angeles
villagestudios.com

BOOST YOUR IMAGE

Introducing the 3 Series Powered Studio Monitors, the first to incorporate JBL's groundbreaking Image Control Waveguide. Matched with high-performance JBL transducers, the patent-pending Image Control Waveguide lets you hear detail and depth like you've never heard before. The micro-dynamics in reverb tails, and the subtleties of mic placement are clearly revealed. And the sound stage is so realistic it creates a tangible sense of a center channel. A broad, room-friendly sweet spot means everyone in the room hears the same level of dimension and transparency in your music. Adopting technology developed for our flagship M2 Master Reference Monitors, the 3 Series are surprisingly affordable, so you can hook up a pair and boost the image in your studio.

Learn more at **jblpro.com/3series**

3 SERIES | POWERED STUDIO MONITORS

Hear the truth.

JBL PROFESSIONAL by HARMAN

Pro Audio Review

The Review Resource for Sound Professionals

JULY/AUGUST 2014

EDITORIAL
Frank Wells, **Editorial Director**
212-378-0400 x535, fwells@nbmedia.com

Strother Bullins, **Editor**
336-703-9932, sbullins@nbmedia.com

Clive Young, **Managing Editor**
212-378-0424, cyoung@nbmedia.com

Kelleigh Welch, **Assistant Editor**
212-378-0423, kwelch@nbmedia.com

Lynn Fuston, **Technical Editor**

Rich Tozzoli, **Software Editor**

Russ Long, **Senior Contributor**

Steve Harvey, Will James, Liz May, Randy Poole, Rob Tavaglione, Tony Ware, Sterling Winfield, Dan Wothke
Contributors

ADVERTISING
Charlie Weiss, **Group Publisher**
212-378-0484, cweiss@nbmedia.com
Tara Preston, **Associate Publisher**
917-331-8904, tpreston@nbmedia.com
Janis Crowley, **Sales Director, West Coast Office**
415-505-1985, jcrowley@nbmedia.com
Zahra Majma, **Specialty Sales Associate**
212-378-0400 x517, zmajma@nbmedia.com

ART & PRODUCTION
Nicole Cobban, **Senior Art Director**
Walter Makarucha, Jr., **Associate Art Director**
Fred Vega, **Production Manager**
212-378-0445, fvega@nbmedia.com

CIRCULATION
Meg Estevez, **Group Director, Audience Development**
Michele Fonville, **Associate Circulation Manager**
Subscriptions: Pro Audio Review, www.MyPARmag.com
P.O. Box 234, Lowell, Ma 01853
Tel: 888-266-5828
(U.S.A. Only, 8:30 A.M. - 5 P.M. Est)
978-667-0352 (Outside The U.S.)
Fax: 978-671-0460
E-Mail: Newbay@Computerfulfillment.com

NEWBAY MEDIA AUDIO GROUP
Adam Goldstein, **Senior Vice President/Group Publishing Director**
Ragan Whiteside, **Web Director**
Robert Granger, **Online Production Manager**
Ashley Vermillion, **Web Production Specialist**

NEWBAY MEDIA CORPORATE
Steve Palm, **President & CEO**
Paul Mastronardi, **Chief Financial Officer**
Jack Liedke, **Controller**
Bill Amstutz, **Vice President of Production & Manufacturing**
Robert Ames, **Vice President of Digital Strategy & Operations**
Denise Robbins, **Vice President of Audience Development**
Anthony Savona, **Vice President of Content & Marketing**
Ray Vollmer, **Vice President of Human Resources**
Anthony Verbanac, **IT Director**

LIST RENTAL:
914-925-2449 or danny.grubert@lakegroupmedia.com

REPRINTS AND PERMISSIONS:
For custom reprints & eprints please contact our reprints coordinator at Wright's Media:
877-652-5295 or newbay@wrightsmedia.com

PRINTED IN THE U.S.A.
Administrative, Advertising, & Editorial Offices
28 E 28th Street, 12th floor
New York, NY 10016
TEL: (212) 378-0400

NewBay Media

technically speaking

On The Importance Of Gear

by Frank Wells

One regular client from my studio days, a superlative golden-eared producer/engineer, had a plethora of exotic analog outboard gear that he married in to a digital workflow. Then one day, he put the whole lot on eBay. He said that while he could discern differences between the hardware and the latest software versions of the devices, the differences were so minute as to be insignificant. In a full mix, he said, any differences were completely insignificant, as they were not audible in that context. None of his artists had any complaints about the transition, nor did the labels paying him for his services.

That same engineer undertook an exhaustive evaluation of analog to digital converters to mate with his DAW. Some of the devices under test ran north of $500 a channel. At the low end was a converter package that ran south of $30 a channel. Converter semiconductors have gotten very good. With care in circuit board design and clocking, and with quality analog circuits on the inputs, A/D performance is possible at that low end that beats out the onboard converters in very expensive digital recorders of decades past. As with his outboard, my friend said he heard performance differences, and the more expensive converters won critical listening shootouts. Yet, when the various packages were used in tracking, the differences were imperceptible in a mix.

I'm not out to discourage high-end gear sales, but to offer another perspective on the bold statement made on this issue's cover, "Why Gear Doesn't Matter." Of course, gear does matter, or my engineer friend wouldn't have chosen from a variety of microphones and mic pres for his recordings. He wouldn't have employed even the digital emulations of his discarded hardware. But his experience does suggest that the value of gear must be considered in a larger context.

If the performances being captured have no emotion, all the gear in the world isn't going to get a listener's toes tapping. With the right technical and aesthetic chops, and a moving performance, a good engineer will make the most of the tools available. Now, given their druthers, they'll still opt for the good stuff; it's just not what makes them good.

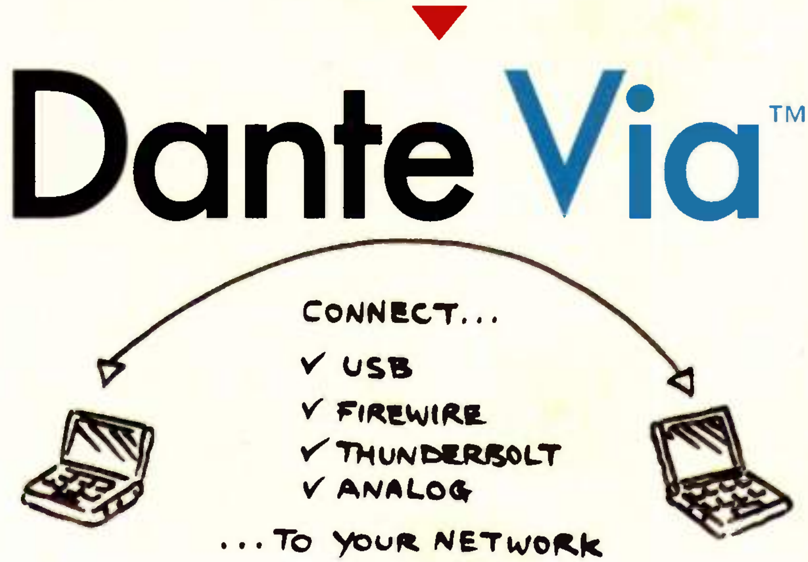
Audinate Dante Via Software Application

Audinate's new Dante Via transforms Macs and PCs into networked I/O devices, allowing for complete, standalone audio systems of networked computers without the need for any dedicated Dante-enabled hardware to be present on the network, and to add such devices to a Dante hardware network.

Legacy USB, FireWire and Thunderbolt audio interfaces are all Dante Via compatible. For example, Dante Via allows users to distribute or loopback audio through the network from any application such as Cubase, Pro Tools, Nuendo, Logic, Reaper or even Skype.

Dante Via will be available later in 2014.

Contact: Audinate | audinate.com

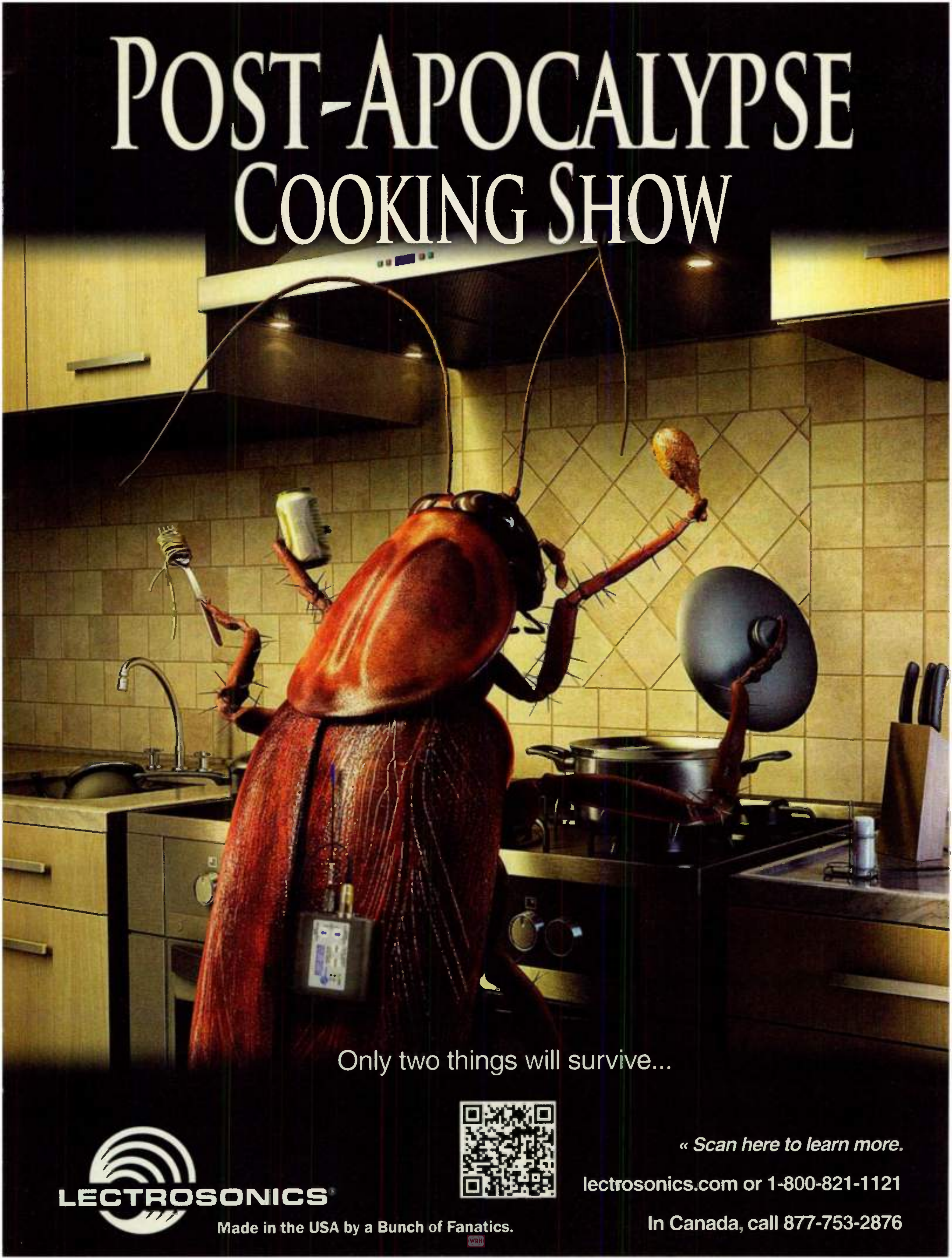


Genelec 8010 Active Monitor: Smallest-Ever 8000 Series Speaker

Genelec's 8010 Active Monitor ($440 each) is the smallest member of its 8000 product range. It is intended for environments such as small studios and OB vans, when paired with portable recording devices and in other mobile production applications. The 8010 features a balanced XLR input, 3-inch bass driver, 3/4-inch tweeter and bi-amped Class D power amplification—one amp per driver. Its bypass-able Intelligent Signal Sensing (ISS) circuitry saves energy by automatically putting the monitor to sleep when an audio signal is absent. As with all of the 8000 Series, the 8010 employs a Minimum Diffraction Enclosure rounded-edge cast aluminum cabinet that incorporates Genelec's Advanced Directivity Control Waveguide and rear-mounted reflex tube opening.

A full range of accessories is available for the 8010, including comprehensive mounting products.

Contact: Genelec | genelecusa.com


POST-APOCALYPSE
COOKING SHOW
Only two things will survive...
« Scan here to learn more.
lectrosonics.com or 1-800-821-1121
In Canada, call 877-753-2876
LECTROSONICS
Made in the USA by a Bunch of Fanatics.

MOTU 1248, 8M and 16A Thunderbolt Audio Interfaces featuring AVB Ethernet Networking

MOTU has announced three new Thunderbolt audio interfaces, equipped with complementary I/O configurations, A/D/A conversion with high dynamic range, 48-channel mixing, DSP effects and AVB Ethernet audio networking for system expansion.

Based on a new, shared technology platform, the 1248, 8M and 16A differ only in analog I/O. The flagship 1248 offers 8 x 12 balanced TRS analog I/O, four mic inputs with digitally-controlled preamps, two front-panel hi-Z guitar inputs, two independent headphone outs and stereo RCA S/PDIF digital I/O. The 8M provides eight balanced TRS analog outputs plus eight mic/line/instrument "combo" inputs with digitally controlled preamps, 48 VDC phantom power, pad and MOTU's hardware-based V-Limit overload protection. The 16A features 32 balanced TRS analog connections (16 in, 16 out). All three units provide two banks of optical digital I/O, word clock I/O and computer connectivity through either audio class compliant USB 2.0 or Thunderbolt (1 and 2 compatible).

Each interface is also equipped with ESS Sabre32 Ultra converters, DSPs and a single AVB Ethernet network port. Analog I/O latency has round trip performance of 32 samples (0.66 ms) at 48 kHz. Each unit provides metering for all inputs and outputs on a large, backlit 324x24 pixel LCD.

Featuring 32-bit floating point precision, each interface offers DSPs that drive a 48-channel mixer, taking signal from the physical inputs on the interface itself, audio channels from host software on the computer, and audio network streams and mixer outputs. The mixer provides seven stereo aux busses, three groups, a reverb bus/fourth group, a Main Mix bus and a separate Monitor bus/solo bus. Various effects and a matrix routing grid are also included.

All three interfaces are equipped with a single AVB Ethernet network port, which can be used to add a second MOTU interface via CAT 5e Ethernet cable or three to five interfaces connected to the five-port, 1-Gigabit MOTU AVB Switch (sold separately). Each MOTU interface on an AVB network, or any computer connected through Thunderbolt, can simultaneously send and receive 128 channels of network audio I/O. A MOTU AVB network supports more than 512 audio channels. Point-to-point network latency is fixed at 30 samples (0.625 ms) over multiple switches, when operating the system at 48 kHz. AVB provides unified, system-wide, precision clocking and synchronization that is measured in nanoseconds. Any available network port, including the extra (sixth) Ethernet port on the MOTU AVB Switch, can be connected to a standard Wi-Fi router or directly to a local Ethernet network, allowing users access to device settings, audio routing features and the 48-channel mixer from web app software.

Contact: MOTU | motu.com

In 1973, at Watkins Glen, live sound went just a bit digital.

"In the beginning, audio was analog. And, almost anything was possible."

Everything but delay. And so the Digital Delay Line (DDL) was born. DDLs were simple and limited to just delay. Today, forty one years after a stack of Eventide DDLs delayed the audio feeds to the speaker towers at Watkins Glen, Eventide introduces the DDL·500. Designed to be 'as analog as possible', the DDL·500 will make your 500 series lunch box "just a bit digital."

DDL·500

Digital: Just Delay

24-bit
10 sec @ 192 kHz
160 sec @ 16 kHz

Analog: Everything Else

Mix
Input Kill
Feedback
Soft Saturation
Low Pass Filter
Send/Return

http://goo.gl/63hOzC

Eventide

by Lynn Fuston
PAR Technical Editor
lfuston@nbmedia.com

Why Gear Doesn't Matter: A Heretical Look At Why We Do What We Do

In the beginning, there was music. It was simple at first: beating on logs, whistles made of bamboo, etc. Gradually the complexity of the instruments as well as the music advanced. Then along the way, someone decided to try and preserve music, by crude means at first. Recording technology developed in order to capture and distribute performances of music and voice, but the music was still the catalyst.

Then in the mid-20th century, that all started to change. As technology progressed, with tape recording and the ability to add tracks to prerecorded performances, it allowed the creation of music that would otherwise never have existed. Now in the 21st century, with digital technology, miniaturization and inexpensive manufacturing, recording has grown from being a novelty—like the wire recorders or phonographs of the late 19th century[1]—into something that most everyone can do at any time on accessories affordable by all.

Recording gear has never been so accessible and affordable in the history of mankind. So has all this progress made music recordings "better" than they were a few generations ago? Sadly, no.

Modern Recordings Aren't Better

We find ourselves at a crossroads, where technology has stolen the focus away from the music it was intended to serve. The goal of writing a good song has given way to programming a good loop or finding a good riff. Programmers/producers passing around beats/loops for other musicians to layer parts on has become the unfortunate norm. In the past decade, we've wondered why there has been such a devaluation of music and why it doesn't connect with audiences like it did in the past. One common theory about the lack of musical involvement points the finger squarely at digital recording and subsequently at MP3s. Some people mistakenly attribute the musical magic that so many writers/musicians aspire to (the classics of the 1950s through the 90s) to analog tape or tubes. The real magic, though, was in the music and the performances, not in the gear.

So how did we get here? It's the result of an industry that pushes merchandise instead of creativity. I see musicians these days that spend more time talking about gear and reading magazines and online forums than they do writing songs or practicing their instruments. It's unfortunate but true.

The dirty little secret that nobody will tell recording novices is you don't need great gear to make a great record. I'll say it again: You don't need great gear to make a great record. The single most important ingredient in making a great record is talent. I hear this sentiment over and over again from veteran engineers and producers that I know. Sure—the gear helps, but it shouldn't be the focus. In the past few months, I've asked numerous big-name engineers, each with credits on multiple millions of records, the same question: "If someone asked you to make a record with one big stipulation, that you can only use SM57s to record everything, could you make a great-sounding record?" The answer from every one of them was, "Sure." It might not be as easy as having their toolkit of microphones, but it can be done and done well. Honestly, a lack of gear often inspires—rather, it demands—greater creativity. So crank up the creativity to make up for the gear you don't have. (I did a record once where we made a guitar talkbox out of an Auratone gaffer-taped to the bottom of a microphone stand and an echo chamber out of a tall stairwell.)

Gear Is Everywhere, Talent Is Elusive

So why is the industry as a whole so obsessed with gear? Here's my theory. You can buy gear. You can't buy talent. Gear is just a credit card and phone call away and you can have it in a matter of days or less. It's easy to buy that vocal chain that McCartney or Michael Jackson used, but much harder to achieve that sound. Know why? Because it wasn't the gear that made that sound, though it did contrib-

> You don't need great gear to make a great record.

PHOTO: HARRY BUTLER

Lynn Fuston at his stack of shootout preamps from 3D Audio's *Preamps in Paradise* DVD (2004) ute. What does Sir Paul McCartney sound like when you take away the Neumann U47 and REDD console and Fairchild compressor and BTR analog tape machine? Like Paul McCartney.

I saw Ed Cherney (Bonnie Raitt, Eric Clapton) recently and he recounted a studio experience that many engineers have had, myself included. "I was trying, mostly unsuccessfully, to get good drum sounds for the first 16 years of my engineering career. Then one day Jeff Porcaro (Toto, Steely Dan) was booked on a session I was doing. I became a genius that day."

PHOTO HARRY BUTLER

The microphone collection from Lynn Fuston's 3D Mic CDs featuring comparisons with male and female vocals, 2001.

My good friend Glenn Rosenstein (Madonna, Ziggy Marley) told me once, "You want to know how to get that James Taylor guitar sound? Hand a guitar to James Taylor. Put up an 87. Easiest guitar sound I ever got in my life."

I got a call once from a person who wanted to know what mic, gear and settings I used to get "those vowel sounds that I hear on Amy Grant's records. They're amazing." He wanted to know what he needed to buy. Well, it's not the gear. Amy's sound comes out of Amy's mouth.

As Bruce Swedien so succinctly puts it, "No one walks down the street humming the console." It's the song. It's the melody.

There is another reason we spend so much time talking about gear. It's fun. We love it. We can't get enough of it.

Technology Is Limiting, Creativity Is Boundless

So, what is my point? Two things, I guess. First, let recording be about the music. Don't let the technology get in the way. Try to find the setup that will allow you to encourage and preserve a musical and emotional performance. Second, don't use gear, or a lack thereof, as a crutch. I hear artists say, "If I only had this, I could get that sound." The truth is, gear isn't standing in the way of you doing something special. It's the mindset regarding gear that is standing in the way. Focus on creativity and overcome the limitations.

I'd honestly rather have Mick Guzauski, Bill Schnee or Al Schmitt mix a record on a Mackie than someone with lesser talents on an SSL. Why? I know they can deliver excellent results regardless of the gear. It's the ear, not the gear.

Aspiring engineers, if you only remember one thing after reading this article, it should be this one, plus two more:

1. Use what you have. I started with one RE20 and two borrowed, no-name condensers.
2. Work hard and long at honing your craft. It took me 10 years before I started getting happy with my sounds.
3. Spend more time developing your skills and less time researching and acquiring gear.

As you master the craft, you'll be able to use those skills with any gear, good or not so good. And great gear only makes your life easier. Any accomplished engineer knows that they still have to deliver superior results, even when working with marginal gear.

Footnote

1 http://www.recording-history.org/HTML/wire2.php

by Russ Long
PAR Senior Contributor
rlong@nbmedia.com

Lynx Aurora, Hilo & LT-TB LSlot Interface Card and AEA NUVO N22 Ribbon Microphone

The Lynx Aurora has been a mainstay in recording since 2005 and is still considered the best I/O option by many top engineers. Last year's news that it was the primary interface used in the making of Daft Punk's *Random Access Memories* (arguably the 21st century's best-sounding album release to date) has brought more attention to the device. In 2012, Lynx released the Hilo (pronounced HE-low) 2-channel converter system, which provides world-class D/A and A/D conversion as well as an audiophile-quality headphone amp. Both interfaces are equipped with Lynx's LSlot expansion slot, allowing use of the Lynx LT-TB Thunderbolt Interface released earlier this year. The LT-TB provides Thunderbolt connectivity, making the Aurora and Hilo perfect options for engineers and musicians seeking outstanding sound quality in a Thunderbolt-equipped device.

I purchased Lynx Aurora after reviewing it for the March 2006 issue of *PAR* (www.lynxstudio.com/nav/getFile.asp?i=9&t=contentfile), and it's been my primary converter ever since.

On the other side of the glass, I've been using AEA's N22 active ribbon mic for the last six months to record nearly everything that's come across my plate. I love it. While it's billed as an affordable ribbon mic offering by AEA, it is truly high-end in every aspect except the price and is sonically unlike any ribbon mic I've encountered.

Lynx Aurora 16

The 1U Aurora 16 A/D-D/A supports sample rates up to 192 kHz and provides outstanding conversion. The sound is open and transparent and almost impossible to discern from the source material at high

Currently On My Desk

1. Alta Moda Audio AM-25 Equalizer
2. Alta Moda AM-30 Mic Pre
3. Focal Spirit Professional Headphones
4. RTZ PEQ1549 Parametric Equalizer
5. Ingram Engineering MPA575 Mic Signal Processor

NOW AVAILABLE for **your** reading pleasure...

Are you a current subscriber to *Pro Audio Review*? Grab your iPad and visit www.prosoundnetwork.com/PARiPad to download your *Pro Audio Review* app.

Not a current subscriber? *Pro Audio Review* is the only U.S. professional audio magazine to focus on product reviews for all major niches. Each exclusive section covers: studio, live, contraction/installation, broadcast production, post-production and project studio. There is no obligation and starting your complimentary subscription to *Pro Audio Review* is easy - Visit us at www.myparmag.com sample rates. The internal clock is spectacular and since the unit operates without an internal fan, there's no issue in having the converters in the control room.

Lynx has followed the Aurora's release with numerous LSlot interfaces that allow the Aurora to be used in various interfacing configurations. The LT-HD (which I currently use in my studio) allows the Aurora to be recognized and controlled from Pro Tools. Initially the LT-HD's firmware tricked Pro Tools into thinking the Aurora was a 192, but with the latest firmware update, Pro Tools now sees the Aurora as an Avid HD I/O (Avid's current premium I/O). The LT-HD card has a DigiLink connector and can be connected to an HD System with a standard DigiLink cable or an HD Native or HDX system with a DigiLink to MiniDigiLink cable. I've experienced flawless operation in all three scenarios. The LT-ADAT, LT-MADI and LT-USB provide ADAT Lightpipe, MADI and USB 2.0 connectivity, respectively. And the latest card introduction, the LT-TB (discussed in this article), adds Thunderbolt connectivity to the Lynx Aurora. Interface to the Aurora can also be accomplished by installing a Lynx AES16 or AES16e card directly in your CPU or expansion chassis.

With all of the available connectivity options, I wish the Aurora and Hilo had been designed with multiple LSlots so multiple cards could be installed at the same time. Obviously only one could be used at a time, but I'd love to be able to pull the Aurora out of my studio that is connected to Pro Tools with the LT-HD card and take it home to record with my laptop and the LT-TB card without having to physically swap cards.

Lynx Hilo Reference A/D-D/A Converter System

The Hilo Reference A/D-D/A Converter System is a mastering-quality, self-contained (8.50" x 3.25" x 10") interface that provides two channels of superb analog-to-digital conversion, six channels of digital-to-analog conversion, Main out and then monitor and headphone amplifiers, each with its own volume control. The 6.75-lb., compact, half-rack box is available in silver or black. While the rear-panel is packed full of connectivity options, the box's front panel is elegant and simple with only a power switch, headphone jack, control knob and 480 x 272-pixel touch-screen. The screen provides easy access to the unit's intuitive feature set as well as great metering options including analog VU, horizontal bar meters, or all I/O. The feature-set includes routing and mixing options, sample rate selection, clock source selection, levels, metering and diagnostic features.

The Hilo was designed with pristine audio as the top priority, which led to the ground-up design of the box's analog stage and converter topology, assuring minimum distortion and maximum conversion transparency. Any or all of the digital and analog inputs can be routed to any or all of the three separate outputs: Line Output (with 8 trim settings), Monitor Output and Headphone Output. All three output pairs have a dedicated DAC, allowing each output to have a unique mix. Output levels can be controlled (in 0.5 dB increments) via faders on the touch-screen or the knob on the front panel.

The Hilo offers a wide range of digital connectivity options, including AES/EBU I/O and S/PDIF I/O (via coax or TOSLINK optical connections). The optical connection can alternatively be used for up to eight ADAT channels and the ADAT I/O is completely independent from the AES/EBU and S/PDIF coax channels. The Hilo has a total of 12 inputs and 16 outputs with 32 possible channels via the LSlot port. DSD audio file playback is supported, as is the DoP V1.1 standard which provides the means for transferring DSD audio over PCM frames. [For more information on the DoP Standard, visit this link: http://dsd-guide.com/dop-open-standard—*Ed.*]

The Hilo is quite complex and certain functionality may require multiple routing and or setting changes. The Scene feature allows the user to save all of the Hilo's current settings and then recall them with the push of a button. There are six storable Scene options that can be used to store these unique setups. In addition to the IEC connector that accepts standard 110 VAC to 220 VAC, the 9 VDC to 18 VDC input allows the Hilo to be powered remotely from a standard video battery pack or automobile cigarette lighter.

In addition to being feature-packed, the Hilo's sonic performance is amazing. I compared a 96 kHz high-resolution recording through the Hilo to the same recording through a Benchmark DAC-1 and was amazed that the top-end was smoother through the Hilo. The imaging was also better with the Hilo. I utilized the Hilo's internal sample-rate converter to down-convert a 96 kHz digital signal and found the result to be quite impressive without any noticeable artifacts or image change.

I tested the Hilo's headphone amp with an assortment of headphones including Focal's Spirit Professional Pro, Audio Technica's ATH-M50 and my Ultimate Ears Custom In-Ear Reference monitors and, in each instance, the amp had enough gain to push the headphones to the point of pain without even a hint of distortion. Not only does it have an abundance of horsepower, it sounds amazing. My Focal headphones have never sounded better than when listening to *Dark Side of the Moon* through the Hilo.

"I purchased the Lynx Aurora after reviewing it... it's been my primary converter ever since."

Lynx LT-TB Thunderbolt LSlot Interface for Lynx Hilo and Aurora Converters

When I stopped by the Lynx booth at Winter NAMM 2014 and saw half a dozen Aurora 16-TB converters simultaneously playing 192 channels of audio from a single Thunderbolt port on the new Mac Pro running Logic, I knew the new Lynx LT-TB Thunderbolt LSlot Interface was worth a closer look.

At 10 Gbs bidirectionally and simultaneously, Thunderbolt is currently the fastest PC connection possible (twice the theoretical speed of USB 3.0). The LT-TB utilizes the latest Cactus Ridge Thunderbolt controller, which ensures maximum bandwidth, minimal latency and full compatibility with the latest computer hardware and applications. While latency varies depending upon buffer size and sample rate, a typical 96 kHz session with a buffer size of 128 samples yields an incredibly low latency of 1.33 mS. The LT-TB was the first Thunderbolt audio product certified for use with both PC and Mac computer systems and it includes an easy-to-use software interface for setting parameters, managing volume levels, monitoring input, metering, etc. Up to six Lynx interfaces (any mix or match of Auroras and Hilos) can be daisy-chained from a single Thunderbolt port, potentially providing 192 channels of audio, or a computer with two Thunderbolt ports can deliver up to 384 channels while still only using 12 percent of Thunderbolt's available bandwidth. In theory, the new Mac Pro, which includes six Thunderbolt II ports, can deliver 1,152 channels with adequate processor speed and system throughput (I admittedly have my doubts, though). Thunderbolt supports hot-plugging, so devices can be added or removed from the daisy chain without requiring a reboot or power cycle of the hardware. This is typically problematic with the software though, so I don't recommend doing it while an audio application is running. The LT-TB supports cable runs up to three meters with standard copper cables and up to 100 meters with the Corning optical Thunderbolt cables. I utilized a 10-meter optical cable for all of my testing and it performed flawlessly (though optical TB will not pass bus power to downstream devices). To facilitate configurations that require a bus-powered device to be daisy chained to the LT-TB, Lynx has added a 12 VDC power input on the LT-TB card enabled by attaching the optional Lynx LYN-ACPS1000 AC Adapter to the card.

Both the Aurora and Hilo are available with the LT-TB LSlot card pre-installed or the LT-TB card can be purchased separately and installed into any existing Aurora or Hilo with a simple (and free) firmware update. The Lynx Mixer application (for the Aurora) and HiloRemote application (for the Hilo), both free and downloadable from the Lynx site, provide routing configuration, zero-latency monitoring and metering. The LT-TB supports WDM and ASIO on Windows computers and Core Audio on OS X computers, so nearly every professional audio application is compatible with the card.

I was recently hired to capture a pair of concerts in a local prison. The gear had to draw minimal power and be easily carried by three individuals (in this instance, myself along with engineers Seiji Inouye and Amanda Blehm). After contemplating several different scenarios, we decided that the best option was recording to Pro Tools via a MacBook Air and utilizing the Aurora 16 for A/D conversion and the Hilo for monitoring while recording to the G-Technology G-DOCK. The Hilo, Aurora and G-DOCK were daisy chained off a single Thunderbolt port. The concert was recorded at 24-bit, 44.1kHz and the system performed beautifully and sounded fantastic. While 8-tracks of 44.1 kHz, 24-bit audio recording for 75 minutes is a nice test, it in no way pushed the envelope, so after getting back to my studio, I recorded 24-tracks of 192 kHz, 24-bit audio for over 90 minutes without a single snag.

Lynx makes great products with no sonic compromise whatsoever. Their manuals are well written and concise, and their tech-support is among the best I've ever encountered. I'm constantly integrating new gear and experimenting with new technology in my studio, and I've called or emailed tech support several times over the years and have always received excellent support. If the technician didn't know the answer to my question, he was more than willing to research it and get back to me in an amazingly brief amount of time. In short, I can't say enough good things about Lynx.

Prices

Aurora 16: $2,995.00
Aurora 16TB: $3,395.00
Aurora 16 VTTB: $3,695.00
Hilo (USB): $2,495.00
Hilo-TB (Thunderbolt): $2,795.00
LT-TB LSlot card: $650.00

Contact: Lynx Studio | www.lynxstudio.com

AEA N22 NUVO Series Ribbon Microphone

The N22 is the first release in AEA's NUVO microphone series and it is definitely worth a listen. Built by hand in Pasadena, California and rooted in the RCA tradition like AEA's other ribbons, the NUVO line provides a renewed incarnation of the ribbon transducer as well as AES founder Wes Dooley's take on the future of the ribbon microphone. At $999, N22 was designed with singer-songwriters, musicians and project studios in mind. But don't let that scare you into thinking it isn't up to par with other high-end ribbons; the mic's magnificent frequency response and durability make it more versatile than most ribbons.

Incorporating the same Big Ribbon technology as the classic R44 and R84 mics, the active N22 provides the warm and smooth sound ribbons are known for with a top end reminiscent of a condenser microphone. The mic's gorgeous champagne-silver finished body is 8.8-inch long with a diameter of 1.6 inches. The mic includes a felt mic bag, shock-mount mic clip and plastic foam-lined case.

The mic's phantom-powered JFET electronics and custom German transformer ensure optimum performance even with lower quality mic pres and the mic's ribbon is enclosed in an internal steel spring cage with multiple layers of blast protection fabric for maximum protection; thus it's a great

"An exceptional acoustic guitar mic," Russ used the N22 while recording a Taylor 514-CE with good results.

option for touring and stage use. I tested this concept by using the mic along with my USB-powered Presonus AudioBox while my computer was running on battery power to record a guitar/vocal song demo; I was surprised at the sound quality I was able to achieve with a USB powered interface.

I first put the N22 to use on UJ Pesonen's amp while tracking The Wrong Kata Trio and had fantastic results. UJ uses a Groove Tubes Soul-o 75 combo amp and the N22 wonderfully captured the full body of the sound as well as the top end sparkle. I used the mic to capture electric guitar several other times and always had great results.

The N22 is also an exceptional acoustic guitar mic. I used it to record a Taylor 514-CE on multiple instances and always had good results.

As with most ribbons, the N22 does a great job recording percussion. I used it to record tambourine, shaker and güiro with wonderful results. It was smooth as silk on hi-hat during a tracking session; I've been using ribbon mics on hi-hat for years now and the N22 is my favorite hat mic to date. I used the mic to record kick drum, placing it about 12 inches in front of the kick, slightly angled towards the floor, and it sounded great; I did find myself using some slight EQ to boost the bottom end, though. I used the mic as a mono room and while the result wasn't the massive drum tone I achieve when using the Coles 4038, it was still good.

I'm no stranger to recording vocals with ribbon mics but, when I do, I'm typically using significant amounts of equalization, usually boosting the top end and/or rolling off the bottom. Not so with the N22—I recorded multiple instances of both female and male vocals with the mic and, in only one instance, I found myself needing to utilize EQ during the process. Quite impressive. Also impressive is the ribbon's isolation from air movement; I still used a pop filter in most instances but the times I didn't, plosives weren't an issue.

I had great results using the mic to record a Yamaha C5 grand piano. I only had one N22 for my review period but I anticipate that a pair of N22s would be great on piano. Cello sounded wonderful with the mic placed about four inches from the sound hole.

While there are a lot of ribbon mics that will work well in most situations, the N22 is the first that I've encountered that will work well in most situations and will do so without the need of EQ. It likes being close to the sound source and if it doesn't sound right, simply move it closer until it does.

Contact: AEA | www.ribbonmics.com

Russ at Work

All of the guitars on the *unfiltered garage jazz sketches* album at the License Lab music licensing site were played by UJ Pesonen through a Groove Tubes Soul-o 75 combo amp and recorded with the AEA NUVO N22:

http://licenselab.com/album/ospk-039

MIX presents

SOUND for FILM

IMMERSIVE SOUND

From Production To Playback

SEPTEMBER 6TH, 2014

SONY PICTURES STUDIOS, CULVER CITY, CA

Immersive sound–Taking Hollywood by storm and rapidly spreading around the world, Dolby's Atmos and Auro Technologies' Auro-3D advanced surround sound formats are impressing audiences with fully enveloping, more natural sound environments for cinema and home theaters.

Not since the advent of 5.1 discrete surround in the early 1990s has there been a more exciting technological breakthrough in film sound production and playback. And it's just getting started. Join *Mix* in the only event of its kind, as we break down the technology and art of Immersive Sound.

This special one-day event includes:

- Keynote Address in the world-renowned Cary Grant Theatre
- Full Day of Panels: The Editing, Mixing, Technology and Business of Immersive Sound
- Film Studio Showcase: 10-Minute Reels, played back in the Atmos/Auro-3D equipped Willam Holden Theater, with introductions by the creative teams
- Sponsor Workshops – hosted onsite in Sony's world-class film sound facilities
- Exhibition Space – the world's leading technology suppliers
- Networking & Sponsorship Opportunities: scheduled face-time with influential professionals from MPSE, CAS, and *Mix*.

The day will be capped by an exclusive screening of a new Sony Pictures feature film in the Holden Theatre.

http://www.mixonline.com/soundforfilm

Don't miss out on the only event of its kind in film sound!

Contact: Charlie Weiss • 212.378.0478 • cweiss@nbmedia.com

Host Sponsor: SONY PICTURES

Event Partners: MPSE SOUND · CINEMA AUDIO SOCIETY

Sponsors: AUDIO HEAD · AURO 3D AUDIO · AVID · DOLBY ATMOS · dts LISTEN · GC PRO · Harrison · YAMAHA commercial audio

by Rich Tozzoli
PAR Software Editor
tozzoli@nbmedia.com

Universal Audio Thermionic Culture Vulture for UAD-2 and Apollo

The Thermionic Culture Vulture is one of those pieces of gear I've heard about for years, but never actually had in my hands. The hardware is a hand-built, 2U rackmount valve unit made by Thermionic Culture that produces varieties of harmonic distortion. With EF86, 5963 and 6AS6 tubes, transformers, capacitors and filters, as well as the ability to adjust bias and drive, it can go from subtle to screaming. [Read *PAR* Contributor Rob Tavaglione's full review of the hardware Culture Vulture here: http://www.prosoundnetwork.com/article/review-thermionic-culture-culture-vulture-/15370.-Ed.] However, with the release of Universal Audio's V7.8 software, the Culture Vulture is available on your desktop for UAD-2 hardware and Apollo interfaces.

Each channel (1 & 2) features Drive, Bias and Distortion Type with choices of Triode, P1 (Pentode 1) or P2 (Pentode 2). Below that resides a switch for Overdrive and Filter (OFF, 9 kHz, 6 kHz). There are dual VU meters, a Master Power button, On/Off switch, Control/Link switch, and a Mix knob for wet/dry control. There is also an Output level knob (1-11) and a Bypass switch for each channel.

In general, Triode mode is for gentle warming. Pentode 1 has more edge and Pentode 2 even more. The Drive knob and Overdrive switch hit the 6AS6 tubes for up to 20 dB of nastiness. The only real difference between the hardware and software versions is the addition by UA of a wet/dry Mix control and the Link function for stereo imaging.

Since I like to add grit to my bass tracks, I thought that would be a good place to start when checking the software out. I jumped into the presets, which are actually all named for engineer/producers. The first one was the Chris Coady > Better Bass. On my '69 Fender P, I loaded it up. Sure enough, it was just that—better bass. A few quick clicks on the cool blue 'Power' knob confirmed it. I like this thing! The Drive and Bias were up around 4 and the Distortion Type was Triode. I lowered the Output just to make sure that "better" wasn't just "louder." No, it wasn't. The bass had a touch a grit, presence and thickness that worked immediately in the track. Pushing up the Drive even a few numbers gave it even more sack but added more gain, so I lowered the Output level to compensate. By pushing up the Bias a bit, you can get almost a sense of compression happening;

it sounds like tubes are being squished. It reminded me of the old James Jamerson Motown sound and I just left it as is—when it works, it works.

Next I tried it on kick drum, recorded with an AKG D112. Loading up a preset called Clearer 808, the Kick got a bit mushy, but it did sound like smacking analog tape, that's for sure. I dialed back the Drive and Bias a bit and then played with the Mix for Wet/Dry control; that definitely added more attitude. I thought this might work great in parallel mode, so I placed an instance on a stereo Aux track and loaded one up. I put my usual Pultec EQP-1A back on the kick and sent some to the Vulture. I put the Mix knob up to fully Wet, and dropped it into P1 with some Bias and Drive. This gave the kick some good analog thump on the bottom and a bit more forward presence. Doing it this way let me retain my original kick sound, but bring in some extra juice.

Since the track I was mixing had acoustic guitar that was doubling the electric, I figured, 'why not try it there and go for an effect?' I loaded it onto the track, a Martin 015M recorded with a Miktek C7 large diaphragm mic. I checked out the preset called Even Harmonics, which sounded like Pete Townshend playing through a Hi-Watt. At this point, I realized the Overdrive switch has to be down to be 'on,' which is the opposite of what I expected. Now I played again with the Mix control until the acoustic had just a sense of grit to it, and once again, I saved a preset—I'll use this again for sure. So in order to get a nasty distorted sound, just dial the Mix up higher; to lessen the effect, just dial it down until you get what you need.

Now to a vocal. On a male lead, tracked with a Neumann U67, I loaded in a preset called BG Vocals Excitement. It was a bit aggressive with the saturation, but bringing down the Drive just a touch nailed it. The before and after actually made it sound like he stood a few inches closer to the mic. It has a real in-your-face thing that I didn't expect to get from this plug-in. It brought up the bass response of the voice a bit as well, but in this case, it was a good thing. I then experimented with the Filter, checking out the 9 and 6 kHz settings. While I clearly heard it working, as expected it sounded better with the Filter Off (which unlike the Overdrive is marked OFF when set to the up position). Once again, I could hear this working great in a bus/send configuration on an Aux for parallel processing.

Finally, for this test, I placed a stereo instance on the Master Bus, and just played around with Distortion Type and Bias controls, as well as the Output Level. Here's where the Bias makes sense to turn up high, because as you do, the Drive has less impact. So you can turn the Bias up, add a touch of Drive, and get a nice tape saturation effect going.

Overall, the Thermionic Culture Vulture is undoubtedly a useful tool. It added something to every instrument I put it on, especially when used sparingly, but you can also use it aggressively if you want that effect. To warm up your tracks and give them thickness and attitude, take this baby for a spin.

Book: *Film Music: A Neglected Art* by Roy M. Prendergast

I've been reading a book, *Film Music: A Neglected Art* by Roy M. Prendergast. Aside from the historical discussion of some great film music, and many of those wonderful cartoons of the past, there were a few other points that caught my attention.

According to Prendergrast's research, the American film business grossed $1.7 billion in 1946. By 1962, those numbers had dropped to $900 million and the production costs of films had gone up. He noted two main issues for the cause of that decline; first, the studio-owned chains of theaters lost their monopolistic control of the film distributions and showings, basically due to antitrust laws going into effect. The second reason for the loss of revenue was the new 'electronic toy', which was being developed in the 1930's. This toy combined picture and sound and at first, he noted, Hollywood laughed at this new invention. But by the late 1940's and early 1950's Hollywood was forced to deal with this new technology: television.

So, as time progressed, more and more Americans bought these new TVs and stayed home to watch shows instead of going out to the movies. Prendergrast noted "at this point, Hollywood declared an all-out war on this new invention. No Hollywood film could appear on television, nor could a film star appear on television. American audiences countered by staying home and watching British movies on television as well as the new stars that television itself developed."

He goes on to discuss that by 1952, Hollywood realized this strategy wasn't working. They had to create a product that would give audiences something TV couldn't. So the first major unique factor was the size of the screen, and the fact that Hollywood had fifty years of research in color under its belt. TV was black and white and at that point, couldn't be recorded for personal playback. At this point, Hollywood tried 3D, Cinerama, Widescreen and finally, the use of stereo sound, which eventually evolved into various forms of surround sound.

Many interesting points caught my attention in his book. With the changing of the American film studio structure and lack of dictatorial production, the independent producer came to power. The independent producer "selected the property, the stars, the director, as well as raised the money for the production and supervised the selling of the finished film. The producer usually owned no lot, no long-term contract with the stars, no staff of writers and technicians. He assembled a company for a particular film, disbanded it when the film was finished and assembled another company for his next film."

After reading through these points and more, I couldn't help but think "what's old is new again." All the format-based issues we've been through—with surround sound, SACD, DVD-A, MP3, the demise of the record companies, production budgets being slashed, and now the push for high-res audio—have a historical precedent. Can we learn from the past? Maybe. I certainly hope so. Television and movies seem to be doing quite well right along side each other, aren't they?

Factory Visit: C.F. Martin & Co., Nazareth, PA

Since a lot of the TV music I compose uses acoustic guitars, I need to have a variety of them on hand. From detuned 12 strings and resonators to high-strung Nashville dreads to small-bodied 0-series guitars, they all have their purpose. But what I didn't have was that classic sound of Americana, which for many is a Martin D Series instrument.

Recently, I purchased a D-18 and, while at the Martin factory in Nazareth, PA, I caught up with the highly knowledgeable Dick Boke, who runs its museum archives and special projects department.

Rich: First, let's start by discussing the different Martin models. Fundamentally speaking, the smaller guitars are going to resonate less. The larger guitars will resonate more.

Dick: Typically that's true. Larger guitars will have more bass response, smaller guitars will have more treble. Mahogany guitars typically have a brighter, crisper, cleaner treble response whereas the rosewood guitars have a deep, more resonant, rich, thick bass response.

Now get more specific in the Martin line.

Well, specifically the dreadnought—the big guitar—has a big, boomier bass response.

Which is the D Series.

Yes, the D Series. But when you build the dreadnought guitar with mahogany back and sides, you still have strong bass, but you have a crisp, clean treble response, very nicely balanced. Generally, I think the mahogany guitars, like the D-18 or any of the mahogany guitars built in the smaller sizes as well, are typically better for record-

ing in the studio.

Studio engineers constantly are telling me that in the studio, the D-28, which is rosewood, produces such a strong bass response they can't subtract the bass response out of the recording. They can add warmth, but they can't subtract it. They prefer the warm, mahogany back and side instruments.

We did a project with George Martin, who said he specifically ran into this problem in the studio. Designing the guitar as his signature model, we were trying to create the quintessential recording studio model. We actually chose the M size. The M size is one size larger than the Triple 0. It has a very tight waist. The tight-waisted M guitars were initiated as a result of David Bromberg. He was having problems with guitars because he was playing notes so quickly that they would resonate, but they wouldn't go away. They had *too much* resonance. He wanted a guitar that would play, resound, resonate and decay quickly enough for the next note. So he gravitated towards this M size, which was actually an arch top Martin guitar known as the F series. Somebody had smashed the top of one of them, and it was re-topped with a flat top, and Bromberg got a hold of it. He brought it to us and showed us what he was looking for, and we made the M model, which became the favorite of folk musicians playing onstage and studio musicians in the studio because it wasn't producing feedback like the big dreadnoughts were. It was really really good in studio, and really *really* good on stage. Guthrie, Bromberg, Steve Howe, etc., are all proponents of the M models. Of course, nowadays with the pick-up technology, you can install a piezo and they've evolved very nicely. You can play Madison Square Garden with a soundhole cover and not worry too much about feedback.

What is the classic Martin bluegrass instrument?

That would be the D-28. Though you can't discount the D-18. Both of them have their role in bluegrass. Peter Rowan, Lester Flatt, Elvis Presley, Neil Young, McCartney—everybody, regardless of the genre, has gravitated toward the D-28 because it's so good for vocal accompaniment.

Talk briefly about the tops of the instrument.

The Sitka tops—Sitka spruce which grows between Oregon and Alaska—is probably the most prevalent wood for guitar tops. The Sitka spruce produces a deep, low boom when you tap the top. It's quite low, like a rumble. Of course, when you build a guitar and string it up, it can produce a very nice balance, but the other types of spruce that we use are Engelmann spruce, Italian Alpine spruce, German spruce, Carpathian spruce and historically, we've used Adirondack spruce. Roosevelt made a huge state park out of the Adirondacks, so logging was discontinued in the 1940s. Our supply of Adirondack spruce got more and more difficult, so we switched in the 50's to German and Sitka spruce. Recently, Adirondack spruce has become more available though, from personal stock.

So conversely with regards to Sitka spruce, which has low tap tone, I was selecting guitar tops for Laurence Juber, who played with McCartney and Wings. These were Adirondack tops, and he wanted me to pick them out. They had a much wider grain, but they were producing a very high pitch, noticeably about half an octave higher. I equate this type of thing to banjoes. If you stretch a banjo head and start adjusting the lugs around the banjo head, the more you tighten them, the higher the pitch. So I'm equating the higher pitch top with a stiffer membrane. And of course, the Adirondack is highly prized for that reason. That's not to say you can't build a guitar with any variety of spruce and get a great sound. So overall, you've got Adirondack with a higher pitch, Sitka with a lower pitch. The rest of the spruces tend to fall between them.

The D-18 records so well. How about the classic D-18 top?

The D-18 uses a Sitka spruce top. These tops are the most available [for] us, so there's great quality of the grain, with the straightness of the grain and the medullary rings, which run exactly perpendicular to the grain rings.

Can you explain medullary rings?

Medullary rings run perpendicular outwards. When you hand split a cant of spruce, you're in effect creating a pie section of the log. First, you're cutting out cylinders, and then you're using a wedge to cut out pie sections of the tree. Those are put on a saw that swings and allows you to cut perfectly vertical grain. The grain rings, as you're viewing the end of the spruce, the grain rings are showing up as perfect vertical lines. They're extremely strong, but also very stiff and vibrant.

The classic recording instruments, the D-18 and the D-28, have the same tops but different backs and sides.

Yes, the primary difference is the back and sides. The D-28 has East Indian rosewood back and sides, which produces a warmer resonant tone than the D-18. The D-18 has a clearer, more focused bass response. Laurence Juber started to compare tone to wine. Maybe this is a good way of thinking of it, that the mahogany—the D-18 has a chardonnay effervescence—whereas the D-28 is more of a cabernet, more aromatic in its tone. And there are other tone woods like Koa wood, which is used for back and sides, and maple. I would consider maple more of a vodka. The Koa wood is falling somewhere between the mahogany and rosewood—maybe like a Malbec or something. Ah, let's go have lunch!

"Designing the guitar for [George Martin] as his signature model, we were trying to create the quintessential recording studio model ... we actually chose the M size."

by Rob Tavaglione
PAR Contributor
rtavaglione@nbmedia.com

This summer has allowed me to enjoy a wide variety of musical styles and some interesting new sonic horizons, challenging my usual work assumptions. It's all a part of this ever-changing landscape of "how an audio engineer survives in this DIYD (do it yourself digitally) age." Thus I try to keep my competitive edge with superior gear and informed ears, and these review subjects all fit the bill, which ultimately helps keep my united nations of clients—from Russia, Sierra Leone, the U.K., from metropolitan suburbs to small country churches—all sonically satisfied.

SPL Crimson Desktop Interface/ Monitor Controller

Up until this review, I've never used a desktop interface or monitor controller, as I drive a console with rack-mounted converters/interfaces. Sure, I've watched my less-fortunate colleagues and students struggle with some pretty bad (and quite popular and cheap) devices that are inadequate for pro use in many ways—including their mic preamps, converters, headphone amps and general build quality. The SPL Crimson appears to solve these problems and offer enough flexibility to claim the top perch in this category.

Features

Input section: two single-transistor discrete mic preamps (with phantom power, HPF, XLR inputs), two pairs of line inputs on quarter-inch balanced TRS connections, two Hi-Z instrument inputs on quarter-inch TS, a pair of RCAs and an eighth-inch stereo mini-plug for -10 dB consumer devices (with an automatic, bypassable gain boost to pro level), and a digital input via SPDIF.

Monitor section: a large unstepped control-room level control, two sets of control room outputs (set A on XLR, set B on quarter-inch TRS with "tweaker" trim controls), two headphone amps with quarter-inch TRS outputs and high output, a balance control for blending between the analog input section and the DAW returns.

DAW implementation: two pairs of DAW returns via one USB 2.0 input (not 3.0, but 2.0 for its faster and more stable drivers with 1 ms of latency), a total of six simultaneous channels of conversion to/from DAW, 24-bit processing, sample rates up to 192 kHz, and low-jitter fixed internal master clock. The Crimson will operate sans drivers (using Core Audio), but high sample rates and low latency requires SPL drivers.

In Use

I started out using the Crimson simply as a stand-alone monitor controller and was immediately struck by its "feel." The steel chassis, the large control room level pot, the trim and headphone level pots, the switches—they all had that firm and smooth operation that inspires long-term confidence. The rear-panel legend was printed twice, once upside down, for easy connecting from either viewpoint. Savvy ergonomics? My interest was immediately piqued.

As I ran through the functions in my mix session, the Crimson did not disappoint. I summed to mono to check for phasing issues; monitor switching was convenient with a single button push between A and B; and I kept two sets of cans (for me and the client) always connected for quick comparison checks. I noticed that the control room level did not go all the way down to muting the hot +4 outputs of my D/A converter, but the

Audio Clips: SPL Crimson

Please follow the link below to audio clips recorded directly to DAW via Crimson's instrument input or mic preamp (for the vocal track using an AKG C451). Note that the vocal track exhibits wide dynamic range, from handclaps that are purposefully slightly clipped with squared off waveforms—exhibiting Crimson's gentle handling of "overs"—to super-quiet whispers with virtually no noise floor.

https://soundcloud.com/pro-audio-review-magazine/spl-crimson-acoustic-gtrbass-di-vox-441

manual explained that these pots offer about -80 dB of attenuation, enough to silence most sources. I did notice that the control room level was not balanced and centered at lower levels; I'd prefer stepped attenuators for such very low-level balance checks.

I downloaded the Mac drivers from SPL's website (they offer Windows drivers, too, for XP and Vista 7 and 8) and suffered a bad install. Once I reinstalled the drivers, I received stable and excellent operation from the Crimson as a front end/DAW companion. For a naked, no-bed voice over, the Crimson mic preamps did a fine job. I'd describe their voicing as "neutral and flat" as they lacked any significant color but were super quiet and distortion free, even with lots (up to 60 dB) of gain. They are maybe not as euphonic as some out there, but wisely clean and non-obtrusive. The instrument inputs were quite similar; with passive basses, active basses, acoustic guitars and electric guitars I received ample gain, low noise and neutral voicing (again, not as "pretty" as my reference preamp, the Millennia-Media STT-1, but smartly flat and flexible) with plenty of headroom.

For overdubbing vocals and such, the Crimson again nailed the job. The blend control allows no-latency monitoring of the analog input signal and acts as a convenient one-knob "more me" control when the singer needs just a little more level over the music. The multiple sets of DAW returns are also quite useful for setting up separate monitor mixes or wet/dry balances.

For more complicated scenarios (two headphone mixes, the need for talkback, and checking reference mixes) the Crimson has a number of advanced routing flexibilities. These are basically achieved with the "Artist mode" which routes analog inputs in real-time, monitoring of DAW returns 1/2 to producer via Phones 1, returns 1/2 or 3/4 routing out to Speaker B for a headphone amp and Phones 2. Talkback is achievable with a externally amplified mic into analog source 1-left; upon hitting the "talk" button, talkback routes to Phones 2 and Speakers B, while Speaker A is dimmed to prevent feedback.

With so many functions available for a number of the jacks and connections, I had to constantly re-patch to achieve different set-ups (e.g. inserting plugs in quarter-inch inputs 1/2 will override mic inputs 1/2, instrument inputs 3/4 override lines 3/4, etc.). Nonetheless, there's enough flexibility and utility here to satisfy the needs of all but a traditional pro facility—and that's a lot more function than typical out of a desktop device.

I used the Crimson at at 44.1 and 48 kHz and it sounded even better at 96 kHz (a bit more open, shiny and precise) with basses, vox, guitars, acoustic guitars and percussion.

To My Ears

In my opinion, the performance of the Crimson measures up to professional grade. Clean and neutral mic pres and converters as well as excellent sonic performance (high headroom, wide frequency response, ample bottom end, excellent imaging) from all the analog inputs are highlights (surely due to the Crimson's high internal 34 VDC operating voltage). Digital capabilities are up to par as well with excellent, stable and fast drivers, low latency and a lack of issues from sample rate conversions or external clocks.

Despite a few technical concerns, I am going to give the Crimson an unmitigated approval for the most basic reasons. For $699, buyers get a steel chassis and a set of strong components that I estimate will last four to five times longer than those plastic toys which populate the desktops of my studio students/interns and newbie clients.

SoundField SPS200 Surround Sound Microphone

Multiple diaphragm surround-sound mics seem like such a luxury to engineers who work in stereo. There are numerous solutions and methodologies for capturing such immersive audio, but the microphone systems on the market are quite expensive and often bulky—and if not bulky, they at least have accompanying hardware for encoding, monitoring, etc. Worse yet, some solutions require users to commit to a surround format and hope that any needed fold-downs or re-formatting will be successful in post. The SoundField SPS200 takes an entirely different approach.

Audio Clips: SoundField SPS200

Please follow the link below to restricted public access in the "*Pro Audio Review* webclip audio" folder, containing outdoor natural sound 5.1 surround audio capture in both interleaved 5.1 and stereo fold-down via SoundField's software. Click each clip for stereo streaming or surround download:

https://drive.google.com/folderview?id=0B9cLrbot3Mw3Vmx6bkdMNXRnNFk&usp=sharing

Features

The mic itself is pretty simple and shockingly portable: four small-diaphragm condensers (the same as in other SoundField mic systems) in one chassis, in a tetrahedral arrangement, mounted on a single small body (no bigger than a C451 with multiple heads). Its proprietary cable fans out to four XLR outputs. The mic can be positioned end-fire or side-address; then (remember how you pointed it) the accompanying Surround Zone software (TDM/RTAS, AU, VST) will encode your audio into the desired format (stereo, 5.1, 6.1, 7.1, etc.) after the fact, in post. More than just encoding, numerous "placement" functions are selectable: variable HPF, M/S encoding, swiveled left or right, tilted up or down, zoomed in or out, and widths of front and back are adjustable; as long as the mic is reasonably placed, numerous options abound.

In Use

I used the mic for some non-surround apps, like capturing an a capella Gospel trio. I placed the 200 in the center of the group, aimed upward (side-address), used four channels of super-well-matched Earthworks 1024 mic amps and received great results. The frequency balance was desirably neutral—similar to a DPA or Schoeps mic array, not euphonic like Neumann or AKG. The cardioid patterns seemed sufficiently wide, with very nice imaging and a palpable sense of "being there." Without using the software, I panned and leveled using only common sense and received great results. OK, I did add a little chesty bottom end in the mix, and that really pleased the clients.

Next, it was time to really test this baby, so I called in local engineer Joe Miller for a second opinion. Joe used it to record orchestra with choir and he praised the ease of set-up. Monitoring only in stereo on location, Miller wondered how his tracks might sound after decoding, but found the software to be indispensable. "The microphone sounded great, but the software took it to the next level," he explained. "I can't imagine using a hardware decoder for this mic." Any worries I had about bottom end response were squelched, as the bass was extended, balanced and more than ample.

Miller also captured the sounds of insects with the 200 and was even more impressed: "My tests were outdoors near a wooded area. After bringing it back to the studio and decoding, it was shocking how natural the surround presentation was relative to the natural environment. Plus, the random motion of sound within the field made for interesting playback in the studio: very lifelike, very cool. One caveat: this microphone eats wind [Ed. note: As will any condenser in an outdoor environment] and a blimp or dead cat windscreen would be mandatory, at least for me."

To My Ears

Any multi-channel mic is only as good as its kit, and full kits are available with Rycote windshield and Pelican case for $3,800 (about $2,750 for mic and software only) and such a rig is a necessity for anything outdoors. That price point is nothing to sneeze at, but is more reasonable than first glance. "You really are buying four mics," Miller offered, highlighting the bottom line. Four mics, world-class sonics and eminently flexible software, plus the deal-maker: If you've ever captured surround audio, you know "placement regret" is possible and how fatal such errors can be. It seems to me the Surround Zone software is worth the price in peace of mind alone.

Coleman Audio RED48 Analog Summing Device/Monitoring Controller

Glenn "Coley" Coleman builds simply designed products, always using high-quality components. His products solve problems without frailty, flashy frills or corporate hyperbole. Upon hearing of the unique feature set of RED48, I was intrigued, as Coleman quality in such a forward-thinking design could make for a potent piece of gear.

Features

The RED48 is really a three-function unit: a transformerless analog summing device, a communications/cue system and a "center section of a console" (with passive electronics, stepped level control), all in a 2U chassis. Three stereo input sources on XLRs (or the summed mix) are selectable for monitoring. There's also a stereo "cue" input (dual quarter-inch TRS) that can be routed to the control room/engineer phones output (front panel quarter-inch TRS) or via "cue" output to talent/performer (with dual TRS to a headphone amp or similar). A small talkback mic is included, connecting via eighth-inch mini-plug, and is engaged with a momentary switch.

Summing is accessed with six DB-25 connector inputs; with 48 tracks of analog summing arranged in stereo pairs only, there are no pan controls, mono switches or faders, and all inputs are summed to L/R at unity gain. Mix output is via two XLRs to the user's master deck or back into the DAW.

Insert points (stereo send and return) are provided for the mix via four TRS jacks.

RED48 also includes a single, wired master L/R fader (a Penny & Giles long throw) in a remote desktop box that is very useful (as the only other level controls you'd be using would be ITB). This remote also wisely houses backlit dim, talkback engage and slate buttons.

In Use

Setup was easy and I was running in no time, other than stumbling upon mismatched threads between my cabling's DB-25 connectors and the RED48's connectors. Did you know there are metric and SAE thread types used for DB25s? Ugh—it's like pin 2 hot vs pin 3 hot all over again (though SAE 4-40 is specified in AES standards—*Ed.*). Anyway, once connected, I took an ITB mix session and spread it out over 24 channels (12 stereo pairs) of output via D/A converters and then into the RED48. Bam! Instant OTB (out of the box) mix.

Immediately, mix differences were present to my oversensitive ears. First, the RED48 has back and bottom: deep bass extension yet a lack of bass "color" with no frequency-based humps, dips or other negative interpretations. Second, I noticed a lack of harshness, as the tracks lost the forwardness that I've heard via other summing methods—either ITB or in my Soundcraft Ghost analog console. Third, and perhaps most importantly, the soundstage took on an entirely different shape and size. Admittedly, my ITB mix was every bit as wide, with even more precise panning, and the RED48 mix had a sense of front/back and depth that was obviously pronounced. In particular, reverbs took on more space and realism, with increased audibility as well as presence.

I certainly missed insert points, as I wished to process individual tracks as well as apply a little NY-style parallel processing to my groups as well. This function is achievable by routing DAW converter outputs to dynamics processors before hitting the RED48; 48 channels is enough room to accomplish a 32-channel mix with eight more channels of subgroup for NY-ing. Nonetheless, I didn't have a patchbay or cabling flexible enough to accomplish all my insert point needs, and summing units with channel inserts are rather expensive.

As I mixed, I found myself wishing for numerous analog desk conveniences, which I have on my console: a mono summing switch for checking polarity, channel faders to give me a little more past unity at crescendos, aux sends to hit my outboard effects and level meters! Other than the total absence of metering and the lack of a mono switch (pretty important needs, IMO) the other conveniences can be worked around. These "no console" sacrifices are largely overshadowed in importance by a little thing known as "total recall," as one can perfectly recreate a mix just like you would ITB with the exception of needing to chart any L/R insert processor settings (or others, if you patched processors inline).

To My Ears

The RED48 is quiet. Those passive electronics don't hiss or hum, even when cranked, so you may forget you are "up full"—a good thing for the backlit fader remote buttons or you wouldn't even know RED48 was powered up.

The RED48 barely has a sound: no color, no attitude, very dynamic and plenty quick, though the extended bass might be mistakened for bass hype. About a third of my clients preferred ITB mixes while the other two-thirds preferred the RED48.

The RED48's headphone amp sounds great, and all its switches feel great—quality components.

The RED48 overloads gently (with a grainy fizz) if hit with too much level—no nasty "clacks," like some analog.

The RED48 is simple, sturdy and proper; I'd expect a lifetime of service, if treated right.

If you place importance on nebulous qualities like mix depth, imaging, a sense of space and low-level-detail, go get a Coleman RED48. They're only $2,500 and it's a wise investment for your work. If you like analog summing, but require a tonal shift and some color to get your bus moving, then the RED48 is not your choice. Others manufacturers in this market category offer character via transformers (Phoenix, SPL, Thermionic Culture); many have them on the L/R, and a few have them at all inputs. If you need insert points, those are available, too, but only on a couple of units that have price tags over $3,500. Many high-end analog summers offer pan pots, faders, level boosts and some even auxes—but then that's basically a mixer, isn't it? Part of our goal here is total recall and decades of durability, and, in general, fewer parts = fewer problems.

For what it does, at this price, with this many channels, the Coleman RED48 is the best deal in analog summing today.

Superb Studio Monitors for $500/Pair from JBL and M-Audio

JBL 3 Series LSR308 Studio Monitors and LSR310S Subwoofer

by Strother Bullins
Editor of *PAR*
sbullins@nbmedia.com

In January 2013, JBL Professional unveiled its large reference studio monitor—the M2—to intrigued pro audio types at the Winter NAMM Show. In private listening sessions, the M2 certainly was impressive to me, due to a number of new JBL technologies, one of which—its patented "Image Control" waveguide—provided truly superb imaging plus flat and even frequency response across a wide listening area, creating a very pleasing, smooth listening experience.

On the opposite end of the price spectrum, JBL soon after unveiled its 3 Series of powered nearfield studio monitors with a very familiar-looking waveguide for its soft dome tweeter—one that, to my eyes, was clearly influenced by the successful M2's "Image Control." I wondered, would the 3 Series be anything like its large and pricey ancestor, especially considering that the largest 3 Series monitor, the LSR308, costs well over $9,000 less than the M2?

Three months ago, I received a pair of LSR308 monitors ($249 street, each) and its accompanying subwoofer, the powered LSR310S ($399 street) for review. And what I discovered over the next few weeks was surprising.

JBL's 3 Series powered monitors are available in two sizes: the aforementioned LSR308 with an eight-inch woofer and the LSR305 ($149 street, each) with a five-inch woofer (a 41W/41W Class D bi-amped design). The LSR308 features bi-amped Class D power (56 W/ 56W for LF/HF, respectively), a 37 Hz to 24 kHz frequency range, 112 dB SPL (C-weighted) maximum SPL, and a ported design. Inputs include XLR and quarter-inch TRS with switchable +4 dBu/-

PHOTO: RHON PARKER (RHONPARKER.COM)

M-Audio's BX8 Carbon monitors sit atop JBL's 3 Series monitoring rig—an LSR308 pair with the LSR310S subwoofer.

Get the most out of your exhibitor investment by marketing directly to AES attendees

PLANAHEAD & PREMARKET

THE AES SHOW RETURNS TO LOS ANGELES THIS OCTOBER

Consult with our marketing representatives: reserve your spot today for premium positions and talk to us about your complete AES marketing package.

AES's much awaited return to L.A. begins this October, and The AES Daily—the official daily of the show—will be all over town: at the entrances, show hotels, buses, and throughout the AES show floor. We will also be distributing copies to many of L.A.'s most influential studios and dealers.

FULL AES DAILY PACKAGE STILL AVAILABLE

here's still time to get into all 4 editions, including the **VIP issue** (mailing in both print & digital versions two weeks before the show), plus all **3 at-show editions**!

CONTACT YOUR REPRESENTATIVE TO GET YOUR CUSTOM AES DAILY PROMOTIONAL PACKAGE:

- Charlie Weiss, Group Publisher, cweiss@nbmedia.com, 212-378-0478
- Tara Preston, Associate Publisher, East Sales/Europe, tpreston@nbmedia.com, 917-331-8904
- Janis Crowley, Senior Account Executive, Western Region, jcrowley@nbmedia.com, 415-505-1985
- Zahra Majma, Specialty Sales, zmajma@nbmedia.com, 212-380-0400 x517

10 dBV input sensitivity, and the cabinet weighs in at a comparably light 19 lbs. Rear panel adjustments include three-position LF and HF Trim parameters, each with -2 dB, 0, and +2 dB settings. The front of the speaker is made of attractive polypropylene while the rear-ported cabinet is constructed with a lighter weight MDF.

The LSR310S is a down-firing powered subwoofer featuring a heavy-duty 10-inch driver, 200 W of Class D amplification, and a low frequency range of 27 Hz with a maximum SPL of 113 dB. It features a compact design (less than 18" x 15" x 16" and 35 lbs. in weight), the patented rectangular "Slip Stream" double-flared front panel port "for accurate bass response at low playback levels," and JBL's proprietary XLF Extended Low Frequency—a 10 dB boost at 60 Hz engaged via switch; XLF is essentially a custom frequency response to simulate club/large PA playback, especially helpful in modern-leaning music productions.

Overall, 3 Series build quality is impressive and attractive, even if a bit spartan yet refined. In a business overly aware of brand names, these monitors also have the value-added benefit of being "JBL"—a name associated with professional studio monitors for decades.

The LSR308 pair satisfied my low-end needs by themselves in most applications. Yet, while I don't typically use a sub during production, the LSR310S came in handy, like when listening to an ultra-low frequency sound source in a mix—a bass "drop" for emphasis. The sub allowed me to accurately gauge where these extreme low-end elements sat in the mix. But largely, I didn't notice the sub, and that's to its credit; the transition between the low-end from the LSR308 pair and LSR310S was seamless. I expect that the sub would be more crucial/useful when paired with the smaller LSR305s, or when working less in pop/rock and more in music ranging from EDM to even orchestral/chamber music.

I've auditioned many solid nearfield studio monitors over the past 15 years, most in the eight-inch woofer category, used within my own workspace. That said, the 3 Series truly impressed me. I could use them as my main monitors indefinitely.

Knowing their low price points before starting this review, I was suspecting to hear things I wouldn't like, based on its composition (of largely materials such as MDF, plastics and low-cost Class D amplifiers), but I discovered that the design trumps materials. To me, the Image Control waveguide obviously makes the LSR308 a low-cost marvel of detailed imaging and controlled frequency response across horizontal and vertical planes, as the tweeter itself seems to be a rather common soft dome. The fact that a recordist can own a pair of large JBL near-fields, clearly borne from M2 R&D and this good-sounding, for under $500 per pair makes the LSR308 perhaps the best bargain available in powered studio monitors today.

"The fact that a recordist can own a pair of large JBL near-fields, clearly borne from M2 R&D and this good-sounding, for under $500 per pair makes the LSR308 perhaps the best bargain available in powered studio monitors today."

M-Audio BX8 Carbon Powered Studio Monitors

M-Audio's BX8 Carbon studio monitors arrived while I was reviewing another eight-inch woofered, powered pair of studio monitors, JBL's 3 Series LSR308 studio monitors—also priced at $249 street, each (see above). I'll start off by saying that either of these competing products are solid choices for discriminating recordists, though different enough in both specifications and performance to represent two decidedly different experiences.

The BX8 Carbon is a two-way powered monitor featuring an eight-inch Kevlar woofer and one-inch silk dome tweeter with a 38 Hz to 22 kHz frequency response; a BX5 Carbon model with five-inch woofer is available, too ($149 each, street). Rather than Class D, the BX8 Carbon is powered by Class A/B amplification with 70 W and 60 W power ratings for its low- and high-frequency drivers, respectively, thus weighing in at a comparably weighty 26.4 lbs. On the rear panel, inputs include balanced XLR and balanced/unbalanced quarter-inch TRS and three-position acoustic space switch; when employed, it compensates for bass build-up in placement near walls and corners with flat, -2 dB and -4 dB LF settings. Cabinetry is made of vinyl-laminated MDF with, for a lack of a better word, muscular front panel gray-on-black styling.

I used the BX8 Carbon pair alongside LSR308 pair during the latter half of the review period, routing through the Dangerous Source Monitoring Controller (reviewed in full, below). I also matched the pair with the LSR310S subwoofer, utilizing the speakers both with and without sub-frequency support.

The BX8 Carbon monitors impressed me right off the bat. Over the past few years, I had noticed a marked increase in apparent quality and features within the M-Audio studio monitoring lines, and the BX8 Carbon is proof of the company's dedication to this affordable segment of the market. Like the LSR 308, the BX8 Carbon performed beyond what you would assume for the price point. Bottom end is nicely full and smooth, yet with notable "punch" response; as with the LSR 308, I didn't rely on the subwoofer that much. To my ears, it reminded me of a "tighter" KRK Rokit 8 ($249 street, each) with the detailed, open, and more "imaging-friendly" top end

of the KRK VXT8 ($599 street, each)—the eight-inch powered monitor I had depended on for nearly a decade. As such, I find the BX8 a true bargain and worth serious consideration for those shopping in that "Kevlar woofer" category of powered studio monitors.

Dangerous Source Monitoring Controller

The Dangerous Source is a portable desktop monitor controller featuring comprehensive I/O—multiple analog and digital inputs, dual speaker outputs and dual headphone outputs—flexible routing options, and premium quality knobs and switches on the front panel. As such, it's a mobile recordist's dream come true, featuring premium components and crystal-clear signal paths. I had the opportunity to use the Source as my mobile studio hub for the past six months—in most every possible source, headphone and monitor configuration—and was thoroughly impressed, enlightened and inspired.

Unlike many all-in-one routing/monitoring products available today, the Source doesn't provide bling in the way of mic/line preamps, inserts, metering features, etc. Instead, its cost reflects Dangerous Music's investment in impeccable manufacturing standards and a tightly focused purpose: to provide very high quality, flexible monitor routing options for professional end users.

Ten multicolored LED-backlit round switches, two rotary level pots, and two quarter-inch headphone outputs populate the Source's front panel. These include four switches for Headphones Source; four switches for Speaker Source; SPKR 1 or SPKR 2 speaker source switches; Headphones Level and Speaker Level volume controls; and single or multiple-source headphone jacks. Monitoring sources can be either AN 1 (two analog Amphenol XLR Combo inputs); AN 2 (analog -10 dBV eighth-inch stereo input); AES/SPDIF (XLR with dual on-board stereo 24-bit/192 kHz D/A converters); or USB (up to 24-bit, 192 kHz, Win/Mac OS 10.6, USB 2.0 compliant). SPKR 1 (dual XLR), SPKR 2 (dual quarter-inch TRS), Selected Line Out (dual quarter-inch TRS) and an XLR AES/SPDIF throughput complete the Source's output options. The Source is also available with an optional 1U rack panel kit/faceplate.

Setup was easy and intuitive with my MacBook Pro for USB-based use, and I quickly discovered the benefits of the Source's Toggle and Latch features. Toggle allows silent switching between four input sources—ideal for comparing mixes or a mix with a reference track; Latch allows silent switching between speaker sources—ideal for monitoring multiple inputs at the same time. Momentoggle, a feature built into all source and selection buttons, allows users to talkback with artists—just add mic and preamp.

The Source easily complemented my recording and monitoring routines—whether, for example, comparing source material between two monitor sets (as I did while reviewing the JBL LSR308 3 Series and M-Audio BX8 Carbon powered monitors), monitoring Logic Pro X mixes via USB with a separate headphone mixes for overdubbing vocalists and instrumentalists or simply listening to iTunes via the AN 2 input while working at my desk.

> "The Source's cost reflects Dangerous Music's investment in impeccable manufacturing standards and a tightly focused purpose: to provide very high quality, flexible monitor routing options for pro end users."

At $899 street, the Dangerous Source really isn't a low-cost monitoring solution, as cheaper controllers with a wide variety of subjectively useful bells and whistles abound in the marketplace. However, if you're not budget-restricted nor interested in an all-in-one solution—you already have a variety of pro-grade input sources to juggle and prefer to use your own preamplifiers/signal chain for tracking—the Dangerous Source is a straightforward and great-sounding monitoring device, pleasantly overbuilt to last for years.

CAD Audio 1600 Series Wireless Systems

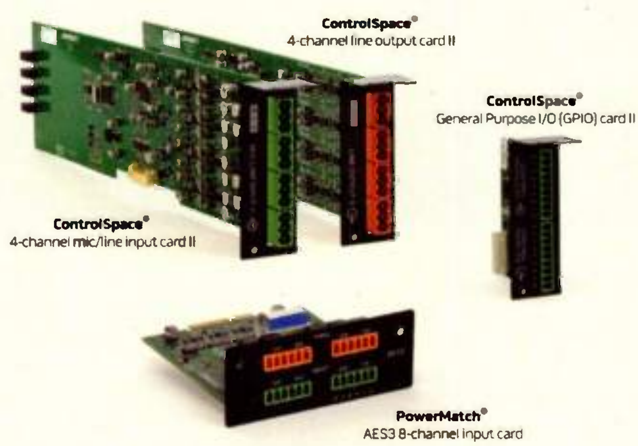
Ideal for house-of-worship, broadcast and corporate applications, CAD Audio's new 1600 Series wireless systems provide 100-channel frequency agile UHF operation for maximum operating range. They also feature proprietary CAD ScanLink technology to precisely scan, select and link to the optimum channel in any RF environment. Other features include True Diversity operation to minimize multi-path interference along with CADLock Automatic Tone Encoded Squelch that eliminates unauthorized transmissions in the signal path.

The WX1610 Bodypack System features CAD's Equitek E19 miniature condenser earworn mic, a Cardioid Lavalier and guitar cable. The TX1600 handheld and TX1610 body pack transmitters have SoftTouch multi-function On-Off/Mute/Low Battery/ScanLink status switches with multi-color LED indicators. Two AA batteries provide more than 15 hours of transmitter battery life. The WX1600 receiver is housed in a metal chassis with quarter-inch and XLR outputs for additional flexibility. System specs include a Frequency Response of 40 Hz to 15 kHz, and Dynamic Range >105 dB.

Stay tuned to the *PAR* for a full review of the TX1610 Series package.

Contact: CAD Audio | www.cadaudio.com

Bose ControlSpace Accessory Cards



Bose ControlSpace and PowerMatch digital audio processing and amplifier systems now have several new input and output accessory cards. The new ControlSpace cards, each for the ESP-00 II engineered sound processor, include the ControlSpace 4-Ch Mic/Line Input Card II, ControlSpace 4-Ch Line Output Card II and ControlSpace General Purpose I/O (GPIO) Card II. For PowerMatch amplifiers, the PowerMatch AES3 8-Channel Input Card offers a direct connection from digital audio mixing consoles.

The ESP-00 II processor features eight available I/O expansion slots that accommodate any combination of analog and digital cards, and supports up to a total of 64 bi-directional channels.

The ControlSpace Mic/Line Input Card II boasts >117 dB dynamic range, improved distortion and crosstalk performance, with adjustable gain up to +64 dB to accommodate both microphone and line level analog sources. The ControlSpace Line Output Card II features >115 dB dynamic range and is capable of +24 dBu analog output. The ControlSpace GPIO II 8x8 expansion card provides enhanced general purpose analog and digital connectivity for easier integration with output devices such as LEDs and relays. Additionally, the GPIO II card will replace the standard GPIO card that ships with the ESP-00 II processor. Up to two GPIO II cards may be fitted.

For PowerMatch configurable professional power amplifiers, Bose is introducing the PowerMatch AES3 8-Channel Input Card in response to customer requests for a direct connection from digital audio mixing consoles. It represents the fourth option card available for the PowerMatch amplifier line, joining the Dante, CobraNet and ESPLink audio networking options currently available.

Contact: Bose Professional | worldwide.bose.com/pro

Mackie SRM450 and SRM350 High-Definition Portable Powered Loudspeakers

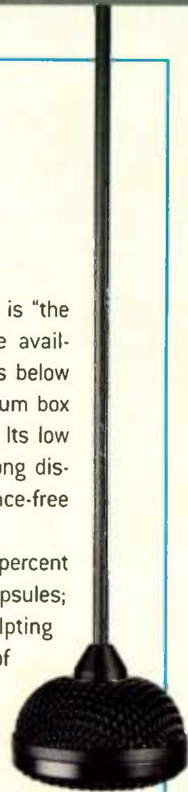
With a new 1000 W amp platform, the redesigned SRM450 ($629) and SRM350 ($519) two-way powered live loudspeakers offer two simple-to-use, built-in DSP-based tools: first, four application-specific speaker modes, each re-voicing the SRM to sound best per application, and second, an intelligent feedback destroyer, which instantly identifies and eliminates feedback using up to four narrow 1/16th octave filters.

Mackie's proven HD Audio Processing—making its top-shelf HD Series so good, as proven in full *PAR* review—is a combination of patented acoustic correction DSP with a precision crossover, driver time alignment and phase correction.

Finally, the SRM450 and SRM350 now feature a handy integrated two-channel mixer with input-friendly Wide-Z inputs—instrument, line or mic-ready, and ideal for singer/songwriters relying on instrument and voice for live performances. Mounting and handling features have been updated as well.

Contact: Mackie | http://www.mackie.com/products/srm450v2/index.html

Audix M3 Tri-element Ceiling Microphone



According to Audix, its new M3 is "the only multi-element microphone available with fully balanced circuits below the ceiling and a UL rated plenum box solution above the ceiling tile." Its low impedance design allows for long distance, cross talk and interference-free cable runs.

Other features include 100 percent RF shielding; gold diaphragm capsules; low self noise; frequency sculpting for voice clarity and rejection of extraneous noise; adjustable cable length; and black or white color options.

Contact: Audix | audixusa.com

One Systems 118|M-SUB High Performance Natatorium Loudspeaker System

The 118IM-Sub is a lightweight copolymer-based, all-weather loudspeaker system with a medium-density polyethylene enclosure, optimized for high UV exposure and long life in harsh environments. The 118IM-Sub may be used in direct weather outdoor installations, as well as indoor applications where extended low frequency response is required. The 118IM-Sub offers a compact design featuring a patented One Systems 18I/O 18-inch extended bass transducer. The 118IM-Sub is available in black or white and offers a complete selection of array and flying/suspension hardware.

The 118IM-Sub's grille is a 3-layer "rain shield" design to minimize direct rain contact with the system's transducers. Its unique vent design also allows for easy moisture drainage from the interior of the enclosure. The enclosure and grille design are rated to IEC 529 IP45 (solid object penetration to 0.04 inches and water jets from any direction). The enclosures and components are also rated to Mil Spec 810.

Contact: One Systems | http://www.onesystems.com/118im-sub.php

live review

Midas Pro1 Digital Console

After 33 years in the production business, we have, as do most production companies, a rather large fleet of expensive, behemoth-like analog consoles. We have a small armada of Midas and Soundcraft analog consoles—each weighing in the 600-800 lb. range, about the size and weight of a small cow. Thus I refer to our shop as "the barn."

by Will James
PAR Contributor
wjames@nbmedia.com

Lately, we have covered a couple of festivals where I marveled at the compact nature and powerful capabilities of the new breed of digital desks. Even the "Big Boy" consoles have come down in price dramatically while offering maximum sound quality and ease of operation.

I recently received a rather pleasant surprise for review in *Pro Audio Review*, arriving in the form of Midas' smallest and most affordable entry in its line of high-end digital desks: the Pro1.

Features

Midas was kind enough to supply us with the Pro1 console in a flight case and with the DL251 snake head, along with 300 feet of Cat 5 cable. The Pro1 took two guys to set up and remove from the case, as its flight case is rather heavy. In usual Midas fashion, the console is quite colorful. It's also quite compact, weighing in at 47 lbs. uncased/200 lbs. cased. The surface is a pleasant steel/slate blue, accented with touches of Midas' complement of pre-school crayon colors. The total surface is broken into five sections: upper left hand screen, lower left hand input faders, lower right hand output faders, upper right hand pre-assigned "hard" keys, and the lower-center's mouse/trackball and clickable buttons.

The bulk of Pro1's FOH mode navigation is done via onboard mouse and left-click key, with other adjustments made at the faders, hard keys and center-located UP/DOWN/ENTER keys. In monitor mode, users mainly live on the hard keys, which flip the channel faders into send mode; when engaged, these remind users, in very large letters in red, that they are in Fader Flip mode.

The Pro1's OS is notably unique and quite a bit more intuitive than many of the digitals I've used. The console arrives as a blank slate; users custom-tailor their initial data settings and preferences based on needs. The center section keys allow instant access to menus and prefs, and building a show via naming, color coding and arranging is quick and painless.

Midas has incorporated some very high-end electronics into this package with a plethora of menu prefs and options. For example, the console can be configured to have six on-board effects and eight DN360 graphic EQs; or users can lower the effect count all the way down to one to have 28 graphic EQs; or users can mix and match anywhere in between. Each of the six effects banks have a variety of reverbs and delays that are instantly malleable to suit—plate, chamber, and vintage room variables plus the DN780 delay.

The output section contains 16 aux masters, six matrix masters, eight VCA group masters, and LEFT/RIGHT/MONO master faders with all names and colors programmable. Both the output section and the automation section allow for page-by-page, scene-by-scene operation. Just above the master faders resides the POP section, short for "Population groups," which allows instant grouping of any faders on the console to a one-touch operation; this is better than sub-groups, and not quite a VCA.

(continued on page 42)

ad index

Midas Pro1

(continued from page 40)

The top right of the console houses hard keys, allowing manual operation of the channel strips, accessed through depressing the channel's name button. The hard keys control input gain, high- and low-pass filters, comp, gate, fully parametric EQ, talk back and monitoring/headphones. The rear panel of the Pro1 offers 24 XLR inputs and 16 XLR outputs, as well as a variety of digital connections including multiple CAT5 connections for multiple digital snakes.

In Use

We unpacked the Midas Pro1 at my house, setting it up in my garage, as this would allow me to learn the OS for a couple of days prior to taking it out to actual gigs. The console arrived with a clean slate, so to speak, knowing nothing. Immediately I discovered how deep yet user-friendly its menus are. In order to assign the DL251 digital snakehead, I had to connect the two provided Cat 5 snakes to the snake head and the Cat 5 connections on the rear panel. My first chore was to build a template, or initial data, for both FOH and Monitor. The channeling menu is basically divided into two halves, input and output.

To create the initial data, I clicked on a series of screen locations, joining the two halves (input and output) together in whatever manner I needed per event. It's not a complicated process, but attention to detail is everything, and this console has every detailed parameter adjustment you can imagine. For example, inserting EQs into monitor masters is a lot like patching them manually, but remember to assign the proper return to the chosen send and turn on your inserted devices. This goes for effects, too, although the on-screen menu allows for instant assigning of the device to whatever master buss you desire, and same goes with the EQs.

As soon as I got through the assigning of everything, I decided to create a template show, naming and color-coding 32 channels worth of inputs, six Effect Returns and two iOS input channels. Connecting outboard equipment (in addition to the full menu of goodies internally provided) was a simple task through use of the channel assign menu, allowing my iPod to be assigned locally—physically at the console—while the DL251 lives on stage where stage I/O takes place. In monitor mode, the console's local 16 XLR outputs connect directly to monitor amplifiers or IEM receivers.

Our first gig with the Pro1 was an outdoor car show and music festival with four bands per day; we provided full production with lighting, roof and staging. The Pro1 was to be used as the FOH mixer. Groups were all tribute acts, ranging between a very good Guns 'N Roses tribute, an excellent Beach Boys tribute, and in between, a band called Winter Dance Party, a tribute to early 60's legends the Big Bopper, Richie Valens and Buddy Holly. The latter was a superb band with great vocals and were excellent test subjects for this console; I was interested to see how much color (or lack thereof) this console added to a mix. My immediate response at sound check was that this console took a boat load of input signals and was relatively forgiving to use.

The user keys at the top of the channel faders serve as analog trims, with additional digital gain available in the hard key section. Unlike some digital consoles, I found that I spent very little time in the EQ department of the channel strip or the overall graphic EQs inserted on the mains. The sound check clipped along quite smartly as the console's hard keys were quite easy to access; the reverbs and EQs in the center menu were, too. The graphic EQs are accessed through the center menu, or through the GEQ key at the bottom of the output select keys; this turns the output faders into various frequency groups, and users may scroll through the EQ frequencies very quickly via arrow up/arrow down keys next to the output fader bank.

My favorite feature of the Pro1 is its POP section, or "population keys," which allow for comprehensive, flexible grouping to one convenient button push. I was using a POP group for lead vocals; I had the vocal channel fader, reverb send and reverb return together. POP group is surely the most convenient way to group together operations, rather than constantly scrolling through layers.

> "Lately, we have covered a couple of festivals where I marveled at the compact nature and powerful capabilities of the new breed of digital desks. Even the 'Big Boy' consoles have come down in price dramatically while offering maximum sound quality and ease of operation."

Summary

I believe that I could go on for 10 pages describing how cool this console is, highlighting the goodies and power contained within it. But I will summarize by saying that the Pro1 provides so many features, it's almost unbelievable considering the price. This console, coupled with the DL251 snakehead, is possibly the best digital mixing system that I've used.

I give the Midas Pro1 my highest recommendation. If you are in the market for a modestly priced digital console that has all the "big boy" sounds, ergonomics, features and clout, this is the one for you. I am going to purchase two of these Midas consoles right away for Atlantis Audio, and we do production for everyone between from Creedence to Korn.

Price: from $7,999 street
Contact: midasconsoles.com

NEXT-GEN 96kHZ USB 2.0

Two beefy high-headroom mic preamps handle the hottest inputs and + 48V phantom power for studio condenser mics.

Studio One® Artist DAW with unlimited tracks, 26 plug-ins, direct upload to free Nimbit® account & gigabytes of 3rd-party loops 'n' samples.

iONE & iTWO INTERFACES RECORD TO iPAD® & MAC® OR WINDOWS COMPUTERS WITH SUPERIOR SOFTWARE FROM PRESONUS

iONE/iTWO™

Beefy high-headroom mic preamp handles the hottest inputs. 48V phantom power.

Both models have rugged metal chassis for rough field use.

	iOne	iTwo
24-bit, 48/88.2/96kHz recording	✔	✔
Class A preamplifiers	1	2
Balanced ¼" line level inputs	–	2
Instrument inputs	1	2
iPad® 30-pin or Lightning I/O	✔	✔
MIDI In / Out	–	✔
Balanced ¼" line level outputs	L/R	L/R
Hi-power headphone output	✔	✔
True +48 V phantom power	✔	✔
Bus-powered for remote use	✔	✔
Kensington® lock hole	✔	✔
Studio One® Artist full DAW	✔	✔

Capture™ Duo (free from the Apple Apps Store*) is the new iPad version of our proven Capture software. One-tap recording... then wirelessly beam your tracks to your Mac or Windows computer and open directly in Studio One Artist! Nobody else's interfaces can do this!

Get the i-Series story and videos @

PreSonus®

PRESONUS.COM/i-SERIES